1998 BROADCAST ENGINEERING CONFERENCE

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FOREWORD

For over 50 years, the NAB Broadcast Engineering Conference has served as the primary source of new information on radio and television engineering. As demonstrated by the new products available on the exhibition floor of NAB '98, digital technologies and techniques are being developed at an unprecedented rate. This year's *Proceedings* contains many of the technical papers presented at the Conference that address these innovative concepts and applications.

It is obvious that the television industry is well on its way to an all-digital future. Conference presentations contained in this *Proceedings* cover the real-world aspects of building a digital television facility, from origination through to transmission. For radio broadcasters, digital technologies have essentially reshaped the studio environment, while DAB carefully moves forward. And, of course, consolidation in the U.S. radio industry is intensifying the demand for new ideas on ways to efficiently operate multiple facilities.

Our partner in the 1998 Broadcast Engineering Conference, the Society of Broadcast Engineers (SBE), once again assisted in developing the conference sessions and presentations. Through its Ennes Foundation, the SBE presented the first NAB Radio Boot Camp. In addition, the National Institute of Standards and Technology (NIST) teamed up with the Institute of Electrical and Electronics Engineers Broadcast Technology Society (IEEE BTS) to present a full-day technical seminar on the digital transition and the Society of Motion Picture and Television Engineers (SMPTE) also organized a seminar addressing the future of digital television based upon their own unique perspective.

We sincerely thank all of the industry professionals who gave their time and energies to make this conference a success. As always, the NAB/SBE Conference Planning Committee welcomes your comments on the program.

It is our sincere hope that this publication will assist you and your organization in making a successful transition to the digital future.

Jerry Whitaker
Chairman
NAB/SBE Engineering Conference Committee

Lynn D. Claudy
Senior Vice President
Science & Technology

1997-98 NAB/SBE CONFERENCE PLANNING COMMITTEE

Jerry Whitaker, Chairman
Technical Press
393 Yellowstone Drive
Morgan Hill, CA 95037

Andy Butler
Public Broadcasting Service
1320 Braddock Place
Alexandria, VA 22314

Jerry Butler
Public Broadcasting Service
1320 Braddock Place
Alexandria, VA 22314

Dane Ericksen, P.E.
Hammett & Edison, Inc.
P.O. Box 280068
San Francisco, CA 94128

Robert Hess
Director, Broadcast Operations/Engineering
WBZ-TV, WODS-FM, WBZ-AM
1170 Soldiers Field Road
Boston, MA 02134

Tom McGinley
Chief Engineer
WPGC-FM/WPGC-AM
6301 Ivy Lane
Suite 800
Greenbelt, MD 20770-1402

Robert Seidel
VP, Engineering
CBS
524 W. 57th Street
New York, NY 10019

Milford Smith
Vice President, Engineering
Greater Media, Inc.
2 Kennedy Blvd.
East Brunswick, NJ 08816

John F. Swanson
Vice President of Engineering, New Media and Technology
Cox Broadcasting
1400 Lake Hearn Drive
Atlanta, GA 30319

Barry Thomas
Director of Engineering
KCMG – Chancellor Media
5900 Wilshire Blvd. Suite 525
Los Angeles, CA 90036

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NAB, Washington, DC

***Presenters**: Robert Graves, Chairman, Advanced Television Systems Committee, Washington, DC and Charlie Morgan, Chairman, National Radio Systems Committee and Susquehanna Radio Corporation, York, PA

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*Papers not available at the time of publication

Creating the DTV Signal: Digital Television Transmission Issues

Sunday, April 5, 1998
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Chelmsford, England

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*Papers not available at the time of publication

HDTV BROADCAST OF MAJOR LEAGUE BASEBALL --A CASE STUDY

Gerald W. Collins
Harris Corporation, Broadcast Division
Quincy, Illinois 62305

ABSTRACT

On September 16, 1997 Harris Corporation sponsored the first ever HDTV broadcast of a major league baseball game. This historic event originated at Camden Yards in Baltimore, MD and was broadcast over stations WETA-HD and WHD-TV in Washington, DC. The broadcast was seen by 300 industry leaders who viewed the game on a 16 by 9 foot screen at the National Press Club. Harris sponsored this broadcast in order to prove the reality of DTV technology by bringing together an entire system from origination through distribution to over-the-air broadcast. To increase the probability of success, alternative delivery systems were demonstrated. In the process, Harris gained key insights into the problems that broadcasters will face when bringing HDTV to their viewers, as well as their possible solutions.

This paper describes the system that was implemented including origination and production at Camden Yards, distribution by satellite and fiber optic cable, broadcast from two stations in Washington, DC and demodulation, decoding and display at the National Press Club. The comparative advantages of the two distribution methods, other lessons learned and issues yet to be addressed will also be discussed.

INTRODUCTION

The atmosphere was perfect - hotdogs, pretzels, cold soft drinks, pennants - and a super wide screen TV. Harris Corporation was committed to helping develop enthusiasm for HDTV among industry leaders. The vehicle was the first broadcast of a major league sporting event in the HDTV format. By sponsoring this baseball game, it would be necessary to assemble a complete system, thus proving that the technology is available to do so. To assure that the program would be available to the viewers, two methods of delivering the signal were used, including dual transmitter sites. This event not only served to develop increased awareness of the potential of HDTV, but in the process the engineers involved gained new insight into the problems broadcasters will face as they proceed with implementation of the system.

Oh yes, the Orioles beat the Indians 7 to 2.

SYSTEM DESCRIPTION

The telecast was originated at Camden Yards in Baltimore, MD. As shown in Figure 1, a production truck was

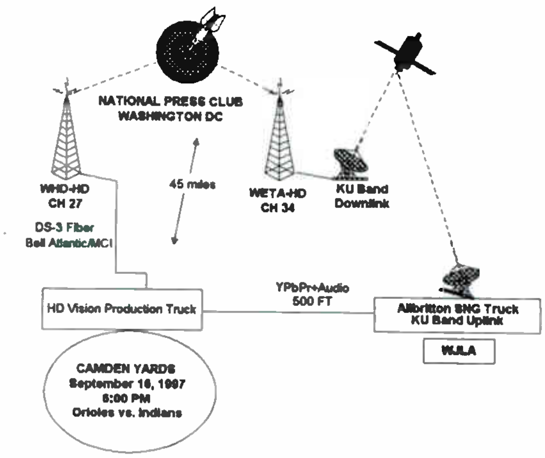

Figure 1
Broadcast System

located there as well as a Ku band SNG uplink truck. From the production truck the signal was fed to WHD-TV in Washington, DC via a fiber optic link. The fiber signal was compressed to fit the bandwidth of the DS-3 link. The signal was also beamed to a geostationary satellite and downlinked to WETA-HD also in Arlington, VA. The two stations broadcast the signal on Channels 27 and 34, respectively. The signal could be received on either of the channels at the National Press Club where it was projected on a 16 foot by 9 foot screen. Even though the distance from the point of origination to the viewing site was only 45 miles, all the elements required for program distribution and broadcast were demonstrated.

Camden Yards

A total of six Sony high definition TV cameras were installed inside the park at Camden Yards. These signals were fed back to the production truck (Figure 2). For the fiber link to WHD-TV, the RGB and audio was encoded

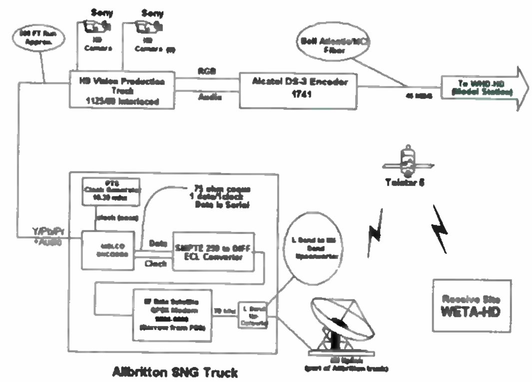

Figure 2
Block Diagram for Camden Yards Operation

to a 45Mb/s digital signal using an Alcatel DS-3 1741 encoder. For the satellite link, Y P_b P_r component video plus audio was sent to the SNG truck to be encoded to 19.3Mb/s serial data plus clock using a Mitsubishi MH1000E encoder. The transport stream output was converted to differential ECL to interface with an EF Data SDM-9000 QPSK modem. The 70 MHz output of the modem was upconverted, first to L band, then to Ku band for up linking to the General Electric K-2 communications satellite and down to WETA-HD.

WHD-TV

At WHD-TV, the DS-3 input was decoded to RGB video plus audio with an Alcatel decoder as shown in Figure 3. The video was encoded to ATSC format with a Grand Alliance encoder. The video and Dolby audio was

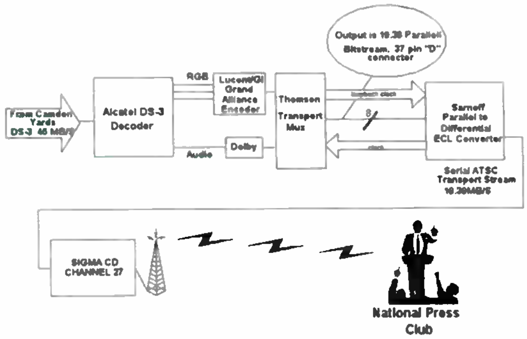

Figure 3
Block Diagram of
WHD-TV Equipment

multiplexed to the transport stream with a Thomson transport multiplexer. The parallel output of the mux was converted to a differential ECL serial transport stream which connected to the input of the Harris CD-1 DTV exciter and Sigma CD transmitter. The signal was routed to the antenna through a channel combiner. This permitted another DTV transmitter to operate on Channel 30 simultaneously with the Harris transmitter on Channel 27.

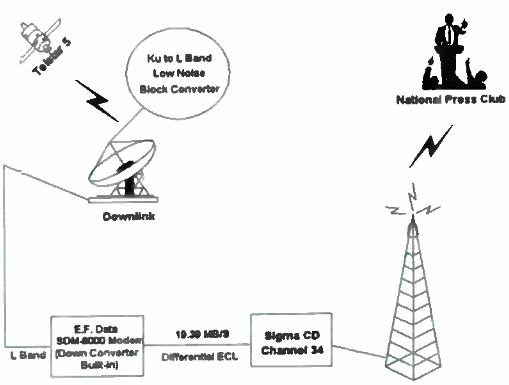

Figure 4
Block Diagram of
WETA-HD Equipment

WETA-HD

At WETA-HD, the received signal from the KU band antenna was downconverted to L band for demodulation and decoding in the E.F. Data SDM-9000 modem as shown in Figure 4. The resulting differential ECL transport stream was fed to the Harris CD-1 DTV exciter and Sigma CD transmitter for broadcast.

National Press Club

The receive site (Figure 5) was located at the National Press Club. An outside antenna was used. A bandpass filter was available in the event there was

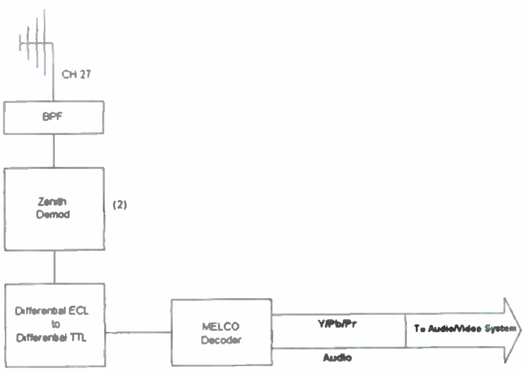

Figure 5
Block diagram of
National Press Club

interference from nearby signals. A Zenith demodulator provided the transport stream on differential ECL output. This was converted to differential TTL to interface with the Mitsubishi MH1000D decoder. The component video plus audio output of the decoder provided the signal for the projection monitor.

LESSONS LEARNED

Aside from making a bit of history, this project was a useful engineering learning experience. For the WHD-TV signal, it was necessary to follow the encoding/decoding for the DS-3 link with ATSC encoding/decoding. The fact that these processes could concatenate successfully without noticeable loss of quality is important. This approach requires more distribution bandwidth but has the advantage of making audio/video break away easier. The ability to insert commercials, promotional material, and screen crawls while in the analog component format is advantageous. On the other hand, distribution of ATSC compressed component video requires less bandwidth. However, with currently available technology, insertion of local content is more difficult making this approach most useful for pass-through operation. In this case, the ATSC compression was done just ahead of the satellite up link. Other points of interest include the lack of significant interference from a lower adjacent NTSC on channel 26 into the DTV on channel 27. In fact, the DTV picture quality was excellent. In addition, there was a lack of noticeable interference to a second lower adjacent NTSC on channel 32 from the DTV on channel 34. Finally, this event successfully combined a pair of DTV signals into a common antenna and transmission line. These are site specific details, but nevertheless will be important considerations when implementing many systems.

A spectral plot of the local signal levels as measured at the base of the WHD tower is shown in Figure 6. It is evident from the distortion to many of the signals that a great deal of multipath was present at this location.

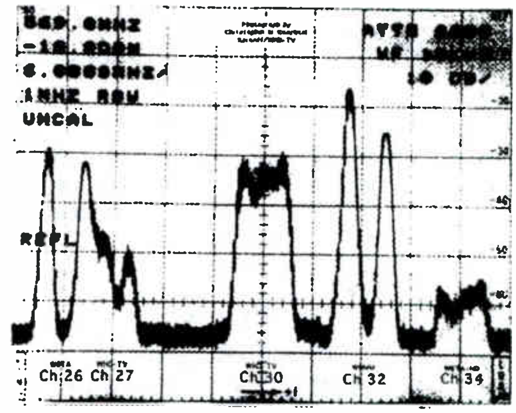

Figure 6
Received Off-Air Spectrum
Under WHD-TV Tower

However, this plot serves to give the reader a sense of the variety of RF signals present in the test.

OPEN ISSUES

A variety of issues surfaced during the course of this project. Some were anticipated, others were not. Some, if not all, of these issues are being worked as DTV development continues.

It is evident from the previous discussion that many interfaces between system components were not yet standardized. A pseudo SMPTE 259 to differential ECL converter was needed between the Mitsubishi encoder and the EF Data modem. A parallel to serial differential ECL converter was needed between the Thomson transport mux and the CD-1 exciter. A differential ECL to differential TTL converter was needed between the Zenith demod and the Mitsubishi decoder. Work is underway to resolve some, if not all of these interface issues. SMPTE recently approved a standard for the transport to transmission interface. It is important that interfaces be implemented as soon as possible as standards are developed. To this end, Harris has implemented the SMPTE transport to transmission interface in the latest version of the CD-1.

The need for additional monitoring was also highlighted. This would seem to be driven primarily by availability of equipment. For this event, additional monitoring at the input to the satellite truck, at the output of the satellite downlink, and at WETA-HD would have permitted measurement of the signal quality at those points in the link. This might have driven changes in system configuration during the broadcast.

One of these problems was that of lip sync. The source of this problem was difficult to assess during the broadcast due to lack of monitoring capability at key points in the links. This was thought to be a software bug (now corrected) in one of the decoders. However, much of the guess work could have been taken from this conclusion with more monitoring capability.

Another issue that arose was that of interference to cable head ends and master antennas. The channel 27 DTV signal is an upper adjacent to WETA's NTSC on channel 26. While the NTSC did not interfere with the DTV signal, the DTV signal interfered with the NTSC, especially in some cable and master antenna systems. For these situations it was necessary to insert a channel 27 notch filter just after the receive antenna. As shown in Figure 7 this filter had a flat passband in channel

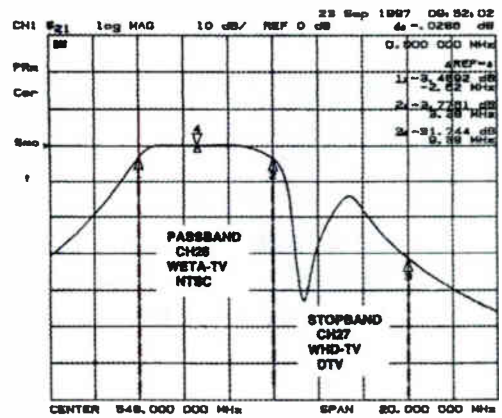

Figure 7
Cable Headend EMI Filter

26 and provided adequate attenuation to channel 27. In addition, adjustment of the alignment of master antennas was found to be effective to reduce the channel 27 signal. The interference of DTV with the NTSC was not a DTV problem, per se, but was just the result of adding a new RF signal where one was not previously present.

Some wireless microphones at RFK stadium had been setup to operate on previously unused channel 34 for a Washington Redskin's game. It was necessary to tune these to alternative unused channels when interference was noted.

Some, but not all, of the HDTV cameras exhibited noticeable lag and ghosting. This was believed to be a matter of proper setup rather than an inherent design deficiency in the cameras.

Some of the equipment was susceptible to RFI. There were also problems with ground loops. These problems were solved by proper grounding.

CONCLUSIONS

Televising the first HDTV broadcast of a major league game not only brought exciting attention to this technology, it was a valuable engineering exercise. It was necessary to design, assemble, test, and operate an end-to-end HDTV system complete with origination, distribution, transmission and reception of the video and audio signals. Although several problems arose, they were solved in a timely manner and the telecast was an unqualified success. Several issues were identified, many of which are now being addressed. Future broadcasts of HDTV can build on the experience of this event.

ACKNOWLEDGEMENTS

I wish to thank my colleagues at Harris Broadcast - Joe Seccia, Rick Proske, Bob Weirather, and Bob Plonka for reviewing and making helpful suggestions to this paper.

ADJACENT CHANNEL COMBINING IN DIGITAL TV

Graham Broad and Robin Blair
Radio Frequency Systems Ltd.
Melbourne, Australia

ABSTRACT

This paper outlines the design principles being applied to a new range of adjacent channel combiners now going into production and intended particularly for the combining of DTV and NTSC channels into a common antenna. The pass and stop band characteristics become very critical when the combining involves adjacent channels, and computer simulations have been necessary to derive designs which adequately preserve the transmitted signal quality. The paper shows however that the ideal of having no distortion cannot be realised in practice. Although the best designs give good results, the overall system performance can be further improved by applying active equalisation around both the transmitter and combiner. The paper proposes a method by which this can readily be implemented.

INTRODUCTION

As the change over from analog to digital TV gathers momentum, the operators of existing TV transmitters will be faced with the problem of running the new digital service in parallel with their NTSC service until the latter is phased out. For many, a viable solution will be to combine together the analog and digital transmissions and radiate them from the existing or an upgraded antenna system. This is relatively easy where they have been allocated a new digital channel in the same frequency band as the analog channel, except where the two are adjacent contiguous channels. The pass and stop band requirements of the channel combiner then become critical, and it is difficult to devise a design which adequately preserves the quality of the transmitted signals.

This paper explores the requirements of adjacent channel combiners and shows that the ideal cannot be realised in practice. Nevertheless, computer simulations of the effects of amplitude and delay distortions have been particularly effective in revealing designs that give much improved results. In some circumstances the better of the residual distortions arising from these designs might be acceptable on the transmitted signal, as they can be compensated by the receiver equalisers. However, if one adopts the philosophy that the transmitted signal should be substantially without distortion, then a further refinement is available involving active equalisation around the transmitter and combiner. This effectively integrates the transmitter and combiner and requires co-operation between the designers of each.

THE REALITIES OF COMBINING

The Ideal Combiner

The ideal combiner would take two signals into two input ports and combine them into an output port without loss to either. We can show, however, that for contiguous adjacent channels this is not possible, and there must be at least a 3dB loss for each at the common band edge. The formal proof is fairly straightfoward[1], and can be summarised into the following argument.

Consider a combiner with two input ports A and B, and an output port C. Ideally we would like zero loss from A and B through to C within their respective channels. Thus if a transmitter injected power P, corresponding to voltage V, into either port at an appropriate frequency, power P and voltage V would appear at the output port.
For the particular case of contiguous adjacent channels there is a frequency F which may be regarded as belonging to both channels - it is the top band edge of the lower channel and the bottom band edge of the upper channel.

Now suppose ports A and B are each fed by a transmitter at power P with a CW signal on frequency F. In principle, we could adjust the phase of one transmitter until the two signals at the output port come into phase. This would mean that the output voltage would become 2V and the output power 4P. This is impossible, of course, and the conclusion follows that we cannot have zero attenuation on both channels at their common adjoining frequency, or for that matter, at any frequency close by. One can derive a relationship for the minimum attenuation at any frequency through each input[1], but the simple obvious one arising from this discussion is that there will be at least 3dB attenuation at the common band edge.

It follows that all channel combiners must exhibit some type of filtering characteristic through the input channels, and the question becomes one of optimising that filter characteristic for adjacent channels.

Combiner Configuration

The above argument indicates that there are practical limits to combining and no "magic" circuit configuration is going to avoid those limits. Hence, a combiner configuration which attains those limits will be as good as any other which might be postulated. Hence, there is good reason to stay with one that has other attractive features and has been extensively developed in the analog world - that is the constant impedance dual filter configuration sometimes known as the "Lorentz Ring".

This well known configuration is illustrated in Fig. 1. We wish to make the point that in the narrow-band channel, which is the one of interest, the band-pass characteristic accurately reflects the parameters of the filters chosen by the designer and inserted in the ring. Hence, optimising the combiner resolves into optimising the character of those filters. This has recently been the subject of computer simulation[2], and the results applying to adjacent channel operation are outlined briefly below.

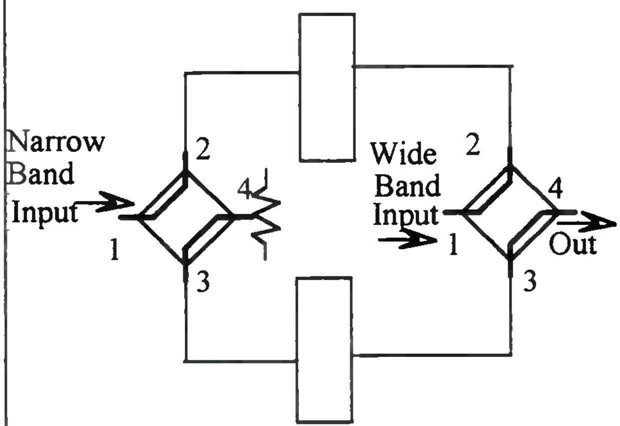

Fig.1 - Combiner Configuration

FILTERING DTV SIGNALS

Symbol Shape

The essence of any digital transmission system is the degree of faithfulness with which it reproduces the intended shape of the signalling symbol at the receiver output. The ideal pulse shape for 8-VSB DTV is shown in Fig.2a, and the manner in which preceding and succeeding pulses overlap is shown in Fig.2b. Hence, the important elements are that there be a high central peak at the time at which an individual pulse is sampled, surrounded by points of zero voltage level at the times when adjacent symbols are to be sampled. This latter, of course, is required to avoid intersymbol interference. Any deviation from zero at these sampling instances manifests itself as noise on the adjacent symbols. It follows that the level of this "noise", compared to the central peak voltage is a good measure of the symbol distortion.

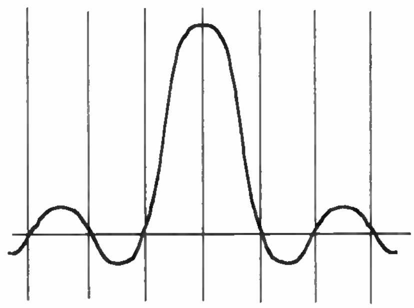

Fig.2a - Symbol Shape

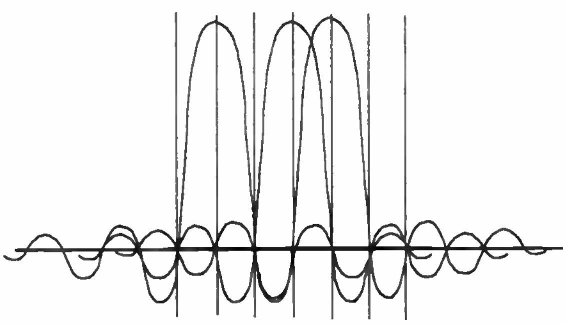

Fig.2b - Pulse Train Overlaps

In the computer simulations the value described above is derived from the calculated symbol shape by summing the square of the error voltages at the sampling times (which ideally should be times of zero voltage) and expressing this sum as a ratio, in dB, to the central peak value. This value defines an RMS error measure (and will be abbreviated to RMS Error in this paper). It can be shown that the value is directly related to the loss in signal to noise immunity in a system, and to the tap energy which it may impose onto the active equalisers in consumer receivers. The latter, combined with observations of tap energies imposed by propagation in typical service areas[3], suggests that a value of around 26dB would be a desirable target to achieve on a transmitter installation.

Amplitude and Phase Distortion

With the computer simulation, a very convenient way of determining what limiting parameters are likely to be practical is to take the type of filter that would most likely be used in practice, and assess the system performance as the bandwidth is reduced. Signal distortion, as measured by the RMS Error will arise from both the limiting of the channel bandwidth and the introduction of phase distortions. The latter can easily be removed by mathematical manipulation of the filter model. Thus we can assess separately the relative importance of the two effects.

To this end, the results for a six section Elliptic Function filter with 0.1dB passband ripple are shown in Table 1. The table shows the values of RMS Error which are realised for various 3dB bandwidths, first with no phase correction, and then with the phase made linear so that the resulting distortion is due to bandwidth limiting alone. The conclusion seems to be unequivocally that the phase distortion in practical filters is likely to be much more of a controlling factor in DTV applications than is the limitation of bandwidth.

Filter Bandwidth	RMS Sampling Error: Natural Phase	RMS Sampling Error: Phase Linearised
10MHz	-27.9dB	
9MHz	-25.1dB	
8MHz	-21.1dB	
7MHz	-16.3dB	
6MHz	-9.1dB	-35.3dB
5.7MHz	-6.0dB	-31.1dB
5.5MHz		-26.4dB
5.0MHz		-17.4dB

Table 1 - Sampling Error for Elliptic Function Filters

In practice, it is not possible to apply the necessary degree of phase correction to high power Elliptic Function filters, but these results led to the exploration of a similar class of filters with cross couplings between non-adjacent resonators. Some of the cross couplings can be designed for partial phase correction[4], whilst others realise transmission zeros to achieve pass-band rolloffs close to those of the Elliptic Function filters. The filters are broadly referred to as Cross Coupled Filters, and the principle is shown in Fig.3.

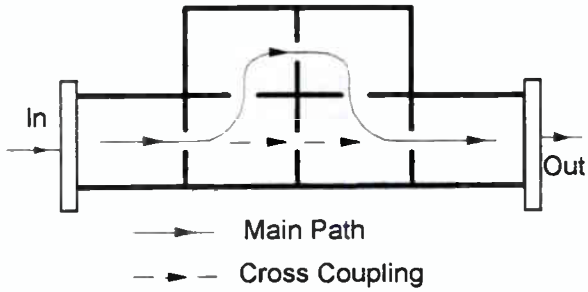

Fig. 3 Filter with Cross Coupling

The improvement is substantial. At 6MHz 3dB bandwidth a six section cross-coupled filter produces an RMS Error of -16.5dB, as against -9dB for the comparable Elliptic Function filter. The RMS Error of -26dB or better is realised for bandwidths down to 7.0MHz, which compares to 9.5MHz for the Elliptic Function filters.

The performance of these filters is certainly adequate for the combining of non-adjacent channels. Let us now look at what can be achieved in adjacent channel applications.

Adjacent Channel Application

The first question a potential user is likely to ask is "What will be the effect on my existing NTSC service?" The answer will largely determine what bandwidth he will accept for his digital channel, inasmuch as wider bandwidths imply more attenuation to components in the NTSC channel. To this end, table 2 shows how the analog carriers are affected by filters of differing bandwidth in the combiner. These are 6 section Cross Coupled filters, with the NTSC signal into the broadband port of the combiner.

Filter 3dB Bandwidth	Attenuation in Analog Channel		
	Va	Vsb	Vc
6.0MHz	1.0dB	0.4dB	0.2dB
5.75MHz	0.5dB	0.3dB	0.2dB
5.5MHz	0.2dB	0.2dB	0.15dB

Va is the sound carrier in the lower adjacent channel

Vsb is the VSB edge in the upper adjacent channel

Vc is the vision carrier in the upper adjacent channel

Table 2 - Analog Channel Attenuation

In the digital channel the RMS Error values for each of the bandwidths listed above is as follows in Table 3.

Filter 3dB Bandwidth	RMS Error Magnitude	
	Uncorrected	Corrected
6.0MHz	-16.5dB	<-35dB
5.75MHz	-13.6dB	-33.5dB
5.5MHz	-10.7dB	-28.4dB

RMSE Uncorrected is the value with no phase correction

RMSE Corrected is with the residual phase corrected in the transmitter

Table 3 - RMS Error in DTV Channel

It is seen from these two tables that a filter bandwidth of 6.0MHz might be considered acceptable, particularly if the NTSC service occupies the upper channel. However, the benefits of phase precorrection can also be seen to be considerable. The group delay in these cross coupled filters is of the order of 35 to 100nS, so that static precorrection in the transmitter IF stages is certainly possible. However, there is a better way, taking advantage of the active equalisers that are an inherent part of the 8-VSB system.

Active Equalisation

The block diagram of the proposed method of equalisation is shown in Fig.4. The "mirror filter" in this diagram reproduces the characteristics of the filter in the combiner, or at least the phase characteristics, but in a low powered form. Hence, by virtue of the feedback loop, the adaptive equaliser adopts a characteristic that linearises the overall phase response of the combiner and transmitter.

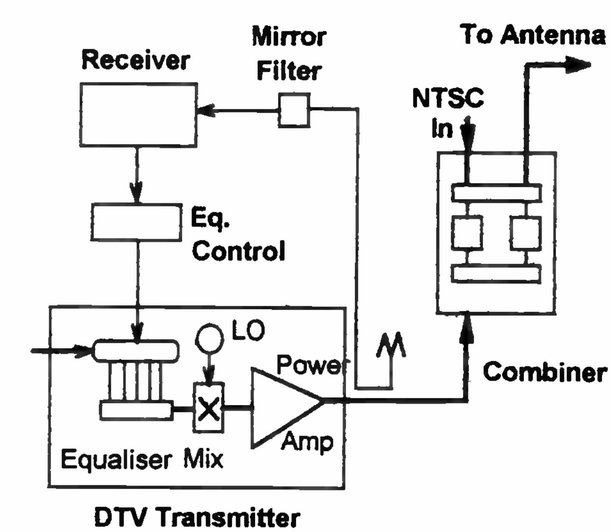

Fig.4 - Precorrection in Transmitter

The RMS Error Magnitudes listed previously indicate the extent to which the equaliser's capacity may be absorbed, and the advantages of the cross coupled filters are evident in this context. Of course, if the range of correction required is still too large for any particular design of equaliser, then a fixed precorrection circuit could be inserted either at the baseband input or in the IF amplifier of the transmitter.

PRACTICAL REALISATION

At UHF the focus is on waveguide techniques because of the low losses and high power handling that this makes available. Waveguide lends itself to a form of construction known as the dual-mode in-line structure[5] illustrated in Fig.5. In this scheme each resonator carries two orthogonal modes of polarisation and effectively contributes two electrical resonances. Coupling between successive resonators is achieved by strategic apertures in the end walls of the cavities, while coupling between the two modes within each cavity is brought about by small screwed probes in the cavity walls as illustrated in the figure. The former coupling accounts for the cross-coupling between the non-adjacent resonances as required by the filter design.

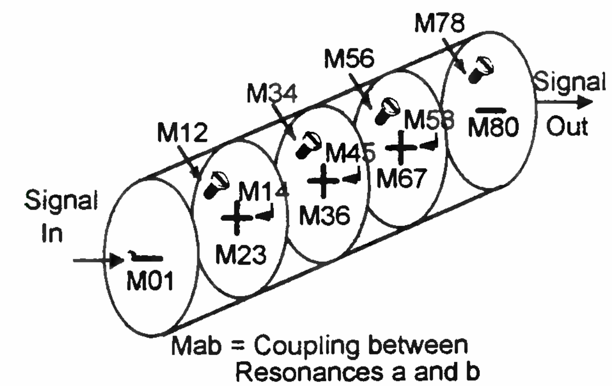

Fig. 5 Dual-Mode In-Line Construction

Additionally, if rectangular waveguide is used to interface to the filter, it is possible to place strategically located coupling apertures in the end walls of the dual-mode in-line structure such that only the forward travelling modes in the waveguides couple to the appropriate orthogonal modes in the end cavities. Hence the two directional couplers shown in the Lorentz Ring schematic in Fig.1 are simply realised, and the one physical filter emulates the two electrical filters in that diagram. The very simple physical form that these combiners take is then as illustrated in Fig.6.

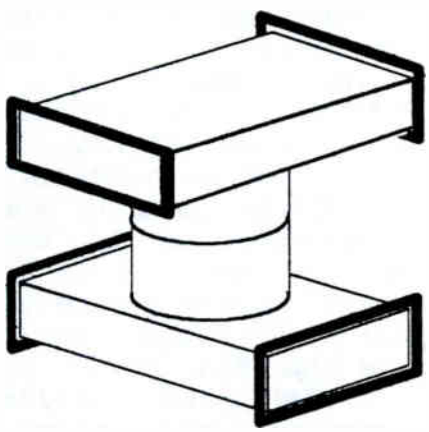

Fig.6 - UHF Combiner Structure

This construction is so effective that co-axial to waveguide adapters are often used in low power operations to preserve this convenient structure, rather than using the more conventional co-axial or strip line hybrids.

The Cross Coupled filters are also being realised as interdigital structures. These offer a more compact structure for the very low power UHF applications found in Europe, and in bands other than UHF, where the dual-mode in-line structures become physically too large. The interdigital filters do give a comparable performance, of course, but lack some of the convenience of their waveguide equivalents.

CONCLUSIONS

This paper has considered the requirements for adjacent channel combining of DTV and NTSC services. It has shown that, as an outcome of computer simulations, conventional filter types fall far short of the requirements. However, by introducing a new class of Cross Coupled filter, designers have realised a very significant improvement and can come close to achieving tolerable distortions on the radiated signals. The residual distortions have magnitudes that are readily within the scope of precorrection in the transmitter. By taking advantage of the active equalisers which are inherently part of the 8-VSB modulation systems, designers have available a very simple means of achieving near perfection in the radiated DTV signal. At the same time, the new filters lend themselves to a very compact and convenient method of construction. Hence, adjacent channel combining is now both feasible and convenient. In conjunction with modem broadband transmitting antennas, it promises to avoid the need for major external plant upgrades, and should be carefully considered wherever DTV and NTSC are to share the same transmission band.

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A TECHNICAL REVIEW OF TRANSMISSION LINE DESIGNS AND SPECIFICATIONS FOR TRANSMITTING TELEVISION SIGNALS

Kerry W. Cozad
Andrew Corporation
Orland Park, Illinois

ABSTRACT

With the introduction of digital television, many of the "rules of thumb" used for analog transmission are being reviewed for their applications to digital transmission formats. The dual impacts that transmission lines have on the RF plant: connection between the transmitter and antenna, and the major contributor to added wind load on the tower, make this component a significant candidate for review. It is the purpose of this paper to review the major design criteria for standard broadcast transmission lines and their correlation to published specifications from manufacturers. In this way, technical performance and applications for DTV can be compared on an "apples to apples" basis.

INTRODUCTION

Transmission lines are one of the main components in the RF transmission plant. They play a critical role in both the quality and reliability of the broadcast signal. Therefore, the proper choice of a transmission line type to be used can have a significant impact on the success of the station.

The choice of transmission line is typically decided based on the following criteria:

Frequency of Operation
Power Handling
Attenuation (or efficiency)
Characteristic Impedance
Tower Loading (size and weight)

For broadcasting, the impedance and size characteristics have been standardized to a fixed set of options: 50 or 75 ohm input/output impedances; 3-1/8", WR1500, 1750 circular waveguide, etc. for sizes. Some changes have been made recently to improve tower loading such as the introduction of seam welded outer conductors for rigid coaxial lines to reduce weight and a 7" rigid coaxial size to bridge the power gap between 6-1/8" and 8-3/16" lines. However, the primary characteristics involved with DTV transmission performance are operating frequency, power handling and attenuation. Associated with these is a new concern over group delay within a waveguide transmission line system.

These characteristics will be reviewed for coaxial lines first and then waveguide.

COAXIAL LINES

A coaxial transmission line consists of two concentric conductors, the inner conductor being supported within the outer conductor through the use of a dielectric material. The dielectric material may be continuous throughout the line or, as in the case of rigid coaxial lines, located at distinct points along the line in the shapes of "pegs" or "discs". Because of the geometry, all coaxial lines follow common guidelines in determining their electrical and thermal characteristics.

Frequency of Operation

For coaxial lines, the frequency of operation for a specific outer conductor size and characteristic impedance is limited by the highest usable frequency before undesirable modes of propagation occur. This is sometimes called the cut-off frequency, f_c. However, each mode of propagation has a unique cut-off frequency and care should be taken not to confuse the coaxial line f_c discussed here with cut-off frequencies for waveguides (which will be discussed later).

The coaxial cut-off frequency is important when power handling versus frequency is being reviewed. A larger size of coaxial line will handle greater power levels, however, its frequencies of operation will be at lower frequency ranges. Since higher frequencies have higher attenuations (see next section), it is desirable to run more power to overcome the losses. This results in a situation where a choice between (1) lower ERPs (operating at lower power levels), (2) using waveguide (typically higher windloads) or (3) risking degraded performance due to higher order modes being present must be made.

The higher mode coaxial cut-off frequency is calculated using the following equation:

$$(1)\quad f_c\ (MHz) = \frac{7520}{\sqrt{\epsilon'} * (D+d)}$$

where

ϵ' = dielectric constant or relative permittivity of dielectric to air
D = inside electrical diameter of the outer conductor, in.
d = outside electrical diameter of the inner conductor, in

Differences in the maximum operating frequency of specific line sizes is sometimes evident when reviewing various manufacturers' specifications. This is typically a result of a different safety factor used when deciding on the specification. A 5-10% reduction in the calculated cut-off frequency is a normal safety factor and will account for manufacturing tolerances and the effects of connections and elbows. For complicated installations (where extensive use of elbows or transitions occurs), additional margin may be necessary to prevent the generation of higher order modes.

Attenuation

The attenuation of a coaxial line is normally expressed in terms of loss per unit length or db/100' (db/100 meters). The attenuation is due to conductor and dielectric losses. As a simple equation this can be expressed as:

$$(2) \quad \alpha = A*\sqrt{f}+B*f$$

where

α = attenuation constant, db/100'
A = conductor losses
B = dielectric losses
f = frequency, MHZ

For rigid coaxial lines, dielectric losses have been considered negligible and with the use of copper conductors, Equation 2 is usually shown as:

$$(3) \quad \alpha = \frac{0.433}{Z_o}*\left(\frac{1}{D}+\frac{1}{d}\right)*\sqrt{f}$$

where

Z_o = characteristic impedance
D = inside electrical diameter of outer conductor, in
d = outside electrical diameter of inner conductor, in

It should be noted that designs using additional dielectric material for better structural support between the inner and outer conductors (i.e. additional pegs or cylindrical discs), should include additional losses of between 1% and 4% in the calculations. Also, the above equation assumes a conductivity rating of the copper conductors of 99% or greater. In practice, the surface conditions rarely approach that of newly produced copper tubes due to oxidation and handling of the materials. To account for this effect on the conductor, a conductivity rating of 95% should be used, resulting in an additional 1-2% increase in attenuation.

When comparing manufacturer's data, these issues should be reviewed for consistency between specifications. Also note that the conductivity varies with temperature. A 20°C ambient is standard for conductivity ratings.

If the temperature of the conductor is different from the standard rating, the conductivity must be adjusted. One area that is not normally taken into account for television broadcast is the actual temperature of the inner conductor during operation. The increased temperature due to power loss in the line results in higher attenuation values. The adjustment factor typically used for attenuation (M_α) is given by:

$$(4) \quad M_\alpha = \sqrt{1+\sigma_o(T_t-T_o)}$$

where:

T_t = inner conductor temperature °C
T_o = inner conductor temperature at standard rating, °C
σ_o = temperature coefficient of resistance at standard rating

For a standard temperature rating of 20°C, σ_o = 0.00393/°C. Then:

$$(5)\quad M_\alpha = \sqrt{1+0.00393(T_t-20)}$$

Therefore, if the rated average power of the line allows an inner conductor temperature of 100°C, during operation at maximum rated power the attenuation will increase by a factor of 1.146. By performing this calculation the author is not indicating a desire to change decades of standard procedure in determining the system power requirements for TV stations. The intention is to review design parameters that are not typically used in system analysis but could be used to better analyze what some may consider marginal configurations for DTV.

Once the attenuation constant has been determined, the efficiency of the system can be calculated. The total attenuation (α_{total}, db) is found by multiplying the attenuation constant by the total length. This is then converted to efficiency:

$$(6)\quad Eff.\ \% = 10^{-\left(\frac{\alpha_{total}}{10}\right)} * 100$$

Power Handling

The power handling capabilities of coaxial lines are based primarily on two factors: the maximum peak power (or maximum voltage gradient that can safely be present) and the maximum average power, which is determined by the allowable temperature rise of the inner conductor.

Peak Power

The maximum electric field strength between two coaxial conductors can be calculated from:

$$(7)\quad E_{max} = \frac{0.278}{d} * \sqrt{\frac{P}{\ln\left(\frac{D}{d}\right)}}$$

where

E_{max} = maximum electrical field strength, volts/in
P = power level of signal, watts

Because voltage breakdown levels are extremely sensitive to effects such as internal surface conditions and environmental factors, the theoretical value should not be used in practice. It has become standard procedure to use 35% of the theoretical value in determining the production test voltage and ultimately the rated peak power value. The DC test voltage is derived from the following equation which includes the derating factor:

$$(8)\quad E_p = 3.17*10^4 (d*\delta) * \left[\log\left(\frac{D}{d}\right)\right] * \left(1+\frac{0.273}{\sqrt{d*\delta}}\right)$$

where

E_p = production test voltage
δ = air density factor =

3.92B/T
B = absolute pressure, cm of mercury
T = temperature, °K

(δ=1 for B=76cm and T=23°C=296°K)

The production test voltage must now be converted to the RF RMS voltage, E_{rf}:

$$(9)\quad E_{rf} = 0.7 * E_p * \frac{1}{\sqrt{2} * SF}$$

where

E_{rf} = maximum RF RMS operating voltage with no derating for VSWR or modulation, but includes a safety factor, SF.
$1/\sqrt{2}$ = RMS factor
0.7 = DC to RF factor
SF = safety factor for voltage (typically 1.4 for coaxial cables and 2 for rigid coax)

The peak power rating, P_{pk}, can now be calculated:

$$(10)\quad P_{pk} = \frac{(E_{rf})^2}{Z_o} \text{ watts}$$

then

$$P_{pk} = \frac{\left(\frac{E_p * 0.707 * 0.7}{SF}\right)^2}{Z_o} \text{ watts}$$

Once the peak power rating has been determined, it is necessary to derate that value for the effects of modulation and VSWR. These deratings are calculated as follows:

AM:

$$(11)\quad P_{max} < \frac{P_{pk}}{(1+M)^2 * VSWR}$$

FM and DTV:

$$(12)\quad P_{max} < \frac{P_{pk}}{VSWR}$$

Analog TV:

$$(13)\quad P_{max} < \frac{P_{pk}}{(1+AU+2\sqrt{AU}) * VSWR}$$

where

P_{max} = derated maximum peak power
M = amplitude modulation index (100% = 1)
AU = aural to visual ratio (20% aural: AU = 0.2)

Note: For DTV, P_{max} should be compared to the +6db peak power levels for 8VSB and not the average power of the signal.

For most installations, the peak power ratings will not be a significant factor as they are typically much higher than a single transmitter system can generate. The primary concern will be for multiple channel installations where two or more TV signals are combined into the same transmission line. If the peak voltages from two or more signals of equal power add together in phase, the equivalent peak power rises as the square of the number of carriers. In this situation, voltage levels can become the primary concern in specifying the transmission line type.

Average Power

The average power rating is determined by the amount of heat created due to line losses. The amount of heat, or temperature rise, is primarily limited by the safe, long term performance of the dielectric material used to support the inner conductor. Since the temperature rise on the inner conductor is greater than the outer conductor, the maximum allowable temperature is normally specified based on inner conductor temperature at the rated power level.

Typical industry conditions have been to allow the inner conductor to reach a temperature of 100°C with an ambient temperature of 40°C. This means the inner conductor temperature is allowed to rise 60°C above the ambient. The average power rating can then be calculated using the following equation:

$$(14)\quad P_{avg} = \frac{16{,}380 * \sigma * D}{M_\alpha * \alpha}\ \text{watts}$$

where

P_{avg} = average power rating for 60°C rise of inner conductor temperature
D_{OD} = outer conductor outside diameter, in.
σ = heat transfer coefficient of outer conductor, watts/in^2
M_α = correction factor for attenuation (relative to 20°C
α = attenuation constant, db/100' at 20°C

Standard heat transfer coefficients are listed below in Table 1 for rigid coaxial line types.

Table 1

Line Size	Z_o	Heat Transfer Coeff. σ
7/8"	50	0.1280
1-5/8"	50	0.1200
3-1/8"	50	0.1070
4-1/16"	50	0.1035
6-1/8"	50	0.0970
6-1/8"	75	0.0770
7-3/16"	75	0.0760
8-3/16"	75	0.0740
9-3/16"	50	0.0900
9-3/16"	75	0.0660

The average power rating is based on the temperature rise on the inner conductor and this in turn affects the long term performance of the dielectric material. Therefore, operation at higher temperatures will result in a reduction in the life expectancy and reliability of the line relative to the lower temperature performance.

Field experience has shown that, barring improper installation or damage, the typical failure mode of coaxial lines is damage to the connection points as a result of excessive heating over time. Based on this observation, long term operation of coaxial lines at elevated temperatures is not recommended.

Velocity of Propagation

A final performance characteristic to review for coaxial lines is the velocity of propagation or phase velocity (V_p). It is expressed as a fraction of the speed of light in a vacuum and is inversely proportional to the effective dielectric constant of the insulating medium:

$$(15) \quad V_p = \frac{c}{\sqrt{\epsilon'}}$$

where

c = speed of light

As can be seen from this equation, for coaxial lines that are effectively homogeneous throughout their structure, the phase velocity is constant for all frequencies. Therefore, group delay is not an issue when reviewing performance for digital signal transmission.

WAVEGUIDE

When using waveguide as the broadcast transmission line, peak power and average power ratings will be much greater than needed for even combined channel operations. Since broadcast waveguide types do not require the use of a center conductor, voltage breakdown and average power levels are controlled primarily by the quality of the installation and any waveguide to coaxial line transitions that may be present. Therefore, these characteristics will not be discussed in this paper.

Attenuation

Attenuation in a waveguide structure is dependent on its shape, the conductor material and frequency, much like it is in coaxial lines. However, because there are various types of waveguide offered to broadcasters: rectangular, circular, doubly truncated, etc., the discussion is not quite so straightforward. One characteristic that should not be overlooked, however, is that the attenuation is inversely proportional to frequency. This means that for a specific size, the attenuation constant becomes smaller as the frequency of operation is increased.

Both waveguide and coaxial lines exhibit very little change in attenuation values across a 6 or 8 MHZ channel. The variation is typically less than 0.05 db for well designed systems and can be considered negligible to the overall performance for both analog and digital signal transmission.

Frequency of Operation

Unlike coaxial lines, waveguides have both lower and upper cut-off frequencies. these define a band of frequencies in which the performance of the waveguide is acceptable for broadcast use. The upper cut-off frequency is based on the same desire as coaxial line in preventing the propagation of unwanted modes. The lower frequency cut-off is the frequency at which true wave propagation begins. Therefore, below this frequency there is no usable propagation of the signal. Again, a 5-10% safety margin is desired to account for manufacturing tolerances and components in specifying the actual frequencies of operation.

Velocity of Propagation

From the previous discussions on attenuation and operating frequencies, it may be guessed that the propagation of signals in waveguide is somewhat different than that for coaxial lines. Due to the lower cut-off frequency, F_{c1}, the velocity of propagation in waveguide is dependent on the frequency of operation. For waveguide:

$$(16) \quad V_p = c*\sqrt{1-(\frac{F_{c1}}{f})^2}$$

As an example of the effect of the foregoing, assume a 1000' long run of 15" diameter circular waveguide with a lower cut-off frequency of 461.13 MHZ. At Channel 44 (650-656 MHZ), the time difference of arrival from the transmitter to the antenna between the upper and lower channel edge is:

(17)

$$V_{p1} = 0.9835*\sqrt{1-(\frac{461.13}{650})^2} = 0.6931\ ft/nsec$$

(18)

$$V_{p2} = 0.9835*\sqrt{1-(\frac{461.13}{656})^2} = 0.6995\ ft/nsec$$

(19)

$$1000/0.6931\ ft/nsec = 1442.8$$
$$1000/0.6995\ ft/nsec = 1429.6$$
$$T_1 - T_2 = 13.2\ nsec$$

For the same waveguide used at Channel 30 (566-572 MHZ), the time difference will be:

$$V_{p1} = 0.5703\ \text{ft/nsec}$$
$$V_{p2} = 0.5819\ \text{ft/nsec}$$

$$T_1 = 1753.5\ \text{nsec}$$
$$T_2 = 1718.5\ \text{nsec}$$

$$T_1 - T_2 = 35\ \text{nsec}$$

Based on the overall system requirements, these time delays may be negligible. If not, pre-correction can be accomplished since the delay can be readily calculated.

SUMMARY

The basic formulae for determining the primary operating characteristics of both coaxial transmission lines and waveguides for broadcast have been presented. For coaxial lines, the broadcast engineer can now perform basic calculations to provide a comparison to manufacturer's data. This should provide for a better understanding of safety factors and risks when analyzing new systems for analog and digital transmissions.

Waveguide was reviewed primarily to provide more insight into its performance that will most effect its use for DTV. Based on simple calculations, no significant impact on DTV transmission should be present in a well designed system.

Mechanical considerations were not a part of this paper, however, tower loading is a significant issue and should be an integral part of the decision process.

Acknowledgments

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Time Domain Characterization of Antennas for DTV Application

Ali R. Mahnad Ph.D.E.E.
Micro-Tek Engineering
Carmichael, CA

Abstract

*Digital format and wideband modulation are the two prominent characteristics of ATSC- DTV. Together they pose challenging questions on the performance of antennas in DTV broadcast environment. Since antenna is a significant component of any broadcast system, an in depth understanding of how an antenna **"pattern" is build up in time domain** and how this process affects **antenna gain** are essential in determining their impact on the overall system performance. Characterization of antenna pattern solely by the use of spectral techniques is incomplete and often misleading. Group delay issues inherently point to time dependence responses but do not lend it to clear understanding of sequential data input to antennas.*
*This paper presents a comparison of **time domain radiation properties** of Variety of modular (such as Panel or Dipole) type and traveling wave type(such as slot) antennas, when the input is **time dependant data pulses.** We discuss radiation pattern build-up as an out put port of a two-port device to determine its true frequency response as well as its bandpass limits.*

Introduction

Digital Television (ATSC-DTV) imposes challenging requirements on a typical RF transmission system. In an ideal situation every component of the system should provide "distortion-less transmission" over a 6 MHz bandwidth. Transmitters, combiners, filters, transmission lines, waveguides as well as antennas are all components of a typical transmission system.

Each and every one of these components, except the antenna, has a **two-port device** representation. An **input port**, an **output port** and a **Transfer or a Response** function representing the frequency response of the two-port. These devices are all Linear[1] Time Invariant[2] devices. However, for these devices to act as "**distortion-less transmission**" channels, their **output** (within their passband, of course) should be replica of the input with the exception of a time delay and a possible change in magnitude.

In a typical analysis of a two-port device, the "distortion-less transmission" requirement is verified by examining the flatness of the device amplitude response and linearity of the phase response over the 6MHz required bandwidth. The maximum variation in group delay

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[1] A **Linear Device** is a device that satisfies the principle of superposition. This means that the device output due to the sum of several signals is the same whether the signals are summed first and then sent through the device or each is sent through the device and then summed.

[2] A linear system is **time invariant** if a time delay in its input results only in an equal time delay in its output.

response is typically used as a measure of the degree of phase linearity.

The ability to perform two- port measurements directly on the system is limited to two- port devices such as filters, combiners and transmission lines, etc.. Antenna, on the other hand, is not a two-port device (at least in a general sense). In general one can not measure the response of an antenna with same ease as one does the same for a two-port device.

It is possible; however, to develop a theoretical model that predicts the response of an antenna as a two-port device. The value of such model is two folds: 1) it provides information on the true **bandwidth**[3] of an antenna from a systems point of view. 2) It provides insight into the **design vs. bandwidth** inter-dependencies in different antenna types.

Antenna Bandwidth, Why all the fuss?

In the world of NTSC, the signal bandwidth has always been considerably narrower than the antenna Bandwidth[4](whether it was recognized at all or not, is beyond the scope of this paper). Under this condition the signal frequency may be considered continuous wave (CW). It is, usually, on the basis of a CW signal that an antenna's **radiation patterns** and **gain** are defined. When the signal bandwidth becomes substantial, as is the case in DTV signals, an apparent radiation pattern and gain result which are dependent upon the transmitted bandwidth. This should not be dismissed as a system problem, since the extent of this problem depends heavily upon the antenna design.

As it will be demonstrated, both antenna type and design affect the transmitted bandwidth tremendously.

From an antenna designer stand point, it is important to understand the nature of the input signal.
So it is appropriate here to review some basic attributes of digital signal transmission in general and DTV signals in particular.

In a digital system, a string of pulses (sometimes at different amplitude levels) are input into the device. The output of the device is sampled periodically at the center of each pulse. If the pulse-width is (T) seconds the rate at witch output is sampled is (1/T) per second. For an Ideal pulse this sampling rate is equal to 2B where B is the

[3] It should be pointed out that the Bandwidth of an antenna as defined by its input VSWR has limited value. Broadband input VSWR provides ideal load condition for the transmission line terminated by the antenna. It, however, says nothing about what happen to the signal when it passes through the antenna. Such information is dependent on the knowledge of the actual antenna bandwidth as defined by its response as a two-port device.

[4] All references made to antenna bandwidth in this paper are meant to be the transmission bandwidth as determined by the response and not the VSWR bandwidth, unless otherwise noted.

Bandwidth of the pulse. For a pulse with a raised cosine amplitude spectrum with B Hertz bandwidth, the sampling rate reduces to B with a pulse width of T=1/B seconds. In general, variety of pulse shapes can be used to provide sampling rates between B and 2B Hertz depending on the pulse shape.

DTV presents a signal to the antenna in the form of 8-VSB. A discussion on the time domain characteristics of this signal is beyond the scope of the present paper. It, however, suffice to say that it should have the characteristics of a string of pulse modulated signal. The 8VSB is an eight signal level VSB Amplitude Modulation scheme. Each signal level is referred to as a **symbol.** The DTV symbols are generated at the rate of 10.7 Mega-Symbols Per Second (MSPS). Symbols are approximately 1/10.7= .0934 microsecond long. At near Nyquist sampling rate the required bandwidth for transmission channel is 1/2T=1/(2*0.0934) = 5.35 MHz.

Antenna as a band limited Transmission Channel

Antenna is the last component of a transmission system before the signal hits the space and the terrain. In order to separate the degradation due to the terrain and broadcast environment from that introduced by the antenna, It is crucial to know exactly what happens to the signal as it passes through the antenna. Furthermore, to satisfy the transmission requirements of DTV, it is necessary to determine if the antenna as a two-port device has sufficient bandwidth to pass the signal without distortion.

To avoid mathematical complexities that will be beyond the scope of this paper, we will attempt to present the idea using graphical means. To this end we consider an ideal rectangular pulse with duration T as an input to different antenna types. We, then, study the output as a function of time. In this treatment we assume a wide-band receiver (and receiving antenna) to simplify the analysis. We further assume that there is no mutual coupling, no feeder dispersion, no multiple reflection in the feed system, and that elements of the array (whether dipole or slot) have frequency independent admittance.

We consider two major antenna types: The Branch fed Type, and Travelling Wave slot antennas.

Branch fed type antennas are assumed to be equal –path-length. By that we mean that all elements receive the signal from the feed system at the same instant. In contrast, Slots in traveling wave antennas receive the input signal not simultaneously, but sequentially.

We begin analysis by considering a branch fed antenna type consisting of a simple N-element array. Fig (1) shows an example of such an antenna. All elements (in this case 10) are fed in phase with equal path length. Fig (2) Shows CW pattern of this 10 element full-wave spaced antenna array. This pattern represents the radiation pattern of the antenna under steady state condition.

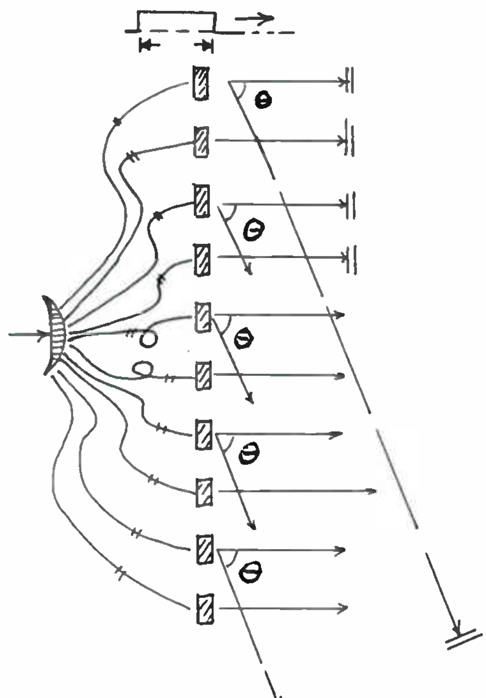


Fig. 1

In time domain it means a single frequency with constant amplitude over a long enough time so the pattern is invariant as the time passes. This is not, However, a true representation of antenna pattern under pulsed input environment[5] or Wide-band input conditions. Suppose a pulse of duration (T) is applied at the input of the array. Since the feeder is dispersion-less, front end of the pulse arrives at all elements at the same time. In the broadside direction all pulses will be received simultaneously. However as the receive point moves to other angles the situation changes.

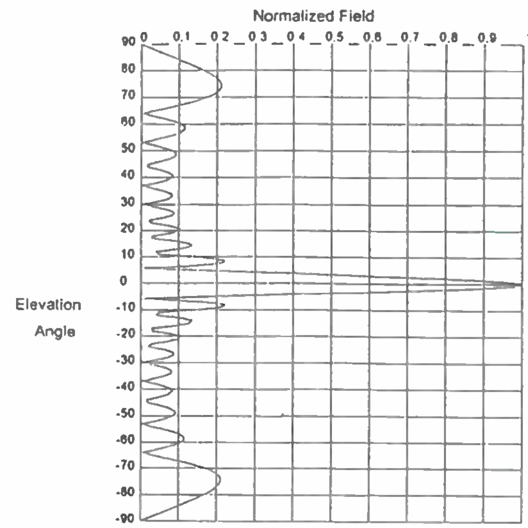


Fig. 2 Elevation Pattern

The signal from the far elements will reach later than the ones that are near. In fact the arrival time of the pulse front from the nth element is delayed by:

$$N * T_0 * \mathrm{Sin}(\theta)$$

Where (T_0) is the time it take for the pulse to travel between two adjacent elements ($T_0 = d / C$ where d is element spacing & C is speed of light).

At extreme angles this delay is maximum for the farthest element. What this indicates is that it takes at least ($N * T_0$) for this antenna to develop the pattern of Fig(2) . Any sampling of the field during this period will render erroneous results. After this minimum time, one can expect the field levels predicted by the CW pattern, provided that ($T >> N * T_0$). Similarly, when the tail end of the pulse leaves the array it takes

[5] D. Polk, " transient Behavior of aperture antennas," Proc. IRE, vol. 48, pp. 1281-1288, July1960.

the same length of time for the field to collapse everywhere.

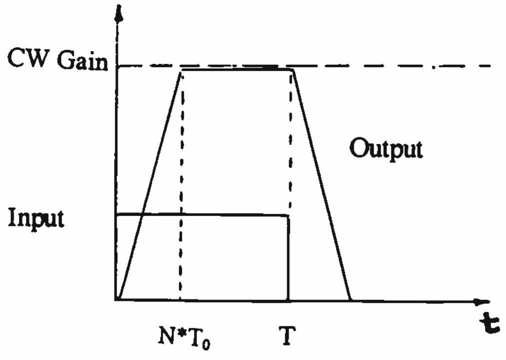


Fig. 3

Fig (3) shows the relationship between the input and out put pulses. Notice that the output pulse duration is: $(T + N * T_0)$.
In cases where (T) is quite large this widening of the pulse is insignificant and has no bearing on the transmission of signal through the antenna. This is the case in narrow-band input signals, where any amplitude change takes a long time compared to $(N * T_0)$. If, however, the input pulse-width (T) is of the order of $(N * T_0)$, the effect is more serious. This becomes more obvious when we consider the output pulse sampling rate:

$$\text{Sampling Rate} = 2B = 1 / (T + N*T_0)$$

This equation connects the maximum sampling rate to the aperture length of the antenna. It also shows the relationship between aperture length and the maximum bandwidth of the signal that may pass through the antenna without distortion.

To demonstrate the value of this formula, let's assume that:

$$T = 2 * (N * T_0).$$

Further we assume that the antenna is a typical UHF antenna with (N=30). The sampling rate for this antenna will be (assume $C=11.81*10^9$ inches / second, and d= 20 inches)

$$S.R. = 2 B = 1 / (3 * N * T_0) = 6.56$$ Mega symbols per second

Or a Nyquist Bandwidth of 3.28 Mhz. It is interesting to compare these results with the DTV requirement of 10.7 MSPS for a distortion-less transmission.

It is important to emphasize that the above formula simply provides limitations imposed by an antenna on the sampling rate and says nothing about how to design proper antenna.

Next, let us consider a travelling wave type antenna. In a traveling-wave type antenna signal is fed from one end of the array. As it travels down a “leaky” transmission line, power is coupled from the wave to radiating elements along the way. Because of this sequential feeding of array elements, there is a delay between radiation from consecutive elements of the array. This delay as we will see has additional effect on the pulse response of the array.

In order to determine the response of this type of antenna; We consider an N-element traveling wave slot antenna.

Fig (4) shows a typical 10-element end fed traveling-wave antenna.

We assume that the input is an Ideal rectangular pulse.

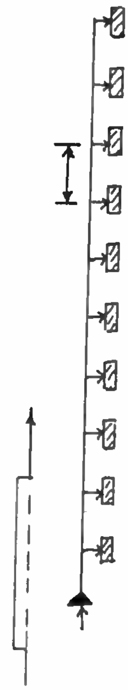


Fig. 4

We further assume that the wave traveling inside the transmission line does not experience any multiple reflection or any frequency dispersion and that the wave travels down the line distortion-free.

As the pulse travels down the transmission line power is coupled to slots in a sequential manner. If ($T_g = d * V_g$) is the time it takes for the pulse front to travel between the adjacent slots and there are N slots in the array it takes ($N * T_g$) for the signal to fill the antenna aperture. Once the aperture is filled, it takes $N*T_0$ for the pattern to develop. If the transmission line is simply a coaxial line: $V_g = C$. & $T_g = T_0$.
For such an antenna it takes ($T_g + T_0 = 2* T_0$) to fully develop the pattern.

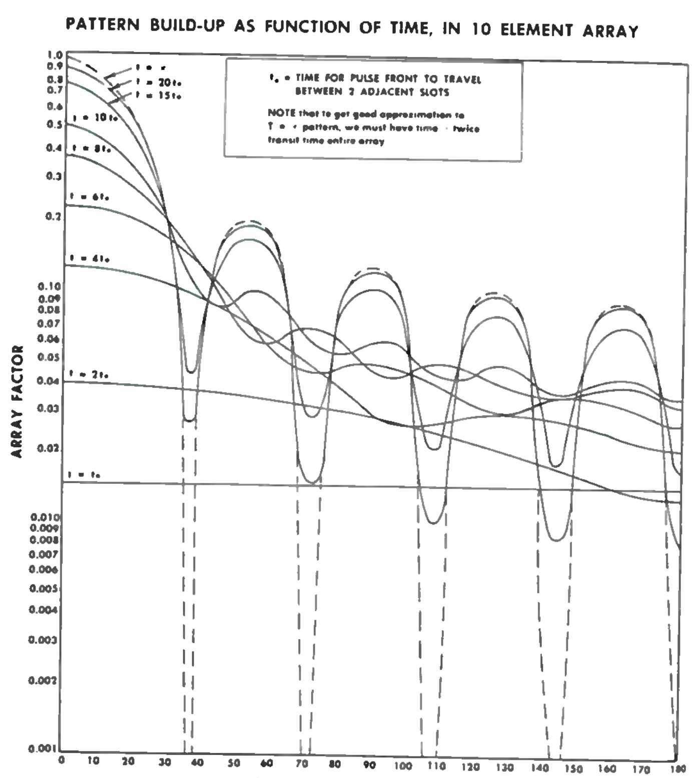


Fig. 5

Fig. (5) [6] shows Pattern Build up as a function of time in a 10-element half-wave-spaced traveling wave slot antenna. Note that it takes $20 * T_0$ for the full development of the pattern and the gain to reach the CW gain level.

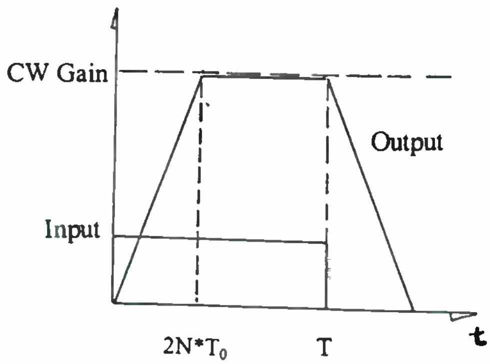


Fig. 6

Fig (6) compares the input and output pulses for an N-element array. The output pulse in this case is: $T + 2 * N * T_0$ wide. The sampling rate for this case is :

$$S.R.= 2 * B = 1/(T + 2 * N * T_0)$$

Again this Formula provides a relationship between antenna aperture and the bandwidth limitation imposed by the traveling wave antenna.

As we stated before, in cases where $T >> N * T_0$, pulse broadening due to the antenna is not significant and may be ignored. In other cases where T is of the same order as $N * T_0$, This broadening of the pulse is significant and may not be overlooked.

[6] Courtesy of L.L. Bailin, Hughes Aircraft Co.

In a typical case we may reasonably assume that $T = 3 * N * T_0$. This provides $(N * T_0)$ of flat amplitude for sampling. Then:

$$S. R. = 2 * B = 1 /(5 * N * T_0)$$

Again, in order to get a feel of numbers in a typical situation, let us assume that the antenna under consideration is a 30-slot traveling wave array with uniform slot spacing of 20" (T_0 = 20/11.811 = 1.694 Nanosec.) and uniform aperture distribution. Then:

S. R. = 2 * B = 1 /(254) Giga Symbols/Sec.

S. R. = 3.93 Mega Symbols / Sec.

Or equivalently:

B= 1.97 Mega Hertz.

None of these figures are encouraging when compared with the symbol rate or bandwidth requirements of DTV. It should be pointed out that the typical antennas used as examples here are high gain version of broadcast antennas. As the aperture gain is reduced or aperture distribution is modified from uniform, higher sampling rate will become possible. The details of these design varieties as well their effects on sampling rate is beyond the scope of this paper and will be discussed in future papers.

Concluding Remarks

An examination of pattern development of antenna arrays as a function of time reveals band limited characteristic of antenna arrays. We used this phenomenon to define pulse response of two types of arrays. We developed formulas relating aperture length and sampling rate and/or bandwidth. Typical antennas were examined for typical sampling rates and bandwidths based on these findings were compared with those of DTV. This analysis indicates the importance of using proper aperture size as well as distribution in achieving proper performance of antenna for DTV broadcast.

SLOTTED CYLINDER ANTENNA DESIGN CONSIDERATIONS FOR DTV

Ernest H. Mayberry
Dielectric Communications
Raymond, ME

ABSTRACT

Various approaches to slotted cylinder antenna electrical design are available. Choices made at the design stage impact antenna performance parameters important for DTV use. The antenna output response performance, radiated signal amplitude and phase frequency response across the channel, is a major concern for digital TV. Most critical to antenna output response are the aperture illumination and the feed method: end or center feed. The effect of these design choices on the frequency response performance of the beam tilt and null fill are examined.

Specifically, the gain variation across the DTV channel versus depression angle for both end-fed and center-fed designs are evaluated. A comparison of calculated and measured results for a center-fed design is presented. Beam tilt variation across the DTV channel as a result of end feeding the antenna is considered. The impact of end-fed antenna output response variations on the selection of antenna gain is discussed. Also, the performance of adjacent channel antenna designs using end and center feeding is compared.

INTRODUCTION

U.S. broadcasters planning their digital transmission facilities must choose a new antenna for DTV. A variety of antenna designs are available to the broadcaster from the many antenna manufacturers located around the world. Complicating the selection issue for the broadcaster is the necessity of locating and operating the DTV antenna system simultaneously with their NTSC antenna system. The broadcasters' goal is to configure the DTV and NTSC antenna systems to provide good signal coverage while minimizing tower wind loading.

The vast majority of UHF antennas currently used in NTSC service are slotted cylinder designs. Performance characteristics that made slotted cylinder antennas the antenna of choice for NTSC UHF service are also desirable for DTV, i.e., excellent omnidirectional azimuth patterns, low wind loads, and smooth null fill. However, the digital TV transmission system will require more stringent performance with regards to output amplitude and phase frequency response. As a result, the antenna output response performance, which was given little consideration in NTSC service, is an important consideration for DTV.

WHY SLOTTED CYLINDER ANTENNAS?

UHF slotted cylinder antennas gained prominence in NTSC broadcasting due to their combination of low wind loading and superior omnidirectional performance. Their small diameter construction, most within the 8" to 14" outside diameter range, provides the minimum wind load reducing the cost of tower structures. Small physical diameter also translates into small electrical radius (R/λ) for the slot radiators, which results in excellent circularity of the azimuth pattern.

Figure 1 shows an overlay of two azimuth patterns. The smooth, nearly circular azimuth pattern, is typical for a slotted cylinder antenna with a circularity of ±0.5 dB. Compare it to the typical azimuth pattern of a panel antenna with a circularity of ±2.0 dB. When both are normalized to unity, the amplitude difference between the patterns at the minimum of the panel pattern reaches 3 dB. Considering approximately 1 mile of coverage loss per dB, this could mean 3 miles of service reduction in some directions for the panel.

Of course, if the patterns were normalized to the same RMS value, the difference would reduce, but would still amount to 2 dB. In the NTSC domain, normalization to the same RMS value is the rule, however, it is not currently clear the same applies to DTV. The FCC has assigned a directional reference ERP pattern for every digital station allotment that sets the maximum radiation at each azimuth heading. The FCC DTV rules suggest that the peaks of the patterns must stay below the reference ERP at those azimuth directions, which would effectively prohibit use of the RMS gain. Further FCC clarification on this issue is expected.

The other principal advantage of the slotted cylinder antenna relative to a panel, low wind load, is demonstrated in Figure 2. This comparison illustrates two approaches to providing an existing UHF broadcaster with DTV & NTSC service from the same tower top. The slotted cylinder stack is typically 2 to 3 times lower in wind area than the wide band UHF panel [1].

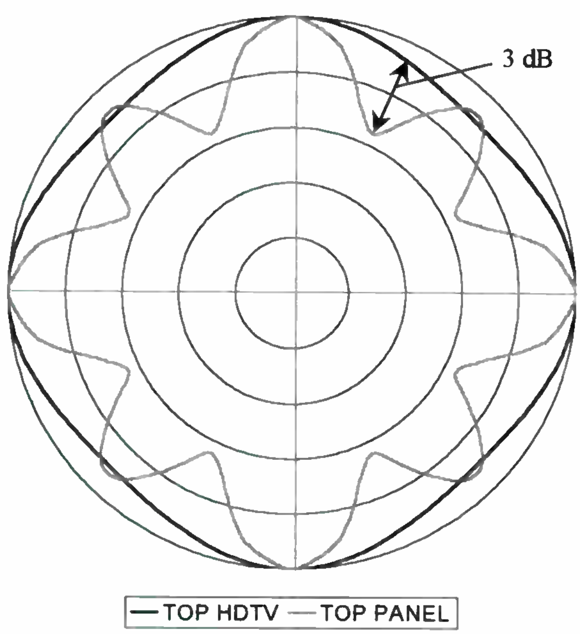


Figure 1

APERTURE ILLUMINATION

All the parameters that describe the elevation pattern are determined by the amplitude and phase illumination of the antenna aperture. The important parameters are beam width, gain, beam tilt, and null fill. Two antennas with the similar number of layers can have greatly different elevation pattern results depending on the aperture illumination design as demonstrated in Figure 3.

While the type of elevation pattern shown in Figure 3a was widely used with success in NTSC, the following investigations into the antenna output response performance will demonstrate why the elevation pattern of Figure 3b, with its smooth null fill response and wider beam width, is far superior for DTV.

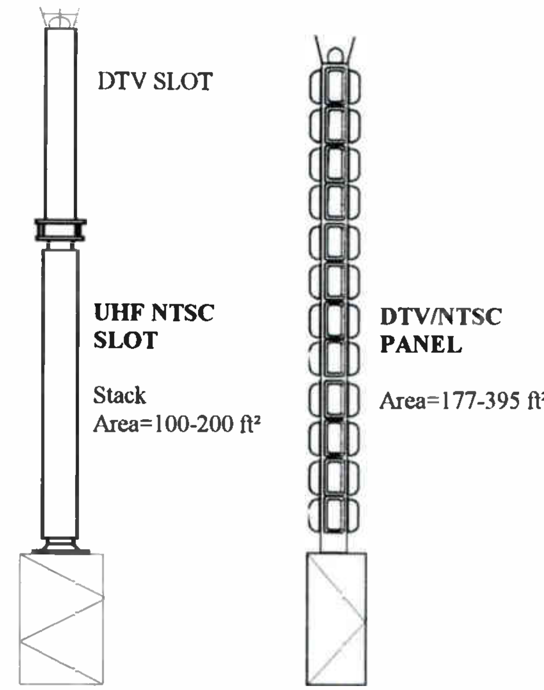
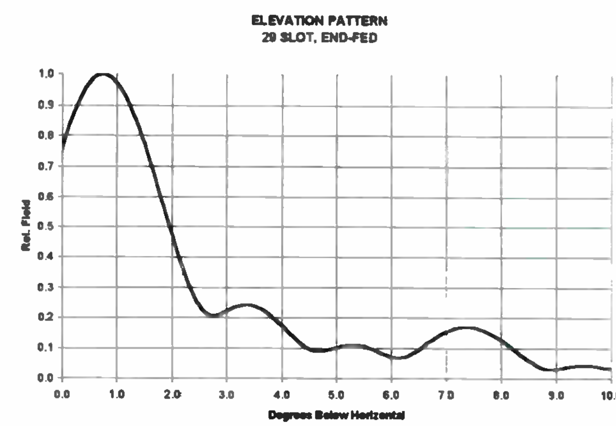


Figure 2



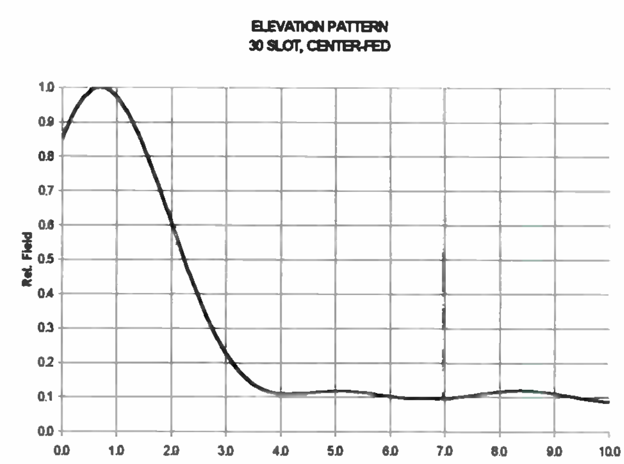


Figure 3b

ANTENNA OUTPUT FREQUENCY RESPONSE

The antenna output frequency response received virtually no attention for NTSC applications, but is of major importance for digital TV. Consider that the visual carrier, color sub-carrier, and the aural carrier dominate the NTSC RF spectrum with signal energy falling away rapidly from the carriers. NTSC antenna optimization was often performed concentrating on the visual carrier + 2 MHz. The effective luminance bandwidth is less than 4 MHz. By comparison, the DTV RF spectrum is flat across of the 6 MHz channel, except for the last 0.3 MHz. The entire channel is of equal importance and of larger effective bandwidth than NTSC. The antenna can no longer be optimized around a 2 to 3 MHz of the channel; DTV antennas should exhibit flat output response over a larger bandwidth.

END-FED ANTENNAS

Many slotted cylinder antennas are designed to feed the RF power from the bottom end of the antenna. This is mechanically convenient, especially for antennas mounted on the tower top. Broadcast slotted cylinder antennas must produce the main beam perpendicular to the vertical axis of the antenna. This requires that each slot level be nominally in phase. With the signal fed from the bottom and traveling towards the top, the end-fed antenna is made with a nominal one wavelength spacing between slots at the design frequency. As the signal progresses upward from one slot level to the next, a phase rotation of 360° occurs putting each successive slot level in phase.

However, the one wavelength spacing is only obtained exactly at the design frequency. As the signal frequency scans above or below the design frequency, the electrical spacing changes causing the beam tilt to vary. The end-fed configuration is depicted in Figure 4.

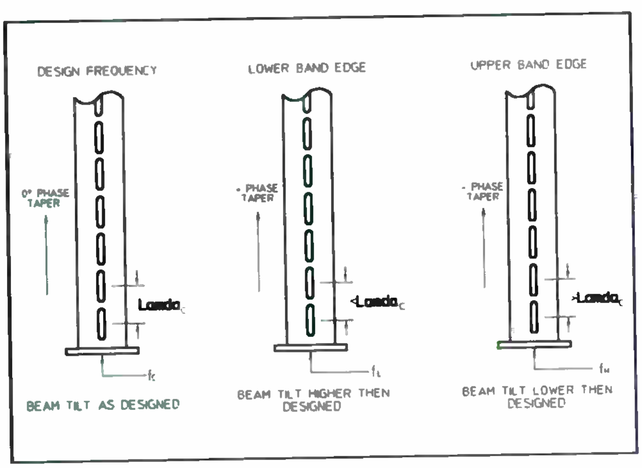


Figure 4

Consider a 30 slot, end-fed design with a smooth pattern, a calculation of the elevation pattern at the center (design) frequency, at the lower edge, and at the upper edge is plotted in Figure 5. Note that the beam tilt varies ± .25° from the design tilt.

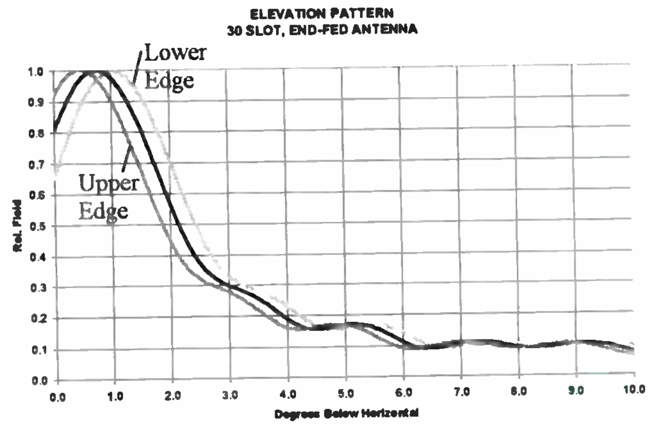


Figure 5

The detrimental effect of this beam tilt sway with frequency is the variations it produces in the antenna signal amplitude (gain) and phase output responses. Figure 6a demonstrates the maximum calculated gain deviation over the DTV channel for depression angles 0° to 10° below the horizontal. Likewise, the maximum calculated group delay variation at depression angles from 0° to 10° is plotted in Figure 6b.

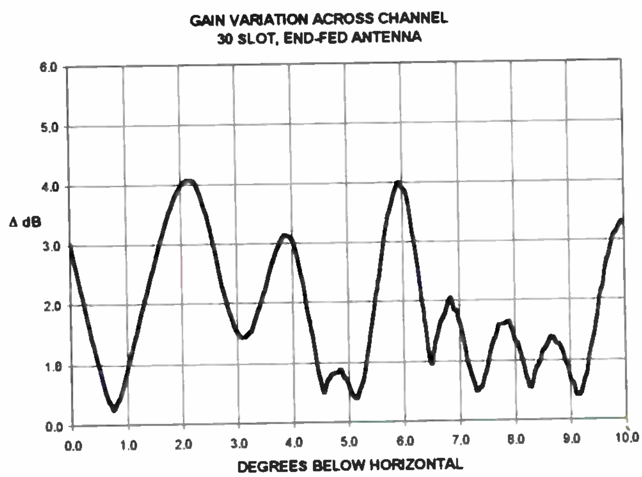


Figure 6a

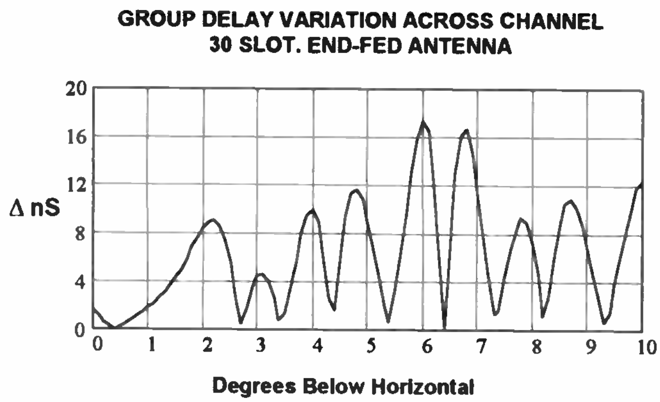


Figure 6b

CENTER-FED ANTENNAS

An alternative feed design employs electrical center feeding. This is simply done with side-mount antennas by using an input 'T' between the two antenna halves. Center feeding of top mount antennas is mechanically more complex, but is accomplished by using a triaxial configuration in the bottom half of the antenna to deliver the RF power to the center. These two configurations are shown in Figure 7.

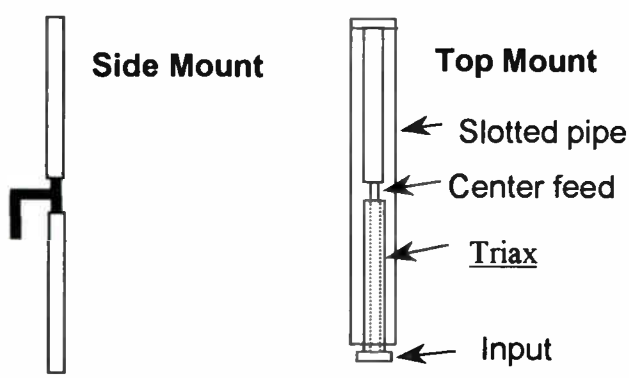


Figure 7

With the center-fed design, the signal travels up the top half antenna and down the bottom half antenna. The beam tilt varies in each half as frequency scans across the channel, but the bottom and top half tilt in opposite directions which produces a constant beam tilt for the complete antenna. Figure 8 demonstrates the electrical considerations of center feeding.

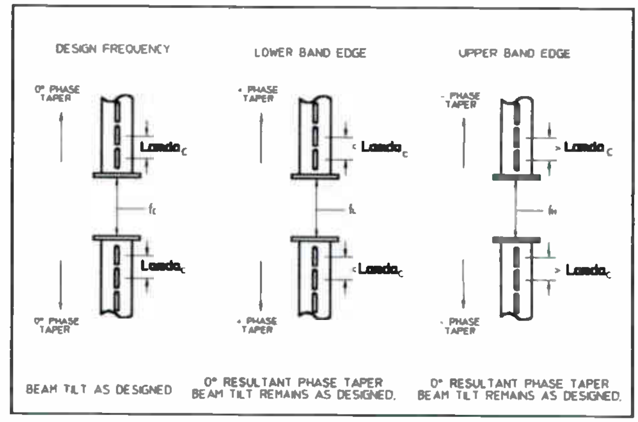


Figure 8

Returning to the center-fed illumination design of Figure 3b, a calculation of the elevation pattern at the center (design) frequency, at the lower edge, and at the upper edge is plotted in Figure 9. Note that the beam tilt variation is insignificant and variations in the nulls are minimal.

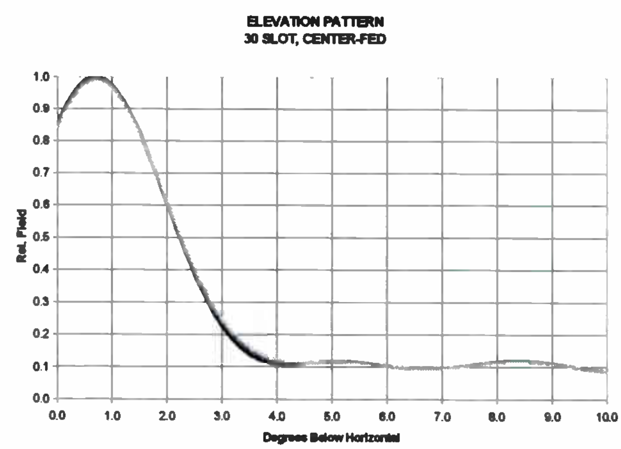


Figure 9

Figure 10a demonstrates the maximum calculated gain deviation over the DTV channel for depression angles 0° to 10° below the horizontal. Likewise, the maximum calculated group delay variation at depression angles from 0° to 10° is plotted in Figure 10b.

All of the previously shown elevation patterns and output responses were calculated. Like most calculations, these results are better (demonstrate less output response variations) than will occur with actual antenna hardware. The calculations do not include the frequency response effects of individual slot radiators that have a specific "Q"

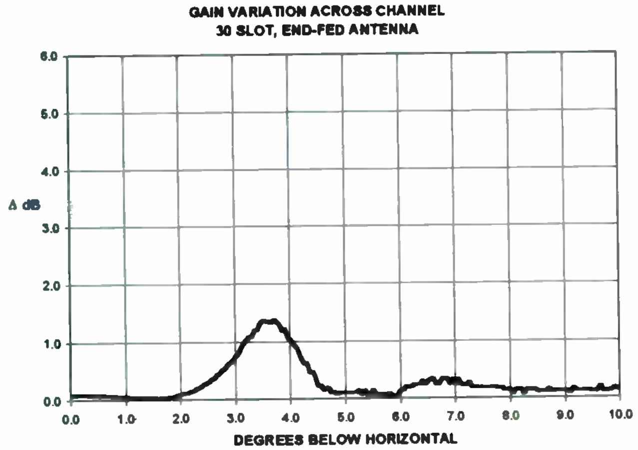


Figure 10a

factor. To demonstrate the effect of the radiator "Q" and other hardware factors, actual measurements of the center-fed illumination design of Figure 3b are

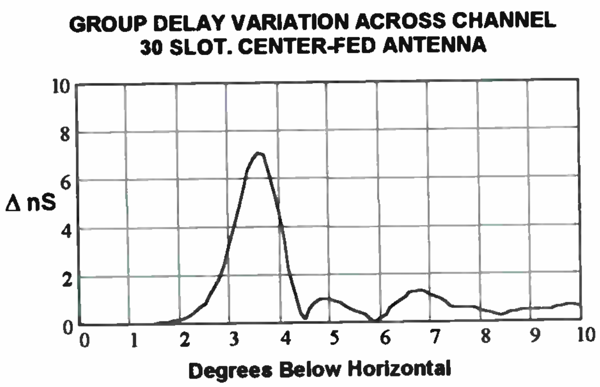


Figure 10b

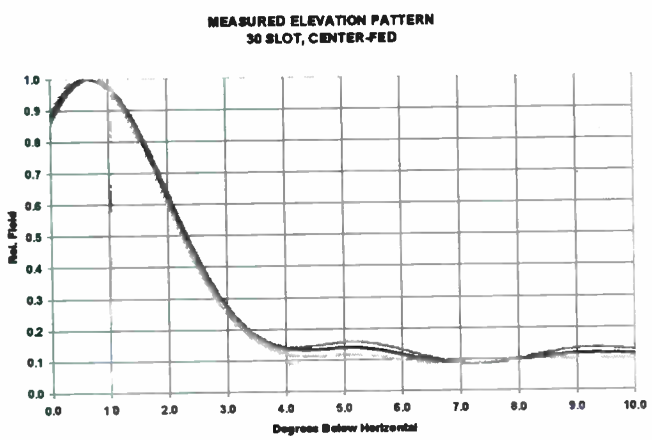


Figure 11a

presented in Figure 11a & 11b. As expected the measured results show an increase in gain variation in the nulls. The beam tilt still has insignificant variation.

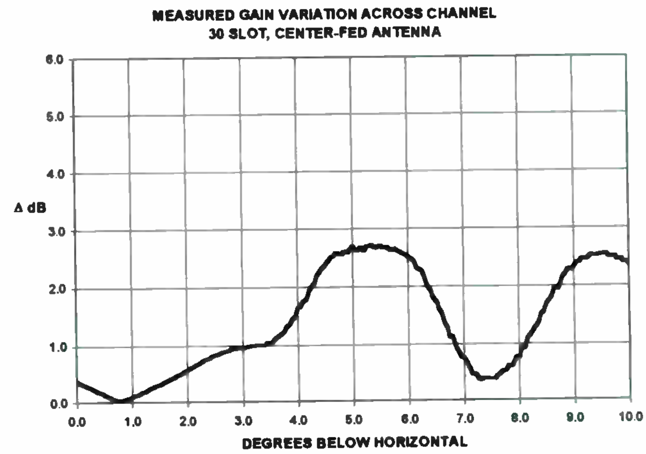


Figure 11b

FREQUENCY RESPONSE COMPARISON

End vs. Center Feed

The above maximum gain variation plots allow us to compare the worst case signal variations versus depression angle; however, they do not reveal the frequency response shape. It is instructive to compare the end-fed and center-fed designs with regards to the frequency response plots for a specific depression angle.

For this comparison, the depression angle of the gain variation maxima nearest the design beam tilt angle for each antenna is used. This represents the worst signal variation that will occur at the greatest distance from the transmitter site for each antenna. The end-fed gain

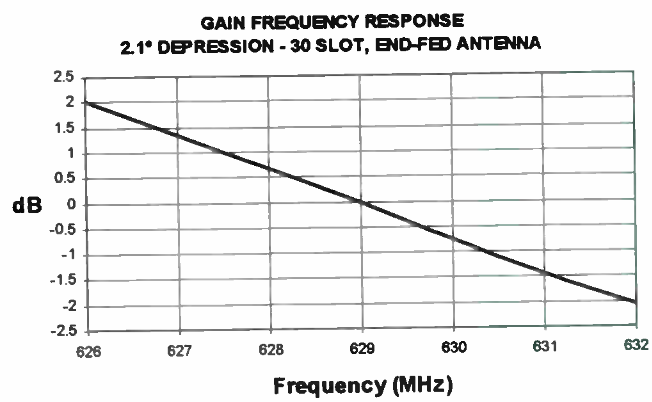


Figure 12a

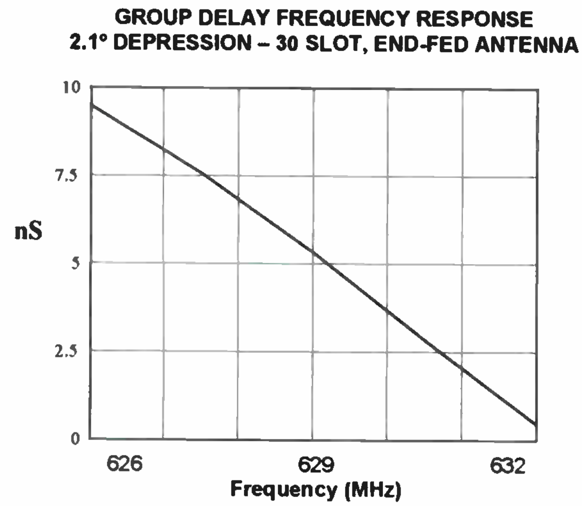


Figure 12b

frequency response at a depression angle of 2.1° is plotted in Figure 12a. Figure 12b shows the end-fed group delay frequency response at the same depression angle. Note that the min-to-max differences correspond with the plots in Figures 6a and 6b.

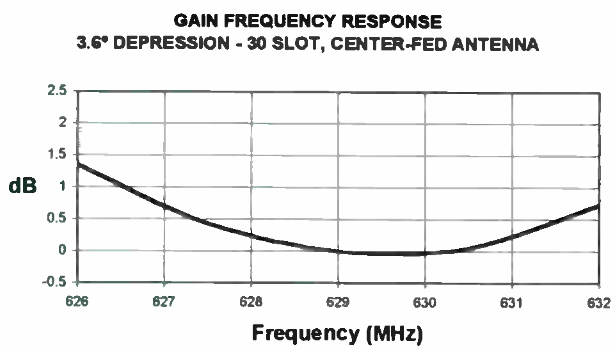


Figure 13a

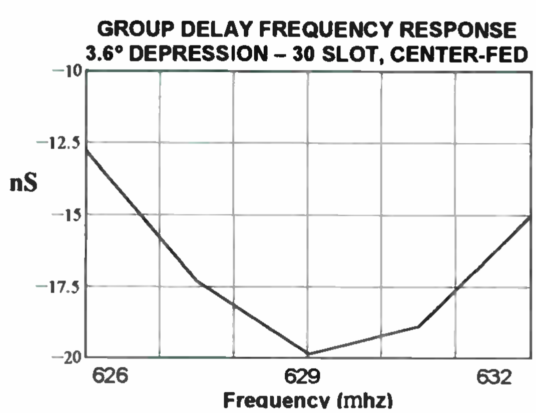


Figure 13 b

The corresponding frequency response plots for the center-fed antenna are shown in Figures 13a and 13b. Note that the end-fed response is a straight line while the center-fed response is bell shaped which will result in a lower frequency response distortion penalty [2]. The calculated min-to-max gain variation of the end-fed design is 4.1 dB, while it is 1.4 dB for the center-fed.

The other significant point of difference between end and center feeding is the location of this first gain variation maximum. The center-fed antenna peaks at a higher depression angle, 3.6° compared to 2.1° for the end-fed antenna. The end-fed maximum at 2.1° occurs on the slope of the main beam as a direct result of the beam tilt sway with frequency.

ANTENNA GAIN & SYSTEM DESIGN

The above analyses clearly show that for a given number of layers the center-fed design is superior to the end-fed design due to lesser antenna output response variations and beam tilt sway. One approach often used to minimize these detrimental performance effects of the end-fed antenna is to design systems with lower antenna gains. Low end-fed antenna gain results in less beam tilt sway and increases the null fill levels to mitigate the impact of signal variations at depression angles below the main beam.

However, the low gain system designs can have significant economic impact on the transmission system costs. Compare two system designs for the following scenario:

DTV Ch. 40 @ 1 MW ERP
Line length = 1800'

A: Antenna: 27.5 Gain Center Feed

6"-50 ohms rigid (58.3% eff.; 70 kW rated)
TPO= 62.4 kW (3 tubes)

B: Antenna: 20 Gain End Feed

1) 8"-75 ohm rigid (68.6% eff.; 102 kW rated)
   TPO=72.9 kW (4 tubes)
   Transmitter and line cost difference to "A": ~ $650 k
   Wind Area> 1.3 x tower load

Or

2) DTW-1500A (77.3% eff.)
   TPO=64.7 kW (3 tubes)
   Line cost difference to "A": ~ $200 k
   Wind Area > 2.5 x tower load

The ability to use higher gain with the center-fed antenna allows the use of smaller rigid line and transmitter. Choosing circular waveguide to maintain the transmitter size increases the tower load 150%. Waveguide adds another source of signal distortion that requires compensation by the transmitter manufacturer due to the signal dispersion (group delay) across the channel.

ADJACENT CHANNEL DESIGNS

Although adjacent channel slotted cylinder antennas have been made using both end-fed and center-fed designs, the performance characteristics described above apply across 12 MHz as well giving the center-fed antenna the advantage.

The calculated elevation pattern performance for a center-fed antenna across the NTSC and DTV channels is shown in Figure 14.

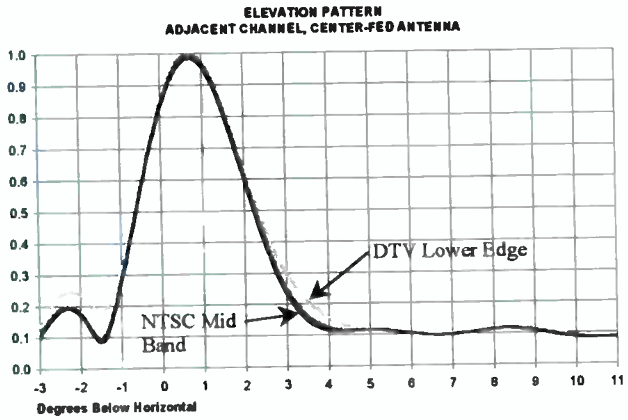


Figure 14.

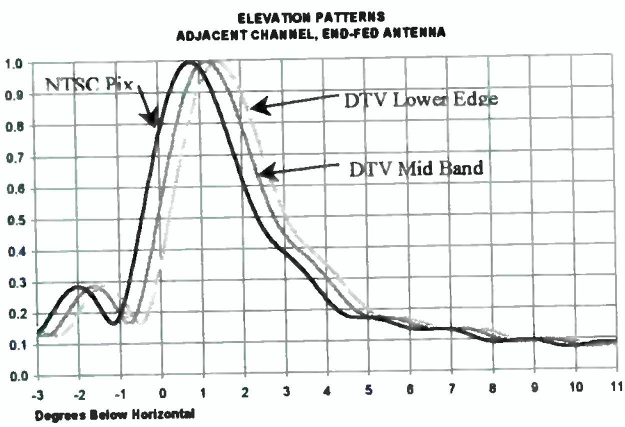


Figure 15.

An example of calculated elevation patterns for an end-fed, adjacent channel design is presented in Figure 15. Note the beam tilt variation and differences for each channel. Using the end-fed design produces a different beam tilt for the DTV channel compared to the NTSC channel.

CONCLUSIONS

Comparing slotted cylinder antenna performance of end-fed and center-fed designs, it is apparent that center-fed antennas provide significant advantages for DTV applications.

- Center-fed antennas have insignificant beam tilt sway vs. frequency.
- Center-fed designs with smooth null fill have low amplitude and phase response variations throughout the main beam and null structure.
- Frequency response shape of the center-fed antenna yields lower penalty due to frequency distortion.
- Systems designs can use higher gain, center-fed antennas to reduce transmission line and transmitter sizes resulting in lower cost transmission sites.
- Adjacent channel, center-fed antenna designs produce the same beam tilt angle specification for both the NTSC and DTV channel.

References

1. Mayberry, E. H., "Stacked Antenna Considerations for HDTV Conversion", Presented at IEEE Broadcast Technology Society 46th Annual Broadcast Symposium, September 27, 1996.
2. Bendov, O., "A New Approach to the Analysis of Adjacent Structure Effects on HDTV Antenna Performance", Presented at the 1995 NAB Convention.

KLYSTRON TRANSMITTER CONVERSION FOR SIMULTANEOUS ANALOG AND DTV TRANSMISSION

R.W. "Sam" Zborowski
David W. Brooking
ITS Corporation

ABSTRACT

This paper will describe an economical option of broadcasting DTV signals available to a large class of klystron transmitter owners. It is possible to modify existing klystron transmitters to operate the visual klystrons in combined aural/visual analog service, and operate the aural amplifier in DTV service. DTV service can be supported by biasing the aural klystron to the original (20 % aural) beam power, tuning to support 6 MHz bandwidth, adding a DTV exciter with appropriate predistortion and adding an output bandpass filter. This modification can be achieved at a small fraction of the cost of a new full-power DTV transmitter

INTRODUCTION

The coming transition from analog to DTV is financially difficult for many stations. During the early years of DTV most consumers will not yet have DTV receivers. Hence, advertising revenue will initially be minimal for DTV operation. Broadcasters are struggling with choices to either:

(1) Buy a new transmitter rated at the full authorized DTV power to achieve the desired coverage at a relatively high initial cost.

-or-

(2) Buy a new low power DTV transmitter to start transmitting over a smaller coverage area followed by a later power upgrade or new transmitter when DTV advertising revenues warrant the investment.

This paper will describe a third option available to a large class of klystron transmitter owners which may seem more attractive than either of the above. Most klystron transmitters were initially designed to support aural operation at 10 to 20 percent of rated peak visual power. More recently, to conserve energy, many stations have biased their aural amplifiers lower and generate aural power in the range of 2.5 to 5 percent of peak visual power. Regardless of what aural ratio is transmitted, CATV headend processors are adjusted to notch the aural down to about -16 dB (2.5 %) to achieve acceptable system intermodulation distortion levels.

It is possible to modify existing klystron transmitters to operate the visual klystrons in combined aural/visual analog service, and operate the aural amplifier in DTV service. With appropriate predistortion and additional output filters, one can operate visual level at about -1 dB from the existing visual power level with a -16 dB aural ratio. The 1dB analog power reduction is essentially imperceptible to existing viewers. DTV service can be supported by biasing the aural klystron to the original (20 % aural) beam power, tuning to support 6 MHz bandwidth, adding a DTV exciter with appropriate predistortion and adding an output bandpass filter. If the new DTV channel is nearby in frequency to the existing analog channel and the same antenna pattern is in use, one can approach or actually achieve the DTV power necessary for equivalent coverage area. (The DTV exciter and output filter may be reusable as part of a new DTV transmitter system following the DTV transition.) This modification can be achieved at a small fraction of the cost of a new full-power DTV transmitter.

DTV POWER REQUIRED

Calculations by members of the Advisory Committee for Advanced Television Service (ACATS) and measurements in lab tests by the Advanced Television Test Center (ATTC) indicated that for similar coverage of DTV relative to NTSC, the DTV average power should be approximately 12 dB lower than the NTSC peak visual power. Subsequent field testing in Charlotte indicated that the -12 dB ratio was conservative, suggesting that even lower DTV powers could provide equivalent coverage [1]. More recently, FCC has been wrestling with channel allocations to facilitate a smooth transition to DTV. The FCC goal is to replicate the grade B analog NTSC contour for DTV coverage while providing adequate interference protection for each station in a given area. Combinations of existing analog service EIRP and channel and new assigned DTV channel frequency define an enormous range in DTV transmitter power output (TPO) requirements from low VHF to the highest UHF channel to replicate the analog coverage. Across the UHF TV band the variation in EIRP is only about 4 dB for equivalent coverage [2]. Since this paper deals with the case of a UHF broadcaster adding a UHF DTV channel, the -12 dB power ratio goal will be assumed for discussion.

EXISTING HARDWARE

Most full-service UHF broadcast stations presently employ transmitters using klystron tubes as output devices. Typically, the klystrons used in these transmitters are rated to produce from about 30 to 70 kW peak power each in visual service. Transmitters built in recent years have generally employed Inductive Output Tube (IOT) devices (earlier versions of this device were called Klystrodes) to achieve better energy efficiency than is possible with conventional klystrons [3]. A relatively smaller number of recent transmitters have employed Multi-Stage Depressed Collector (MSDC) klystrons to achieve better energy efficiency than conventional klystrons [4]. A significant percentage of existing klystron transmitters employ an energy saving technique called pulsing. Pulsing introduces a step change in beam current to provide the input beam power necessary to support the peak power level of the sync pulse only during the time sync is actually present (8% of the time) and a somewhat lower beam power at all other times [5]. Most of the operational klystron transmitters now in service in the U.S.A. are believed to employ some version of the ITS-20 or ITS-20A exciter/modulator. Hundreds of these units were delivered to broadcasters for retrofit into transmitters built by a variety of vendors including at least RCA, Harris, Comark, GE, Townsend, TTC, Marconi, CCA and Ampex. Some exciter/modulators were provided as OEM equipment incorporated within transmitters manufactured by Advanced Broadcast Systems and Astre Systems. The conversion described in this paper applies to transmitters that employ either conventional klystrons or MSDC klystrons.

Figure 1

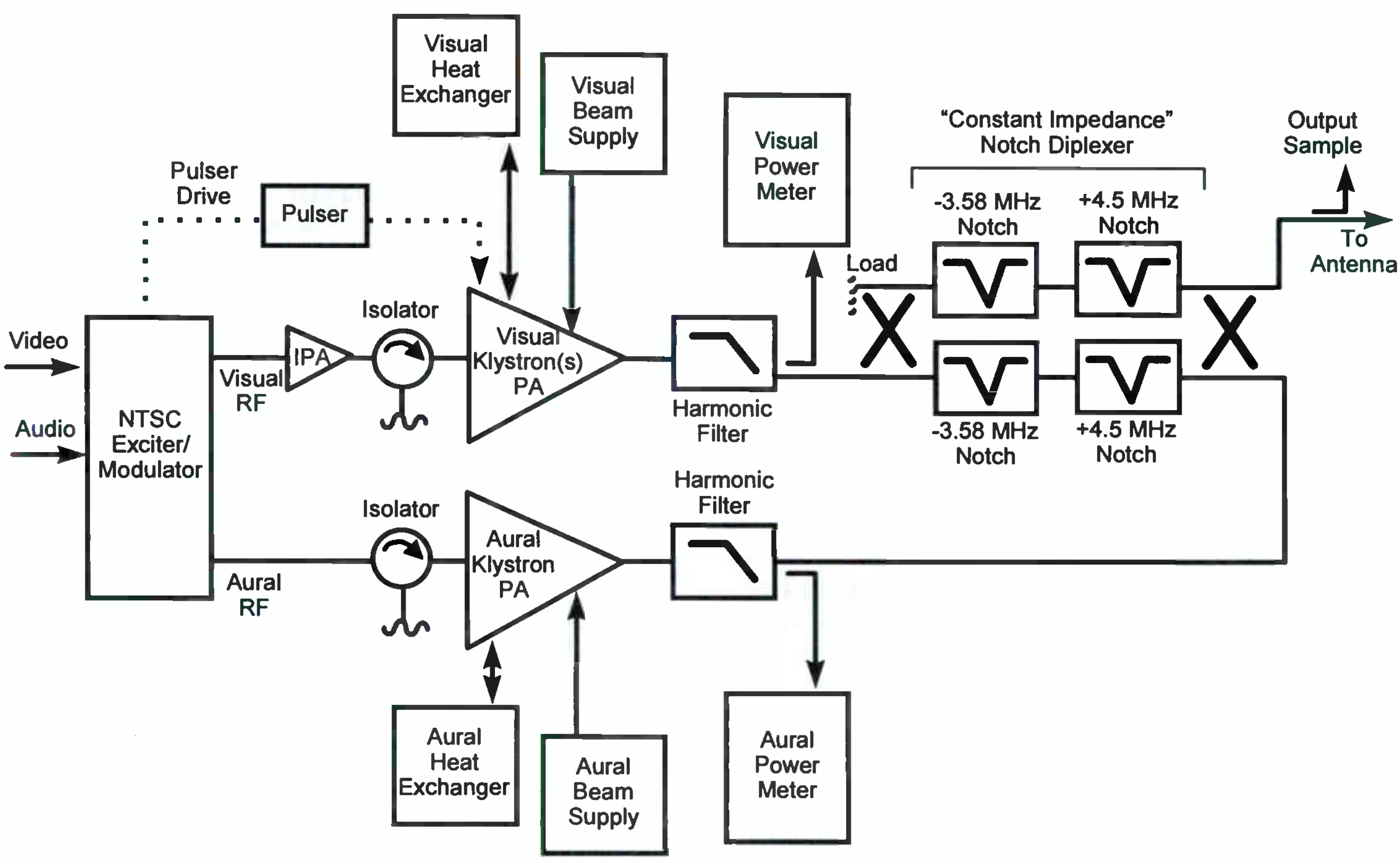


Figure 1 shows a high level block diagram of a typical klystron transmitter.

Video and audio signals are applied to an exciter/modulator which generates the visual and aural RF drive signals for the power amplifiers. ITS-20/20A processing will be described for illustration. The exciter/modulator includes oscillator/ multiplier circuits to generate the carrier frequencies used in IF modulation and to convert the IF signal to the final output RF channel frequency.

The visual side of the exciter/modulator includes the following video functions: clamping, level control, peak limiting, group delay predistortion, differential gain predistortion and differential phase predistortion. A visual IF modulator generates a double sideband AM visual signal that is passed through a surface acoustic wave (SAW) filter to create the vestigial sideband (VSB) AM NTSC visual signal. If pulsing is employed, the VSB AM signal is AM detected to drive a sync separator. The sync separator output is used to drive the pulser which changes the klystron bias point. Sync separator outputs are also used to drive IF gain and phase predistortion circuits to complement the distortions introduced by pulsing the klystron.

Following the pulsing predistortion circuits (if used), the VSB AM signal is passed through an automatic level control (ALC) attenuator followed by IF predistortion circuits for amplitude linearity correction, incidental phase correction and frequency response correction.
These circuits complement the AM-AM, AM-PM and amplitude vs frequency response distortions introduced by the klystron power amplifier. The visual IF ends with a peak detector which is used to close the ALC loop around the IF corrector stages and which forms the start of an automatic gain control (AGC) loop around the up-converter and RF drive amplifier stages.

A double balanced mixer is used to convert from IF to RF. UHF bandpass filters select the desired frequency conversion product to pass through an AGC attenuator and RF drive amplifiers. The visual drive output is monitored by a peak detector, which completes the AGC loop.

The aural side of the exciter/modulator starts with an aural IF VCO that is held on the correct center frequency by a phase locked loop (PLL). The aural VCO is frequency modulated by the audio input signal. The FM IF signal is passed through ALC, upconverter, RF filter, AGC and RF amplifier stages which are similar to those on the visual side. There is no need for linearity and frequency response predistortion circuits since the aural operates at constant level (FM vs AM) and occupies only a tiny fraction of the klystron bandwidth.

The exciter/modulator includes jumpers to add the aural IF signal to the visual IF path upstream of the IF predistortion circuits. This aural/visual multiplex feature provides a convenient way to temporarily transmit the combined A/V signal through any one klystron in the event of a klystron, beam supply, heat exchanger or RF diplexer failure. The backup mode requires appropriate RF patch connections to be available in the transmitter RF output transmission lines.

The exciter/modulator RF outputs are rated at several watts peak power, which is typically sufficient to drive the aural klystron directly. On the visual side an intermediate power amplifier (IPA) is usually fitted to provide tens of watts peak power to the visual klystron power amplifier (PA). The IPA typically employs class-A biased bipolar junction transistor (BJT) devices. On some low UHF channels with particular tube types as much as 100 watts peak visual drive is required. Many systems include two or more visual klystrons that are combined at the output to provide higher power and a measure of redundancy to the system. The klystrons in popular use are typically water-cooled or vapor-phase (steam) cooled. Both integral cavity and external cavity types are in service. An appropriate heat exchanger is provided either one per tube or shared by multiple tubes.

Klystron electronic operation may be described as follows. The klystron includes an electron gun with a heater and cathode operating at -15 to -26 kV relative to chassis (ground) potential, dependent on the tube type in use. The electrons emitted by the cathode are accelerated by the electrostatic field toward the RF cavities, drift tubes (body) and collector of the tube which are all essentially at chassis potential. The beam is initially focused by the shape of the electrostatic field in the region of the concave-shaped cathode. Further focusing is accomplished by an electromagnet assembly which forms a linear magnetic field in the same direction as the hollow center (axial direction) of the tube. The resultant electron beam is tightly focused and of uniform density prior to application of RF drive. The electron gun also includes a doughnut shaped electrode through which the beam passes between cathode and body. This annular ring electrode is called a modulating anode that is biased generally in the range of 0 to -10 kV to select the quiescent beam current. Mod-anode type pulsers use this electrode to toggle the beam current between two levels as described earlier. Many recent model klystrons also include another electrode (either annular or grid-shaped) in close proximity to the cathode which is used by "low voltage" pulsers to accomplish the same effect.

RF amplification is accomplished by velocity modulation of the electron beam as follows. The RF drive signal is coupled to the beam via the input cavity which includes a pair of capacitive rings that form a structure called a gap. Electrons passing the gap are accelerated on one half cycle of the RF waveform and those passing during the other half cycle are decelerated by the electric field across the gap. The electrons continue drifting toward the collector along the drift tube to the next RF cavity and associated gap. The faster electrons overtake the slower ones, forming a slight "bunching" by the time they pass the next cavity gap. The bunched electrons passing the gap induce an RF signal in the second cavity which "rings" as it is resonant near the frequency of interest. The "ringing" induces a field across the gap which accelerates electrons from the sparse intervals of the beam to amplify the bunching phenomena during the drift to the following gap. This process continues for each cavity and drift space. When the electron bunches arrive at the final gap and cavity, RF power is coupled out by the cavity and a coupling loop to the output transmission line. The electrons continue past the final cavity gap to deposit their remaining energy by collision with the collector structure. Non-linearities arise largely due to the electrostatic repulsion of the electrons from each other which tends to oppose bunching at higher RF signal levels. The bunching would be proportionally concentrated at higher signal levels absent this repulsion/spreading effect. Detailed description of klystron operation is provided elsewhere [4~13]. Typically, the same klystron tube type is employed in visual and aural sockets. The conventional klystron can be thought of as a class-A amplifier whose AM-AM and AM-PM transfer characteristic exhibits no crossover distortion at low drive levels and exhibits a substantial but smooth monotonic compression characteristic at the highest drive levels. This transfer characteristic is also stable over time provided that sufficient beam voltage regulation and coolant temperature regulation is incorporated in the transmitter. This smooth, stable non-linearity is easily compensated by predistortion in the exciter/modulator [5,6,7,9,13]. The transmitter visual and aural outputs are monitored by peak detectors which drive power meters.

The visual and aural PA outputs are applied to a constant-impedance diplexer assembly which combines the two signals with low loss into one transmission line to drive the antenna. The diplexer works as follows. The visual signal is applied to a quadrature (90 degree offset outputs) splitter called a hybrid coupler. The signal is split equally in power at the outputs. The two visual signals flow over notch filters centered at the lower color subcarrier frequency (visual-3.58 MHz). The –3.58 MHz filters are needed to achieve the FCC specified spectral mask. Signal components in the –3.58 MHz region "see" an impedance mismatch and reflect back from the filters to the hybrid. The 90 degree phase shift of the hybrid adds to the 90 degree shift of the first pass to cancel at the visual input and sum to the termination port. Signal components in the NTSC visual band from –1.25 to +4.18 MHz pass through to combine in the output hybrid. The aural signal is applied to the aural port of the output hybrid, reflects back from the +4.5 MHz notch filters and sums to the antenna port similar to the operation of the –3.58 MHz filter described earlier.

HOW TO IMPLEMENT THE CONVERSION

Analog NTSC side.

The exciter/modulator is jumpered to the internally diplexed mode. The pulser (if used) is disabled or removed. In the internally diplexed operation, the klystron step changes in gain and phase would modulate the aural signal and cause unacceptable levels of video (sync) crosstalk into the stereo and SAP channels of multichannel sound service. It may be possible to add complementary gain and phase modulators to the aural IF upstream from the aural/visual summing point to allow continued pulser operation in future. A new power metering circuit is added to display the relative power of aural and visual signals at the PA output. The common amplification mode generates more consequential intermodulation products in the klystron amplifier than the visual-only service. In-channel intermod products are largely cancelled by products generated in the IF linearity and incidental phase correctors of the exciter/modulator.

An improved incidental phase corrector, optimized for the common amplification service is available to achieve more complete in-channel intermod correction. If the klystron were a broadband device, it would be possible to cancel out-of-channel products as well. Unfortunately, the relatively narrowband klystron cavities significantly attenuate complementary out-of-channel products prior to the final drift space and cavity where most of the PA non-linearity occurs. The major out-of-channel products that exceed the spectral mask lie at –4.5 MHz, -3.58 MHz and +9.0 MHz relative to visual carrier. A series of notch filters must be added to meet the spectral mask. A possible alternative is to retune the existing notch diplexer and add only the additional –3.58 MHz notch as shown in figure 2.

Figure 2

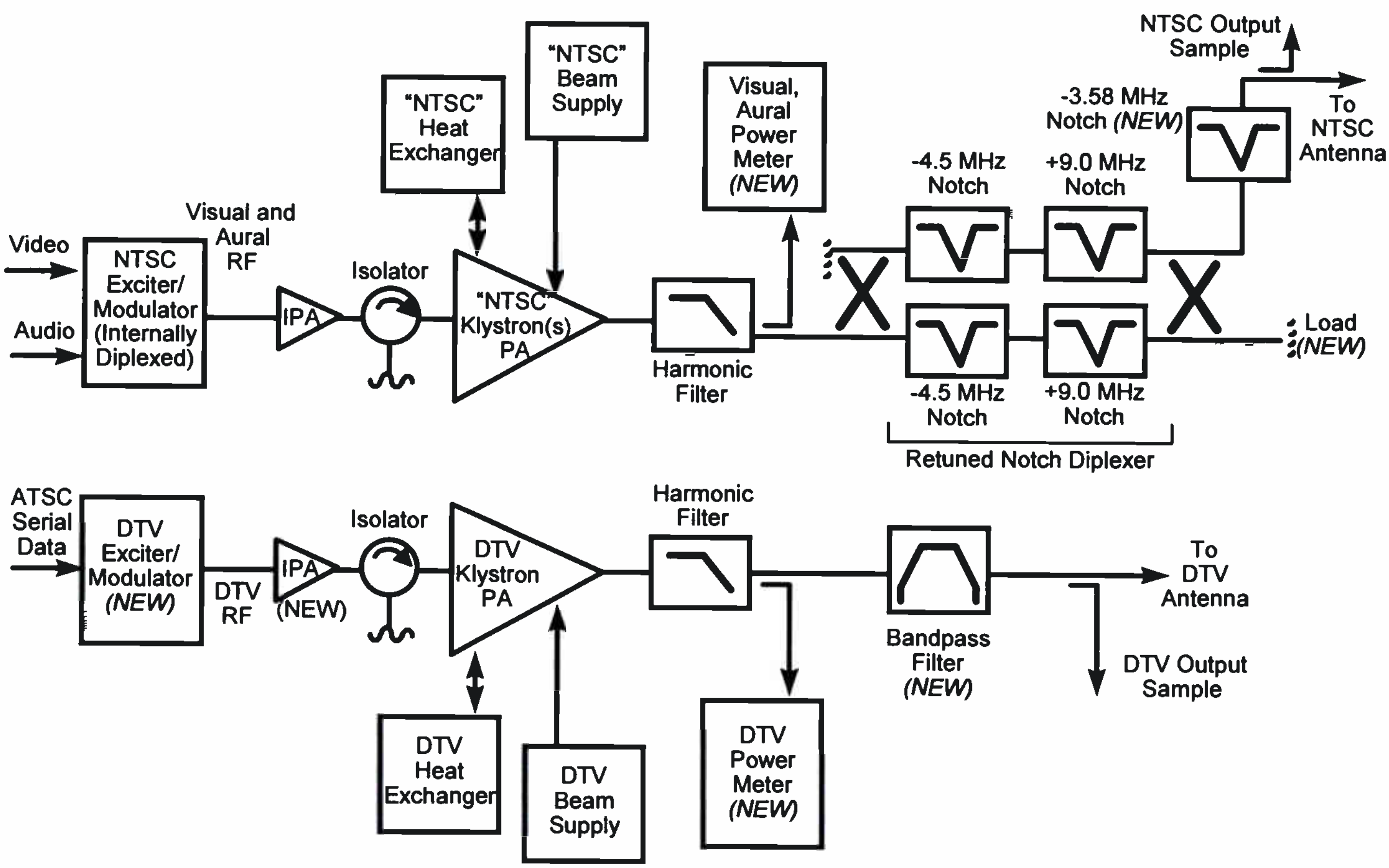


Figure 2 is a block diagram of the proposed transmitter conversion

IF predistortion, ALC, upconverter, AGC and RF drive amplifier stages similar to the corresponding sections of the internally diplexed NTSC exciter/ modulator. The former aural klystron must be retuned to support the full 6 MHz channel bandwidth of the new DTV channel. Note - If the new UHF DTV channel assignment lines outside the tuning range of the present klystron, an exchange will need to be negotiated with the klystron vendor. The broadband tuning reduces the RF gain, so an additional IPA stage is required to amplify the drive to the DTV klystron. The existing peak detectors of the transmitter are likely to have slow rise time relative to the DTV symbol rate and fast decay time relative to frequently occurring peak values of the DTV signal. A new power metering circuit must be fitted to display the

DTV side.

A new DTV exciter/modulator is introduced to drive the former aural klystron in DTV service. The exciter/modulator includes an ATSC compliant 8-VSB DTV IF modulator. The DTV IF signal is applied to

average power of the DTV signal. The DTV signal passing through the non-linearity of the klystron generates intermodulation products similar to those described in the common amplification case. The spectral shape of the DTV signal is very different from the NTSC in that a data randomizing function disperses the spectral energy about equally across the channel. The DTV spectrum looks like bandlimited noise on a spectrum analyzer. The flat noise-like signal generates broadband intermod products that fall off gradually from channel edge into the adjacent channels; these intermod products are also called spectral regrowth. In-channel intermod products are generated as well which are not noticeable on a spectrum analyzer display. As in the NTSC common amplification case, the in-channel intermod products can be nearly cancelled by products generated in the IF pre-distortion circuits. Again, out-of-channel products cannot be complemented well by the IF correctors due to klystron bandwidth limitations. The broadband nature of the intermod products dictates the use of a bandpass filter which must be added to meet the spectral mask requirements.

DTV MEASUREMENTS

Tests were conducted to investigate the DTV performance of a klystron biased at various beam power levels and tuning conditions. A Varian (CPI) model VKP-7553S housed in an RCA TTU-110B transmitter operating on channel 25 was made available for one night of testing. To save time in tuning for 6 MHz bandwidth, and to avoid having to substantially retune for aural service before sign-on, the test series employed one of the visual tubes.

This transmitter employs three integral cavity klystrons, two in visual service and one in aural service. An ITS-20A exciter with separate 50W amplifiers for each tube comprises the visual drive system. The exciter output directly drives the aural tube. ABS low voltage pulsers are in use to improve energy efficiency. A variable coupler [11] is fitted on each visual tube output and a fixed impedance transformer is fitted on the aural tube output.

Some initial parameters include:

- 23.5 kV beam voltage
- 3.2 A beam current (pulser operating), 4.5 A (pulser disabled) for each visual tube.
- 1.4 A aural beam current.
- 98 kW visual TPO, 6.3 kW aural TPO.

An 8-VSB DTV signal was applied to the IF processing circuits in place of the analog NTSC signal. The improved incidental phase predistortion circuit was fitted in place of the original ICPM corrector. The visual 1 tube output was patched to a water-cooled load which serves as a calorimeter for average power measurement. Beam power was disconnected from the visual 2 and aural klystrons during visual 1 testing. The visual 1 beam current was adjusted by changing taps on the mod-anode bias string. A variety of tuning patterns [12,13], beam current and predistortion settings were tested according to the following process:

1. Adjust beam current to the new setting.
2. Tune the cavities to the desired response using a network analyzer. Record gain change.
3. Increase drive power and adjust predistortion for maximum power with less than –35 dB spectral regrowth (Intermod) at channel edge.
4. Measure output power with the calorimeter.
5. Measure digital signal to noise ratio (SNR) using a vector signal analyzer (VSA).

MEASURED RESULTS

Test number	1	2	3	4	5
Beam current (A)	4.5	2.6	1.5	1.5	2.6
Tuning pattern	S	S	S	H *	H *
Output power (W)	8400	5500	1600	1600	5500
Relative gain (dB)	REF	-8	-17	-10	+1
SNR (dB)	33.7	31.9	29.5	32.2	32.2

(*) Note- could not quite make –35 dB spectral regrowth in H-mode.

DISCUSSION

The S-mode tuning pattern produced less spectral regrowth but the H-mode pattern seemed to have an edge on overall digital SNR. SNR is a measure of total digital "noise" including effects of both linear and non-linear distortions, phase and amplitude noise. As a comparison, the DTV modulator measures 38.5 dB SNR with about –55 dB spectral regrowth. The 50W drive amplifier uncorrected spectral regrowth worst case was –40 dB for test #3 where it was running at 110% of peak rated power. It is likely that adjustment of the variable output coupler would have facilitated an increased output power capability at the lower beam currents. Time constraints limited the number of parameters that could be examined in this test. The variable coupler performs an impedance matching function between the output cavity and the 50 ohm output line. When the beam current is reduced, the beam impedance is increased. Hence, a different output coupler setting is required for optimal power transfer to the output line. This same function is available by adjustment of the output coupling loop in external cavity klystron systems.

CONCLUSION

The operation at 5500 W average power with 2.6 A beam current in this example is –12.5 dB from the present 98 kW peak visual power and is within original transmitter ratings of the beam supply and cooling system. The out of band spectral regrowth can be attenuated sufficiently to meet the DTV spectral mask with available bandpass filters (for the S-mode tuning case). The digital SNR achieved is similar to new high power DTV transmitter offerings. The increased power consumption cost needs to be weighed against the initial cost of a new transmitter that would be more energy efficient. In this example the transmitter was already rather efficient with aggressive pulser operation. The conversion would increase total energy consumption by 49%. If the starting point is a transmitter that is not equipped with operating pulsers, the overall increase in energy consumption is only about 12%. This conversion appears to be feasible as a method to achieve the desired DTV coverage with minimal initial hardware cost.

ACKNOWLEDGMENTS

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The 8-VSB DTV Performance that can be expected from Klystron Amplifier Systems used in existing Analog U.H.F. TV Transmitters

Dr. Roy Heppinstall, Alan E. Wheelhouse, Geoffrey T. Clayworth, Neil O'Sullivan, Mark Keelan
EEV Limited, Chelmsford, England.

ABSTRACT

This paper describes an investigation into the performance of a u.h.f. TV klystron when amplifying 8-VSB modulated digital signals. Both theoretical and experimental results are presented. It is shown that a klystron can transmit digital signals successfully but only at very low efficiency – approximately one quarter of the efficiency of an EEV Inductive Output Tube.

INTRODUCTION

During the last six years the Inductive Output Tube (IOT) has become established worldwide as the final amplifier for new high power u.h.f. TV transmitters. However, many u.h.f. TV broadcasters in the USA continue to transmit NTSC programmes with an analog signal produced by an older transmitter fitted with high power klystrons as the final amplifiers. U.H.F. TV klystrons have high gain and are rugged, reliable devices. They have been used routinely in this application for over 30 years. During this period there have been a number of improvements both in the design of the tube itself and in its mode of operation. These have resulted in enhanced performance and much improved operational efficiency when amplifying analog TV signals.

The advent of digital terrestrial television in the USA is approaching rapidly. Consequently, it is a topic of immediate interest to the operators of these existing klystron powered transmitters to know the performance which might be expected from a klystron used to amplify an 8-VSB digital signal. This is particularly important since one of the recurring demands on the transmitter operator is to have high transmitter operational efficiency and hence low operational costs.

This paper presents the results of computational and experimental work aimed at establishing the true position with respect to operating costs. It then discusses the impact of these results upon the options available to the operator of a klystron powered transmitter during the transition to the digital regime.

TRANSFER CHARACTERISTICS

The amplitude modulation to amplitude modulation (a.m.–a.m.) and amplitude modulation to phase modulation (a.m.–p.m.) transfer characteristics of an amplifier are, respectively, the change in output power and the change in phase of the output signal as a function of the change in input power. The linearity of these characteristics has a major influence on the performance of any device amplifying analog or digital TV signals. The amount of pre-correction which the transmitter manufacturer needs to supply is directly affected by the non-linearity of the transfer characteristics of the amplifier.

The IOT has excellent characteristics, both a.m.–a.m. and a.m.–p.m. The central portion of its a.m.–a.m. characteristic is very linear but some non-linearity occurs in both the low power and high power regions. Nevertheless, its performance is so good that the vast majority of EEV IOTs installed in analog TV transmitters are operated in common amplifier mode, in which a single tube is used to amplify both visual and aural signals.

In contrast to that of the IOT, the a.m.–a.m. transfer characteristic of a klystron is very linear at the lower power levels but exhibits a marked saturation at the high output power levels. It is reasonable to represent the a.m.–a.m. conversion by a graph of the square root of the output power against the square root of the input power. To a good approximation, this is a sine curve. This is shown in Figure 1, where both the output and input powers have been normalised to 100% at the saturation point. Such a representation can be used readily to predict the basic performance of a klystron when amplifying digital signals. In particular, estimates can be made of the peak-to-average ratio of the output signal as a function of the peak-to-average ratio of the input signal and the average digital output power. Table 1 shows calculated values for the output signal for a range of input signal power levels, for two values of peak-to-average ratio on the input signal. Also listed

Input Signal			Output Signal			Clipping (dB)
Peak Power (%)	Average Power (%)	Ratio (dB)	Peak Power (%)	Average Power (%)	Ratio (dB)	
100	20	7	100	41.7	3.8	3.2
90	18	7	99.4	38.2	4.2	2.8
80	16	7	97.3	34.5	4.5	2.5
70	14	7	93.6	30.7	4.8	2.2
60	12	7	88.0	26.8	5.2	1.8
50	10	7	80.3	22.7	5.5	1.5
40	8	7	70.2	18.5	5.8	1.2
30	6	7	57.5	14.1	6.1	0.9
20	4	7	41.7	9.5	6.4	0.6
10	2	7	22.7	4.9	6.7	0.3
100	12.6	9	100	28.0	5.5	3.5
90	11.3	9	99.4	25.4	5.9	3.1
80	10.1	9	97.3	22.9	6.3	2.7
70	8.8	9	93.6	20.2	6.7	2.3
60	7.6	9	88.0	17.6	7.0	2.0
50	6.3	9	80.3	14.8	7.3	1.7
40	5.0	9	70.2	11.8	7.7	1.3
30	3.8	9	57.5	9.1	8.0	1.0
20	2.5	9	41.7	6.0	8.4	0.6
10	1.3	9	22.7	3.2	8.5	0.5

Table 1. Calculated power characteristics for input signals having peak to average ratios of 7dB and 9dB.

is the degree of clipping which is occurring – that is the difference between the peak-to-average ratio on the output and input signals.

It is worth noting that the maximum peak digital output power obtainable is the saturated output power of the tube. Thus for any particular value of peak-to-average ratio on the output signal there is a maximum value for the average digital output power at which the tube can be operated. For example, for a 6 dB ratio the maximum value is 25% and for a 7 dB ratio it is 20%. Alternatively, for any particular average digital output power, there is a maximum peak-to-average ratio which can be obtained. Corresponding values of these parameters – which define a limiting curve – are given in Table 2. This limiting curve is applicable to all amplifying devices, not simply klystrons. Figure 2 shows a plot of the limiting curve and also plots of predicted output power peak-to-average ratios as a function of the average digital output power for three values of input signal peak-to-average ratios: 6 dB, 8 dB and 10 dB. For each of these three cases a truncation point has been defined, being that point at which the output peak power level has reached the same value as the saturated output power.

COMPUTER SIMULATIONS

A number of simulation tools have been developed for evaluating the performance of devices processing digital signals. One of these – the Alta Group's Signal Processing Work System employing library routines developed by the GEC-Marconi research department – has been used for the simulation of the performance of an IOT when amplifying 8-VSB digital signals[1]. With small modifications this model has been used together with a representation of klystron non-linearities to assess the symbol error rate (SER) performance of the

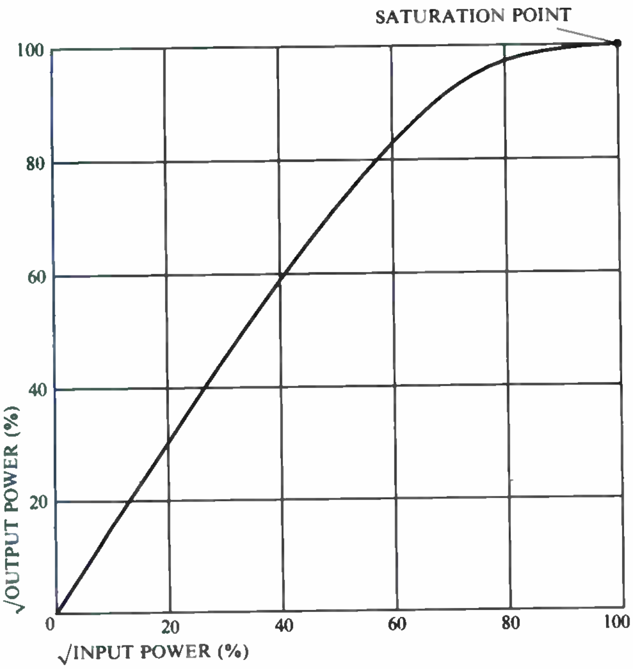

Figure 1. Square Root Transfer Characteristic

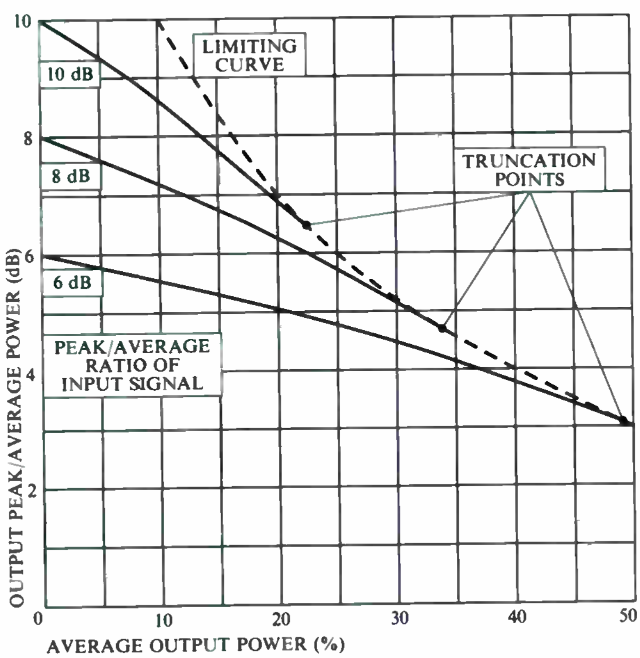

Figure 2. Digital Output Power Characteristic

tube when amplifying an 8-VSB DTV signal. Briefly, the model provided 8-VSB signal generation, klystron transfer characteristics and received data detection functions (see Figure 3).

Average Output Power (%)	Maximum Peak to Average Ratio (dB)
50.1	3
39.8	4
31.6	5
25.1	6
20.0	7
15.8	8
12.6	9
10.0	10
7.9	11
6.3	12

Table 2. The Limiting Curve

The data source is generated as a random eight level pulse amplitude modulation, followed by Nyquist pulse shaping (root raised cosine). This is passed to a single sideband modulator using a Hilbert transformer (digitally implemented) having a transition band set to simulate the VSB modulator.

The klystron transfer characteristics used were modelled as a look-up table providing both amplitude and phase non-linearity information. The information was derived from a.m.–a.m. and a.m.–p.m. measurements taken on a high power wideband u.h.f. K3672BCD klystron, shown in Figure 4. The model can be extended to include variations in non-linearity across the signal bandwidth if necessary but for this simulation only a single pair of a.m.–a.m. and a.m.–p.m. characteristics were analysed.

A Fast Fourier Transform (FFT) analysis of the output data stream is sufficient to provide an evaluation of intermodulation noise. However, to establish the receive error rate performance due to transmitter non-linearity, a receiver noise source is added, together with an IF filter, root raised cosine filter and a SER measuring facility. No simulation of error correction coding was included in this computation. Results are shown in Figures 5 and 6.

Figure 5 illustrates the variation of computed SER as a function of the ratio of energy per symbol to noise spectral density. The reference curve gives the results obtained from the model without including the simulation of the klystron characteristics. The additional three curves are for different signal levels, corresponding to average digital output powers of 40%, 19% and 10% of the saturated output power.

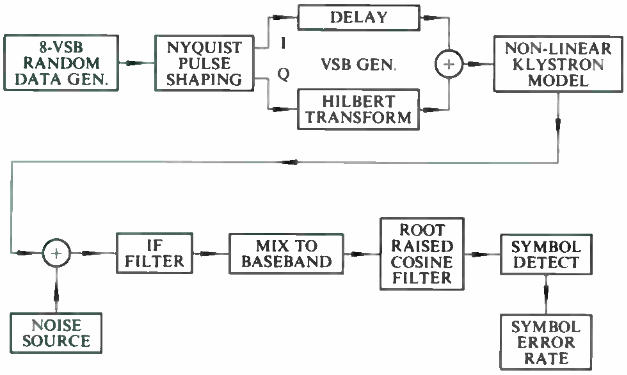

Figure 3. Computer Simulation Model

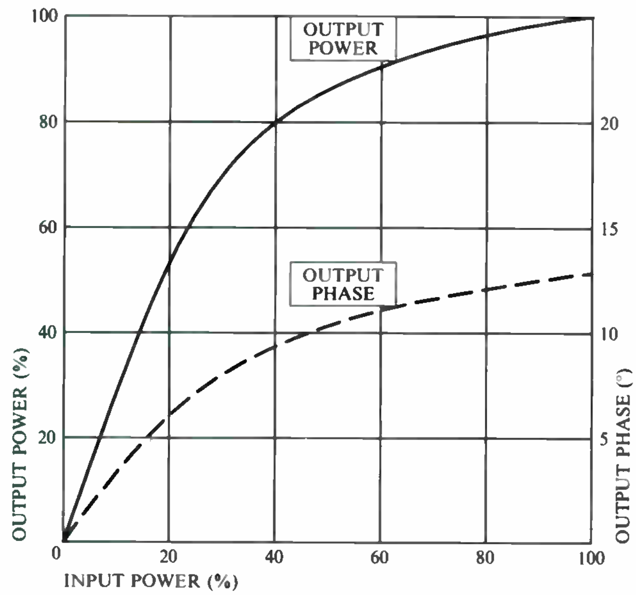

Figure 4. Klystron Characteristics

The effects of the non-linearities of the klystron are clearly seen as an increase in the SER, especially at the higher values of E_S/N_0. The characteristic tails due to the non-linearities are evident and the simulation predicted irreducible SERs of 1.7×10^{-2}, 1.0×10^{-3} and 4.6×10^{-5} for each of the loading levels analysed. The predicted peak-to-average ratios on the output signal were 4.2 dB, 6.3 dB and 6.9 dB, whereas that on the input signal was 7.9 dB. These values are in excellent agreement with those predicted from the sinusoidal square root transfer characteristics given in the previous section.

The results shown in Figure 5 were obtained by including both amplitude and phase non-linearities in the simulation model. A further investigation was made to establish the effect of phase non-linearities alone on tube performance. A digital signal corresponding to the low average output power level of 10% of saturated output was used so that the effect of amplitude non-linearities was low. Three different levels of phase distortion were assumed – zero, the characteristic shown in Figure 4 and phase distortion twice that shown in Figure 4. The results of the simulation are shown in Figure 6. They predict that at this signal level, amplitude non-linearity alone (the zero phase curve) produces relatively little increase in the SER. However, at the measured phase distortion level, an appreciable increase in SER is observed and at twice that level the increase in SER is large. These results emphasise the

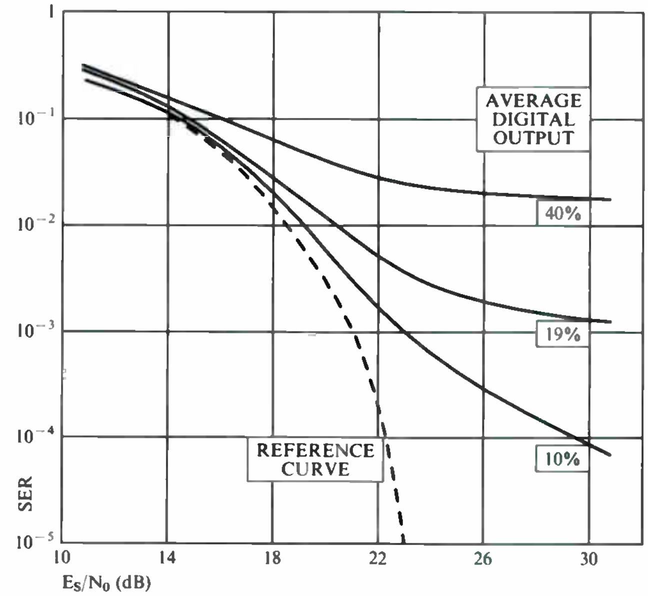

Figure 5. Klystron Simulation Characteristics

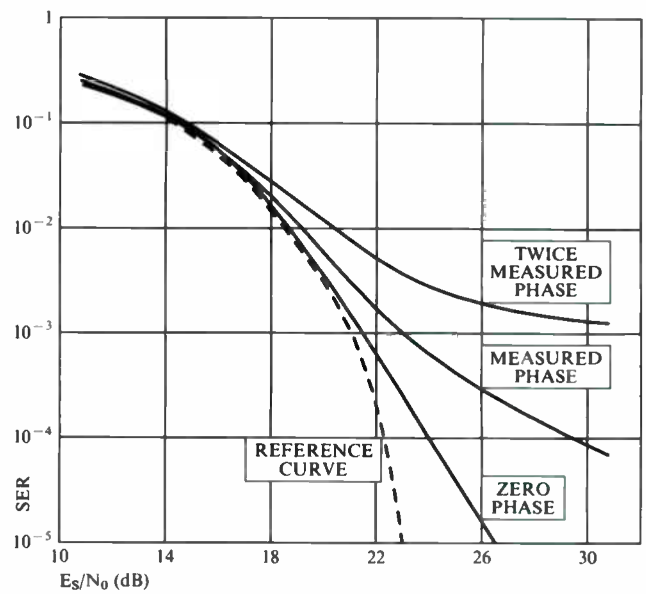

Figure 6. Klystron Simulation Characteristics

impact of phase distortion on the digital performance of a klystron and the importance of ensuring that appropriate pre-correction is available in the transmitter.

EXPERIMENTAL EVALUATION OF A KLYSTRON

EEV high power wideband u.h.f. TV klystrons type K3672BCD are installed in many terrestrial TV broadcast transmitters in the USA. The 8-VSB digital performance of such a klystron has been investigated using an experimental arrangement similar to that used for the investigation of the performance of an IOT and described elsewhere[2,3]. In measuring the klystron's performance, the 8-VSB signal was generated by a Harris CD1 modulator and detected and analysed using a Hewlett Packard HP89441A Vector Signal Analyser. A Boonton 4500 Digital Sampling Power Analyser was used to make peak-to-average power ratio measurements. The klystron was operated at US Channel 34 with a beam voltage of 25.8 kV and a beam current of 5.6 A. The r.f. conversion efficiency at saturation was 48%, giving a saturated output power of 69 kW. The peak-to-average ratio of the 8-VSB input signal was 7.45 dB.

Table 3 lists various parameters of the digital signal at the output of the tube. Power levels are quoted as percentages of the saturated output power and the efficiency is the ratio of the average digital output power to the klystron beam power. The measured sidebands are also quoted.

DISCUSSION

The peak-to-average ratios predicted from the sinusoidal square root transfer characteristic and those measured are generally in good agreement, especially at the lower digital average output power levels. This is shown in Figure 7, where it is seen that the two curves deviate as the output power is increased. Nevertheless, a reasonable evaluation of the performance of a klystron can be made from these results.

The operating efficiency of the klystron for digital operation is obviously of crucial importance, as this is the major parameter which determines transmitter power consumption and hence transmitter operating costs. If it is assumed that the peak-to-average ratio on the output of the tube must be a minimum of 6 dB, then the maximum average digital output power which can be obtained is 25% of the klystron saturated output power. Due to clipping arising from tube a.m.–a.m. non-linearity this would require a peak-to-average ratio on the input of the tube of about 9.5 dB. In the experimental work described here, the peak-to-average ratio on the input was 7.45 dB. This gave a 6 dB ratio on the output at an average digital power level of 17.4%, corresponding to a tube operating efficiency of 8.3% and an average output power of 12 kW. Under these conditions the sidebands were at −28 dB. The beam power consumption was 144 kW which, at a fuel price of 10c/kWh and a broadcast time of 8000 hours per year, corresponds to an annual bill of $115k. Such a large bill is because, although the peaks of a digital

Peak Power (%)	Average Power (%)	Peak/Average Ratio (dB)	Efficiency (%)	Sidebands (dB)
92.5	31.3	4.7	15.1	23
84.9	26.5	5.1	12.8	25
76.4	21.2	5.6	10.2	27
68.7	17.4	6.0	8.4	28
66.1	16.2	6.1	7.8	29
64.9	15.1	6.3	7.3	30
61.7	14	6.4	6.7	32
46.7	9.8	6.8	4.7	34
43	8.9	6.8	4.3	34
34.5	6.7	7.1	3.2	35
16.8	3	7.4	1.4	34

Table 3. Klystron Digital Performance

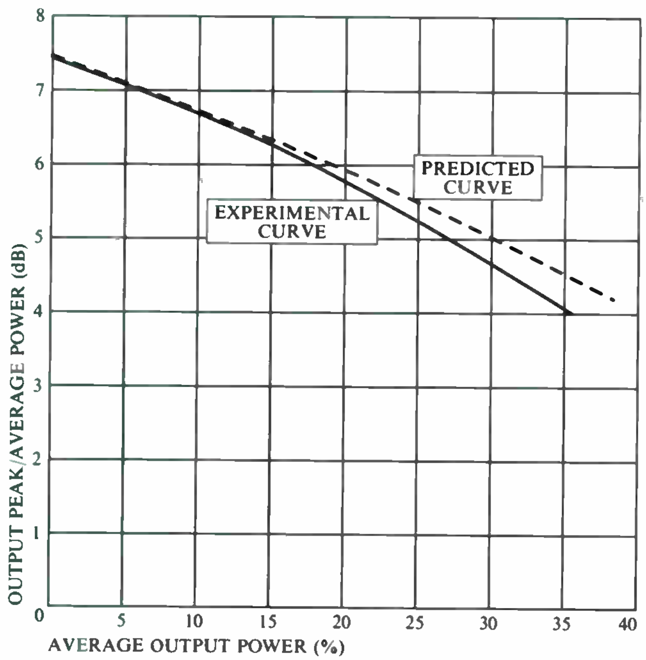

Figure 7. Klystron Digital Performance

signal are of short duration and present only infrequently, the beam power of the klystron must always be present at a sufficiently high level to be able to amplify those peaks – beam current pulsing as used in analog TV transmitters is not an option. In contrast, the mode of operation of an IOT is such that the beam power present at any particular time is only that required to amplify the instantaneous r.f. signal. Experiments on IOTs have shown that in relation to the average digital output signal, an efficiency without pre-correction of somewhat over 40% can be obtained. In practice after pre-correction the operating efficiency is lower. Even if the operating efficiency is as low as 30%, the beam power consumption is still only 40 kW. This corresponds to an annual fuel bill saving of $83k compared with using a klystron. Consequently, it is imperative that the operator of a transmitter using klystrons as the final amplifiers takes this factor into account when considering the options available during the transition period from analog to digital television. Two of the options available are:

a) To purchase a new IOT equipped transmitter. This will include an immediate capital outlay.

b) To transmit digital signals using the existing transmitter. This will involve a significant but lower capital outlay for digital equipment but has the disadvantage of very much higher operating costs in the future.

The above analysis demonstrates that the ultimate decision on the timing of the purchase of a new transmitter by each broadcast operator should not be driven by considerations of technical performance but by the financial implications of a larger initial capital outlay against a very substantial reduction in future running costs. Individual financial circumstances must guide this choice.

As there is no significant practical field experience of the use of klystrons for transmitting 8-VSB signals, it is recommended that broadcasters contemplating operating in this way should contact the transmitter and tube manufacturers for advice before converting the transmitter to digital operation.

ACKNOWLEDGEMENTS

The authors thank their many colleagues at EEV Limited in England and EEV Inc. in the USA for valuable contributions to the work described. In particular they thank Mr. Ian Alston of the GEC-Marconi Research Centre for performing the computer simulation work and technical discussions on the results obtained.

The views expressed are those of the authors and not necessarily those of the General Electric Company of England.

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A Fresh New Look at 8VSB Peak to Average Ratios and Practical N+1, N-1 Combining Systems

Robert J. Plonka, Principal Engineer
Harris Corporation, Broadcast Division
Quincy, Illinois

Abstract

This paper will revisit the 8VSB peak to average ratio issue to get a better look at the signal conditions that create the high signal peaks. This is for reassessing the 6dB or 7dB planning ratios as a means of estimating peak voltage levels in transmitter PA's, transmission lines, tuned cavities and antenna feeder systems. Actual waveform plots under typical transmitter operating conditions causing various amounts of compression will be examined. The effects of bandpass filters, transmitter linearization and broadband operation will be included, as well as the influence of the peak to average ratios on N+1 and N-1 combining systems.

Practical N+1 and N-1 adjacent channel combing systems will also be discussed with additional details on system equalization. Included in this discussion, will be data showing the results of single amplifier mode of operation for active N+1 and N-1 combining.

Is the 8VSB Pk /Av Ratio 6dB or 7dB?

There has been a lot of industry discussion on this issue with the result that both are being used depending on how one looks at the problem. And in doing so, there are several important features to note.

1. The RMS value is constant.
2. Everything else about the signal is unknown.

Item 2 above says that due to the pseudo random nature of the 8VSB signal, peak levels can not be determined with any degree of certainty. Demodulating 8VSB, of course, will bring out the high quality aspects of the baseband signal, but in terms of the transmitted RF envelope, all that is known for sure is the RMS value. The peaks can only be described on a statistical basis. This presents a dilemma to those who are planning a transmitter system where a single valued Pk/Av number would be a very convenient planning factor to allow calculation of the correct voltage head room necessary in major system components for safe operation.

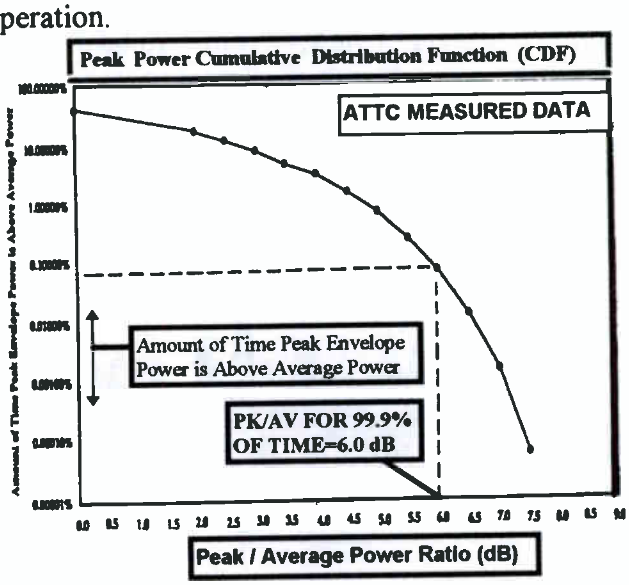

Figure 1 8VSB Pv/Av dB vs % of Time

Figure 1 shows the ATTC measured Pk/Av data normalized to a cumulative distribution curve. The dotted line indicates a typical operating point where the Pk/Av ratio is about 6 dB or less for 99.9% of the time, or for .1% of the time, the peak envelope power is 6 dB or above the average power.

This operating point, 6dB Pk/Av , has been used at Harris as a simple guide line to estimate DTV transmitter power. It is a guide line that has been found to agree with modest levels of peak compression resulting in a spectral spread level that just meets the FCC -35 dB mask (41 dB referenced to total RMS power). It also results in operating the transmitter at a better efficiency point. But, this is not the last word on this subject , since others have used 7 dB Pk/Av.

The 7 dB Pk/Av guideline will call for about 26% more transmitter power if the average power is intended to remain constant between the two Pk/Av planning factors. This says the Pk/Av ratio is a cost driven factor that requires further analysis to understand all the parameters necessary for proper systems design taking into account, important costing.

From this point of view, it is time to take another new look at the well used phrase, "peak to average ratio "(Pk/Av).

A Fresh New Look at 8VSB RF Envelope

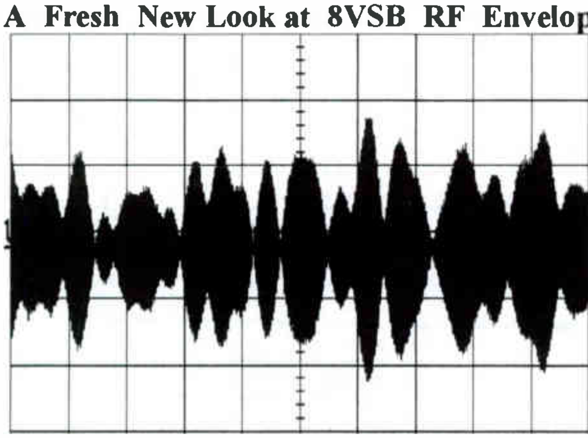

Figure 2. Ch33 exciter output voltage waveform over 50 ohm load. H=.5us V=40mv

The scope readings for Figure 2 were 30.7 mv (rms) and 109 peak averaged for 5000 scope sweeps to get a cumulative peak value. The Pk/Av ratio is 20log(109/30.9) = 10.95 dB. This is significantly above the 6 dB planning ratio by about 5 dB.

Ch33 TX 8VSB Output IMD = -37 dB

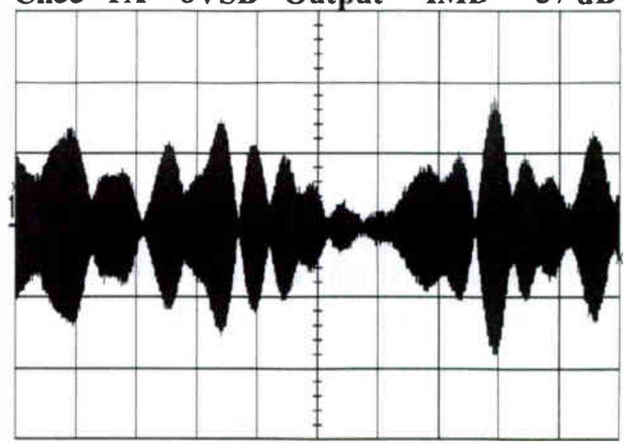

Figure 3. RF sample from TX output. H=.5us /div V=400 mv/div

In Figure 3, a sample from the transmitter output, the rms voltage was 240 mv and the peak voltage was 715 mv peak after cumulating the peak value over 5000 scope sweeps. The Pk/Av ratio is 20log(715/240) = 9.48 dB. Both the exciter output and the transmitter output have high Pk/Av ratios as seen above in the voltage waveforms. This Pk/Av ratio, which seems high, is the result of using a cumulated peak voltage divided by an (rms) voltage. Is this realistic, yes, because it is the peak voltage that causes component breakdown in transmitting equipment. But how does this relate to the standard 6 dB Pk/Av ratio?

Power Meter Calibration is Important

There are several power meters available on the market that measure peak power but what is not explicitly stated is that these meters are calibrated to read (rms) power. This is in line with industry standards that call for power measurements to be read as an (rms) value.

The peak power reading is also an (rms) value measured during the peaks of the waveform, or better stated, over the high points of the RF envelope. This is shown in Figure 4.

Expanded view of 8VSB envelope

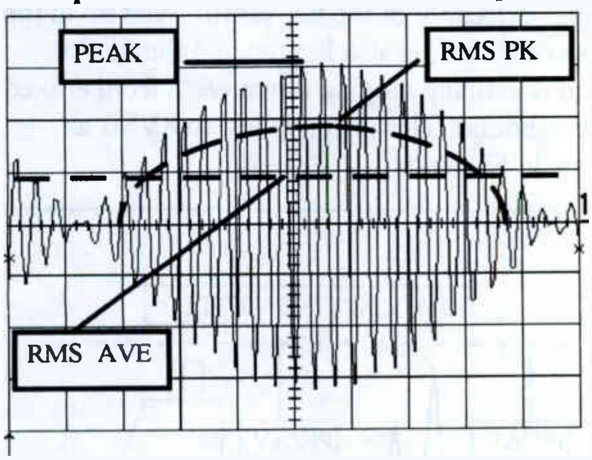

Figure 4. RF envelope showing Peak voltage, RMS Peak and RMS average.

A peak power meter will read the power in the zone at the peak waveform as shown in Figure 4 (RMS PK). The average value will be read at the (RMS AVE) level. The voltage peak at the (PEAK) icon is the value used in this paper for analysis, because it is the item that can cause a voltage breakdown in high power transmitting equipment and initiate a destructive arc. The voltage peak analysis is also useful to determine safe operating levels in transmission lines and antenna feeders, particularly in the case when two or more stations are combined together for a common antenna feed. This will be discussed a little later in this paper.

The voltage analysis was chosen because new high speed sampling scopes are available that can effectively capture the UHF RF envelope accurately to display the true peak voltage. From this, the peak or average power can be easily calculated given the fact the waveform presented on the scope display is taken over a 50 ohm load. The waveform voltage measurements shown here are not intended to start a new Pk/Av method but rather to show the voltage extremities of the 8VSB are 3 dB higher than the standard 6 dB Pk/Av power ratio for planning purposes.

For example, the Pv/Av value calculated from Figure 3 data was 9.48 dB voltage peak to (rms). To convert this to a power ratio, simply subtract 3 dB from the Pk/Av ratio. This is the same as multiplying the peak voltage , 715 mv, by .707 to get the peak (rms) value.

The formula is $20\log(715)(.707)/240 = 6.47$ dB derived from voltage data from Figure 3 where the 240 mv number is true (rms). For the purist, working from power data, convert to power by using the following. (Note: it sometimes pays to go back to the basics.)

A) Let $P_K = ((715)(.707))^2/50$
B) Let $P_A = (240)^2/50$
C) Then $Pk/Av = 10 \log(P_K/P_A) = 6.47$ dB

This is the peak (rms) power over the true (rms) power, the present way of looking at things.

The 6.47 dB number is not too far from the 6 dB planning factor considering the IMD spectral spread was -37 dB for this data instead of -35 (FCC) which will slightly increase the Pk/Av ratio, so the 6.47 number compared to the 6 dB planning factor is reasonable. Also from previous measurements, the Pk/Av ratio can vary from 6 to 7 dB depending on the transmitter setup and the degree of linearization. It is a reasonably accurate statement to say, use 6 dB for purchasing a transmitter and then linearize it to 7 dB for operation.

Cumulative Distribution Function ?

The original ATTC cumulative distribution curve was a way of trying to relate the occurrence of peak power to average power as a function of time. This has some virtues but I believe a transmitter does not know what a cumulative distribution function is and it probably does not care. The transmitter instead, will compress the high signal levels and alter the cumulative distribution curve as seen at the output. It is the purpose of this analysis to use the best available techniques to capture the occasional peaks of the 8VSB signal before and after compression to determine safe operation.

The RF envelope waveforms displayed in Figures 2, 3 and 4 were taken from a 4 GHz sampling scope . One of the important features of the sampling scope method is the availability of statistical data on the waveform at the time of measurement. This includes the true (rms) and maximum peak voltage that is a running sum of peak values over a selected period of time. This provides a running account of any peaks exceeding the previous peak values and is very effective to capture and store the pseudo random peaks of 8VSB as a function of time.

What has been observed so far in testing is that the 8VSB voltage peaks are limited by the transmitter as noted earlier where the exciter output had a Pk/Av (voltage) of 10.95 dB while transmitter output was 9.48 dB. This shows some soft clipping occurred The point here is, the alarmingly high exciter output peaks were safely clipped in the transmitter PA and in most cases, the PA will serve as a limiting device to protect down stream components from high voltage levels. Of course, this leads to the concern about voltage breakdown in the PA amplifying devices. Surprisingly, no real damage, over many months of testing, can be directly attributed to the high 8VSB peak as noted here. This may seem to be in conflict with earlier statements herein , particularly when 3 dB may be added to the standard power Pk/Av ratio to size up the potential voltage levels.

This should not be alarming since the NTSC waveform has a typical 4.5 to 5 dB Pk/Av power ratio or 7.5 to 8 dB voltage ratio.. This will vary according to picture content. The key observation here is that current transmitters are already dealing with wide ranging peak to average ratios. An important point to note is the upper extremity of the waveform, sync tip in the case of NTSC, is also limited and the Pk/Av ratio is actually ranging down ward from a fixed upper bound. This also applies to 8VSB as show in Figure 5.

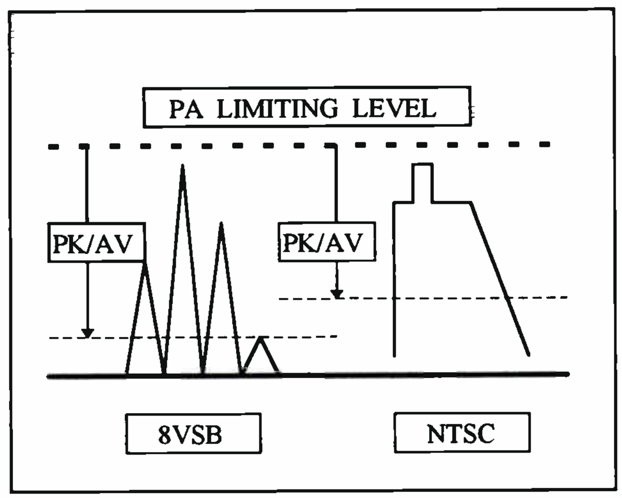

Figure 5. Illustrates Pk/Av ranges downward from a fixed upper level.

Noting the PA limiting level in Figure, the Pk/Av ratio is fixed at one end, the high side, and the average level varies below it. This limiting action makes the system safe for operation by holding occasional high peaks, as forecast by the cumulative distribution curve, to safe normal levels. The output, however, should add the 3 dB to the output Pk/Av power ratio to estimate downstream voltage requirements.

What is the Pk/Av for two combined DTV's

This is a very interesting case since two pseudo random signals will be combined together resulting in a signal whose likelihood of summing up two maximum peak voltages will be the product of the probability occurrence of the maximum voltage on each signal. This should be very small. The question is, how small? This item is under study and in lieu of an in depth statistical analysis, actual measurements were made to get an idea as to the magnitude of the problem. Shown below is the sum of two DTV signals each compressed to -35 dB for analysis.

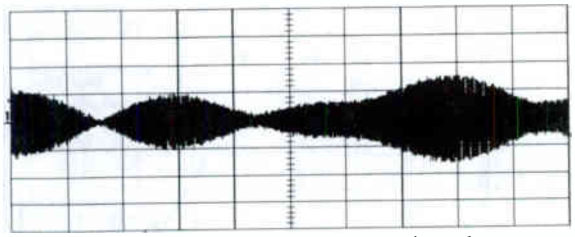

Figure 6. Ch29 8VSB Signal.

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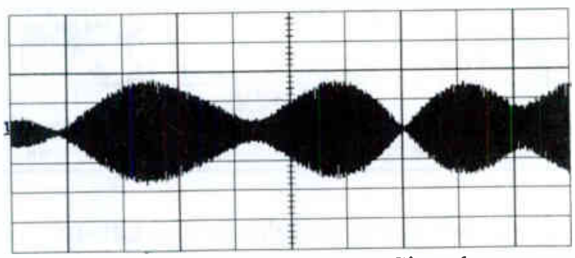

Figure 7. Ch33 8VSB Signal

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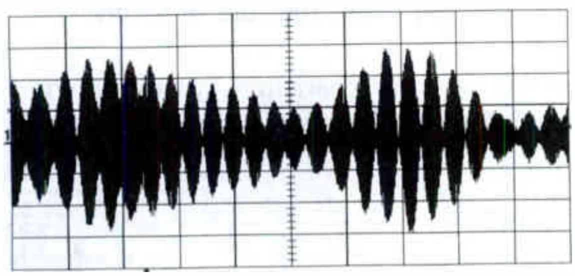

Figure 8. Combined sum of Ch29 and Ch 33. Simulating output of two transmitters

The sampling scope gain for Figures 6, 7 and 8 was held constant to get an idea how the peak amplitudes added up. It is tempting to simply add up the vertical scale values to get the peak amplitude, however, this is not a meaningful thing to do because the signals are not synchronous. The time capture of one signal is not at the same point for the other signal and probably never will be because they are random.

Hence, the reason behind the opening comment in this paper stating nothing is known for sure about the RF 8VSB signal except its (rms) value.

Scope statistical readings are necessary to get the peak summed amplitude.

Figure 6. Vp=.78 volts cumulated peak.
Vrms = .296 volts
Pk/Av = 9.45 dB voltage

Figure 7. Vp=.869 volts cumulated peak
Vrms = .292 volts
Pk/Av = 9.48 dB voltage

Figure 8. Vp =1.462 volts cumulated peak
Vrms = .418 volts
Pk/Av = 10.88 dB

Note: The above data is based on standard transmitter output compression -35 dB.

The results are surprising because the summed Pk/Av ratio increased only 1.42 dB rather than 6 dB for summing up the maximum peaks. The scope readings were taken after 5000 samples. The 8VSB signals were nearly equal in amplitude by noting the Vrms values above. The Pk/Av ratio were also very similar. The signals were summed through a well isolated hybrid so the conclusion here is that the peak sum of two 8VSB signals is noticeably less then anticipated.

This is good news for antenna systems, but as a check, the test was repeated several times using adjacent DTV channels and N-2 channel spacing with about the same results, give or take a few tenths of dBs. Only the beat pattern changed as expected. In addition to the channel spacing parameter, the test was run over night (12hrs) with the same result. More time will be necessary for further testing to determine if there is a long time function here that still might add up the peaks, on a statistical basis which will more than likely be in units of tens of years.

Passive and Active Combining Systems for N+1 and N-1 Assignments.

A classical approach to combing is shown below in Figure 9. This is a constant impedance combining system intended for N-1 assignment.

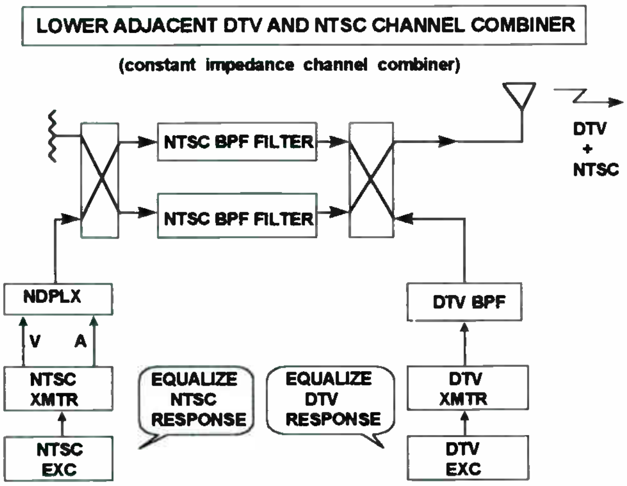

Figure 9. N-1 Combining System

The item to take note of is the requirement for transmitter response equalization on both the NTSC and DTV side of the combiner. This can be substantial due to the sharp tuned filters for the NTSC bandpass filter.

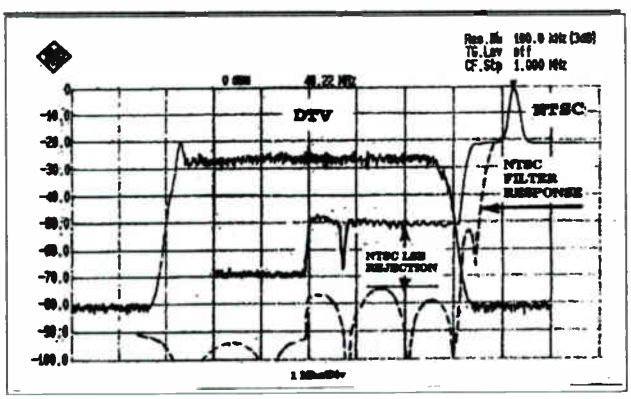

Figure 10. N-1 DTV Spectrum Below NTSC. The arrow pointing to the dotted line in Figure 10 shows the amount of NTSC lower sideband cutting. Also shown is the desirable suppression of the NTSC reinserted lower sideband.

About 150 to 250 kHz of the lower NTSC sideband is cut away to allow more room to combine in the DTV signal. The result of this sharp tuning and sideband cutting on the NTSC and DTV paths is shown below in Figures 11,12.

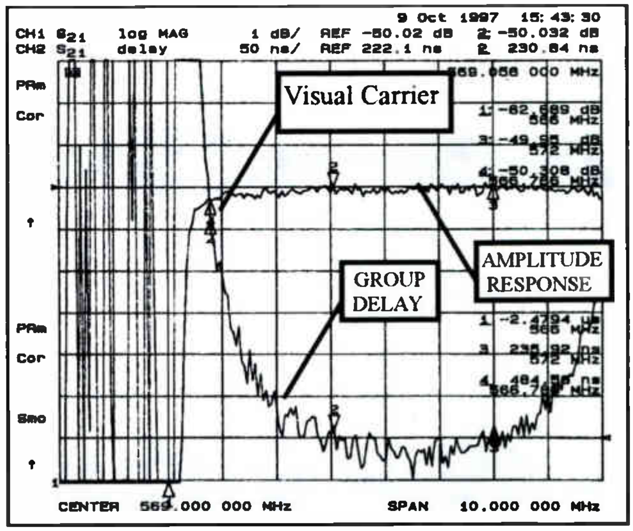

Figure 11. Measured Amplitude and Group Delay Response Through NTSC Path of 60 Kw N-1 Combiner. Plot scale, 50ns/div, 1 dB/div

The DTV response characteristics are shown below in Figure 12.

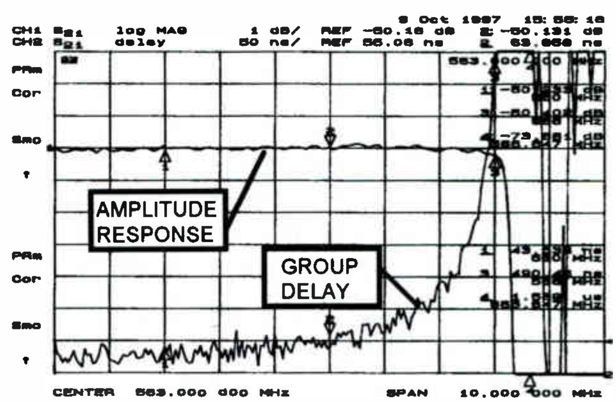

Figure 12 . Measured Amplitude and Group Delay Through DTV path, of 60 Kw N-1 combiner. Plot scale, 50 ns/div, 1 dB/div

The NTSC equalization task for the N-1 combiner can be readily see by looking at the composite VIT signal shown below in Figure 13. The significant 2T pulse distortion is the result of over 240 ns of group delay curvature through the NTSC lower sideband and carrier regions.

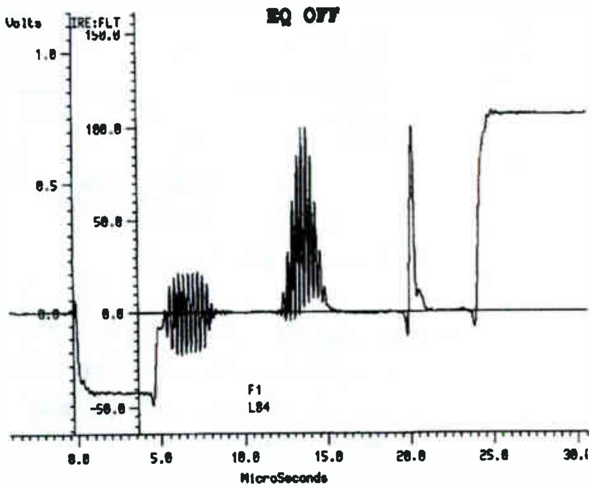

Figure 13. NTSC Composite signal showing 2T, 12.5 T pulse distortion N-1 combiner Equalizer off.

Figure 14 below shows the result of using a 6 pole active group delay equalizer at IF to correct the NTSC output to excellent performance. The same equalizer can be used on the DTV side.

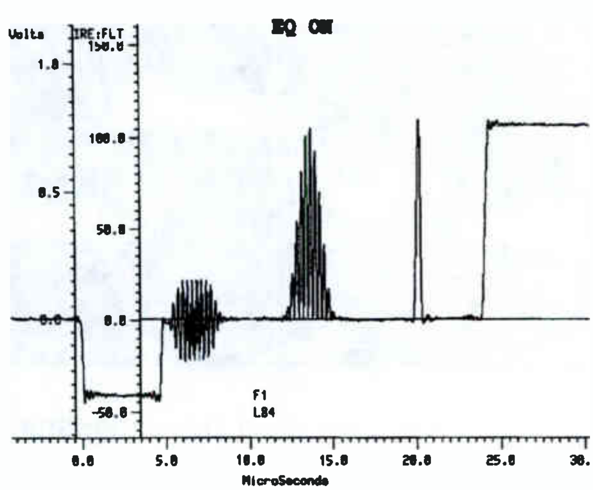

Figure 14. NTSC composite VIT signal at output of N-1 combiner. Equalizer on.

The N+1 combining requirement is a particularly challenging situation because of the very close frequency spacing between the NTSC aural carrier and the DTV pilot. The aural carrier is only .25 kHz below the upper channel edge and the DTV pilot is .308 above the lower channel edge. There is just not enough room to use tuned cavities for a reactive combining system.

However, certain antenna configurations will allow broadband combining of very close spaced channels., i.e., the N+1 assignment. This requirement can be done in a dual fed batwing antenna as shown below in Figure 15.

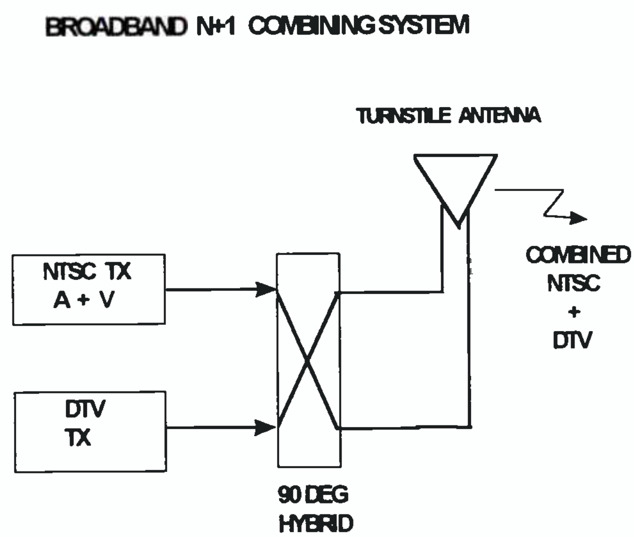

Figure 15. Broadband Batwing antenna

The batwing turnstile characteristics requires a dual line feed system with a 90 degree phase shift but this can be easily achieved by using a broadband quadrature hybrid.

The system consists of feeding diplexed NTSC aural and visual signals into one port of the hybrid as shown above in Figure 15 and feeding the DTV signal into the other port. This works basically on the combining and isolation of the hybrid and the isolated, independent feeds to the antenna. The antenna does the final combining on a broadband basis.

Figure 16 provides a functional view of the batwing antenna system.

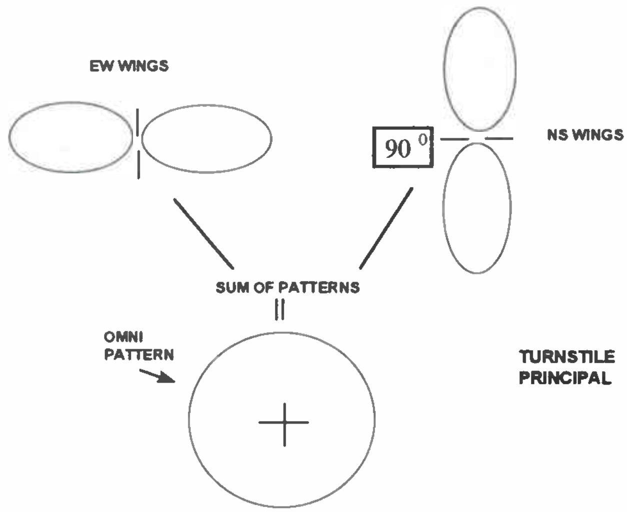

Figure 16. Batwing functional concepts.

The batwing antenna consists of two sets of independent dipole arrays to provide the figure eight patterns shown above. The dipole arrays are mounted at right angles to each other to position the pattern of the E/W array in the null portion of the N/S array . This provides the isolation for combining signals applied to the overall antenna. A 90 degree phase shift, provided by the hybrid causes the two figure eight patterns to form an omni pattern.

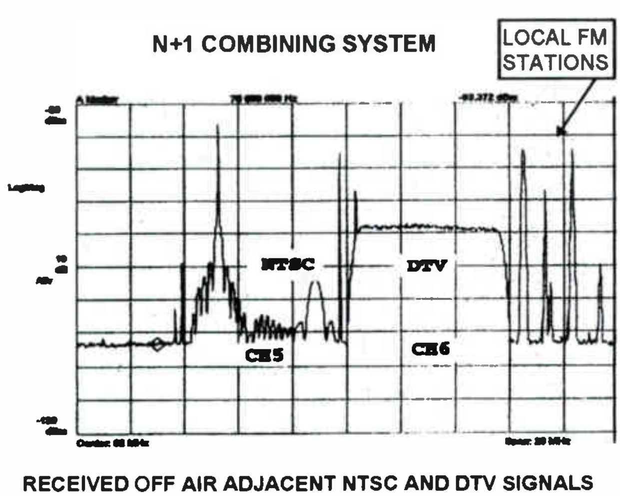

Figure 17. Measured Results of Combining NTSC and N+1 DTV radiated from a VHF batwing antenna.

Active Combining Using a Single PA

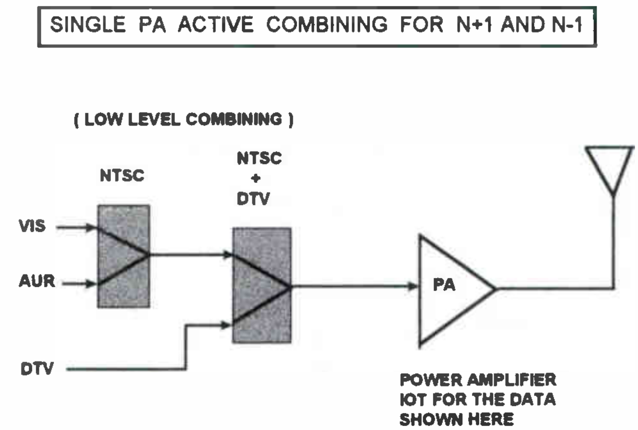

Figure 18. Active combining System

A very simple approach to combining either N+1 or N-1 assignments is to low level combine NTSC and DTV signals together first before applying the composite signal to the PA, as shown above in Figure 18.

This method is simple enough, however, there are a number of artifacts. To begin this analysis, see Figure 19 for a spectrum view of an IOT transmitter tuned for 12 MHz and amplifying simultaneously NTSC and DTV .

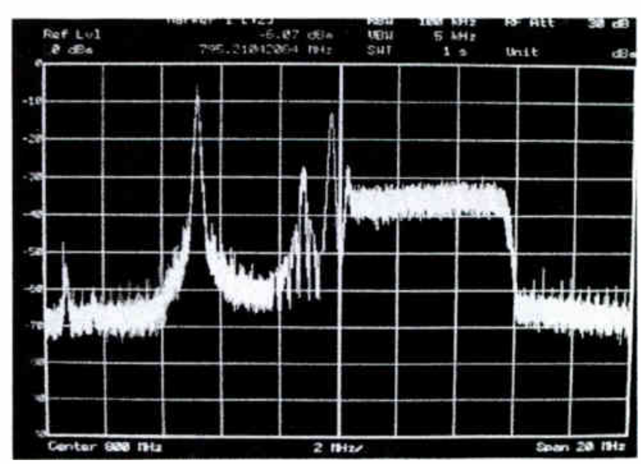

Figure 19. IOT Transmitter Output Spectrum Amplifying NTSC shown on left and DTV shown on right.

The NTSC performance in this combined mode is shown next in Figure 20.

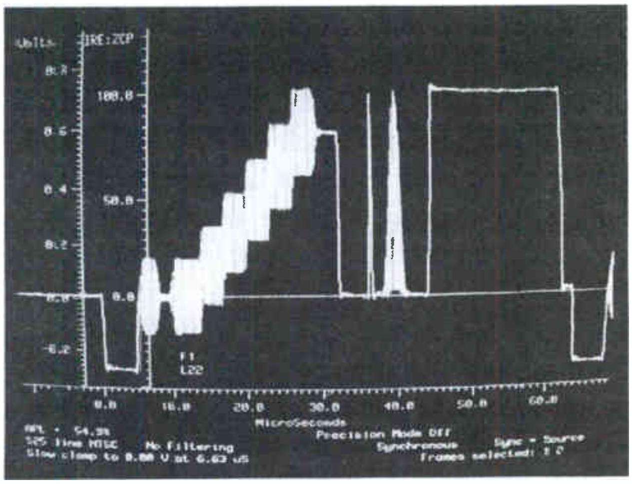

Figure 20. IOT output, tuned for 12 Mhz with DTV off to see NTSC performance with 10% aural on.

The peak of sync of power was about one half of the normal NTSC power, with 10% aural, of that for a 6Mhz tuned IOT transmitter. The performance was quite acceptable as shown above in Figure 20.

One of the surprising artifacts that came out under testing when the DTV signal was added to the NTSC aural and visual signal, was the significant increase in video noise. To observe this, the following waveforms were used.

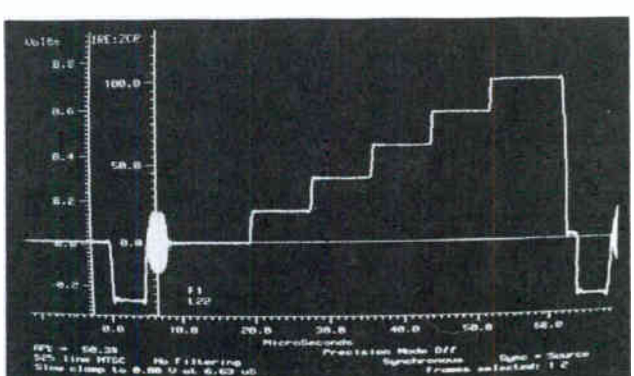

Figure 21. IOT output aural on, DTV off.

Figure 21 shows a good clean signal where the detected video S/N was -56.2 dB.

When the DTV signal was added in at 6% NTSC sync power the noise dramatically came up. See Figure 22 below.

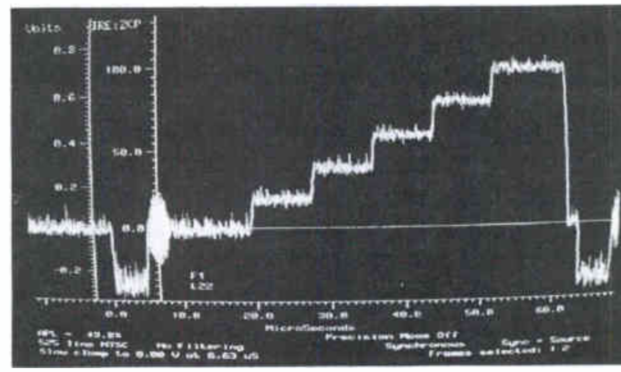

Figure 22. IOT output 10% aural on and 6% DTV on, detected video S/N = -31.5 dB.

The video noise appears to be the result of the DTV signal mixing with the aural signal and wrapping around it to put the unwanted IMD spectrum into the visual signal. This appears to be a very sensitive mechanism since a variation of different aural, visual and DTV power ratios were tested in an attempt to find a "sweet spot". Only about 6 dB improvement could be obtained on the above video S/N ratios by reducing the aural or DTV power about 3 dB.

The N+1 active mode of combining produced a new birdie at .56 Mhz in the visual path due to the aural mixing with the DTV pilot. The low side N-1 was a bit better but it produced an increased level of out of band products below the DTV channel

In either case, active combining will require a significant amount of output filtering to attenuate out of band products below the FCC mask for components 6 Mhz above or below the band.

Broadband Solid State Combining for N+2

The following is shown here as an experiment to see the effects on two DTV signals, N+2 spacing, amplified together in a single amplifier whose characteristics closely follow the non-linearity of a high power solid state PA

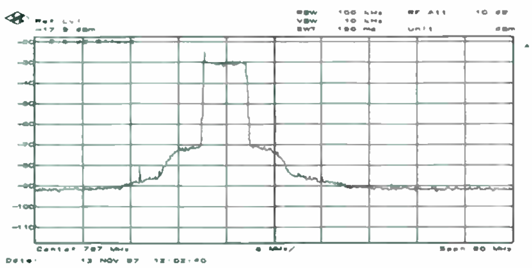

Figure 23. Channel 67 output set for power level that produced IMD levels at -40 dB.

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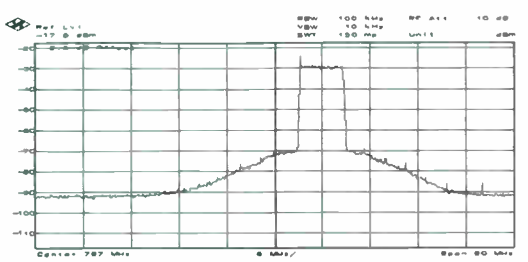

Figure 24. Channel 69 output set to power that produced IMD level at -40 dB.

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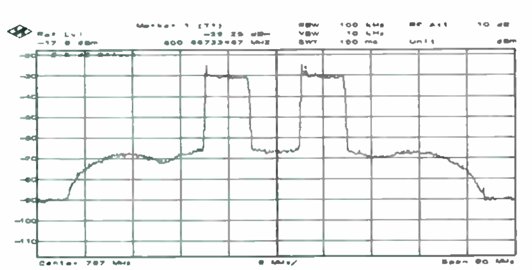

Figure 25 Output spectrum of two active combined DTV signals showing -35 IMD.

The key item in this experiment was to get an idea of power levels to determine the practicality of this approach. To do this, a single carrier was increased in power to reach the FCC spectral spread limit of -35, without linearization, then noting its power level, add in the second DTV signal and adjust the aggregate level until the FCC IMD level is again -35 dB on each carrier. The individual power can then be measured in the dual operation mode and compared to the power level in the single mode. This was about -3.5 dB while the total power remained the same. This says the individual carrier power level, in the combined mode, will be about 45% of that in the single amplified mode. This is an interesting result for high power, broadband, solid state PA's to further investigate. This mode of active combining has some practical aspects, while noting that special output filtering will be required to remove the unwanted spectrum components.

Conclusions

1. The voltage waveform of an 8VSB signal shows a higher peak/average ratio than a cumulative distribution curve.

2. The higher voltage peaks are simply compressed in the PA without serious effects except for out of band spectral components which must be limited to fit under the FCC spectral mask (-35dB).

3. Active combining of N+1 and N-1 systems causes a significant increase in video noise.

4. The sum of two equal level DTV signals will double the average value and increase the peaks about 1.5 dB while still holding the possibility the maximum peak could reach 6 dB.

Understanding & Testing the 8VSB Signal

Linc Reed-Nickerson
Product Development Manager
Tektronix, Inc., Beaverton, Oregon

Abstract

This paper looks at the 8VSB signal in detail. Starting with the interface to the exciter. We look at data randomizing, Reed-Solomon Coding, Convolutional encoding, Trellis encoding and Viterbi Decoding. Each scheme is described in a simplified matter so the audience will have an understanding how each works. We then explore why a pilot signal is used, and how segment sync and the training sequences add to the robustness of the signal. Spectrum compatibility with System M NTSC is discussed as well as why closed loop correction is required in a transmitter. In conclusion the paper will address the 8VSB measurements required to assure optimum transmitter performance, how the measurements should be made, and what is acceptable performance.

Introduction

When you've attended the National Association of Broadcasters Convention over the last decade you've watched the interest in HDTV peak and wane. You've seen technology changes and name changes, the adoption of a standard and now the mad rush by broadcasters and vendors alike to meet the FCC deadlines for the debut of service.

This year many attending the convention are here to make decisions about equipment and to learn as much as we can from our peers, vendors, and technical sessions to determine how best to proceed across the chasm into the emerging technologies of DTV.

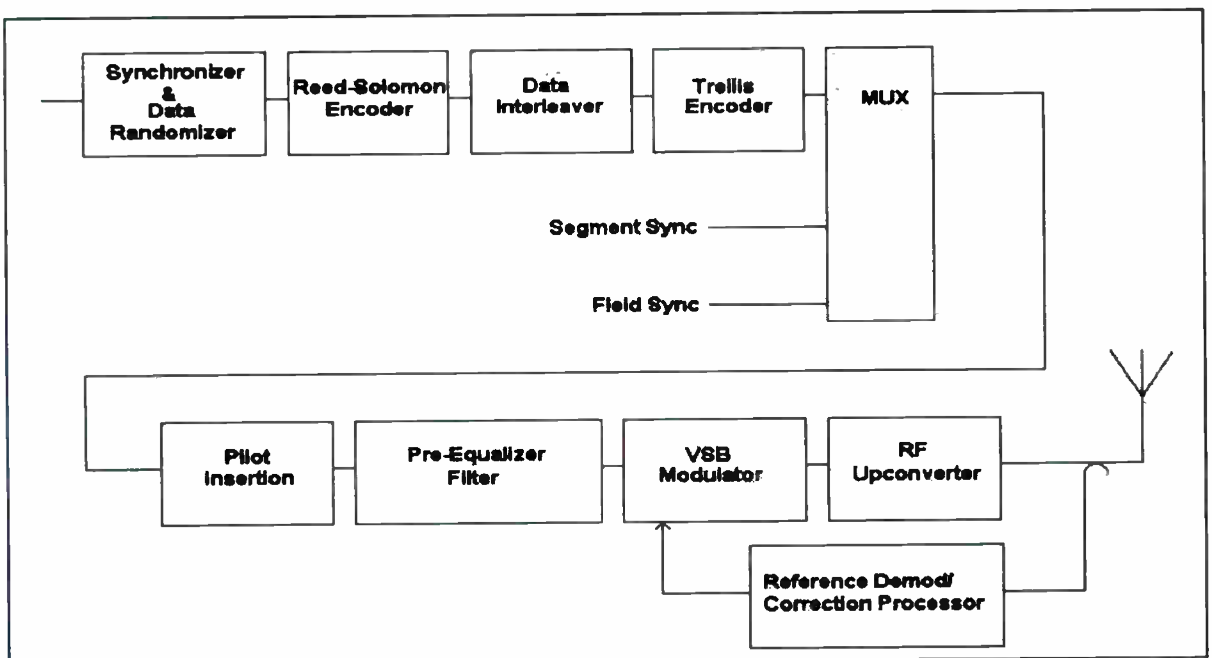

Figure 1. Block Diagram of an 8VSB transmitter.

This paper addresses a single facet of the many details we must concern ourselves with during the transition to digital, the 8VSB transmitter. At the time of this writing the United States, Canada, Korea and Taiwan have adopted the ATSC standard, and it is likely all of North and South America will choose 8VSB. Australia is seriously considering 8VSB, and there is interest in Asia as well. DVB-T, a transmission scheme based on Coded Orthogonal Frequency Division Multiplexing (COFDM) will likely be adopted by all of Europe. COFDM is very complex, using either 1705 carriers (known as "2k"), or 6817 carriers (known as "8k"). These carriers are orthogonally spaced, each carrying a portion of the data at a very low symbol rate.

As this paper is concerned only with the 8VSB transmitter we will start at the signal input. The digital signal fed to the exciter of an 8VSB transmitter is referred to as the "Grand Alliance" (bit) stream or DTV Transport Layer. The Transport Layer contains MPEG-2 encoded video, Dolby ™ AC3 audio and data. A bit rate Of 19.39Mbit/Sec is used and the data is sent in a stream of 188 byte data packets. Each packet starts with a sync byte, followed by a byte packet header (information about the packet), an adaptation packet of varying length, and the data payload. The packet length was chosen for optimum coding performance in the exciter.

The Transport Layer may contain a single HDTV channel, multiple standard definition channels, and data. TV receivers will automatically identify and decode the signal into the appropriate formats.

Synchronization to the incoming data stream is the first thing that takes place in the exciter. This enables the exciter circuits to identify the beginning and end of the 188 byte data packets. At this point the sync byte, which is the first byte in the data packet is discarded, to be replaced by the 8VSB segment sync in the MUX before being transmitted.

Following synchronization the signal, now 187 byes in length, is conditioned by Data Randomizing. The purpose of data randomizing is to guarantee a flat, noise-like spectrum. The data randomizer assures that if the input stream is lost, resulting in long streams of 1's or 0's, or if a number of high power symbols occur in a row the transmitter will not output a signal that would cause interference into NTSC channels. Data Randomizing is an FCC requirement. The data randomizer uses a known pattern of pseudo-random number generation to change the byte values, the process being reversed in the receiver. Data randomizing also provides for optimum performance in the receiver recovery loops.

After randomizing the data is sent to a Reed-Solomon coder. Reed-Solomon coding is a Forward Error Correction (FEC) scheme operating as a byte-wise encoder. Irving S. Reed and Gustave Solomon developed their original coding scheme in 1960. Today Reed-Solomon Coding is used in Compact Discs, Hard Drives, Telephone modems and Digital transmission systems.

At the Reed-Solomon encoder 20 parity bytes are added to the 187 byte packet that will allow the decoder to identify and correct errors. Reed-Solomon coding works best for short "bursty" errors that may be caused by noise, brief signal fades or transmitter non-linearities. For the 8VSB signal the number of errors Reed-Solomon coding can correct is 10 byte errors per packet.

In terrestrial broadcasting it is probable that bursts of noise will occur that are longer than the Reed-Solomon decoding can handle will occur. To protect the R-S coding the data is interleaved, meaning bytes are no longer in order consecutively. A very simple example of interleaving is show in figure 2.

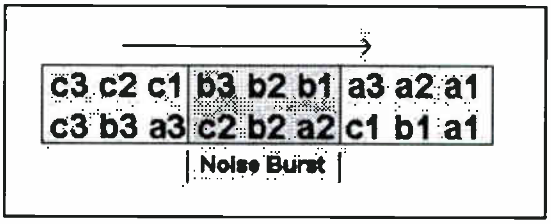

Figure 2. A simplified data interleaver, the top line is an example before interleaving, the bottom line is shown after.

If all bytes are transmitted consecutively a noise burst (the shaded are in figure 2) could be long enough to obliterate all the "b" bytes, but with the interleaved data only one "a", one "b", and one "c" byte would be lost leaving enough data for the decoder to reconstruct. This a very

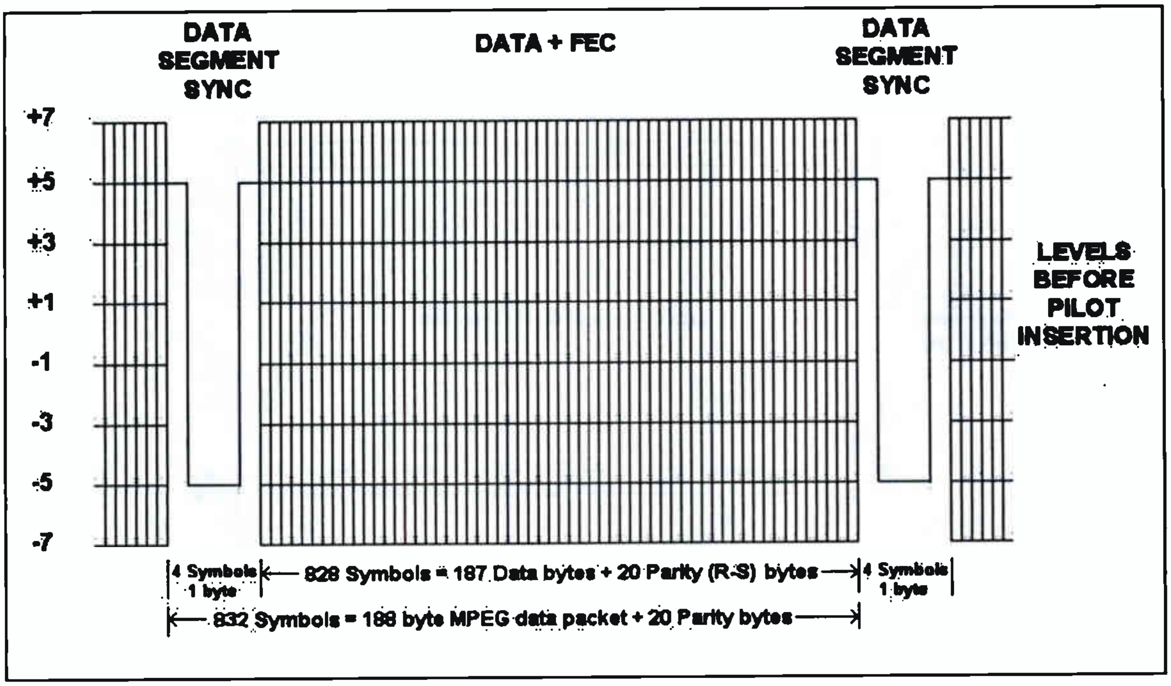

Figure 3. The 8VSB signal showing the segment sync, data, and levels.

simple example; the data for 8VSB is interleaved to a depth of 52, which allows for corrections to data that has been covered by a 193 μs noise burst.

Another form of Forward Error Correction is called convolutional coding. In 8VSB Trellis encoding is used in the transmitter. The complimentary decoder in the receiver is a Viterbi decoder. These convolutional coding and decoding schemes are most effective in handling white noise.

Trellis Coding has gain, which is expressed in dB, but there is a trade off, the complexity of the system. Each bit going into the coder, goes into a shift register that produces 2 bits at the output. There are a finite number of shift registers that produce a finite number of input/output combinations. In 8VSB, out of the 8 possible states in one interval, only 4 will be valid. Which 4 points are valid is dependant on the previous intervals. If the valid state changes are diagramed, the results resemble a garden trellis, hence the name Trellis Encoding or "Trellis Coded Modulation."

16VSB which was proposed for Cable Television broadcast differs from 8VSB by not using convolutional coding. This allows for a higher data rate (38.57 Mbps) but less transmission robustness (28.3 dB for 16VSB vs. 15.8 dB for 8VSB)

The Viterbi decoder not only brings the data back to the original state but is a powerful error correction scheme. The simplest analogy would be to compare a Viterbi decoder to an accomplished musician listening to a piece of music. It is likely the musician can detect a "sour" note, and replace it with the correct one. By looking at the data over time Viterbi decoding can recognize an invalid transition and will replace it with the most likely correct path (state transition). A Viterbi decoder is called a maximum likelihood decoder.

The encoded signal is now sent to a Data Multiplexer where the supplementary sync signals are added to form the 8VSB baseband signal. The supplementary signals aid in signal acquisition, timing and level identification. The 8 in 8VSB indicates the 8 discreet levels, each level having a symbol value. Every 828 symbols two level binary data called Segment sync is added. Segment sync is 6 data levels in amplitude and 4 Symbols long, making each 8VSB Data Segment 832 symbols in length. Segment Sync has a dual role, it replaces the

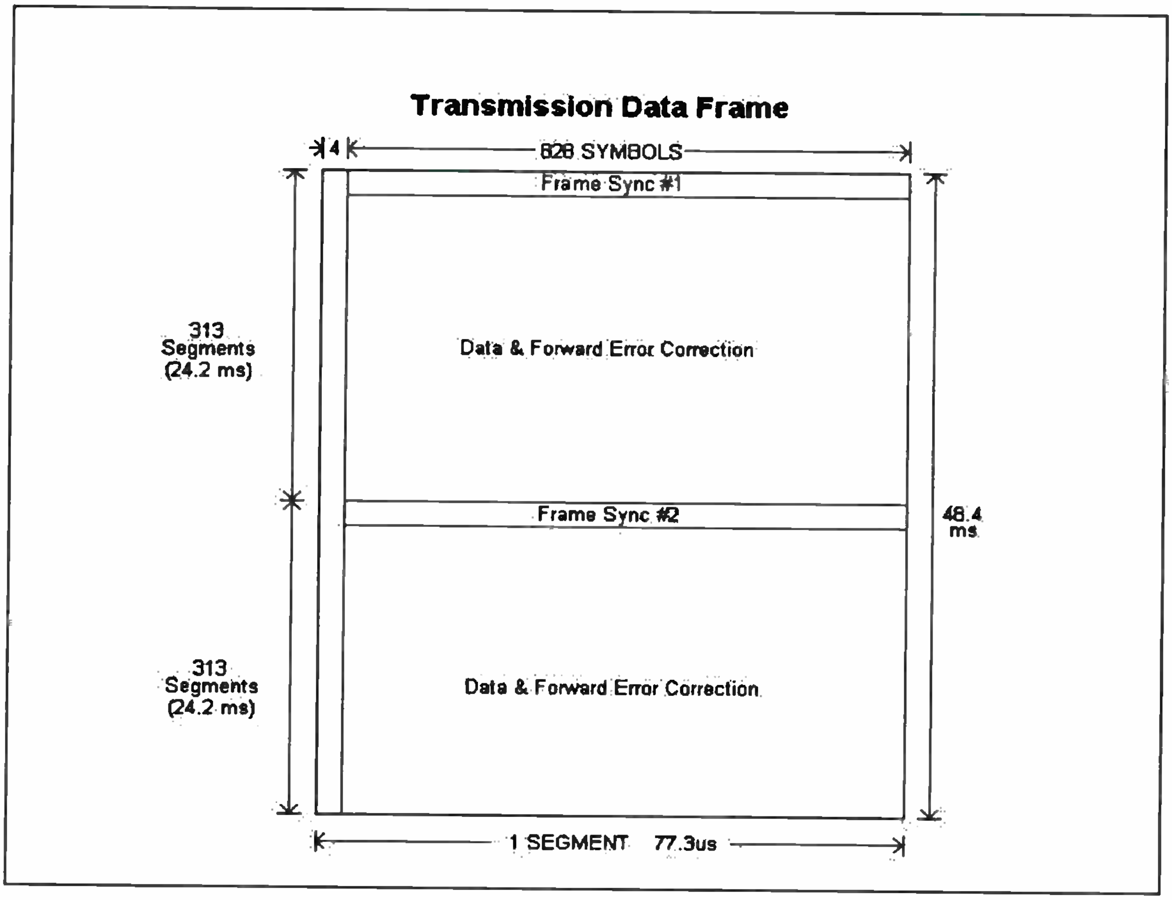

Figure 4. An 8VSB Transmission Data Frame. Showing the relationship of Frame Sync & Data Segments

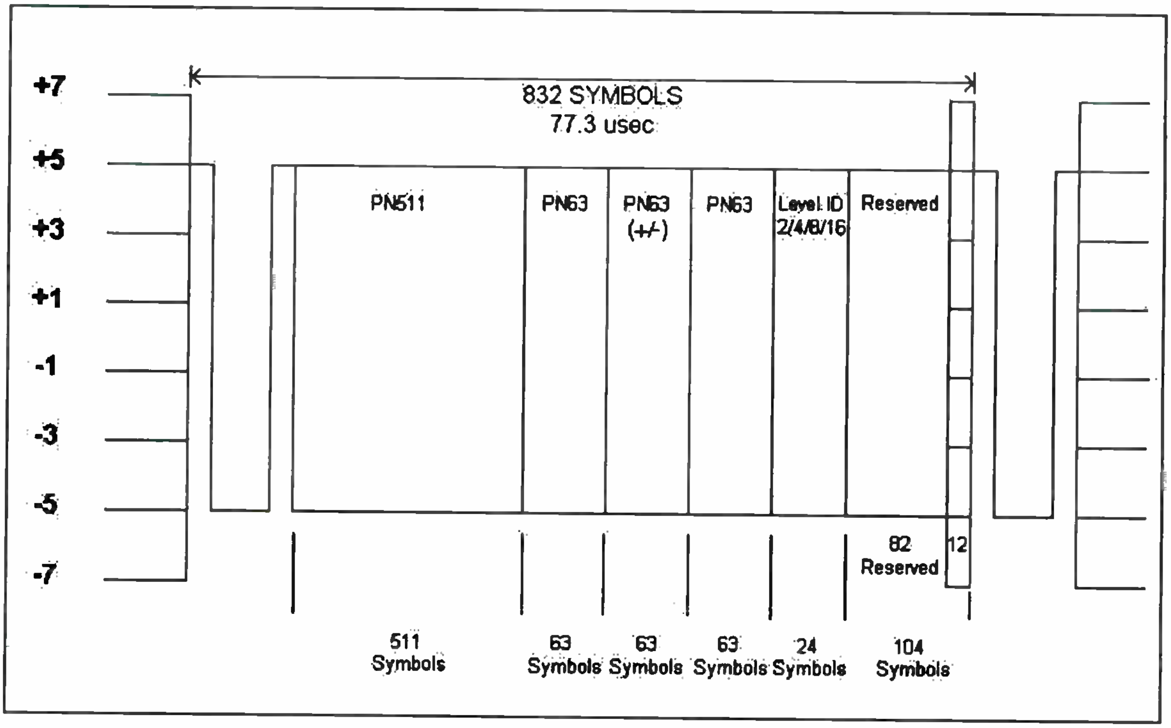

Figure 5. Frame Sync Segment, showing levels and training sequence

sync byte that was removed prior in the synchronizer, bringing the data packet length back to 188 bytes (828 symbols). The high amplitude of segment sync provides robust sync decoding in the presence of noise.

The 8VSB Data Segments are now assembled into a Transmission Data Frame which is 616 segments (48.4 ms) in duration. At the beginning of the Transmission Data frame, and repeated after 312 Segments a Frame Synchronizing segment is sent. The Frame Sync segment carries the training reference signals for the receiver equalizer. It consists of the 4 byte segment sync, followed by 511 reference symbols which the receiver uses for adjusting long equalizer taps, three sets of 63 reference symbols for short equalization, 24 symbols for VSB level ID, 82 reserved symbols and 12 symbols that are repeated from the previous segment.

Some similarity to NTSC is apparent when looking at the 8VSB signal. The 8VSB segment sync, like the analog sync pulse, provides a high level data pulse to help the receiver achieve initial lock in the presence of noise or multipath. The training sequences (PN511 and PN63) set the taps in the receiver equalizing circuits to deal with multipath and other signal aberrations, and the Level ID provides a DC reference to enable the decoder to determine the level for each symbol.

Following the Data Multiplexer a DC offset is added to the baseband signal; this constant offset voltage generates the pilot signal in the modulator. 8VSB is transmitted as a modified single sideband surppressed carrier signal with the pilot added. Unlike System M, where the visual carrier is a very large part of the signal, the pilot consumes only about 7% of the transmitted power.

The baseband signal is then split and passed through a root-raised cosine filter and converted to an analog signal by a high-speed D/A converters. The signals are then input to two mixers, which are phase shifted by 90 degrees. The output is a 44 MHz IF signal, upper sideband only, with root-raised cosine response. The signal is now ready to be up-converted to the channel frequency and amplified in the final stages of the transmitter.

DTV transmitters require a closed feedback correction loop not often found in the analog world. Analog transmitters were very forgiving during times of antenna icing or transmission line problems; typically output power was reduced until the event was over with little or no effect on coverage. With the DTV transmitter closed loop pre-distortion is a requirement and is being provided as part of the transmitter. Analog closed loop correction was a feature of the Tektronix 1440 which was used by some stations with older transmitter to improve remote control operation. Pre-distortion for the DTV transmitter will be required to maintain the desired operating parameters. A separate method of verifying performance in the form of a precision test set is highly recommended.

Operating Considerations

Now that there is an understanding of how the signal is generated, lets look at the reason for the pilot, segment sync, and the training sequence. The receiver looks for the pilot to phase lock to the incoming signal. Once phase lock is achieved the decoder looks for segment sync to achieve an initial data lock. Even a fairly impaired signal can be phase locked and segment sync located. Now the decoder looks for the training sequence. The training sequence provides the information to the decoder that it will use to set the equalizer taps in the receiver to achieve flatness in the incoming signal. At this point the signal has been condition to a state that allows it to be decoded back into the 19.3 MHz DTV Transport Layer. The pilot and sync signals not only help the receiver lock initially, but will aid in maintaining lock should a disturbance occur, such as a log burst of impulse noise or airplane flutter. In many cases the signal may only lose data lock, but keep pilot and sync lock that aid in quick recovery.

Unlike analog signals that degrade gracefully, digital signals are subject to the "Cliff Effect." An analog signal will often remain viewable even while severely degraded, a DTV signal has two states, a near perfect picture or nothing.

Viewers who have been tolerating marginal picture quality because of poor antennas, low signal, or noisy reception may find they don't receive a digital picture at all. Those viewers living in the Grade B contour and fringe areas may find they have picture one day and no picture at all the next. It will be important to educate the viewer on what must be done to have satisfactory reception. Bob Plonka of Harris has recently published a paper on circularly polarized transmitting antennas for DTV. Circular polarization may make a significant difference in DTV reception for those using Rabbit ears and unipoles. Further work is being done to improve receiver performance; Nikhil Deshpande of Tektronix has suggested that equalization techniques can be improved in the receiver that could significantly improve performance in the presence of multipath.

With an analog transmitter a number of faults could be tolerated with the very little effect visible by the viewer, even when the fault was fairly severe. With DTV, a transmitter that is experiencing a linearity problem, for instance, could mean the loss of a significant number viewers in the Grade B coverage area, and those closer in that have marginal receiving conditions.

Transmitter Monitoring

It will be very import to continuously monitor a DTV transmitter with a measurement set capable of providing an alarm when there is signal degradation. The only FCC requirement for DTV that we don't interfere with other stations or other services. Out-of-Band emission testing is required to be certain there is no leakage into adjacent channels.

While we still measure the signal-to-noise ration with DTV we look at it in a slightly different way. We measure the "Desired to Undesired" signal ratio. Desired is, of course, the signal of interest, the pure output of your transmitter. Undesired is any other signal or noise component that does not belong there.

Flat frequency response across 6Mhz has required that we "broadband" an analog transmitter. Just like analog, the first step in setting up a DTV transmitter is tune for flat frequency response and group delay, and no leakage into adjacent channels. With an analog transmitter the effects of group delay usually resulted in Chroma/Luma delay, which if serious could degrade the picture, but the picture would remain be viewable. Group Delay inequity in a DTV transmitter will result in intersymbol interference (ISI). ISI will cause the Bit Error Rate to rise, and many TV sets may drop in and out of lock. Even low levels of ISI may cause TV sets that were operating near the edge of the cliff to lose picture completely.

Amplitude and phase nonlinearity can cause similar problems, and the worse they become the more viewers are lost.

Eye Patterns and Bit Error Rate have become buzz phrases with digital signals, but they may not be the best way to monitor. These measurements don't provide the best information about system health. What is needed is data about signal degradation early enough so that correction can be made before viewers are lost.. Observing the Constellation Diagram and making Modulation Error Ratio measurements will provide early detection of system problems. A rise in BER means you are already transmitting a flawed signal. A Constellation of nice tight vertical dot patterns with no slanting or bending indicates proper operation. You can learn to interpolate the constellation and a glance, and a change in transmitter health will be quite obvious.

If you continuously monitor the Modulation Error Ratio you will see indication of degraded performance before BER is excessively affected. In most cases you will have time to correct a problem before it turns into lost viewers. A good 8VSB measurement set will allow you to continually monitor your transmitter and alarm on both cautionary limits and outright failure.

DTV brings both opportunity and a new set of problems. Opportunity in the ability transmit HDTV pictures, multiple standard definition images, and data services. Picture and sound quality are better than the viewer has had with analog NTSC. In order to sell the viewer on DTV/HDTV the stimulus to the eye and ear are of equal important.

If DTV is to succeed it is paramount that the broadcaster educate the viewer help him or her to get the most from their investment in DTV. Part of that challenge is to be certain your transmitter is providing the optimum signal performance.

Recommended Reading

There are two books that provide interesting reading about the development of standards for television in the United States.

Joel Brinkley, "Defining Vision - The Battle for The Future of Television" (1997, Harcourt Brace)

Brown, George H., "Part of Which I Was - Recollections of a Research Engineer" (1982, Angus Cupar)

Brinkley's book chronicles the science and political science of developing the ATSC Digital Standard. Brown's book describes in detail the struggle to adopt the NTSC color standard , and the battle between RCA and GE over the color wheel.

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DIGITAL ADAPTIVE PRECORRECTION (DAP): A MUST IN DIGITAL BROADCAST TRANSMITTERS

Brett Jenkins
Comark Communications, Inc.
Southwick, MA

Gérard Lemaître
Thomcast
Conflans Ste Honorine, France

ABSTRACT

New digital television standards, both the ATSC and DVB-T standards, use sophisticated modulation schemes allowing the transmission of high data rate streams in a relatively small, fixed channel bandwidth. These very efficient modulation schemes subject systems to a high random peak to average power ratio which requires a Digital TV Transmitter to be designed very carefully with respect to non-linearity effects. Indeed, the intermodulation caused by non-linearities will produce in-band products, which will degrade the END (Equivalent Noise Degradation) or the EVM (Error Vector Magnitude). Another effect is the generation of out of band products (or "shoulders") which must be limited in order to prevent interference with signals in adjacent channels. Not only do these digital modulation schemes have a high sensitivity to non-linear distortions, but they are also harmed by linear distortions. In the past with analog systems, these distortions have been corrected with manual adjustments performed while looking at suitable test signals inserted into the analog signal in order to precorrect the signal. It is clear that in the absence of such test signals in the digital world, there is a great need for a stable and adaptive process of correction that avoids the need for tedious periodic alignment.

INTRODUCTION

It is now well recognized that digital television standards impose some very difficult constraints on a transmitter due to their intrinsic characteristics and the level of performance required for fitting them into an already crowded spectrum. The causes and the effects of the distortions in a High Power Amplifier (HPA) have been presented in numerous papers as well as several candidate correction schemes. This paper will first review some of the most important constraints of these new standards. [1] Then it will explain why the DAP is the best solution for digital transmitters and what advantages it can bring to a transmitter system. Finally, after an overview of the basic DAP principle and the integration of the DAP in a transmitter, some examples of the results will be presented.

TRANSMITTER CONSTRAINTS IN DIGITAL TELEVISION BROADCAST

The basic difficulty of broadcasting digital TV is making sure that the signals are amplified linearly. This requirement is complicated by the following characteristics:

- These new digital standards, both the ATSC and DVB-T standards, have a high random peak to average ratio. This is the price to pay for a high spectral efficiency. The consequences of this are a high sensitivity to the non-linear and linear distortions of the HPA and a large Output Back-Off (OBO) implying poor efficiency and an expensive amplifying structure.
- The digital standards require significantly better out of band product performance than analog TV. The requirement for digital television can vary from 35 to 50dB shoulders [2] [3], compared with 20dB for analog TV (regenerated sideband) [4].
- The coverage of the digital TV standards are subject to the so called "cliff effect". This means that the program at the edge of the coverage area can be completely lost with only a small drop in the Carrier to Noise ratio (C/N). Even a drop as little as 1dB can make the difference.
- These standards do not offer any integrated test signal like the test lines in analog TV which facilitate the correction adjustment and help to verify that the correction is adjusted at its optimum point.

Classical analog correction is certainly able to perform some correction of digital TV standards with a specific and careful design (full band), a long period of tedious and time consuming adjustments, a high level of operator expertise, and all this without the assurance that you will end up at the optimum correction point. Furthermore if we want to improve the performance without degrading the efficiency, this requires a more sophisticated (i.e., more expensive and complicated) correction capability. Taking all this into account, the analog process appears insufficient due to its inherent limitations, the limited approximation of the complementary curve (that is, limited by the typically few segments that can be implemented into an analog corrector), the difficulty to correct some HPA's transfer curves which have inflection points or sharp slopes, and the sensitivity to varying environmental conditions.

ADVANTAGES OF THE DIGITAL ADAPTIVE PRECORRECTION

Improved performance and efficiency

The DAP yields better performance and efficiency since it is able to automatically and (nearly) perfectly correct for any type of non-linear and linear distortions. This allows you to lower the biasing for solid state or tube (IOT) amplifiers. The problem with doing this when analog correctors are used is that it produces significant distortions in the gain transfer curve as shown in Figure 1. The curves shown in the figure depict the influence of the quiescent on the gain/amplitude curve for a solid state UHF amplifier (in this example, LD MOS technology is shown). We will see the result of corrections on the worst case curve later.

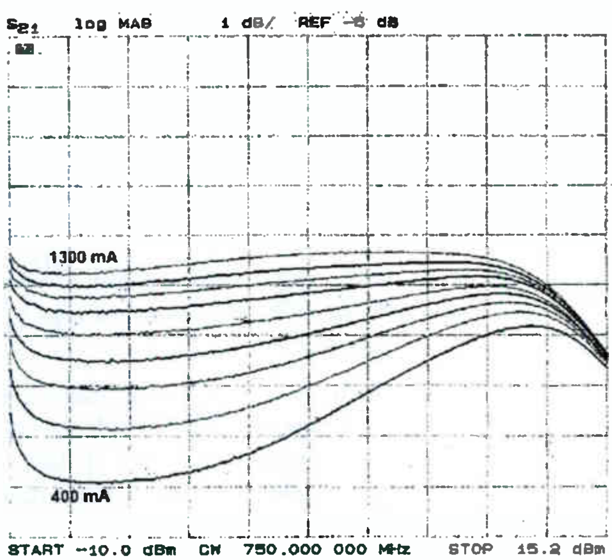

Figure 1 - *Examples of Gain compression curves of a double class AB UHF Solid State HPA (LD MOS) with biasing as parameter*

The DAP is able to correct for the peaks of the signal extremely far off on an oblique compression asymptote such as for the class AB HPA. Therefore it allows a significant reduction in the output back-off. Figure 2 shows the transfer curves of a real amplifier corrected by a DAP.

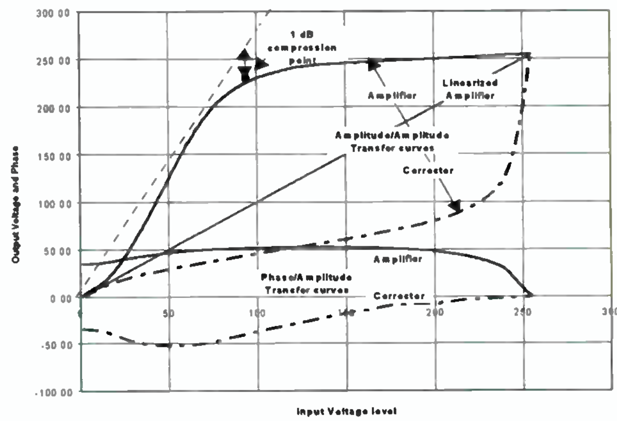

Figure 2 - *An example of a transfer curve corrected by DAP*

In this example, we can see that the HPA has been linearized about one dB beyond the amplifier's 1dB compression point. This allows the peaks of the signals to be transmitted without distortion, which means without in-band or out-of-band products. This represents a gain of about 20% in the output power of this transmitter.

Guaranteed performance with no required skill

The DAP basically performs both linear and non-linear equalization without human intervention, in a manner which maintains stable output performances in the face of varying environmental conditions (temperature, aging, AC mains fluctuation, etc.). The correction is based on the analysis of the transmitted signal itself without using any type of ancillary (training) test signal or pattern. So the "on air" signal is continuously monitored and corrected.

The stability and precision of digital processing

The DAP takes advantage of the well known benefits of the digital technology. Operations in the digital domain can have better precision and increased stability compared with equivalent analog techniques. Also, there is no need for internal alignment of the various circuits.

Multi-standard and multi-technology solution

This digital precorrector can be used in solid state transmitters or tube (IOT) type transmitters with 8VSB or COFDM TV standards. So as shown in Figure 3, the DAP is a multi-standard and multi-technology solution. This allows users to have a measure of uniformity across various transmitter networks which might have varied requirements for power levels. It also allows the transmitter delivery times to be accelerated as product designs converge on a single solution.

Digital Exciter

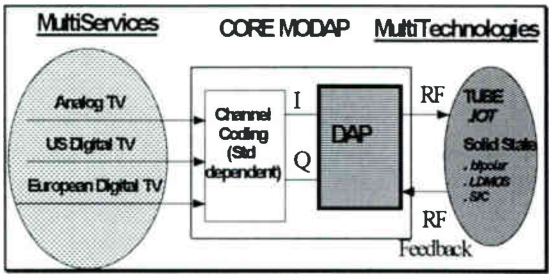

Figure 3 - *DAP application and environment*

Ability to correct linear distortions

In addition to the non-linear distortions, the linear distortions caused by the components following the HPA (e.g., cavities, output filter, RF combiner) may have to be corrected, depending on the standard and the quality required. Certainly, for the best possible coverage, this is the case for the ATSC transmission system which is particularly sensitive to linear distortion. Any linear distortion produced in the transmitter will tend to use up the receiver's capability to compensate for the terrestrial channel. The DAP includes an option, the Adaptive Linear Equalizer (ALE) which can restore a flat frequency amplitude response and a linear phase response at the output of the channel filter, regardless of the amplitude and group delay ripples of this filter within the useful bandwidth. Depending on the need and the configuration of the system, these linear defects may be compensated with adaptive or fixed correction.

Maintenance

From the maintenance point of view, this process allows the customer to change an amplifier module or setting without any need to readjust the correction. The correction will be automatically performed, no skill required. This should dramatically reduce the cost for stations to maintain their transmitters.

Reliable service

As a result of its adaptive feature, the DAP allows the signal to be corrected even in the presence of minor failures in the transmitter (for example, the failure of some of the transistors). Even in these failure modes, the quality of service is maintained. And remember that in the digital world, quality means quantity as even a small change in distortion can result in complete loss of coverage at the fringe areas.

Coverage and implementation issues

The DAP's capability to dramatically reduce both "in-band" and "out of band" products allows not only to limit the Error Vector Magnitude (EVM) (or Equivalent Noise Degradation) in the transmitter, thereby increasing coverage, but also to facilitate transmission using an adjacent channel. As most broadcasters are aware by now, the issue of adjacent channel broadcast has been a critical one over the past several years in the US. It is likely that this digital technology will enable us to successfully reduce interference to levels which are more acceptable to broadcasters.

IMPORTANT DESIGN CONSTRAINTS

As seen in Figure 3, the digital corrector is an interface between the output of the modulator and the input of the HPA. Its role is to transform the characteristics of this properly formatted and filtered signal in order to counter the effects of the HPA. The changes are made in such a way that a perfect signal is recovered at the output of the transmitter system (including the cavities for IOT transmitters and some RF components depending on the system). So the design of the DAP will mainly depend on the characteristics of the digital standard, some minimum requirements that the HPA must meet and the level of performance required at the output of the HPA.

Main characteristics of digital standards

The main parameters influencing the design of the DAP and the HPA are the output symbol rate and the bandwidth, see Table 1. In order to achieve a product which can handle any standard, the design must be based on the most difficult of these parameters.

Parameter	ATSC	DVB-T
Symbol rate	10.76 MS/s	9.14 MS/s
Useful Bandwidth	5.38 MHz	for 8 MHz 7.61 MHz for 7 MHz 6.66 MHz

Table 1 - *Main characteristics of the digital signals to be transmitted*

The output symbol rate fixes the minimum frequency required in the DAP. This frequency must be high enough to be able to produce the precorrected I and Q digital output components without any aliasing, taking into account the fact that the precorrection will extend the I and Q signal's original bandwidth. To allow for the most powerful processing, the corrector operates on the low pass representation of the bandpass signal. Recall that a band pass signal can be represented by a low pass signal using a Hilbert Transform [5]. This low pass signal can be broken down into its in-phase (I) and quadrature (Q) components.

$$x_l(t) = x_c(t) + jx_s(t) \qquad (1)$$

Each component is now band limited to 1/2 the bandwidth of the original bandpass frequency, but is centered at 0 Hz. The original bandpass signal is related to the I and Q components by this equation:

$$x(t) = x_c(t)\cos(2\pi f_0 t) - x_s(t)\sin(2\pi f_0 t) \qquad (2)$$

where f_0 is the center of the original bandpass signal. These I and Q components are the signals which are used in the DAP.

The bandwidth influences the so called "correction bandwidth" required for an efficient correction of the in-band and out of band products. This "correction bandwidth" depends upon the following constraints :

- the useful bandwidth of the signal
- the order of distortion to be corrected (increasing the useful bandwidth)
- the level of shoulder reduction required; this in turn depends on the level of shoulders without correction and the level of shoulders desired at the output of the transmitter
- the capability and complexity of the digital circuitry

HPA requirements

Despite the great potential of correction using DAP, the HPA must still meet certain criteria in order to fully exploit the corrector's potential. For example the HPA distortions must be nearly frequency independent. It must not present short term thermal problems (although in digital TV applications, this is not an issue since the average power is constant). Also, the amplitude and phase response must be flat within the correction bandwidth. Finally the entire system must be able to provide enough power to make the peaks of the expanded corrected signal.

Performance requirement

Out of band performance (shoulders): The desired out of band performance impacts the design by requiring a certain number of bits and a minimum frequency at which the circuits must work (i.e., the correction bandwidth). In Figure 4 we see the mask currently specified by the FCC for the 8-VSB signal [2], while Figure 5 shows the requirements for the DVB-T standards [3].

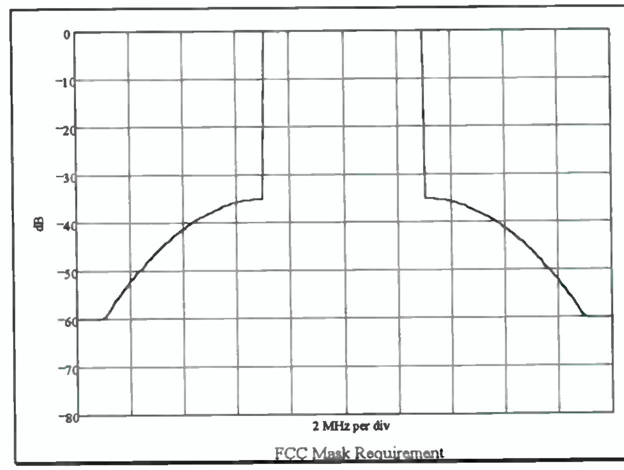

Figure 4 - *8-VSB spectral mask requirement*

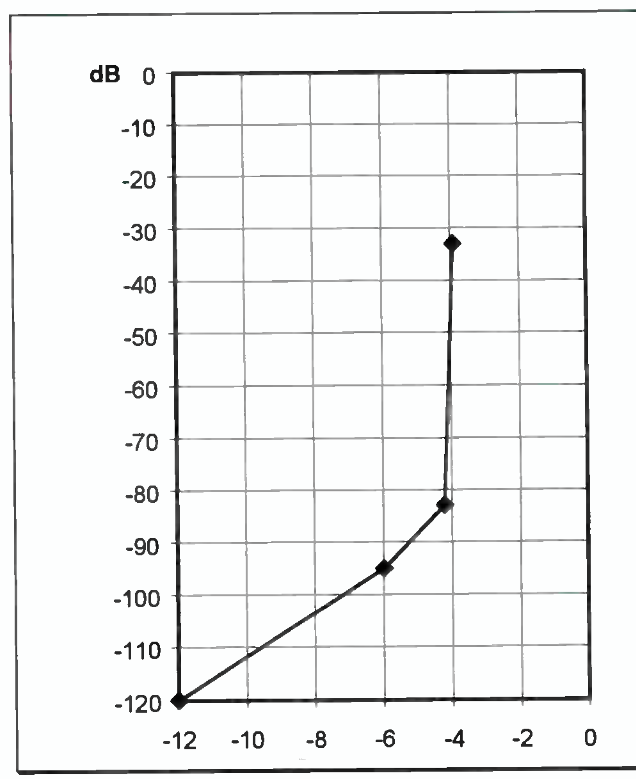

Figure 5 - *DVB-T spectrum mask for critical cases (power measured in a 4 kHz bandwidth)*

In-band performance (EVM or END) : The in-band products appear as additional noise within the signal bandwidth. This effect can be quantified by measuring the deviation of the BER curve f(C/N) from the theoretical curve. This method, used widely in Europe, gives a number called Equivalent Noise Degradation (END). In the US, a measurement which has become widely accepted is the Error Vector Magnitude (EVM). The EVM is sometimes expressed in dB and called SNR. The final result of degradation of these numbers is similar to a loss of useful power and so a loss of coverage.

The DAP offers a great flexibility for finding the best trade-off between both in-band and out of band performance and the useful output power. Depending on the application, it is possible to obtain the best performance with a given output power or to meet just the minimum performances but with some more output power. The ultimate goal is to get the best coverage area possible with a given transmitter system.

BASIC PRINCIPLE

Figure 6 shows the basic principle behind precorrection. The basic but essential function of the precorrector is to generate a perfect complementary amplitude/amplitude and phase/amplitude curve in such a way that the response of the precorrector and the response of the HPA cascaded together results in a perfectly linearized amplifier.

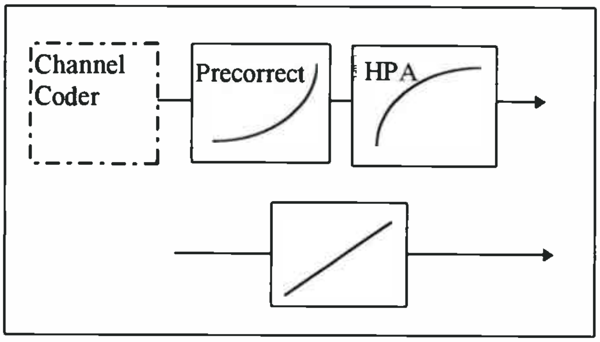

Figure 6 - *Basic principle of precorrection*

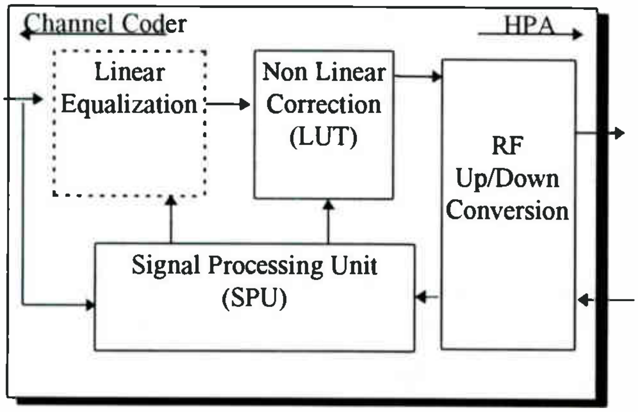

Figure 7 - *Functional block diagram of the DAP*

Figure 7 shows the general functional block diagram of the DAP. The linear equalizer is responsible for pre-distorting the forward signal in such a way as to cancel out linear distortions that occur further on in the transmitter system. It is essentially a complex digital filter whose coefficients are determined by the signal processing unit. The signal processing unit calculates new coefficients by comparing the input signal from the forward path with the reference signal fed back from the transmitter system output.

In much the same way, the non-linear corrector is made up of a large look up table. The values of the look-up table are updated from the signal processing unit. This allows both the phase and gain characteristic of the forward signal to be manipulated. The signal processing unit calculates the HPA's complementary curve. That is, it first finds the transfer curve of the HPA by comparing the input signal and the feedback signal. It then finds the inverse curve and feeds the look up table with new values. The result of pre-distorted signal passing through the HPA is a "perfect" signal at the output, with all distortions canceling.

This idea has been used in analog transmitting equipment for many years now. The difference between those analog correctors and the DAP is the methodology used to "tune" the correction. Because of the digital processing involved, the operator involvement is eliminated since the correction is now computed. Also, the computations are very precise, allowing for the overall result to be much more accurate.

The final piece of the DAP is the RF up/down-converter. It is here where the forward signal is finally converted to an on-channel RF signal. The other function of this section is to receive the RF feedback sample from the transmitter output and translate it to its complex low pass representation.

If the principle of such a process seems quite straightforward at a first glance, its success relies on a deep knowledge of the precorrection principles, a sound global transmitter system approach and a mastery of leading edge digital technology and simulation tools.

INTEGRATION WITHIN A TRANSMITTER

The DAP resides in a digital exciter as depicted in figure 8. This digital exciter can be thought of as a black box whose input is the digital transport stream (TS) and the output is the precorrected RF low power signal.

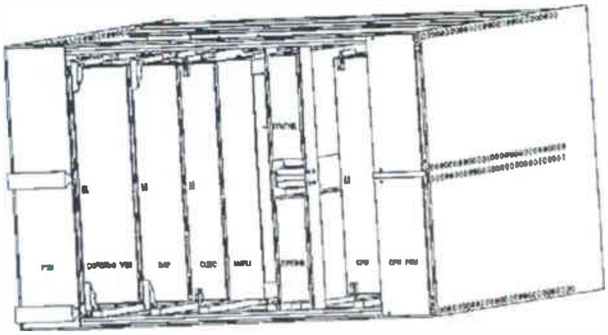

Figure 8 - *The digital exciter*

The integration of the all critical functions (the channel modulator, Digital Adaptive Precorrection, and RF conversion) all within the same rack, facilitates the transmission of the high data rate stream from one board to the next. This compact arrangement is also easier for monitoring, controlling and safety of the transmitter.

EXAMPLES OF RESULTS (8VSB STD)

UHF Solid State LD-MOS Amplifier

Looking back to Figure 1, we can see that for this particular amplifier there is an optimum biasing current of about 1300 mA which minimizes the distortion, before correction. An experiment of correction with the worst case of these curves, I = 400 mA, is summarized in Table 2.

	Conditions	Average Power (W)	OBO (dB)	Shoulders (dB)
Figure 9	without correction	95	≈ 6.8	30
Figure 10	with DAP	95	≈ 6.8	45
Figure 11	with DAP trial to push the power	120	≈ 5.8	40

Table 2 - *DAP correction on a solid state transmitter*

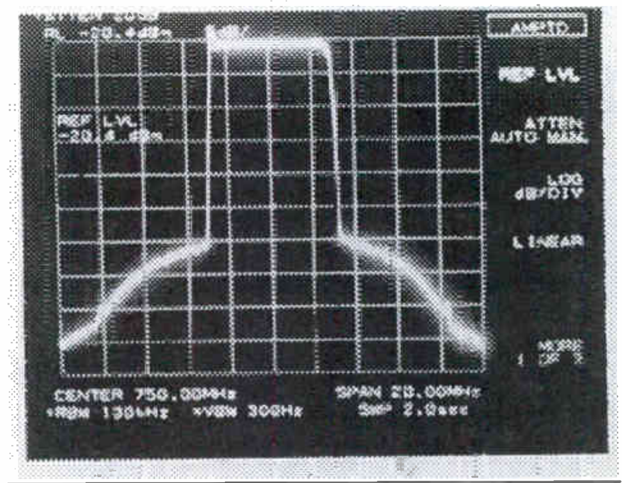

Figure 9 - *Solid state amplifier without correction: 95W*

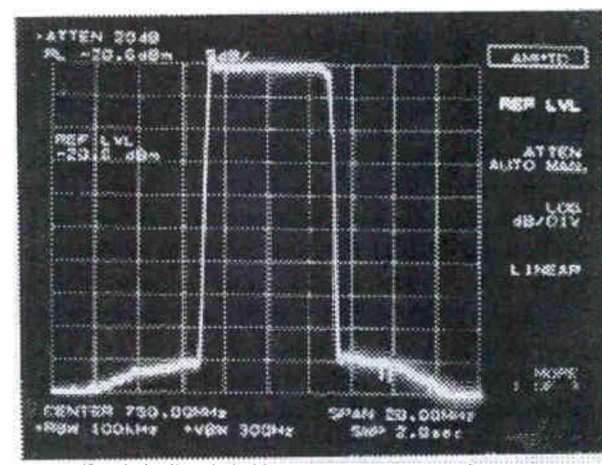

Figure 10 - *Solid state amplifier with DAP correction: 95W*

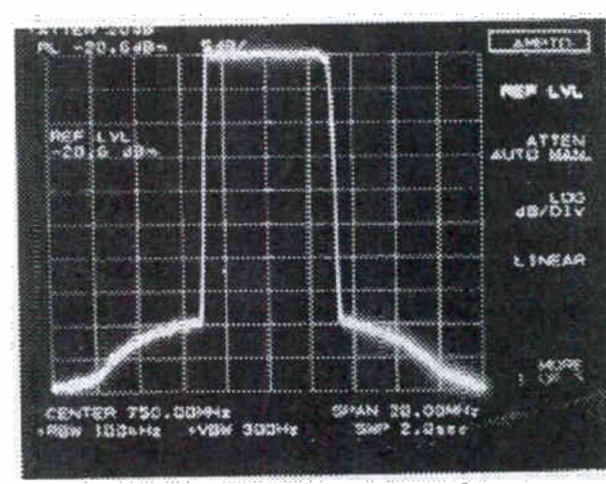

Figure 11 - *Solid state amplifier using DAP to increase output power: 120W*

5 kW Tetrode

Figure 12 shows the results of DAP on a 5 kW tetrode tube transmitter.

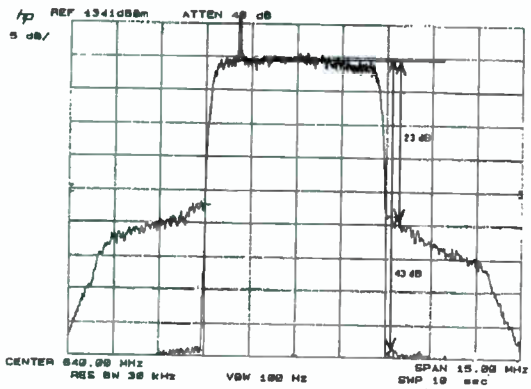

Figure 12 - *Example on a 5 kW Tetrode with 7 dB back off*

CONCLUSIONS

The digital era brings, on the one hand, new standards which are difficult to amplify, but on the other hand it brings about digital solutions to face this problem. It has been shown that Digital Adaptive Precorrection is a very efficient, convenient and cost effective way for facilitating the linear power amplification of these new standards. At the same time it brings about many advantages, such as automatic correction without human skills, and permanent monitoring and correction of the "on air" signal. This is in fact a natural and logical solution: digital processing for digital standards.

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[1] Lemaître, G., "Planning Aspects for Digital TV Transmitters", Montreux 1995.

[2] *Sixth Report and Order*, MM Docket No. 87-268, adopted April 3, 1997, FCC 97-115, pp. 90-91.

[3] Digital Video Broadcasting (DVB) ETSI specification ETS 300 744.

[4] IUT-R, R 624 table III, n°6.

[5] Proakis, John G, and Salehi, Masoud (1994), *Communications Systems Engineering*, Prentice Hall, Englewood Cliffs, NJ, pp. 116-119.

THE CONSTANT EFFICIENCY AMPLIFIER -- A PROGRESS REPORT

Robert Symons, Mike Boyle, John Cipolla,
Holger Schult, and Richard True
Litton Electron Devices Division
San Carlos, CA and Williamsport, PA

ABSTRACT

We will describe details of the design of a constant efficiency amplifier (CEA), including features of the vacuum envelope assembly, the r-f cavities and circuitry, and the collector, including calculations of electron trajectories in the collector at various r-f drive levels. In this article we report successful operation of a 60 kW Inductive Output Tube (IOT) we have developed as part of the CEA development. We expect to have some measurements of r-f performance with and without collector depression when the oral presentation is made at NAB98.

INTRODUCTION

Everyone seemed to realize that a multistage depressed collector would increase the efficiency of an Inductive Output Tube (IOT). No one seemed to realize that the combination of an inductive output amplifier and a multistage depressed collector, when operated at appropriate voltages, could be made to operate as a constant efficiency amplifier over a wide range of power outputs. The inventor[1] had to write a several-hundred line computer program to convince himself that this was, in fact, the case. The combination of an IOT and an MSDC will use technology from each device to obtain new high levels of performance. Nevertheless, the two pieces of hardware that make up this invention, that is the inductive output amplifier and the multistage depressed collector, are not new. For this reason, in reducing the idea to hardware at Litton Electron Devices Division, we decided it would be a good idea to examine not only how the performance of the combination of ideas could be optimized, but also whether or not the requirement placed on the individual parts should change the design philosophy used. We also were well aware that Haeff[2] invented the inductive output amplifier in 1939 and that Charles V. Litton[3], who also provided the name for Litton Industries, invented a multistage depressed collector for a klystron at about the same time.

When one asks oneself the question, "how should one change ideas that have been around for 50 years," sometimes the correct answer is, "very carefully."

THE GOAL

Last year at this conference we presented calculations[4] which showed that the peak-to-average power ratio of an 8-VSB signal is about 4:1. When a conventional inductive output tube processes this signal, the efficiency averaged over the modulation is about half of the peak efficiency of the tube at peak power output. Based upon these calculations, we estimated that the efficiency of a conventional IOT amplifying a digital TV signal would be about 25% and that the constant efficiency amplifier, simply because it is a constant efficiency amplifier, would provide at least 50% average efficiency over the modulation cycle. Thus, the CEA would halve the power input and would save the broadcaster at least $25,000 per year in power costs per 60 kW tube. The saving in power costs over the most modern silicon-carbide solid-state technology is even more impressive. Because the efficiency of silicon-carbide amplifiers at their peak power output is about 33%, the average efficiency for an 8-VSB signal will be only 16%. A transmitter using a constant efficiency amplifier, based on our most pessimistic estimate, would use only one-third of the power of a silicon-carbide transmitter.

During the last year we have made additional calculations of the efficiency of inductive output amplifiers with multistage depressed collectors. As we pointed out last year, inductive output amplifiers never reach the maximum theoretical efficiency of a class B device based upon the calculation of the fundamental component of current in a half-sinusoid current waveform. We pointed out that because of transit time variation with grid voltage and because of space-charge forces between the electrons in the bunch, by the time the bunch of electrons reaches the output cavity, it has been spread out over a much larger angle of flow. In our more recent calculations, we have varied the angle of flow and have allowed it to become as much as 270 degrees of phase. We have also reduced the effective r-f voltage that the electrons see in the output gap from a peak value equal to the beam voltage to a value equal to nine-tenths of the beam voltage. These two changes reduce the peak efficiency calculated for conventional IOTs from 78.5% to a more realistic 50%. It is interesting that when a multistage depressed collector is used to recover the energy from this electron beam with a less-than-perfect energy distribution function, the multistage depressed collector increases the efficiency by a factor larger than that which can be achieved when the beam has an idealized class B current variation with time. Figures 1 and 2 compare calculated efficiencies with and without multistage depressed collectors for ideal and non-ideal current waveforms respectively. It is easy to see that there is a much greater difference in the efficiency of the tubes without multistage depressed collectors than there is in the efficiency of the tubes with this feature. We have therefore raised our goals and we hope that we can provide even greater power savings than those we promised last year.

We hope to be able to offer an air-cooled version of the 60 kW CEA because it will only have to dissipate 15 kW instead of the 45 kW a conventional IOT must dissipate. While we have nearly completed the development of the IOT portion of our CEA, as this paper is being written, we do not yet know the shapes of the depressed collector electrodes, so we do not yet know if air-cooling will be possible. We hope we will have an answer to this question when the oral presentation is made.

FEATURES OF THE INDUCTIVE OUTPUT AMPLIFIER

As was pointed out earlier in this article, the inductive output amplifier has been around for almost 50 years and incorporates the best thinking of Haeff, Preist and Shrader[5], Clayworth, Bohlen and Heppinstall[6], and others. As a result, it will be difficult to make improvements, but nevertheless we may try to make a few, very carefully.

At Litton heretofore, we have built a number of very-high-current-density gridded electron guns. Most of these have employed metallic grids in a so-called "shadow-grid" configuration. Here, a grid tied electrically to the cathode with bars aligned with those of the control grid is interposed between the control grid and the cathode, thus preventing the control grid from intercepting any of the beam current. This has given us an excellent grasp of the technology required to deal with high dissipation grids. However, the shadow-grid technology cannot be used directly in a cathode-driven UHF amplifier because, without using excessive grid voltages and drive power, the transit angle of electrons from cathode to control grid would be excessive.

The pyrolytic graphite grids used in inductive output tubes possess the advantage of having a very low coefficient of expansion. Thus, when heated by radiation from the cathode and by dissipation of r-f drive power when the drive is applied between the cathode and the grid, the spherical grid does not expand into the cathode and short circuit. This permits very close spacing of the grid to the cathode with attendant low r-f drive voltage and high gain. However, the low expansion of the grid also creates a problem. Heat must be carried away from the grid with a high expansion metal support frame and so most manufacturers provide some kind of a spring contact between the mounting frame and the grid flange. This maintains good thermal and electrical contact between the grid and the frame. At Litton we realized that the pyrolytic graphite itself is an excellent spring up to very high temperatures, and so we devised a supporting structure which maintains good electrical and thermal contact by slightly deforming the outer edge of the grid. This structure is shown in Figure 3, and we have applied for a patent on it. We have also had some success with rigid clamping of the grid, so some of the sophisticated grid mounting schemes that have been used in IOTs may not be all that necessary.

Because we had no prior experience with pyrolytic graphite grids and did not want to develop this technology from scratch, we decided to buy it. We were fortunate in finding two suppliers with many years of experience, one in growing pyrolytic materials and the other in the laser cutting of such materials. After about a half a dozen trials with each supplier we were able to produce the grid shown in Figure 4 which meets all of our requirements.

Litton has a sophisticated computer capability for analyzing the performance of gridded electron guns and electron beams. We use a computer program, which was developed by Dr. Richard True[7] who heads our electron beam analysis activities. Dr. True wrote the first version of this program when he was a graduate student, and he has continued to improve it since. It was one of the first programs to use a deformable triangular mesh for solving Poisson's equation and the electrodynamical problem.

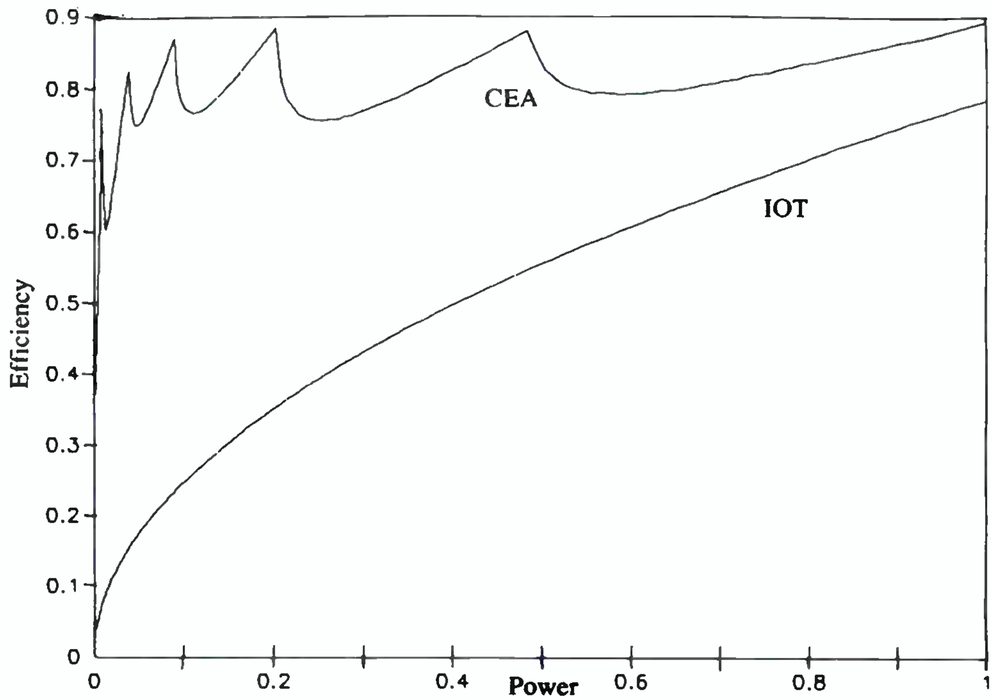

Figure 1 -- Calculated efficiency of an IOT with six collector stages at potentials of 0.1, 0.2, 0.3, 0.45, 0.7 and 1.0 times the beam potential and a peak r-f gap voltage equal to the beam potential

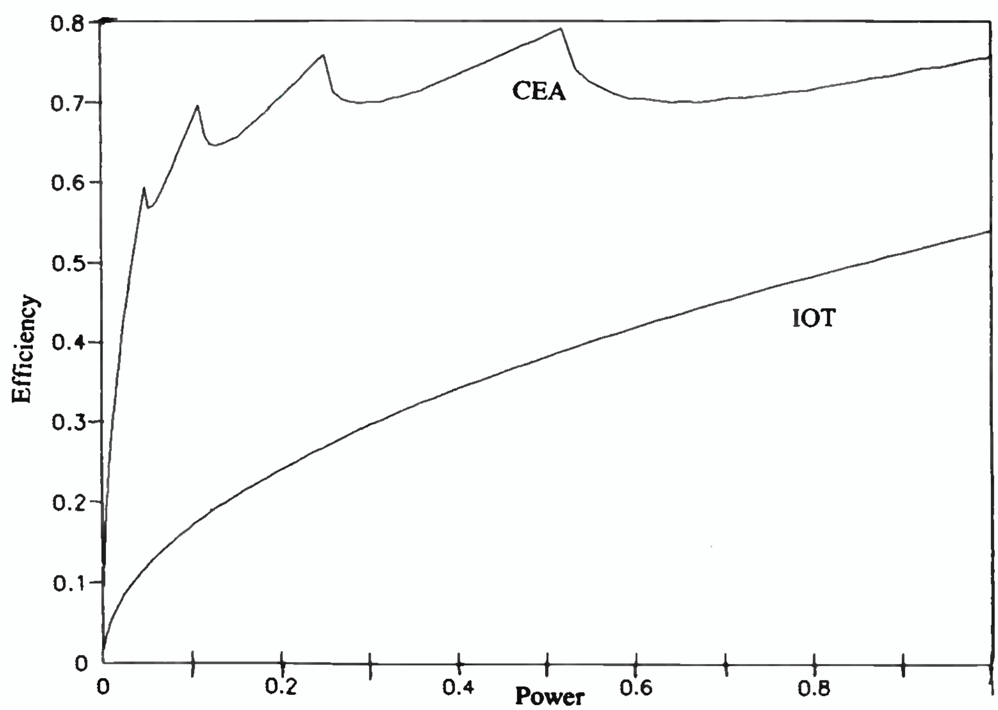

Figure 2 -- Calculated efficiency of an IOT with five collector stages at potentials of 0.2, 0.3, 0.45, 0.65 and 1.0 times the beam potential and a peak r-f gap voltage equal to 0.9 times the beam potential

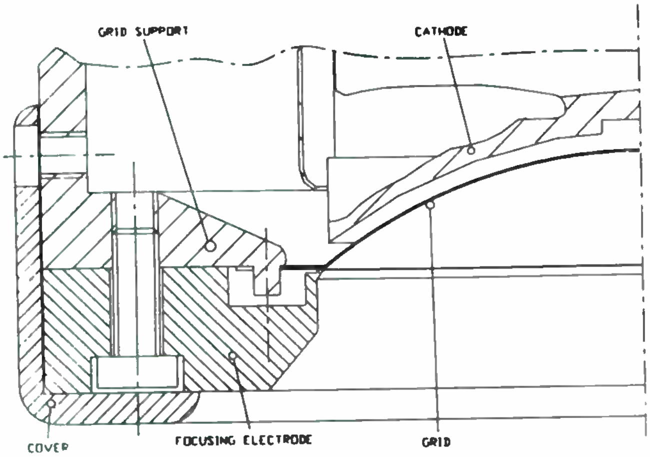

Figure 3 – Possible grid support method.

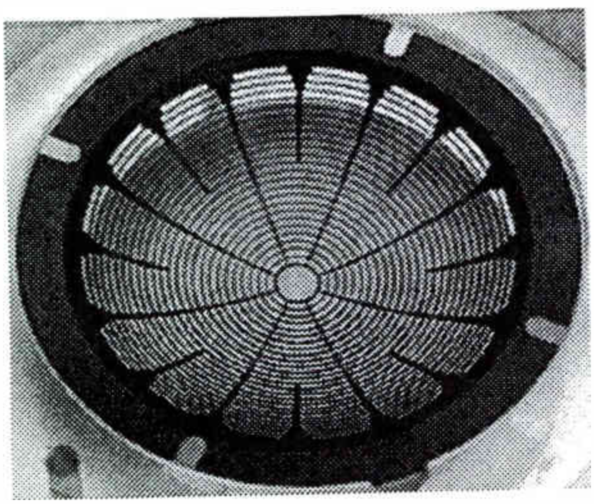

Figure 4 – Litton pyrolytic graphite grid.

Because of the use of a triangular mesh it is possible to match the curved surfaces of cathodes, anodes and grid wires which so typically form the boundary conditions of electron gun problems. We used this electron gun simulation program not only to analyze the performance of the electron gun at a number of different beam currents, but as we will discuss later, we are using it to analyze the trajectories of the electrons in the spent beam as they approach the collector electrodes. Not only does the code use a triangular mesh, but it is also possible to vary the density of the mesh points. The mesh can be very fine where the fields are changing direction rapidly and the electrons are moving slowly, and it can be coarse where the electrons are moving quite rapidly in directions nearly parallel to the electric field lines over long distances. In an IOT in which electrons move between fine grid wires with low velocity the grid wires form strongly convergent lenses that make electron trajectories cross at a distance slightly beyond the grid and give the beam a great deal of transverse energy. The magnetic focusing field that threads the cathode of an IOT and more or less converges with the electrons, guiding them through the anode must overcome this transverse energy. Figures 5 and 6 show trajectories of the electrons calculated for a low beam current and for a higher beam current.

There is one feature in the cathode designs of some inductive output tubes, which we would like to change in order to improve the performance of our IOT. This feature is a heat shield which is connected to the cathode support cylinder at the base of the support cylinder and extends upward to the edge of the cathode. There is a gap between this heat shield and the edge of the cathode, so that it forms a coaxial line. A calculation indicates that this coaxial line has an inductive reactance of about 8 ohms at 800 MHz. A calculation based on the grid cathode capacitance indicates that this may present a capacitive reactance of about 8 ohms at the outside edge of the grid-cathode space. Thus, when the input cavity is tuned to the high end of the television band, there may be a voltage minimum between the edge of the heat shield and the grid and a current maximum between the hot cathode support and the heat shield. A cleaner cathode support design, when looked at as an element of a resonant input circuit, might possibly reduce the required drive power at the high end of the television band.

We have now built several 40 kW and 60 kW IOTs with specified performance on our way to the CEA, and we will soon offer tubes of these designs for sale. We are still developing our r-f circuitry however, because we want it to be compatible with both our IOTs and our CEAs.

THE R-F CIRCUITRY

High power linear beam amplifiers such as IOTs and klystrons have a certain number of insulators through which a modulated electron beam passes. These insulators fall into two classes: those through which one wants r-f energy to flow (cavity insulators) and those through which one wants no r-f energy to flow (high-voltage and collector insulators). Sometimes those insulators that are not in resonant cavities radiate r-f. Not only can this r-f leakage exceed OSHA limits, but it can also cause regeneration if it leaks into low level resonant cavities or even leaks through the braid of

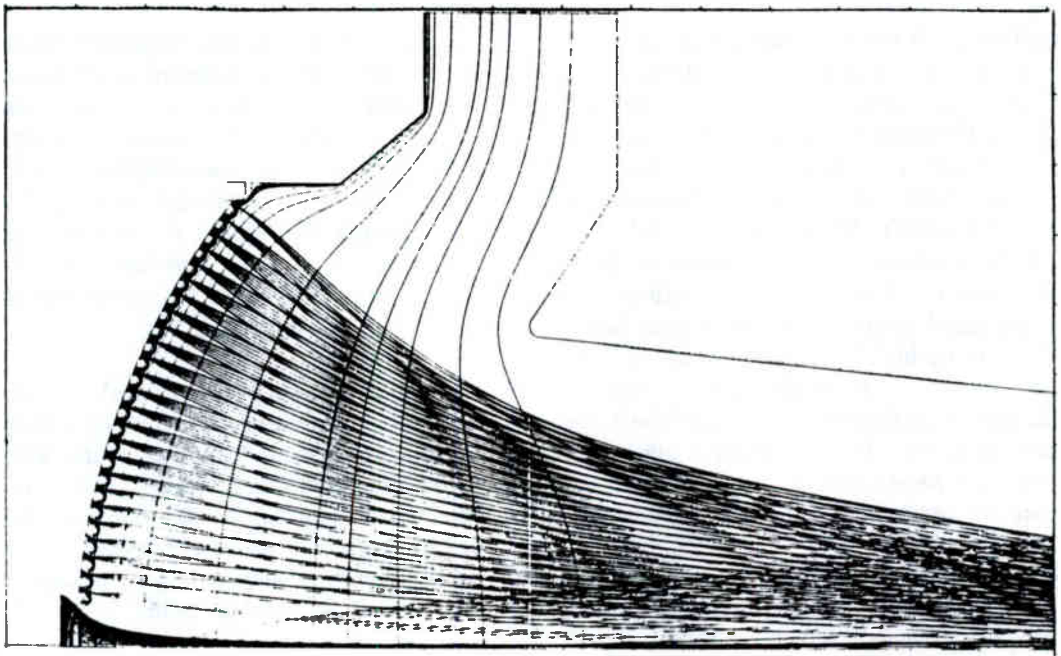

Figure 5 -- Electron gun simulation with grid at -30 volts, anode at 32 kV and 1.958 amperes of beam current.

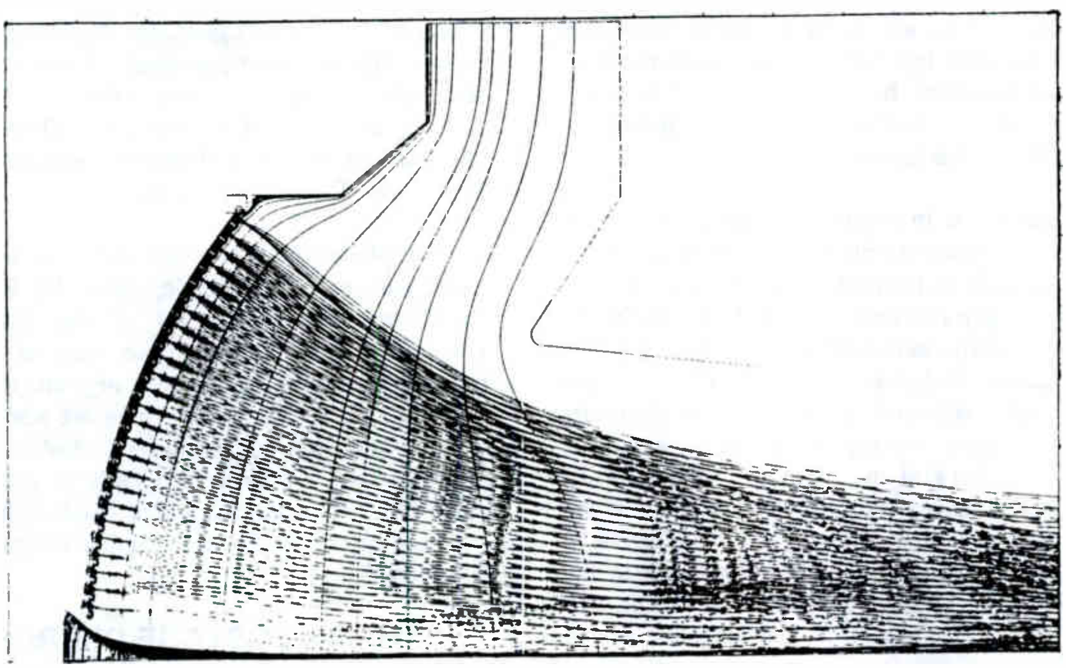

Figure 6 -- Electron gun simulation with grid at 0 volts, anode at 32 kV and 5.176 amperes of beam current.

flexible coaxial cable. It is not uncommon to use coaxial cable with two layers of braid on the outside to achieve stable operation in transmitters with a great deal of gain. The best way to eliminate r-f leakage from insulators that are not in cavities is to reduce the impedance the modulated electron beam sees across the interaction gap within the insulator to zero. Sometimes this can be done by surrounding the insulator with what one might call an "antiresonant" cavity which puts an electric field node, or if not that a very small reactance, at the electron beam. Under other conditions, building structures, which match the electron beam to "free space" through the insulator, will minimize the interaction impedance and the power extracted from the beam. If none of these approaches work, one sometimes must create one's own "free space" by building an enclosure and damping its resonances with lossy material.

One of the authors of this paper was involved with the conceptual design of the first television klystrons with multistage depressed collectors. The very first possibility considered for dealing with r-f leakage from the collector insulators was that of wrapping the water hoses, carrying cooling water from one stage to the next, around the insulators and surrounding the whole structure with a shield can. The water absorbed any r-f that leaked through the seals. This idea survived and is a feature of every MSDC klystron that has been manufactured since that time. In summary, however, it is much better to minimize the amount of r-f extracted from a beam than to soak it up after it has been generated.

When the insulator is in a cavity, the situation is quite different. In this case we want to contain the r-f in the cavity as completely as possible to maximize impedance at the interaction gap and minimize feedback. R-f choke joints have frequently been used in the cavities attached to both tetrodes and inductive output amplifiers. In this way different potentials can be used on the two electrodes which form the interaction gap (for example, a cathode and a grid or a screen grid and an anode). Alternatively, as in some IOT input cavities, part of the cavity can operate at ground potential. Unless the circumference of the choke is small compared to the wavelength in the insulating dielectric, one can get into serious trouble with azimuthal modes. Haeff, in fact, used a choke, which employed the glass envelope of his inductive output amplifier as the dielectric medium of the choke. Two sleeves inside the glass envelope were maintained at the electron beam potential, one on each side of the output interaction gap. The circumference of this choke was small enough to have no azimuthal modes. As the voltage a choke structure must hold off becomes higher, the thickness of the dielectric must become greater and the higher the impedance of the transmission line forming the choke must become. This makes it very difficult to design a good r-f choke, which will provide a solid short circuit for r-f while holding off a large amount of dc voltage. Such joints in input cavities can provide a path for r-f to leak into input cavities and cause instabilities. This is a good reason for not using choke joints in input cavities of IOTs.

To summarize, we believe in dealing with r-f leakage at its source: primarily by avoiding unintended resonant circuits around an electron beam, and secondarily, by the use of loss if necessary. We also believe in designing low level resonant circuits having as few extraneous openings or bad contacts at joints as possible. We also believe in double-shielded cable or, better yet, cable with a flexible tubular wall wherever possible.

In line with these beliefs we intend to use an input cavity which is directly connected to the grid. We will use only one low-voltage bypass capacitor or choke joint in the cathode lead to minimize the diameter of the joint while allowing the cutoff bias to be applied to the grid. Cavity tuning will be accomplished by actuators operated with insulated shafts, and a grounded shield will surround the input cavity to protect personnel. We are leaning in the direction of using a dc block in the drive line similar to commercial units that are available. These can be made small enough to be free of spurious azimuthal modes that might cause feedback problems.

Early textbooks on microwave and ultra-high-frequency techniques as, for example, those by Brainerd[8] and Reich[9] or certain volumes of the MIT Radiation Laboratory Series show various ways of building and coupling cavities for ultra-high-frequency power amplifier tubes and filters. There are also a number of articles and expired patents on double-tuned output circuits which are easily adaptable to inductive output amplifiers and will give ample bandwidth and power-handling capability (See for example Yingst[10], Beaver et. al.[11], and Symons[12]).

THE MULTISTAGE DEPRESSED COLLECTOR

As we mentioned earlier, Charles V. Litton conceived the multistage depressed collector shortly after the klystron was invented. Since that time, multistage depressed collectors have been used on both klystrons and traveling-wave tubes to increase their efficiency, first at a single power, and later in tubes which operated at variable power levels, as first demonstrated by Neugebauer and Mihran[13] at General Electric, later as proposed by one of the authors[14] of this paper for television use, and finally as developed by Earl McCune[15] at Varian Associates. There is an excellent review article by Kosmahl[16] which describes the state of multistage depressed collector development for microwave tubes as it existed in November 1982, when development of MSDC klystrons for television began.

The problem of developing a multistage depressed collector for an IOT is not quite the same as that of developing a collector for any of the applications discussed above. It is different because the dc component of beam current rises and falls in proportion to the square root of the output power of the tube. The d-c beam current is not constant as it was in either traveling-wave tubes or klystrons. As a result, the energy spread is low because the output cavity r-f voltage is low at the same time that the r-f and d-c beam currents are low. Thus, there will be small space-charge forces, and the beam will not spread as much as it travels deep into the collector toward electrodes having the lowest potential. For this reason, we expect that the collector may be rather long and skinny when compared to the multistage depressed collector of a TV klystron. We are starting our multistage depressed collector design effort by putting beams having the energy distribution functions of an IOT operating at various r-f levels through a series of "mathematical," idealized, fully-transparent grids which can produce a potential profile in the axial direction that can have either various gradients between them, or by using them in pairs, can produce a staircase of descending potential. We will examine the trajectories of the electrons of various energy in the various potential profiles and then attempt to design electrodes, which will collect the electrons at angles as nearly as possible normal to the surfaces. In designing electrodes for multistage collectors it is an axiom that one cannot recover any energy from an electron that has momentum directed tangential to the collector surface. Figure 7 shows some electron trajectories for an IOT beam entering a collector which was originally designed for an MSDC klystron. From the various angles that the electron trajectories make with the collector surfaces one can easily see that this collector is less than ideal for the specified electron beam, but even this case shows a considerable degree of constructive sorting of the electrons by energy class.

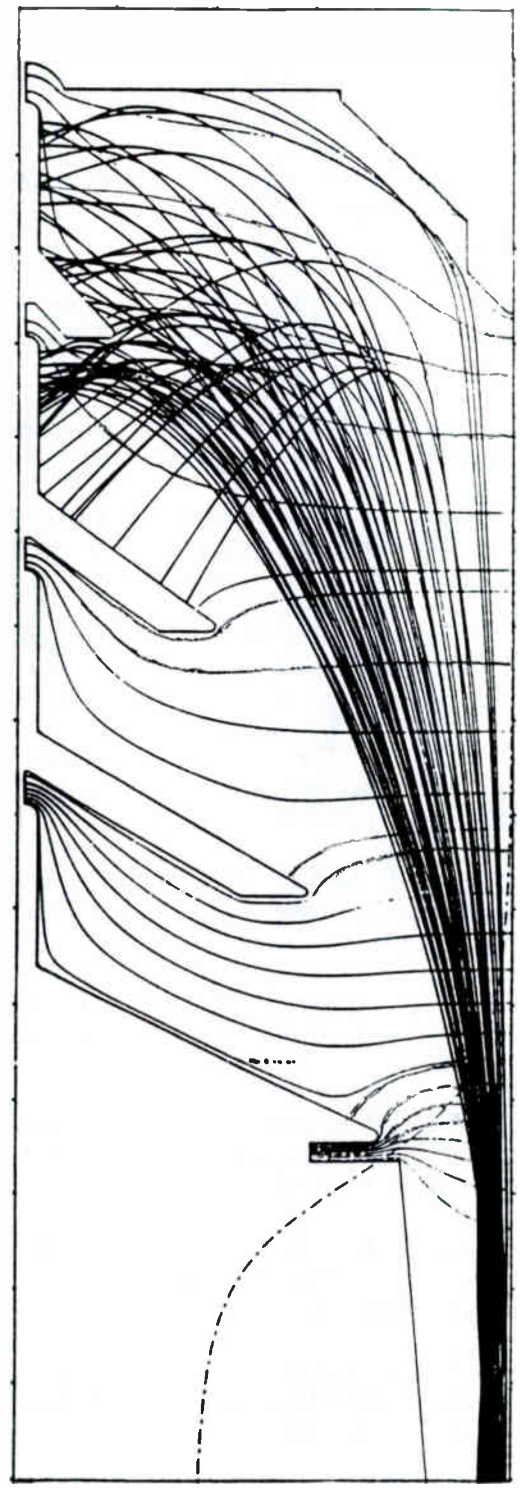

Figure 7 – Electron trajectories in multistage depressed collector at low drive level.

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On Channel Repeaters for Digital Television

Charles Einolf and Walt Husak
Advanced Television Technology Center
Alexandria, VA

ABSTRACT

The On Channel Repeater (OCR) for Digital Television (DTV) offers an obvious advantage in frequency congested markets where broadcast licenses for over-the-air channels are limited. The ability to avoid frequency shifting by receiving and retransmitting in the same channel will greatly enhance flexibility for the broadcaster to ensure adequate coverage throughout a service area. The OCR for DTV can offer many other advantages to broadcasters such as coverage extension, tailored radiation patterns, and reduced main transmitter power consumption. The Advanced Television Technology Center (ATTC) has initiated a program to develop and evaluate the feasibility of OCRs. This paper analyzes repeater coverage as well as discusses implementation issues.

INTRODUCTION

With the conversion to Digital Television many broadcasters are faced with moving their VHF channels to UHF. The characteristics of UHF are significantly different from VHF. UHF is strictly line-of-sight and has higher propagation losses. Although the 8-VSB digital modulation offers better performance at low signal-to-noise and in the presence of multipath, broadcasters are concerned whether existing coverage areas will be maintained. A repeater, that can receive and transmit on the same channel, offers the advantage of providing adequate coverage without adding further congestion to the already crowded frequency spectrum. DTV coverage extension can be realized under the classic terrain isolation case in addition to extension beyond the radio horizon. A tailored radiation pattern could be used to fill in or augment main transmitter antenna nulls due to situations such as tower blocking. The broadcaster may also be able to increase signal strength in locations where there are population concentrations suffering from a weak signal.

NTSC repeaters have been used to overcome terrain obstacles for many years. These repeaters have taken the form of channel translators or microwave relay stations. The NTSC repeater relies upon the reception of one carrier frequency and retransmission on another. The use of separate frequencies is required due to the poor performance of NTSC receivers for co-channel and multi-path interference. These two interference mechanisms arise from antenna mutual coupling inherent in co-located receiver-transmitter systems. The high electrical field strength of a transmit antenna in close proximity to a receive antenna will inevitably cause undesired reception of the transmitter.

The ATTC has initiated a study of On Channel Repeaters as a mechanism for DTV coverage extension. The goal of the study is to demonstrate an operational On Channel Repeater and to evaluate its performance. Initial analyses indicate that coverage areas can be extended using low power repeaters[1]. However, implementation issues need to addressed[2]. In particular, close attention must be paid to antenna design and configuration in order to minimize mutual coupling.

This paper discusses a propagation analysis for the repeater as well as critical elements and implementation issues for the design. The ATTC is in the process of a full implementation of the On Channel Repeater.

REPEATER MODEL

In order to illustrate the concept a simplified model is proposed. The model has a main transmitter located at the center of an omnidirectional radiation pattern as illustrated in Figure 1. A repeater is located at the periphery of coverage (55 mi) such that the radio horizon of both antennas are just in sight of each other. The viewer receiver can be located on a line extending from the main transmitter through and beyond the repeater. The system parameters assumed are listed in Table 1.

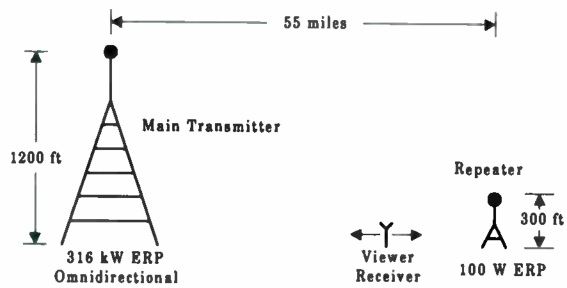

Figure 1. DTV repeater concept placing the repeater near the radio horizon of the main transmitter.

Table 1 On Channel Repeater Model Assumptions	
Main Transmitter Antenna Height (HAAT)	1200 feet (365 m)
Main Transmitter ERP	316 kW (25 dBk)
Main Transmitter Antenna Pattern	Omni
Main Transmitter Radio Horizon	48 miles
Main Transmitter Channel	42 (638-642 MHz)
Repeater Antenna Height (HAAT)	300 feet (91 m)
Repeater Transmitter ERP	100 W (-10 dBk)
Repeater Radio Horizon	20 miles
Repeater Receive Antenna Gain	10 dB
Repeater Receive Antenna Front-to-Back Ratio	14 dB
Repeater Receive Antenna Beam Width	80 degrees
Receiver Sensitivity	43 dB above 1 µV/m
Receiver Noise Threshold	16 dB S/N

Critical elements in the analysis include adequate signal strength and minimal mutual interference. Signal strength must be adequate under all conditions for the repeater and the viewer receiver. Two predominate sources exist: receiver front-end noise; and multipath interference which falls outside of the capture window of the receiver's adaptive equalizer. In order to maintain a Bit Error Rate (BER) of less than $3 \cdot 10^{-6}$, test ready DTV receivers must see a signal field strength of at least 43 dB above 1 µV/m. Noise limited contours are based upon an F(50,90) field strength with a receiver antenna height of 30 feet†. Figure 2 illustrates the signal strength as a function of distance from the main transmitting antenna. The repeater effectively extends the receivable field strength beyond the radio horizon of the main transmitter.

In order to ensure adequate reception, test ready receivers also require a Desired-to-Undesired ratio (D/U) above 16 dB. Furthermore, the correctable range of multipath interference in the test ready receiver is -5 µsec to +20 µsec. Figure 3 shows that for the OCR model, the repeater signal will begin to encroach on the main signal 35 miles out. From 35 miles to within several miles of the repeater, the multipath delay is on the order of hundreds of microseconds. Consequently, the multipath delay is outside of the capture range and will interfere with the main transmitter signal as co-channel noise.

It is apparent that omnidirectional antennas for the repeater transmitter and the viewer receiver contribute to co-channel interference at the viewer receiver. Two dimension contour plots allow the impact of repeater design on coverage area to be visualized. Figure 4 illustrates the effect that directional antennas have on signal strength. Directional antennas with a 10 dB gain, 14 dB front-to-back ratio, and a 80° beam width are used for the repeater receiver, repeater transmiter, and the viewer receiver. Note that the apparent received

† The notation F(x,y) means: the field strength exceeded at x% of the locations at least y% of the time.

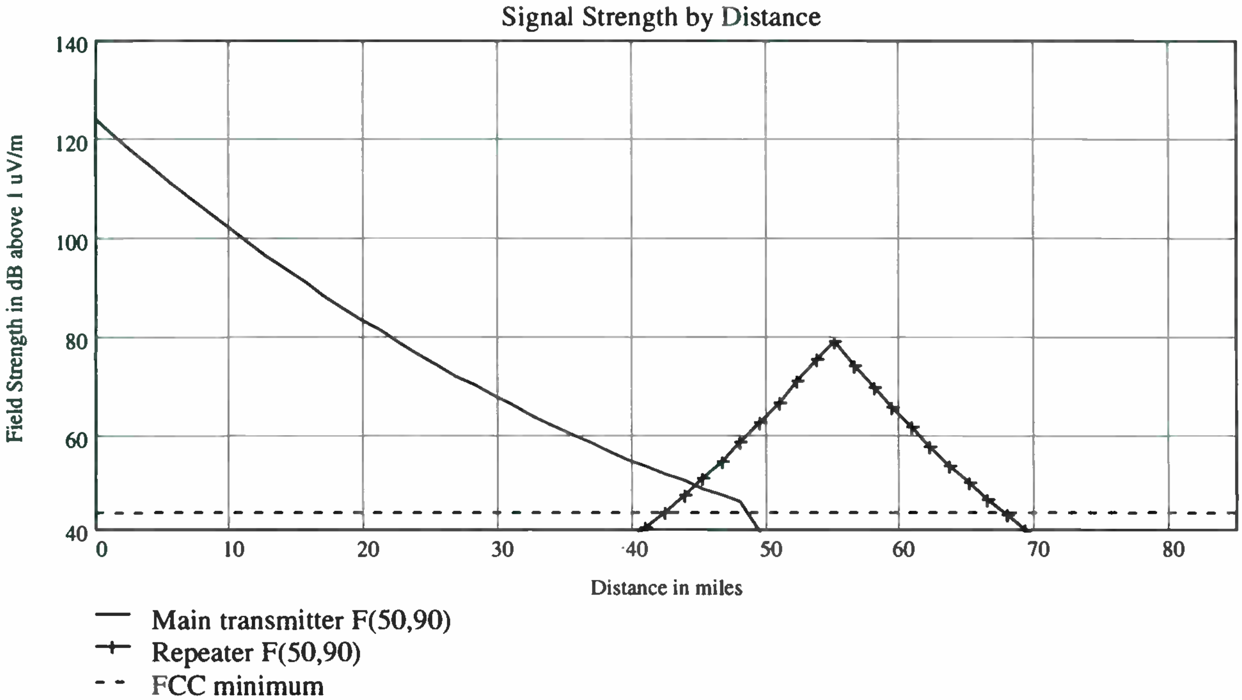

Figure 2. An On Channel Repeater can greatly extend the receivable DTV signal strength beyond the radio horizon of the main transmitter. The repeater is using an omnidirectional antenna.

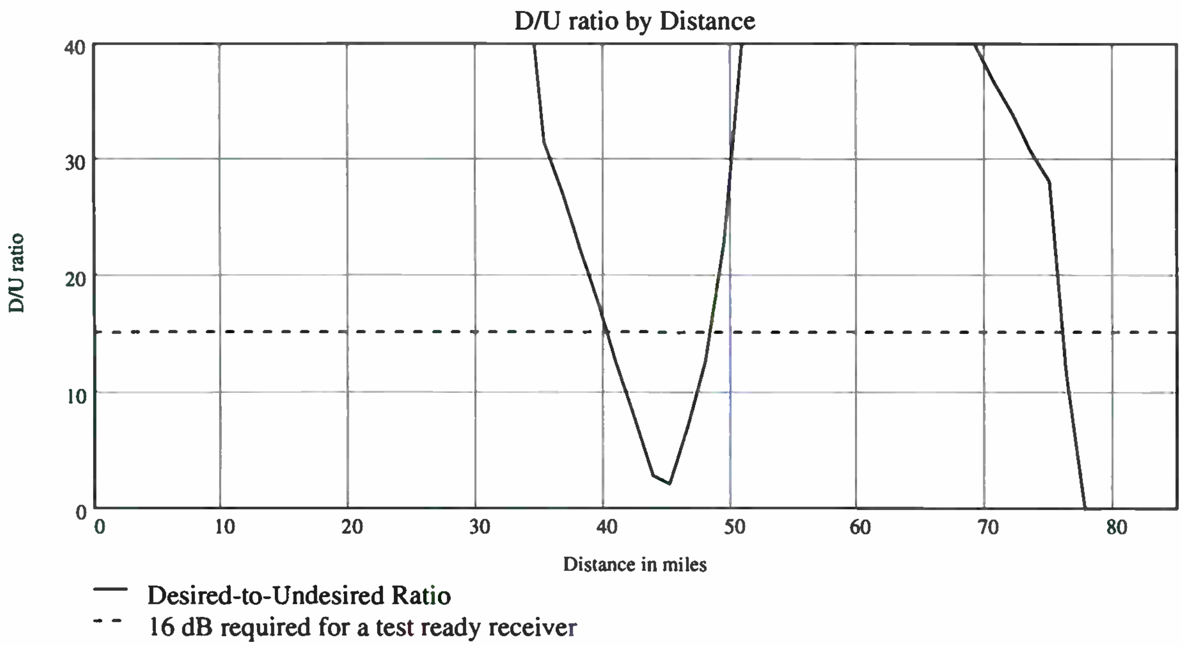

Figure 3. Although the DTV signal strength is extended, the use of an omnidirectional antenna limits reception in regions where the D/U ratio drops below 16 dB.

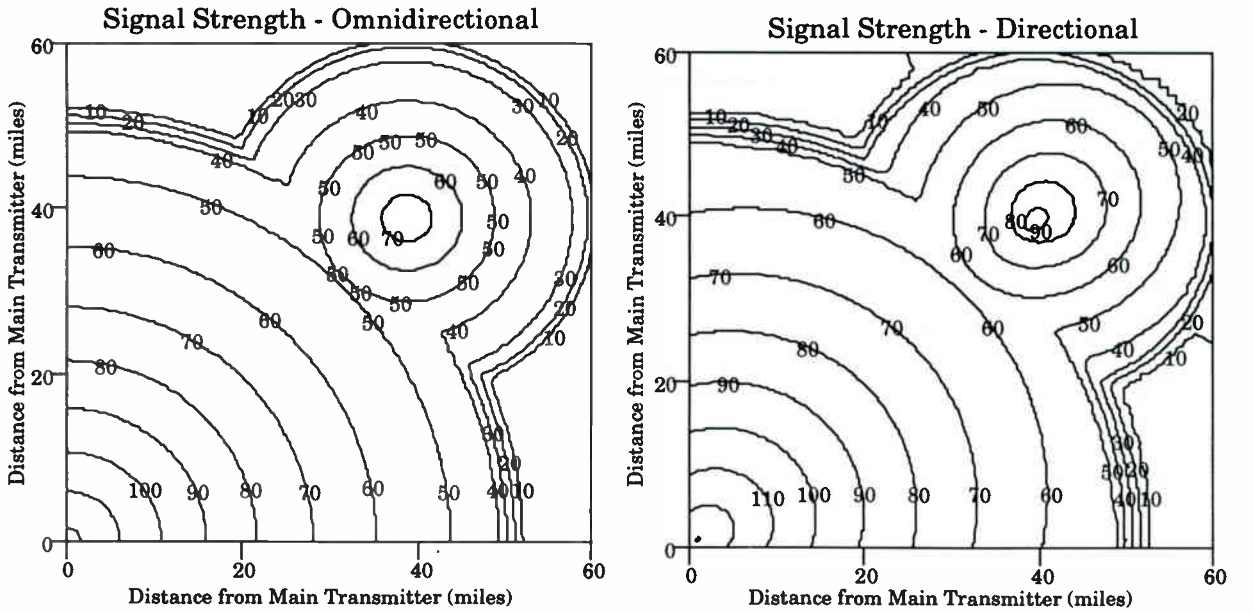

Figure 4. The directional antenna for the repeater transmitter slightly extends the coverage area over the area obtained with an omnidirectional antenna. The signal strength (in dBµV/m) shown on the contours must be above 43.8 dBµV/m for adequate reception.

field strength is higher due to directivity of the system. Furthermore, the repeater transmit pattern is no longer circular. Although the signal strength is not greatly affected, the D/U ratio is significantly impacted. Figure 5 shows the effect of the directional antenna on D/U ratio. The directional antenna significantly reduces the area between the main transmitter and the repeater where the D/U ratio is inadequate. This reduction is especially due to the directivity of the viewer's antenna. Since the antennas have relatively high gain on-axis and less gain off-axis, the less powerful signal is more easily rejected.

It is the combined effect of signal strength and D/U ratio that will ultimately determine the performance of the system. Figure 6 illustrates the difference between omnidirectional and directional antennas at the repeater and viewer sites. The use of directional antennas nearly doubles the coverage area.

OCR APPROACHES

Both analog and digital approaches are being considered for the On Channel Repeater as illustrated in Figure 7 and Figure 8, respectively. The digital system is more complex to implement and more expensive than the analog system. The over-the-air signal is received and demodulated to the transport stream level. The digital system can correct transmission errors which occur between the main transmitter and the repeater receiver. Reception errors can be corrected using the Forward Error Correction (FEC) circuitry. Multipath interference can be removed by the adaptive equalizer in the receiver. The transport stream has the error correction reapplied and the signal remodulated and retransmitted.

The analog approach is less costly than the digital system. The disadvantage of the approach is that the system allows errors to propagate through the repeater. The viewer's receiver must then correct these errors.

ANTENNA MODEL

The repeater receive and transmit antennas form a critical part of the OCR system. A simplified model for the OCR antenna system is illustrated in Figure 9. In this model all loss elements are shown as attenuators and all gain elements are amplifiers. Transmission lines are ignored. The over-the-air received signal is combined with the signal fed back from the retransmitted signal. The combined signal is amplified and supplied as the input to the On Channel Repeater. Similarly, the output of the OCR is amplified and transmitted. However, a portion is fed back through a series of attenuators. These attenuators represent the characteristics of the antennas in the system.

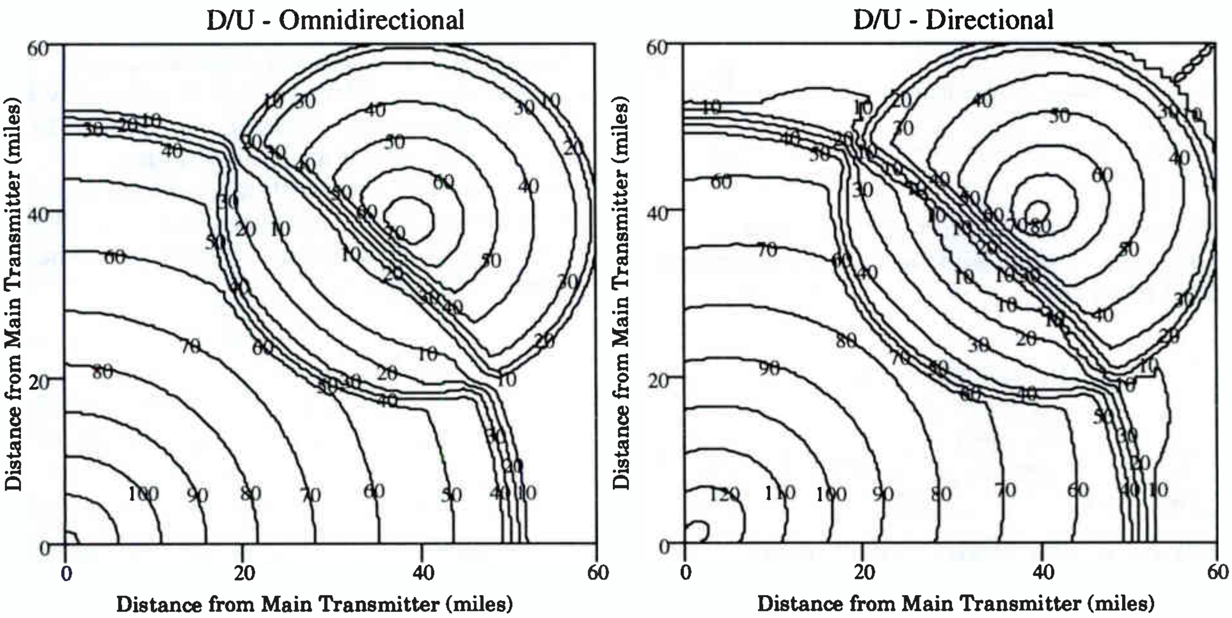

Figure 5. The directional antenna for the repeater transmitter significantly reduces the area where interference occurs between the main transmitter and the repeater transmitter. The D/U ratio (in dB) shown on the contours must be above 16 dB for adequate reception.

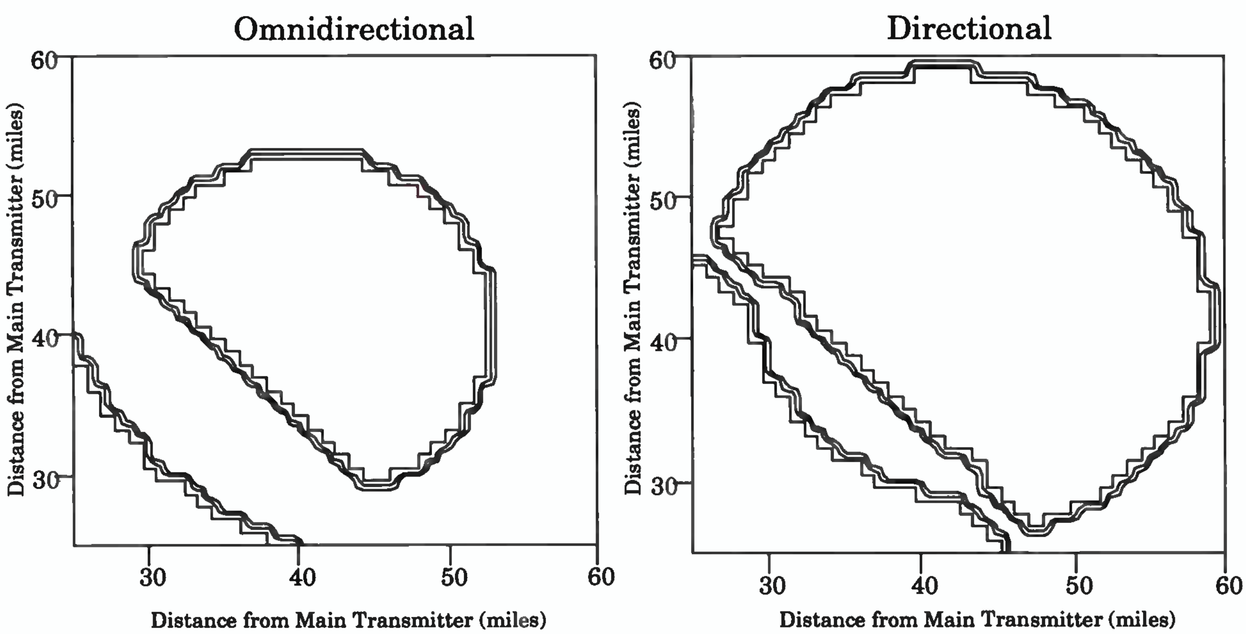

Figure 6. The combined reception criteria, signal strength and D/U ratio, shows that directional antennas can nearly double the area of satisfactory performance for the On Channel Repeater.

Analog Approach

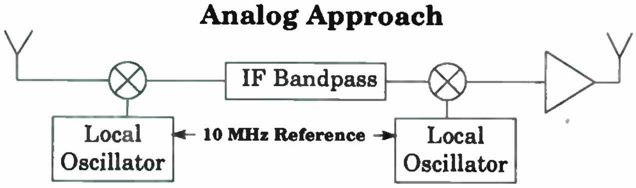

Figure 7. The Analog On Channel Repeater provides a simple and inexpensive implementation.

Digital Approach

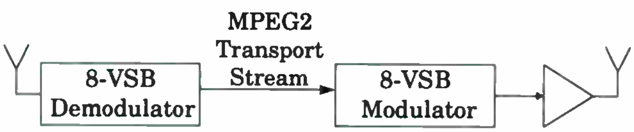

Figure 8. The Digital On Channel Repeater provides full reception and error correction to the transport stream level.

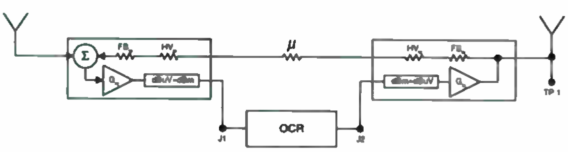

Figure 9. An antenna model for the On Channel Repeater which illustrates the dependence of retransmit power on the mutual coupling.

The mutual coupling, μ_c, is given by:

$$\mu_c = HV_{rx} + FB_{rx} + \mu + HV_{tx} + FB_{tx}$$

where FB are the front-to-back ratios, HV are the polarization factors, and μ represents the physical separation of the transmit and receive antennas.

The effective radiated power, ERP, is given by:

$$\text{ERP in Watts (dBW)} = FS - 180 + \mu_c$$

where FS is the field strength in dB above 1 μV/m. The 180 is the dipole factor scaled to Watts (dipole factor = 130 dBm/dBu; 20 dB reception margin; and 30 dB dBm to dBW conversion).

This analysis of the antenna system shows that the relative gain of the repeater receive antenna is not significant. However, the mutual coupling between the repeater receive and transmit antennas will determine the upper boundary of performance for the OCR. The choice of design will impact the reception margin. The antenna design, selection, and placement will be vital to the repeater performance. Figure 10 shows the results of antenna coupling measurements made within an anechoic chamber using back-to-back directional antennas. These tests confirmed the need for high front-to-back ratios in the antennas.

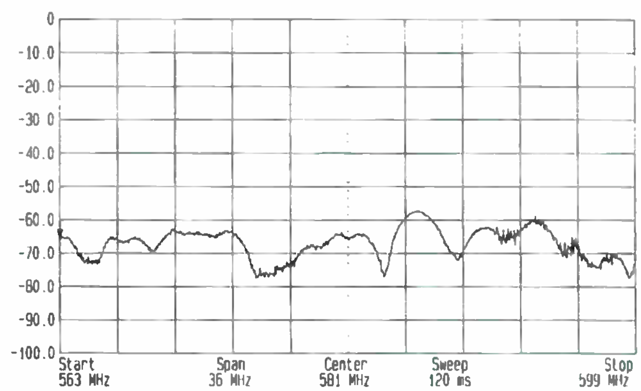

Figure 5. Back-to-back directional antenna coupling measurements confirm the need for high front-to-back ratios and careful placement of the OCR antennas.

CONCLUSIONS

The use of On Channel Repeaters for digital television broadcast coverage extension is feasible. An analysis of coverage extension shows that a significant area can be added by the introduction of a repeater. In addition, the use of directional antennas can nearly double the useful coverage area.

The robust nature of the digital 8-VSB communications channel used in DTV allows repeaters to be used even in the presence of co-channel noise. Digital communications makes it possible for repeaters to operate in both the terrain shielded environment as well as in the presence of mutual interference. An analysis of the system design has identified antenna mutual coupling as the most critical parameter which needs to be addressed. Sufficient isolation must be achieved between the co-located receive and transmit antennas.

The ATTC is in the process of constructing an On Channel Repeater test site and plans to conduct a series of field experiments. These experiments will include both the analog repeater as well as the digital system. The digital repeater system will provide improved performance through error correction between the transmitter and repeater.

ACKNOWLEDGMENTS

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The Challenge Of Testing DTV Systems

Stephane Billat & Richard Hall
Sencore Electronics — Adherent Systems LTD
Sioux Falls, SD — Cambridge, UK

ABSTRACT

In December 1996, the FCC ruled on the implementation of digital TV, giving less than two years for broadcasters to make their move to Digital Television. The DTV system is based on MPEG-2. The ATSC standards define how MPEG-2 will be used in order to meet the terrestrial broadcast and cable market requirements. Since the first of 1996, manufacturers of encoders, decoders, transmitters and other equipment providers have been busy designing systems suitable for the implementation of DTV. At the same time broadcasters and cable companies have been experimenting to see how they can best implement DTV without having to break the bank. On top of it all, not all the standards have been defined yet. Many manufacturers are still scrambling to be sure their equipment will meet the ATSC standards that are being updated on a regular basis. Satellite providers have not moved to ATSC yet, as they have chosen to implement DVB standards, and as a result, broadcasters are forced to deal with many different signal formats. The DTV revolution is here and we need to be sure it is going to be as smooth as possible. With the number of different signals being offered, manufacturers and broadcasters will need to insure they are compliant with the various standards and compatible with the other equipment. In this paper, we would like to review the fundamentals of MPEG as well as look at the DVB and ATSC standards. We will primary focus on the new ATSC standard, reviewing what the signal contains as well as what standards are being developed by the standards committee to insure proper interoperability of the equipment. We will also review what needs to be tested by the manufacturers as well as the broadcaster and what test methods are available in order to insure a good, reliable, high quality signal we expect from Digital Television.

INTRODUCTION

In December 1996, the FCC ruled on the implementation of Digital TV, giving less than two years for broadcasters to make their move to the technology. This revolution is based on MPEG technology. Just a few months ago, at CES, TV manufacturers of sets introduced their first HDTV compatible sets. By the end of 1998, many stations will be broadcasting Digital Television signals with more following in 1999. The digital television revolution is well underway.

Market Situation

The market is in full transition. Many decisions need to be made and the choices are difficult. Only one thing is certain, broadcasters will convert to digital TV. The question of implementation, when and how much is it going to cost remain open issues. They are taking the necessary steps to assure a smooth transition even though there are numerous unanswered questions at this time.

What Challenges Are Broadcasters Facing?

Some of the challenges include the choice of a correct format. Digital television as it has been defined by the FCC, may have multiple formats (18 to be exact). Each format is different and therefore requires different equipment. Some of these formats are close to the NTSC standard with 525 lines and 720 samples per lines at 29.94 Hz interlaced. Others use 1080 lines at 60Hz progressive. The need for equipment to support each format is very different and each broadcaster needs to decide what best fill their needs. The equipment selection process will also be a challenge as most equipment is, for the moment, not available or in beta testing. Testing product quality as well as the compatibility with other equipment will be a major challenge.

What Potential Pitfalls Are There?

There are of course many pitfalls along the way especially with the amount of equipment that will need to be implemented. The first pitfall would be selecting proprietary equipment that will not operate properly with the rest of the industry. The system would perform inside the facility satisfactorily, but prevent the broadcaster from adding other manufacturer's equipment and could prevent interfacing with other facilities. Another obvious pitfall is the amount of investment required to make the move can quickly become overwhelming. The business aspect is beyond the scope of this paper but should be carefully examined.

What Will We Discuss In This Paper

In this paper, we would like to provide information about some aspects of the implementation of DTV. We will review the fundamentals of MPEG, as well as look at the DVB and ATSC standards. We will primary concentrate on the new ATSC standards and review signal contents as well as what standards are being developed to assure proper interoperability of equipment. We will also examine what test methods will help manufacturers develop DTV products as well as what the broadcaster will need to test to insure a good, reliable, high quality signal.

WHAT IS MPEG TECHNOLOGY

MPEG technology has been around for several years. It is a standard that specifies a method of compressing data (mostly video and audio) in order to reduce the amount of transmission bandwidth required to broadcast the information. These techniques have been recognized worldwide and are based on a DCT based algorithm. MPEG is often describe as a tool box, but does not provide information to the manufacturer on how the data should be compressed but provides guidelines on what the final product should be. This is one reason why there are so many differences in terms of features and performance between encoder manufacturers. MPEG technology is not only used in the broadcast domain but is is also found in the computer industry.

Why It Is So Attractive?

The reason MPEG is so attractive is that it reduces the amount of bandwidth required to transmit the signal. For example in an NTSC analog system, 1 TV channel includes 1 video, 1 audio and close caption information that can be transmitted over a 6 MHz channel. With MPEG technology, it is possible to broadcast five video channels each of them with four distinctive audio and close captioning in the same 6 MHz. On top of it all, the quality of those five compressed channels is better than a single NTSC channel! Broadcasters now see a method of increasing revenues by packing more information per channel. The same principal applies to the computer industry, they can easily provide video information over network for example.

What Is It Used For?

MPEG is really used in many different ways. The application of MPEG that most interests us here is broadcast applications. The first large scale, commercial application has been DBS or direct broadcast. Satellites have been able to send to people's homes hundreds of channels of video and audio. There are over four or five organizations that provide direct satellite broadcasting and they have millions of subscribers. MPEG technology is also being used over the Web or with the DVD (Digital Versatile Disk).

The Video

As mention earlier MPEG is a scalable technology and as a result, MPEG offers different levels of quality of compression. They are known as the Levels and Profiles. Each level and profile relates to picture resolution and encoding method, and depends on the use of the compressed video information. Broadcast is now typically using Main Level at Main Profile (ML@MP). However, new HDTV formats specify resolutions that will require the use of higher levels and profiles.

	LEVEL	PROFILE simple	 main	 snr	 spatial	 high
1	**High** 1920 pixels 152 lines	X	80M bit/s 128Mb RAM	X	X	100M bit/s
1	**High** 1440 pixels 152 lines	X	60M bit/s 64Mb RAM	X	X	80M b/s 128Mb RAM
	Main 720 pixels 576 lines	15M bit/s 8Mb RAM	15M bit/s 16Mb RAM	15M bit/s 32Mb RAM	X	20M b/s 32Mb RAM
	Low 352 pixels 288 lines	X	14M bit/s 4Mb RAM	4M bit/s 6Mb RAM	X	X
		No B frames 4:2:0 Not scaleable	B frames 4:2:0 Not scaleable	B frames 4:2:0 SNR scaleable	B frames 4:2:0 SNR & spatial scaleable	B frames 4:2:0 or SNR & SNR & spatial scale

Table 1: MPEG Level and Profiles.

The Audio

MPEG specifies two types of audio: MPEG-1 audio and MPEG-2 audio. Most current systems use MPEG-1. In new DTV systems, none of the MPEG 2 standards have been selected and Dolby AC-3 is to be used with DTV.

The Data

One interesting aspect of the MPEG system is that data can be broadcasted along with video and audio. This is known as Private Data. This data can be as simple as machine control or as complex as upgraded software for the set-top box. In fact any data that is not MPEG video or audio is labeled Private Data. This is how DTV Dolby AC-3 audio will be treated.

Principals Of Operation For Broadcasting

MPEG is used in broadcast with the main intention to fit as much video and audio information in a 6Mhz channel while still maintaining the best possible picture quality. The following processes will occur: The video information will be compressed by an MPEG encoder. The audio channels will then be encoded. The audio information is compressed by the same encoder that compressed video, or done independently. Then all the audio channel, data, and videos are then combine into the transport stream. As an example most DBS systems offer 5 video with 2 or 3 audio per video in a one 6Mhz bandwidth.

APPLICATION OF MPEG IN REAL LIFE

As mention earlier, MPEG is a toolbox, and more constraints need to be implemented in order to develop a usable widespread broadcast system based on MPEG technology. If this is not achieved, it is possible and very likely to have equipment that will be MPEG compliant but will not be compatible.

The first standard that proposed a broadcast solution to the use of MPEG-2 is the European DVB (Digital Video Broadcast) project. This proposed solution was then followed by the North American ATSC Project. (Advance Television System Committee). There are also other real life applications for MPEG such as the DVD (Digital Versatile Disk) but we will not go into detail for these applications.

DVB: Digital Video Broadcast Project

The DVB project started in Europe over 3 years ago and was the first world wide MPEG solution. Most of European broadcasters are complying to the DVB standard but this standard is not limited to Europe. It is used in the U.S. by Satellite providers and in other part of the world such as Asia. It uses mostly ML@MP. DVB is a standard that specify, special control tables, conformance parameters, physical interfaces of equipment in order to achieve interoperability between manufacturers of broadcast equipment.

ATSC: Advance Television System Committee

The ATSC standard was developed in North America in response to the need of a standard that would allow American broadcasters to provide DTV to the market. The FCC has ruled that the ATSC standard will be used in the U.S.A. Since that time, the ATSC and DVB committees have been traveling the world trying to persuade each country to adopt their respective standards. The ATSC standard is newer than DVB and many of its aspects of conformance are still under development. The goal however, is the same as DVB, to provide a complete standard for the U.S. broadcast industry that will provide interoperability of equipment while providing U.S. households with digital TV.

Other: DVD, Video Conferencing.

Another aspect of MPEG is its use in the computer world specifically DVD. The DVD (Digital Versatile Disk) uses MPEG-2 video and Dolby AC-3 audio. Note that DVD uses MPEG-2 Program Stream while broadcasters are using the MPEG-2 Transport Stream. The difference is the multiplexing layer. The program stream is not designed for transmission purposes.

Another MPEG application is video conferencing. Its resolution and bitrate are not suitable for broadcast applications. Those applications are beyond the scope of this paper.

MPEG FOR BROADCAST

In this paper we will concentrate on MPEG broadcast applications. Most of today's applications are using ML@MP or ML@4:2:2 profiles. The 4:2:2 profile is a new profile that answers the need of contribution. ATSC uses a ML@MP or better profile. As we will see, the PES and Transport layers are more or less level and profile independent.

Layering System

MPEG is a layered standard. This means that from raw baseband uncompressed video or audio to the transport stream, we can separate the data into three main area: The ES (Elementary Stream), the PES (Program Elementary Stream) and the TS (Transport Stream). Each of these layers provide a significant step in the encoding and transmitting process of the MPEG video. Note that we will not address in this paper, the PS (Program Stream) that is an alternative of the TS layer and used, for example, in DVD applications.

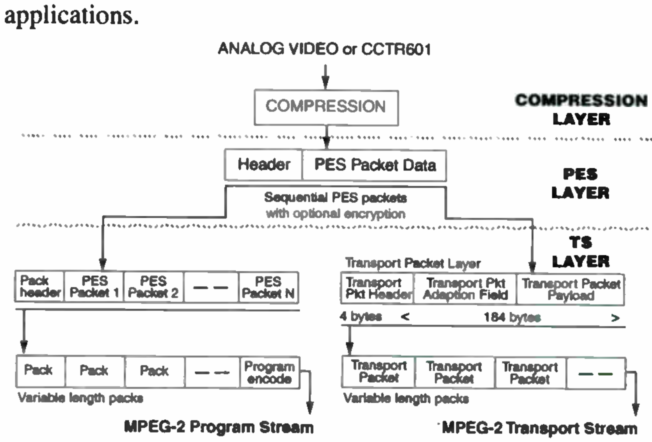


Figure 1: MPEG-2 System Layers

ES (Elementary Stream)
The Elementary Stream is the first step of the encoding process. It takes baseband video or audio and creates raw MPEG data. We will not go into the details of the encoding process as it is beyond the scope of this paper. We will say that a frame is broken into blocks with each being processed. For each block the DCT process is applied and the DCT coefficients are then quantified. The final result of this process is an amount of data representing the frame but is using less information than the original frame. This step is typically performed by chip-sets, giving the broadcaster little control of the process. Each frame can be encoded as a I, P, or B frame depending on the application. B frames contain less information than P frames, which have less information than I frames. The GOP (Group Of Picture) gives the sequence of I, P, B frames used in the encoding process. P and B frames require information from other frames in order to be decoded. The broadcaster can set the GOP depending on its application. The key is to insert as many B and P frames (to reduce bandwidth) without compromising the quality of the video.

PES (Program Elementary Stream)
The PES layer comes directly after the ES layer. The ES raw MPEG data is formatted into PES packets. For video, the PES packet generally represents 1 video frame. The PES packets are of variable length and have a header that provides information about the video frame, timing of that frame, and how the video was encoded. They are also audio PES packets. The syntax of the header is defined by MPEG in the system Part 1 of the standards. One of the critical parameters of the PES packet is timing. As seen in the ES layers, some frames require other frames in order to be decoded. As a result, decoders need to hold in buffer some of those frames and when they need to be presented to the screen. This timing is critical and decoders use DTS (Decoding Time Stamp) and PTS (Presentation Time Stamp). The time stamps are derived from the 27 MHz clock of the encoder. Other important information that can found in the PES packets include: Start of Picture, Splicing Points, and GOP structure.

Splicing of two streams represents a huge challenge for the broadcaster, as it is one of MPEG's most difficult tasks.

TS (Transport Stream)
The Transport Stream is the most critical stream for the broadcaster. The Transport Stream is transferred between equipment, such as between the encoder to the multiplexer to the modulator. The concept of the Transport Stream is to provide a syntax that allows the broadcast of multiple compressed programs over one transmission channel. The PES packets (multiple audio, multiple video) are slices into 188 bytes packets. Each 188 byte packet has a 4 byte header and a 184 byte payload.

The header provides information about the ID of the program (called PID) as well as Sync bytes that are used in the bit stream as packets indicators. The payload can carry useful data (video, audio) or information about the system (table) or about the timing (PCR in the adaptation field). The adaptation field is an extension of the header and allows information to be added about the program. The adaptation field has a variable length. The most important information found in the adaptation field is the PCR (Program Clock Reference) The PCR is a 33 bit sampled value of the multiplexer's 27Mhz clock and is used to transmit timing information to the downstream decoder. Each independent program requires its own set of PCR. Tables data are transmitted in the TS and are very important. It allows the decoder to know which packets belongs to which video and audio. It also provides system information and a program guide. Along with MPEG video and MPEG audio, other data can be broadcasted such as machine control, stock quote, etc. This data is consider private data and placed into data packets as specify in the table. MPEG only specifies the format of the data packet, but provides user freedom to implementation. As a result, it is difficult to find a manufacturer of common private data.

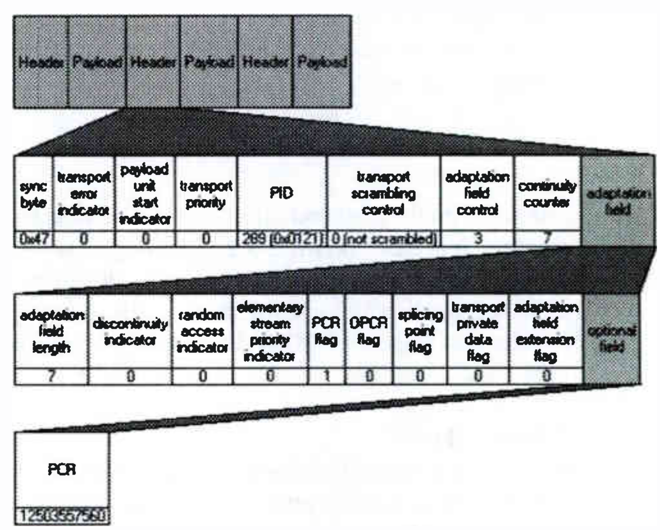


Figure 2: Transport Stream Header

Modulation
The final layer is the Modulation Layer. The main modulation techniques for MPEG are 8, 34, 45 Mbits/s DS3 which are used for studio to the transmitter, QPSK or 8-PSK for satellite delivery (direct or contribution), QAM 64 for the cable industry. An 8-VSB technique is used for terrestrial broadcast. Other delivery techniques include ATM over DS3 or OC-3.

The need for different modulation techniques comes from the constraint of the transmission method. For example noise level. Satellite is noisier than terrestrial broadcasting, which is noisier than cable delivery, which is noisier than a short direct link. Each modulation technique offers different protection schemes such as RS (Reed Solomon) coding, interleaving or framing.

OTHER MPEG PARAMETERS

Tables

MPEG requires the use of three main tables:

The PAT (Program Association Table) provides information about where the table program is located. It is always located on PID 0.

The PMT's(Program Map Table) provides information about the contents of the stream and associate PID to a particular video or audio. The PID of a PMT is defined in the PAT table.

The CAT (Conditional Access Table) defines the scrambling parameters and is located on PID 1. Every other MPEG table is optional.

Conditional Access

The Conditional Access is a key point of DTV. Most programs are or will be encrypted in order to select who will be able to access the data. The MPEG system provides much better security than conventional broadcasting methods. It is very powerful and as a result can be difficult to manage properly.

THE DVB IMPLEMENTATION

The DVB implementation offers a complete broadcast solution. It is based on MPEG-2 technology. DVB specifies the use of multiple formats of levels. However, at this date most applications use ML@MP. DVB specifies a set of rules that make the MPEG-2 technology applicable to network distribution.

Where is it used

DVB is the standard that has been adopted by Europe. It was created by satellite providers such as Astra Satellite or Canal Plus and was later adopted by many cable operators. It will soon be used for terrestrial broadcasting of off-air transmissions. DVB is also being used in the U.S. An example is Echostar, they broadcast a bouquet of programs over satellite. Most point to point MPEG contribution feeds for SNG (Satellite News Gathering) are based on the DVB format right now. Some Asian countries have also adopted the format.

Added Tables

The tables are the heart of the TS layer. They provide the decoder with the information it needs in order to find a certain program and properly decode it. There are three main MPEG tables (PAT, PMT, CAT) and 7 extra DVB tables. Not all of the tables are required in the DVB stream.

Here are brief descriptions of the required tables:

The NIT (Network Information Table) provides information about all the TS in the network.

The SDT (Service Description Table) provides information about the services offered by the stream

The EIT (event Information Table) provides information on programming.

The TDT (Time and Date table) provides time and date information.

All other tables are optional: RST (Running Status Table), ST (Stuffing Table), TOT (Time Offset Table).

Added Constraints: Conformance Issues

As stated earlier, two systems can be MPEG-2 compliant without having interoperability between them. This is why DVB adds more parameters than need to be tested in order to ensure a DVB compliant stream. DVB organizes these tests into three priority levels.

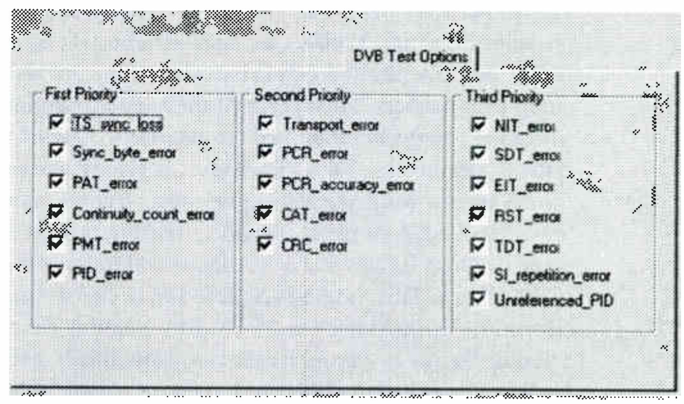


Figure 3: DVB Transport Stream Compliance Parameters

Future in the U.S.

The future in the U.S. for DVB is unclear. There are large systems in the U.S. that will probably keep DVB equipment in operation. Companies made the equipment investment and manufacturers will continue to support and develop DVB equipment. On the other hand, as ATSC makes its way into the broadcast industry it is doubtful that many new DVB systems will be implemented in North America. It will not be long until the amount of ATSC systems outnumber DVB systems.

THE NEW ATSC IMPLEMENTATION

In December of 1996, the FFC ruled that U.S. broadcasters will be broadcasting Digital Television before the end of the century. ATSC will be the standard that will be used. It is based on MPEG-2 technology. There is some confusion existing between terms. DTV and HDTV. DTV (Digital Television System) represents the overall system that will be implemented. HDTV is one of the potential formats for the DTV system. However, most of the DTV format is NTSC resolution similar. ATSC (Advance Television Committee) is a committee that set standards for the use of DTV systems. It is composed of both manufacturers and broadcasters, and is not limited to North America. Broadcasters will send over the air a 19.4 Mbits/s bitstream modulated with 8-VSB technique.

ATSC FORMATS

Spacial Format	Aspect Ratio	Progressive 59.94 / 60 Hz	Interlace 29.97 / 30 Hz	Progressive 29.94 / 30 Hz	Progressive 23.970 / 24 Hz
1920 x 1080	16:9		✖	✖	✖
1280 x 1080	16:9	✖		✖	✖
704 x 480	16:9	✖	✖	✖	✖
704 x 480	4:3	✖	✖	✖	✖
640 x 480	4:3	✖	✖	✖	✖

Table 2: ATSC Formats

What Is The New ATSC Standard?

The New ATSC standard is equivalent to the DVB standard. It makes the MPEG-2 technology applicable to large distribution network. It is geared for the U.S. market and is applicable for satellite, cable, and terrestrial broadcasting. It specifies a set of rules such as tables, program guides, parameters and limits. There are 18 formats defined by the ATSC standard. See Table 2. ATSC uses the Dolby AC-3 standard for Audio. Note that AC-3 is seen as private data from an MPEG-2 point of view.

Where It Will Be Used?

ATSC was developed for the U.S. market much like DVB was for the European market. It is very unlikely that the ATSC standard will be used in Europe, but other parts of the world such as, South America or Asia may look at the ATSC standard. The ATSC committee is traveling around the world doing demonstrations, trying to add additional countries to the ATSC user group. Note that DVB is doing the same thing....

Typical Broadcast Center

It is difficult to say what a typical broadcast center may look like in the future since local broadcasters operate differently than networks or DBS from cable operators. One thing is certain, everybody will get some ATSC data in and out of their facility. Two of the most popular scenarios include full decoding of every incoming source or passing it through "as is." We will probably see both configurations implemented. It is not clear at this time which format (s) the broadcaster will use. Most likely 1920 x 1080P and 740 x 480I.

Implementation Choices

Difficult choices are being made by the broadcaster and their affiliates. The following scenarios are likely to happen: The Network will provide to their affiliates, an ATSC stream compressed to a lower compression ratio than 19.4 Mbits/s (Contribution Quality). Typically we will see bitrates between 45Mbits and 65 Mbits/s transmitted over satellite. These bitstreams will be decompressed locally, edited and recompressed to 19.4Mbits/s and sent over the air. Another possible scenario includes the pass through option. The network will broadcast over satellite a 19.4Mbits/s stream that the affiliates can pass through. This scenario is inexpensive but does not offer much flexibility to the local broadcaster, unless splicing becomes a reality.

Technical Specification

The first step in understanding the new ATSC standard and how it will be used is to study the standard itself: The following documents should be downloaded from the ATSC web site www.atsc.org. The main documents are listed at the end of this paper.

ATSC Table ID Ranges & Values

Table ID Value(hex)	Tables	PID
	ISO/IEC 13818-1 Sections:	
0x00	Program Association Table (PAT)	0
0x01	Conditional Access Table (CAT)	1
0x02	TS Program Map Table (PMT)	per PAT
0x03-0x3F	(ISO Reserved)	
	User Private Sections:	
0x40-0x7F	(User Private for other systems)	
0x80-0xBF	(User Private)	
	Other Documents:	
0xC0-0xC6	(Used in other systems)	
	PSIP Tables:	
0xC7	Master Guide Table (MGT)	0x1FFB
0xC8	Terrestrial Virtual Channel Table (TVCT)	0x1FFB
0xC9	Cable Virtual Channel Table (CVCT)	0x1FFB
0xCA	Rating Region Table (RRT)	0x1FFB
0xCB	Event Information Table (EIT)	per MGT
0xCC	Extended Text Table (ETT)	per MGT
0xCD	System Time Table (STT)	0x1FFB
0xCE-0xDF	(Reserved for future ATSC use)	
0xE0-0xE5	(Used in other systems)	
0xE6-0xFE	(Reserved for future ATSC use)	
0xFF	Inter-message Filler	

Table 3: ATSC ID Ranges and Values.

Tables
As in DVB, the tables are a large portion of the new ATSC standard. All the information relating to the ATSC table can be found in A/65 often called PSIP (Program and System Information Protocol)

The ATSC still requires the MPEG-2 table (PAT,PMT) even though they are not used.

See Table 3 for a list of Tables and ID Values.

For the Terrestrial Broadcasting the following tables need to be included:

The TVCT (Terrestrial Virtual Cable Table) defines the MPEG-2 program embedded in the TS Layer.

The MGT (Master Guide Table) defines the type, packet identifiers, and version for all the other PSIP tables in the TS Layer except for the STT.

The SST (System Time Table), defines the current date and time of day.

The RRT (Rating Region Table) defines the TV parental guideline system.

For the Cable Broadcast the following tables need to be included:

The CVCT replaces the TVCT but has the same function

The MGT, SST, RRT are the same as that for Terrestrial Broadcasting.

The EIT, ETT are other tables that are included into the PSIP

Program Guides
Each channel can carry multiple EIT (Event Information Table) that will provide upcoming programming information to the user. Each program will have four EIT's that will cover 3 to 4 hours of programming. This information will be used by the decoder to provide program information. If the data in the EIT needs to be extended (long description) an ETT (Extended Text Table) can be attached to each EIT to provide more detailed information on the program.

Note that four EIT tables per program are required for terrestrial broadcasting, but are optional for cable. All ETT tables are optional.

Added Constraints
Like DVB, ATSC specifies additional constraints to help with compatibility. The constraints for the TS layer are not defined yet.

Program Paradigm
The Program Paradigm specifies a way to assign numbers to all the programs in a TS layer.

It is as follows:

The base PID is defined by base_PID= program number<<4 where the program number is the 16 bit identifier in the PAT and PMT.

For that program the PMT_PID= base_PID+0x0000, the Video_PID= base_PID+0x0001 (Video PID for the program),

PCR_PID= base_PID+0x0001 (PCR timing PID is the same as the video),

Audio_PID= base_PID+0x0004 (Primary Audio for the program),

Data_PID= base_PID+0x000A. (For the data of the program)

Conformance Issues
Right now, there are no conformance tests that the ATSC requires for broadcast equipment. Most of the compliance tests are offered for the receiver products. ATSC defined the ATSC Certification Program for consumer DTV receivers.

MONITORING AND TESTING THE NEW FACILITY

Why A Need For Testing And Monitoring
Conformance to all of the relevant standards is vital in assuring that all of the components in the broadcasting chain will interoperate and deliver a the desired service to the consumer. The complex protocols which form the basis of DTV have forced changes in testing methodology and test equipment. When applied to a digital TV system, traditional analog test and measurement techniques provide virtually no useful diagnostic information. Errors introduced during encoding, transmission or decoding may produce a blank screen, which provides no indication to the cause of the problem.

The path to implementing, installing and finally broadcasting the new digital TV standards raises a variety of testing and measurement problems. These problems can be roughly broken down into two sets of requirements. Detailed in-depth analysis of the entire bitstream from the transport stream down to the individual video, audio and data elementary streams which forms the basis of the service to be broadcasted. The other is monitoring the broadcast in real-time to ensure that it is conformant and can be decoded for viewing by the target audience. These two sets of requirements are addressed below.

In-Depth Analysis
In depth analysis of the transport stream is vital for the testing of the equipment which forms the digital

broadcasting chain from encoders and multiplexers through to the Consumers Integrated Receiver Decoder (IRD). Throughout the design, development and deployment phase there is a requirement to perform in depth analysis of the Transport Stream. Designers and developers of the encoding and transmission equipment need to analyze the output of their equipment to ensure conformance. Installers have to be able to analyze the Transport Stream throughout the entire system to solve any interoperability problems. In order to meet these demands, test equipment must be able to record the entire Transport Stream at bit rates up to 60 Mbit/s. The recorded stream can them be checked against a comprehensive list of parameters, for example:

- Correct transport stream packet structure
- PCR repetition rate and accuracy
- Correct table syntax
- Adherence to the T-STD buffer model as defined in ISO/IEC 13818-1

In addition to recording and analyzing the Transport Stream, developers, manufacturers and installers require a reliable and repeatable source. The test instruments should provide the facility the ability to playout Transport Streams at bit rates up to 60 Mbit/s.

Monitoring

Once the commissioning phase is completed, the broadcaster needs some means of assuring that the system is still operating correctly. This raises the need to permanently monitor (in real-time) the transmission in order to assure an uninterrupted, high quality service.

Cable and terrestrial broadcasters take their material from a variety of sources and there is a need to re-multiplex the transport stream, including materials which may have been previously encoded. The necessity to re-multiplex the Transport Stream forces the regeneration of the tables, taking information from all of the sources and combining it. The process may also necessitate the recalculation of the timing information that is embedded in the Transport Stream. This leads to the potential of introducing errors due to the re-calculation of time stamps and re-composition of the tables. The diagram below shows a generic system which applies to all of the medias over which Digital Television may be broadcasted. It highlights potential monitoring points. The primary monitoring point is at the output of the chain just prior to transmission, however, the other monitoring points throughout the chain can provide vital diagnostic information in the event of a failure allowing the rapid selection of a redundant component. See Figure 4.

The problem of what to monitor is complicated by the enormous volume and variety of information carried in a digital TV broadcast . Simply monitoring the Transport Stream against a subset of conformance tests will not guarantee the viewers set top box will correctly decode all

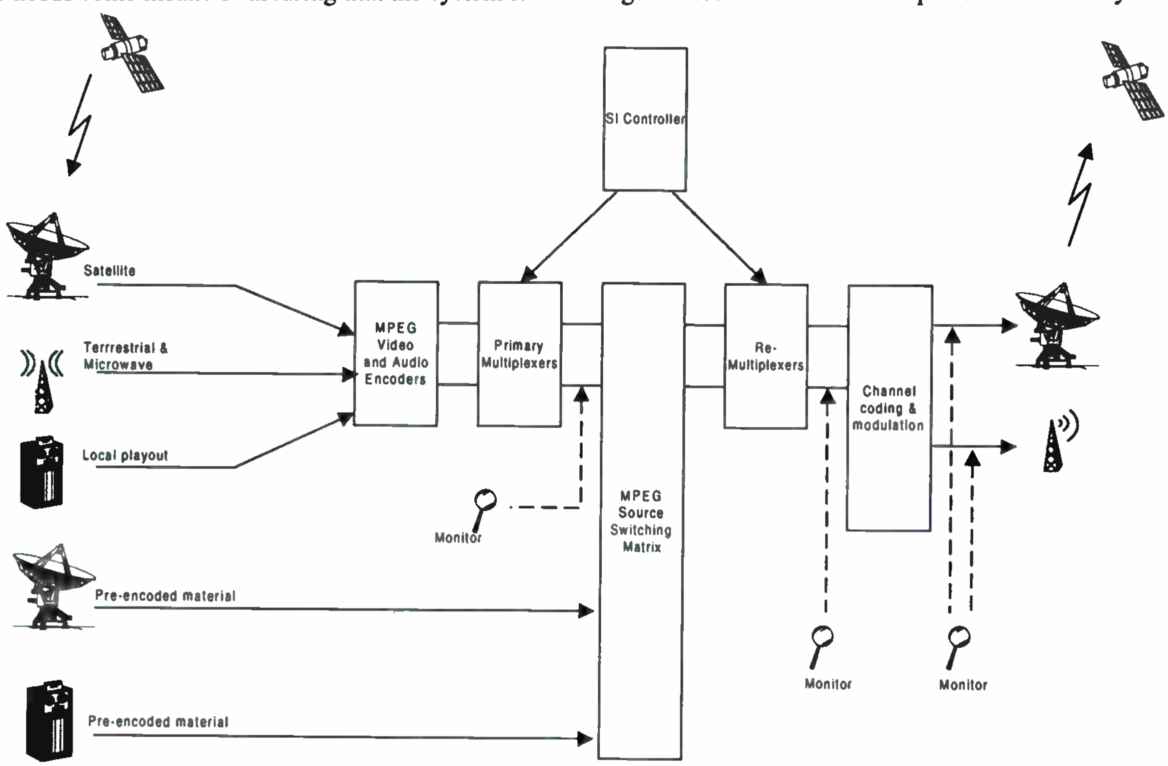


Figure 4: Possible Station Implementation

of the information in the Transport Stream. The ATTC conformance test laboratory will help to ensure that interoperability problems are minimized, however, for the broadcaster there are two fundamental questions which need to be addressed

Is the service present ?

In order to determine whether the service is present the real-time monitor has to successfully interpret the PAT and PMT, then locate the video and audio PIDs. However, this alone is not sufficient to prove that the service is present. While the service might be deemed present by occurrence of all of the correct PIDs, it is equally important that each of the video and audio components of the Transport Stream multiplex are present at the correct bit rate. The real-time monitor must provide the broadcaster with an easy to read display of the relative Transport Stream Multiplex occupancy and bit rate of each PID. Refer to Figure 5.

In addition to providing a display of the relative multiplex occupancy the real-time monitor has to warn the broadcaster when the bit rate of each PID falls below a value which would indicate the service was no longer operational.

Can the viewer satisfactorily decode and watch the service?

The complexities of encoding and broadcasting Digital Television makes this a very difficult question to answer. Unlike analog transmissions, where there are a small number of programs being broadcasted, monitoring of the programs can be done by a number of human observers watching a wall of video monitors. The potential to broadcast a large number of Digital programs renders this approach ineffective.

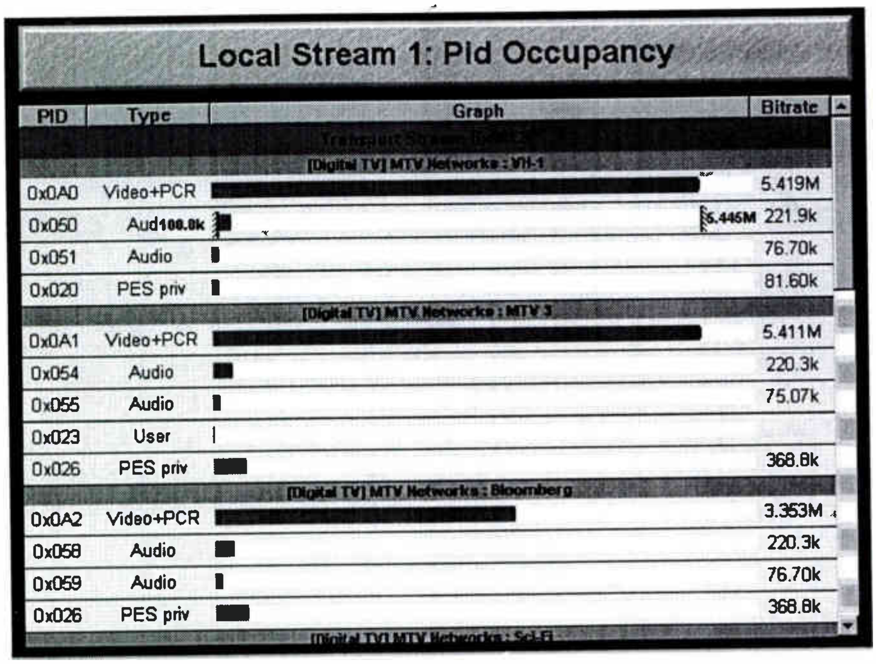


Figure 5: Multplex Occupancy

A pragmatic approach to resolving this question is to assume that during the commissioning of the Digital Television broadcast system it is fully tested and shown to be conformant against the specifications. The monitoring problem is therefore reduced to assuring that nothing has changed after commissioning has been completed. There are a variety of parameters which can be measured, these are summarized in the table.

Problem Reported	Detected by monitoring equipment
No transport stream	✔
Incorrect CRC	✔
Incorrect SI syntax	✔
Reserved bits incorrectly set	✔
Missing service components	✔
No video or audio input to encoder	
Inconsistent information across SI tables	
Inefficient bandwidth utilization	✔

Multiple signals to monitor

The deployment of DTV will not eliminate the other signals in the station. Even if some old signals will have a tendency to disappear, economic reality proves that the broadcaster will have to deal with old technologies as well as brand new ones. It is important to keep in mind that analog NTSC, CCIR601 Digital video, new HDTV non-compressed signals and MPEG ATSC TS will need to co-exist. As a result, the broadcaster will need to monitor carefully all the signals in its facility. The new modulation techniques such as QAM and 8-VSB will also bring some challenges to the user.

EVOLUTION OF THE ATSC STANDARD

The time line for the implementation of ATSC is very short. It is expected that the standards will not be complete yet. It seems that most of the work focuses on the receiver end where it seems that the big market is. It is expected however, that in 1998 the ATSC will develop and offer more and more standards for the broadcast industry. Some of the work being done includes Conditional Access, Data Broadcasting, and hopefully TS compliance. SMPTE will also offer some help in this big task as it did in 1997 by developing the interface standard for professional DTV equipment called the SMPTE SSI interface.

CONCLUSION

The DTV revolution has started and it will not be easy. It is clear from the FCC ruling and the commitment of the broadcasters and manufacturers that DTV will happen. Unfortunately the new technologies are very complex and challenging. The broadcaster will need to get as educated as possible on the subject in order to make the correct decisions when buying equipment or selecting systems. With the new implementation, the broadcaster will have to ensure the complete system is performing as it should and the final user will be able to decode high quality audio, video, but will also enjoy all the new advantages of DTV such as real time program guides, Internet data, etc. In order to achieve the high quality level that everybody expects from DTV, the signal must be monitored at every stage from the creation to the final modulation over the air. Therefore it is important for the video professional to realize that the testing methodology needs to be thought of at the design stage, before the system is implemented. By doing that they will avoid or at least minimize the headaches and frustrations that can be expect with any major studio redesign.

REFERENCES

MPEG Standards

ISO 13818	-1	Systems
ISO 13818	-2	Video
ISO 13818	-3	Audio
ISO 13818	-4	Compliance testing

DVB Standards:

ETS 300 468: Specification for Service Information

ETR 162: Allocation of Service Information (SI) code.

ETR 211 Guidelines on implementation and usage of service information

ETR 290: Measurement guidelines for DVB systems.

ATSC Standards:

A/53: ATSC Digital Television Standard that gives the system characteristic.

A/54: A guide to the use of the ATSC Digital Television Standard.

A/52: Digital Audio Compression- Dolby AC-3

A/65: Program and System information Protocol for terrestrial broadcast and cable

Digital Sound Broadcasting: Worldwide Expectations and Progress

Sunday, April 5, 1998
9:30 am - 4:30 pm

Chairperson: Milford Smith
Greater Media Inc, East Brunswick, NJ

***Digital Sound Broadcasting Becomes a Global Reality**
D. K. Sachdev
Worldspace, Inc.
Washington, DC

***An IBOC Lab and Field Testing Program**
Glynn Walden
USADR
Linthicum, MD

Digital Broadcasting in AM Bands: A Reality
Patrick Bureau
Thomcast
Conflans -Sainte Honorine, France

***In-Band On-Channel Digital Broadcasting: Delivering on Promise**
Derek Kumar
Digital Radio Express, Inc.
San Jose, CA

Critical Issues and Considerations for the All Digital Transmission Path
Frank Foti
Cutting Edge
Cleveland, OH

***Satellite Direct Radio Broadcast Implemetation**
Robert Briskman
CD Radio Inc.
Washington, DC

***Results of Experimental L-Band EUREKA/147 Multimedia Transmission in Toronto, Ontario**
Richard Zerod
Visteon Automotive Systems/Ford Motor Co
Dearborn, MI

***AM Hybrid IBOC DAB System Design and Performance**
David Hartup
Xetron Corporation
Cincinnati, OH

Advanced Audio Coding for Digital Sound Broadcasting
Karlheinz Brandenburg
FhG-IIS
Erlangen, Germany

Technical Development of S-band Satellite Mobile Broadcasting and Communication Systems
Shigetoshi Yoshimoto, Takashi Ito, Toshiaki Sato and Yoichi Kawakami
The Advanced Space Communications Research Laboratory
Tokyo, Japan

***Transmitter and Antenna Requirements for AM DAB (Digital Audio Broadcasting Systems)**
John Delay and Hilmer Swanson
Harris Corporation/Broadcast Division
Quincy, IL

***Tremendous Comeback of AM Bands by Digitalization in Sight**
Michael Pilath and Dietmar Rudolph
Deutsche Telekom AG
Bonn, Germany

*Papers not available at the time of publication

DIGITAL BROADCASTING IN AM BANDS: A REALITY

Patrick BUREAU and Pierre LAURENT
THOMCAST
Conflans-Sainte-Honorine, FRANCE

ABSTRACT

Since the launch of ideas for the introduction of digital modulation within the AM frequency bands, several systems are under development with the common aim to act positively for the digital future of AM Radiobroadcasting. Skywave 2000, the Thomcast contribution for NADIB (Narrow Band Digital Broadcasting) and DRM (Digital Radio Mondiale), has reached the experimental phase and has already proven interesting capability and flexibility during field tests performed under real environmental conditions in Long, Medium and Short Waves.

The present paper, mainly based on the results obtained during these testing campaigns, will provide an accurate idea about the feasibility and limitation of today's existing solution, and the promising expectations based on the availability in the near future of technologies and techniques now under development.

INTRODUCTION

The advantages of digital transmission methods are being increasingly exploited in broadcasting to provide new, better or previously undeliverable programme services. The use of digital techniques is spreading throughout the broadcasting spectrum.

The advantages accruing from digital techniques are mostly of a technical character (in the sense of optimization of quality or a better use of the spectrum) and only represent a real improvement for the customer if they can be introduced with reasonable cost and effort.

This is especially true for AM radio broadcasting in general, and particularly for international short-wave broadcasting, because the customers of the broadcasting services are distributed world-wide.

THOMCAST PROPOSED SOLUTION

Introduction

Skywave 2000, the digital system proposed by Thomcast, is the result of an **optimisation** between possible data rates, bandwidth, channel coding, complexity, quality and flexibility.
Its **incremental architecture** allows easy and transparent adaptation to bandwidth, bitrate and level of protection which are required in all present and future implementations, without any change on the receiver side.

General overview

Skywave 2000 provides a **single solution for all AM frequency bands** (LW, MW, SW) which will bring benefits to the listener and to the radiobroadcaster in terms of simplicity of receivers, economies of scale, and a wider introduction of the new digital transmission mode.
It is the **result of a global system approach** considering both existing receiver and transmitter techniques and easily implemented at low cost with the technology available today.

In addition, Skywave 2000 approaches the problem of the transition period between the introduction of digital AM Radiobroadcasting today and the future fully digital multi-services by offering a **progressive, compatible** (digital and analog) signal which can be received by both today's conventional consumer receivers and by future low cost digital receivers.

Skywave 2000, is based upon a **parallel modem** which has been proven to be an efficient, reliable and flexible solution in presence of disturbed propagation channels especially within the Short Wave frequency range or Medium Wave during night-time.

Basically the system is adaptable to **any multi-bit rates source coder** standard: it has been tested with generic MPEG 2 Layer II and MPEG 2 Layer III.

The next step will be an advanced source coder providing a scalable structure needed to permit graceful degradation of reception.

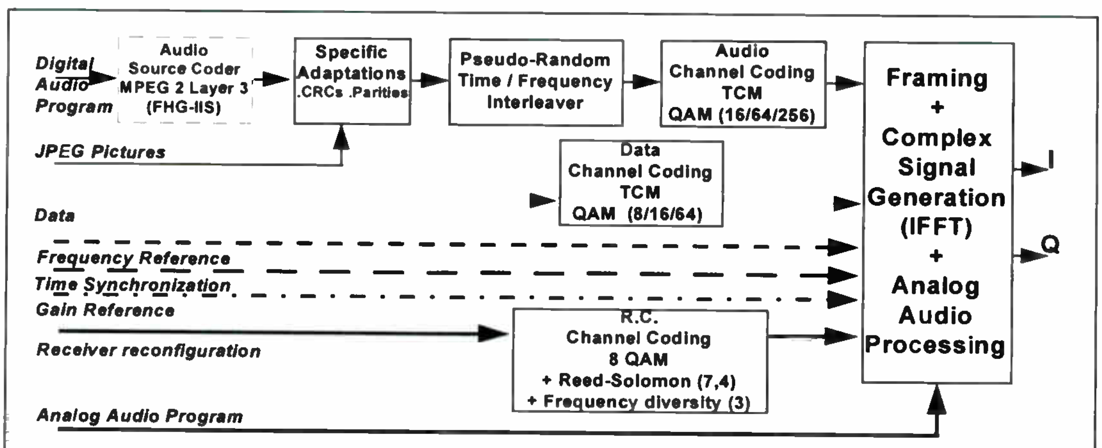
Nevertheless, the final coder has to be one which provides the best perceptual quality in real transmission conditions. Taking into account the effect of transmission errors, some specific adaptations such as CRCs and parities have to be added to critical parameters at the transmitter site to be associated with bad frame indicators at the receiver.

In order to offer a high spectral efficiency of 2 to 3 useful bits/s per Hertz, without increasing considerably the output bandwidth, the parallel modem uses a multi-carrier transmission scheme with trellis coded modulation (TCM) associated with a digital QAM modulation mode in an OFDM structure.

A coherent demodulation is performed by permanent channel estimation.The system is designed to provide fast synchronization and to be insensitive to Doppler shifts (for example ± 300 Hz to 500 Hz).

The system is able to carry a multiplexed frame containing multi-services with different levels of protection according to the sensitivity of the carried information (see below figure n°1: Functional Block diagram):

- digital audio program and associated pictures
- data services: text, messages, still pictures
- data for internal system information. This highly protected low bit-rate data stream is devoted to the receiver reconfiguration. It conveys the transmission parameters:
  * modulation format (16 QAM, 64 QAM or 256 QAM)
  * the interleaving depth
  * the total bandwidth, ...
- an analog program in simulcast mode.



Functional Block Diagram FIGURE N° 1

System Architecture

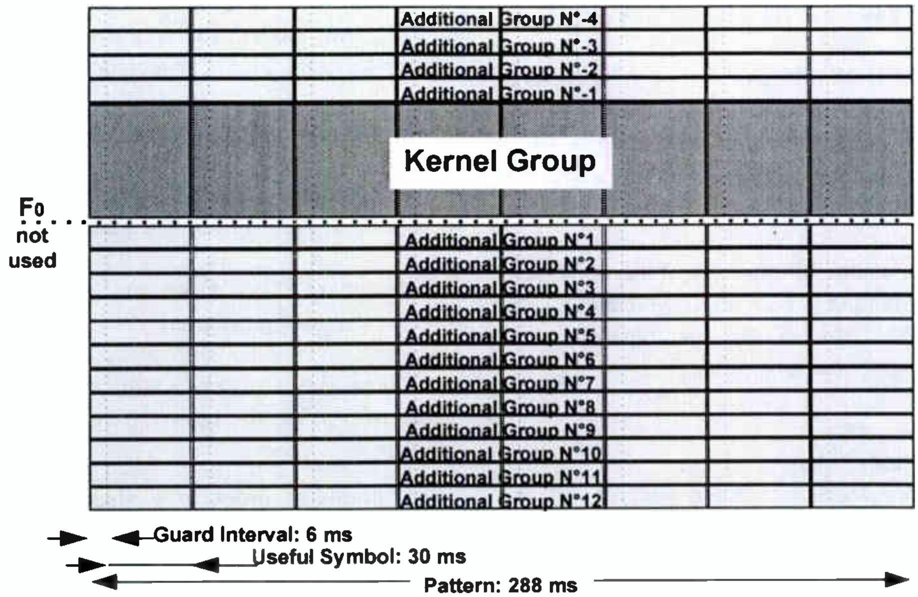
The incremental architecture (see below figure n° 2: Incremental architecture) of the system allows easy and transparent adaptation to bandwidth, bit-rate and level of protection which are required in all present and future implementations, without any change on the receiver side. The transmitted signal consists of:

- one kernel group of carriers of 3200 Hz total bandwidth containing all the signals which are necessary for frequency reference, time synchronization, gain references and receiver reconfiguration.

The kernel group of carriers also includes a minimum of audio and picture symbol blocks comprising 576 audio and picture symbols in 288 ms.

- a number of additional groups of carriers, each of 366.6 Hz bandwidth, each conveying a nominal bitstream of 72 audio and pictures symbols and 5 data symbols.

The number of additional groups depends upon the total available bandwidth.



Incremental Architecture

FIGURE N° 2

Such incremental architecture associated with different QAM modulation modes (16, 64 or 256) offers a maximum of flexibility as it can be easily adapted to the available bandwidth or the propagation channel characteristics.
As an example:

- 64 QAM as the standard modulation mode
- 16 QAM as fall back mode for Short Wave during difficult transmission periods (sunrise, sunset)
- 256 QAM for Long and Medium Waves day-time transmission for maximum audio perceptual quality.

The capacity of the presented system in terms of useful bit-rate versus bandwidth is given in the following chart:

	KERNEL GROUP	Additional GROUP	Kernel Group + 4 Additional Groups	Kernel Group +16 Additional Groups
Bins	96	11	140	272
Bandwidth (Hz)	3200	366.6	4666.4	9065.6
Audio/Picture Symbols:				
- 16 QAM (kbit/s)	6	0.75	9	18
- 64 QAM (kbit/s)	8	1	12	24
- 256 QAM (kbit/s)	12	1.5	18	36
DATA Symbols:				
- 8 QAM (kbit/s)	-	34.7	138.8	555.2
- 16 QAM (kbit/s)	-	52.1	208.4	833.6
- 64 QAM (kbit/s)	-	69.4	277.6	1110.4
Remote Control Symbols for Receiver Reconfiguration	21	-	21	21

THOMCAST'S SKYWAVE 2000 DEMONSTRATOR

Throughout 1996 and 1997 different demonstrations have been performed showing **Skywave 2000** capabilities and experiments progress in order to **demonstrate the short term feasibility** and to inform a large public about the promising possibilities of digital AM.

From 3rd Radio Symposium Montreux in June 96 to IBC Amsterdam in September 97, Skywave 2000, the Thomcast Digital AM system has demonstrated fast progress and real life, over the air results.

General Overview

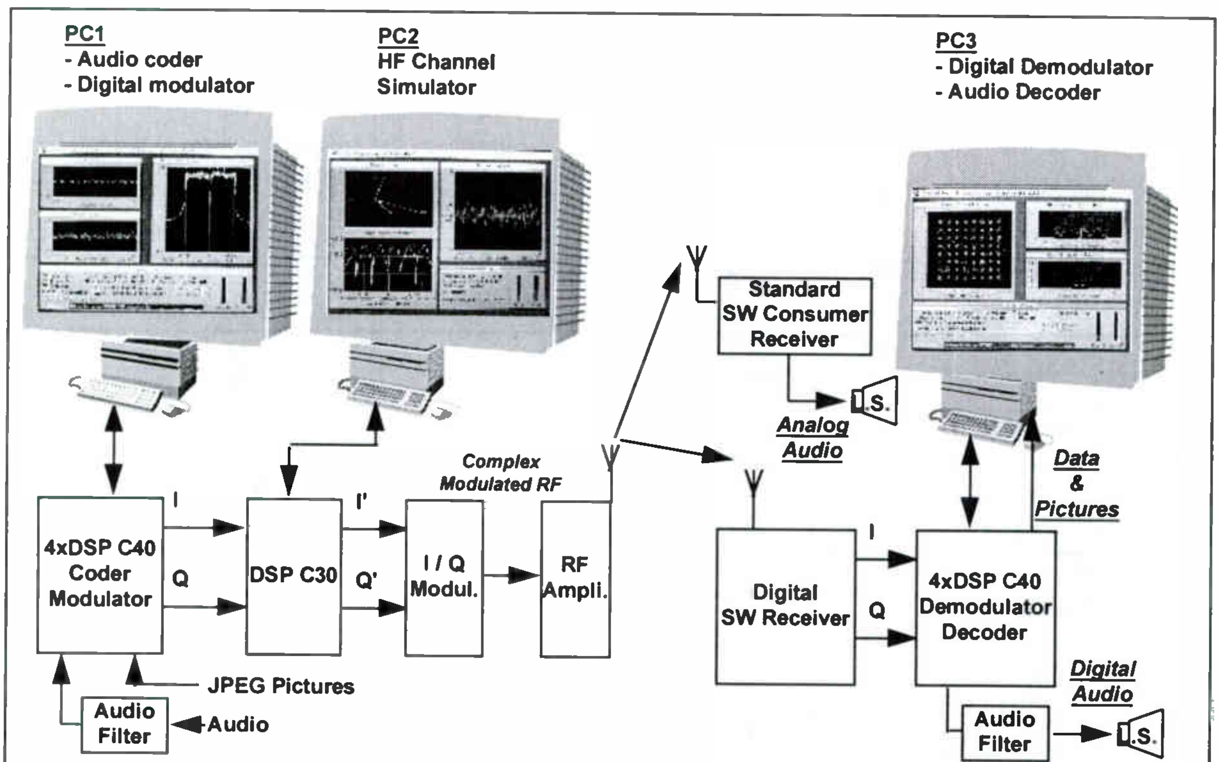
Various demonstrations have shown the system capabilities and demonstrated the rapid progress of the development of the system in order to prove the short term feasibility and to inform a large public about the promising possibilities of digital AM.

The demonstrator, as shown in the following block diagram (see below figure n° 3: Demonstrator Block diagram), consists mainly of:

- a **real time digital processing systems** for coder, decoder and modem (already described), implemented on standard Digital Signal Processor (DSP) PC boards
- an **HF channel simulator**
- a combined analog / digital low power exciter operating in the SW frequency band
- a standard Short Wave consumer receiver dedicated to receive the analog program
- a professional receiver with selectable filters is used in order to provide the reference receiving capability of the system.

In addition, in order to evaluate the capability of the existing receiver technology, a standard Short Wave consumer receiver has been modified to receive the digital program.

The modification to the consumer receiver consists mainly in an implementation of an IF2 output with a wider IF2 filter feeding to the digital decoder.



Demonstrator Functional Block Diagram

FIGURE N° 3

Demonstrator flexibility
The demonstrator developed for demonstrations and experimentations allows working with real AM broadcasting operating conditions and offers maximum flexibility in the choice of transmission channel characteristics (already described) and transmission modes.

Among the large choice of transmission modes:

- standard AM DSB

- SSB: USB or LSB

- simulcast (Analog compatible AM + Digital) with two versions:

  * AM Simulcast

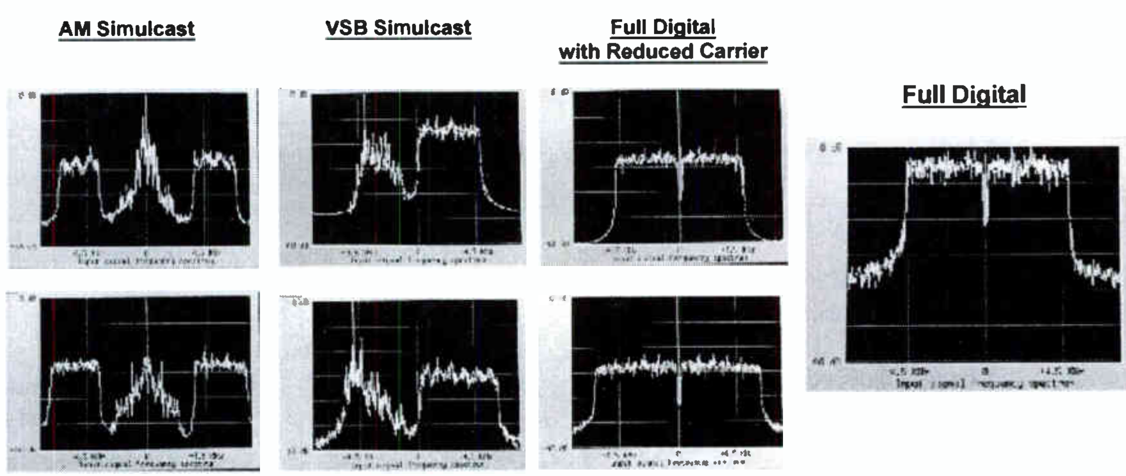
  * VSB Simulcast

- full digital with different versions:

  * with reduced carrier for implementation in some existing transmitters

  * without carrier for future implementation in transmitter dedicated for digital.

The flexibility of this demonstrator allows adaptation of the useful transmitted bit-rate according to the available bandwidth whatever is the chosen modulation mode (see below figure n° 4: Modulation Modes).



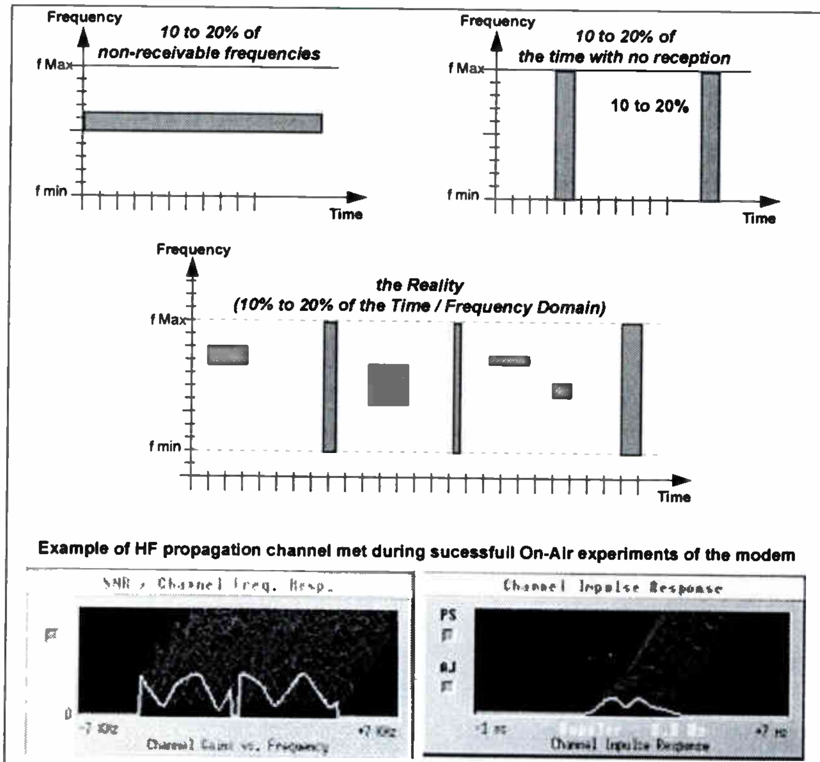
Modulation Modes

FIGURE N° 4

SYSTEM PERFORMANCE

Overall performance
The performance given in this chapter, except when indicated, are not simulated results. They have been obtained through real on-air operation of the already described demonstrator.
Overall performance can be illustrated by means of the following figures (see below figure n° 5: Overall Performance of the Modem in the Frequency and Time Domains).

This figure illustrates the general ability of the modem to operate correctly with 10 to 20% of the Time domain and / or Frequency domain unavailable. In real conditions, the modem will encounter a mix of time and frequency domain impairments as well as SNR impairments.
The modem's capability to deal with these impairments is mainly dependant on the global SNR values.



Overall Performance of the Modem in the Frequency and Time Domains

FIGURE N° 5

Experimental results

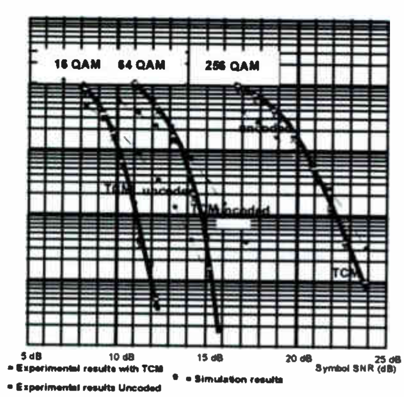
The receiver of the demonstrator having the capibility to perform real Symbol Signal to Noise Ratio (SNR) measurement, the indicated values correspond to the real measures and not to the SNR selected on the HF simulator (see below figure n° 6: Symbol SNR within AWGN Channel and figure n° 7: Eb / N0 within AWGN Channel).

Experimental results of the modem in presence of multipath channel has been performed with the standard modulation mode (64 QAM) and using the Medium Time Interleaving (4 x 288 ms).

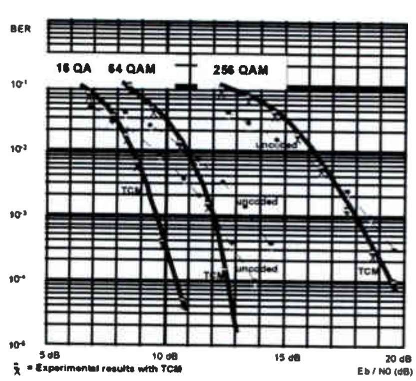
The measured Symbol SNR is the one measured at the reception (real SNR) and not the SNR selected on the HF Channel Simulator.

Symbol SNR is given for 4 different well known characteristic multipath channels.

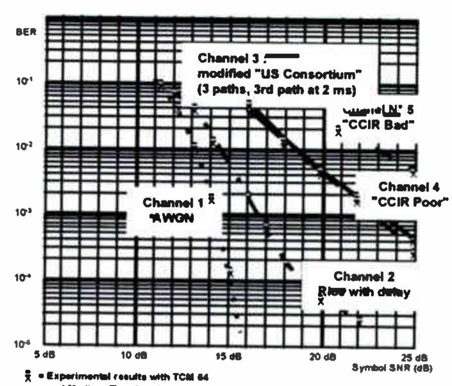
(see below figure n° 8: Symbol SNR with Multipath Propagation)



Symbol SNR within AWGN Channel *FIGURE N° 6*



Eb / N0 within AWGN Channel *FIGURE N° 7*



Symbol SNR with Multipath Propagation *FIGURE N° 8*

Field tests

In addition to these experiments performed with the demonstrator on the HF channel simulator, the demonstrator has been tested On-Air during public demonstrations in 1997 with real HF transmissions using non-modified existing transmitters (with both modern transmitter types: PDM and PSM high level modulation).

For these tests, the following test bench was used (see below figure n° 9: Test Bench Functional Block Diagram):

For these On-Air tests, the following technical characteristics can be mentioned:

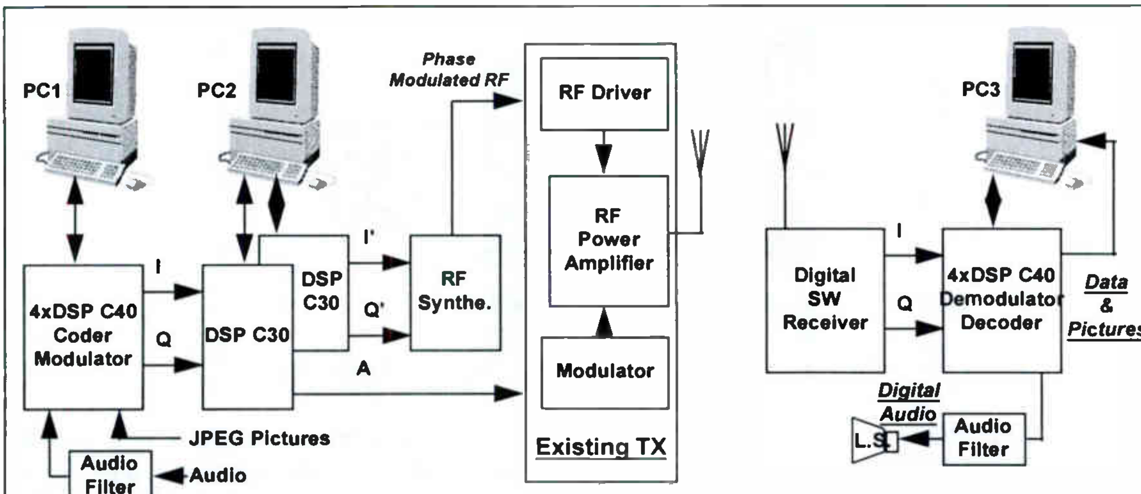
- transmitter modulated in phase and amplitude by a complex signal from the demonstrator

- transmitted power:

* useful power with the digital program: 30 to 100 kW

* carrier power: 60 to 100 kW

- Symbol SNR at the reception site: 17 to 25 dB.



Test Bench Functional Block Diagram

FIGURE N° 9

During these On-Air tests, the modem has proven its capability to operate in presence of multipath propagation, as an example these extreme conditions were met:

- up to 4 paths with fading between 25 to 30 dB:

* the first and second path at the same level with a delay of less than 0.5 ms

* the third path 10 dB below the two first with a delay of less than 2 ms

* the fourth path 10 dB below the third one with a delay of less than 3 ms

- analog program transmissions on the adjacent channels.

The results of these On-Air tests can be summarized as follows:

- good permanent reception of the audio digital program at nominal TCM 64 QAM modulation mode with « grade 4 or 5 » for any Symbol SNR $\geq$ 22 dB

- no audible perturbation of the occupied adjacent channels.

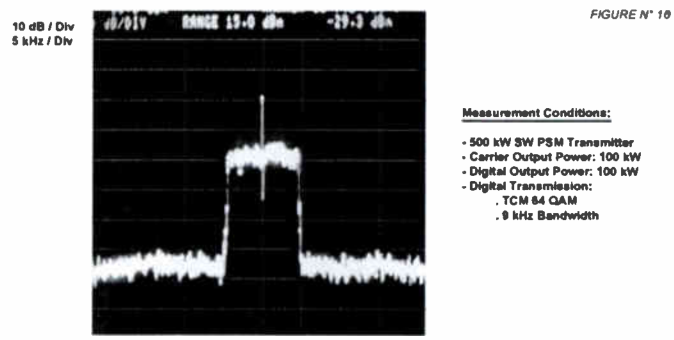
TRANSMITTER HARDWARE

The signal at the output of the digital modulator is a complex type signal represented by its Cartesian coordinates I for the In phase and Q for the Quadrature. To be transmitted by modern transmitter, it has to be converted into polar representation with Amplitude (A) and Phase (φ). Such a technique called « Kahn », is already in operation in all modern Class C transmitters equipped with SSB (Single Side Band) modulation.

Nevertheless, even if experimental tests can be performed with any type of modern transmitter without hardware modification, the caracteristics for a digital signal in terms of, bandwidth, crest factor, linearity, group delay and incidental phase modulation due to the envelope, have to be improved.

The current results as concerns in-band (Signal to Noise Ratio) and out-of-band (Shoulder) are not fully compatible with a new audio digital service providing a « commercial quality » under severe propagation conditions. In addition, the protection for the adjacent channels should be improved to comply with the existing recommendations as shown in the following spectrum photo taken during field tests (see below figure n° 10: Spectrum at the Output of a Non-modified Existing Transmitter).

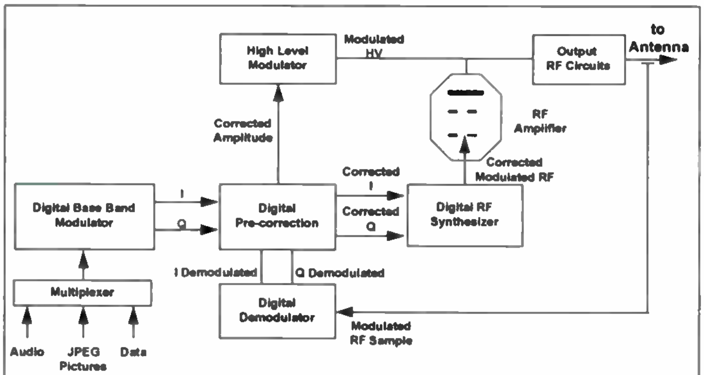
FIGURE N° 10



Spectrum at the output of a Non-modified Existing SW transmitter

At this stage of investigation, we can quantify the remaining improvements which have to be applied to the audio frequency bandwidth of the envelope modulator (minimum 20 kHz, preferable 30 kHz) and the neutralization system of the final RF power stage.
We are confident that these improvements can be easily implemented and that a satisfactory solution will be deliverable soon.

To start the improvement phase with minimal cost of hardware modifications, digital signal processing, such as dynamic pre-correction can be applied as described in the following block diagram (see below figure n° 11: Implementation on Existing Transmitter).



Implementation on Existing Transmitter

FIGURE N° 11

CONCLUSION

The above described experiments have proven the principal capabilities of **Skywave 2000 as a system for digital short-wave broadcasting.**

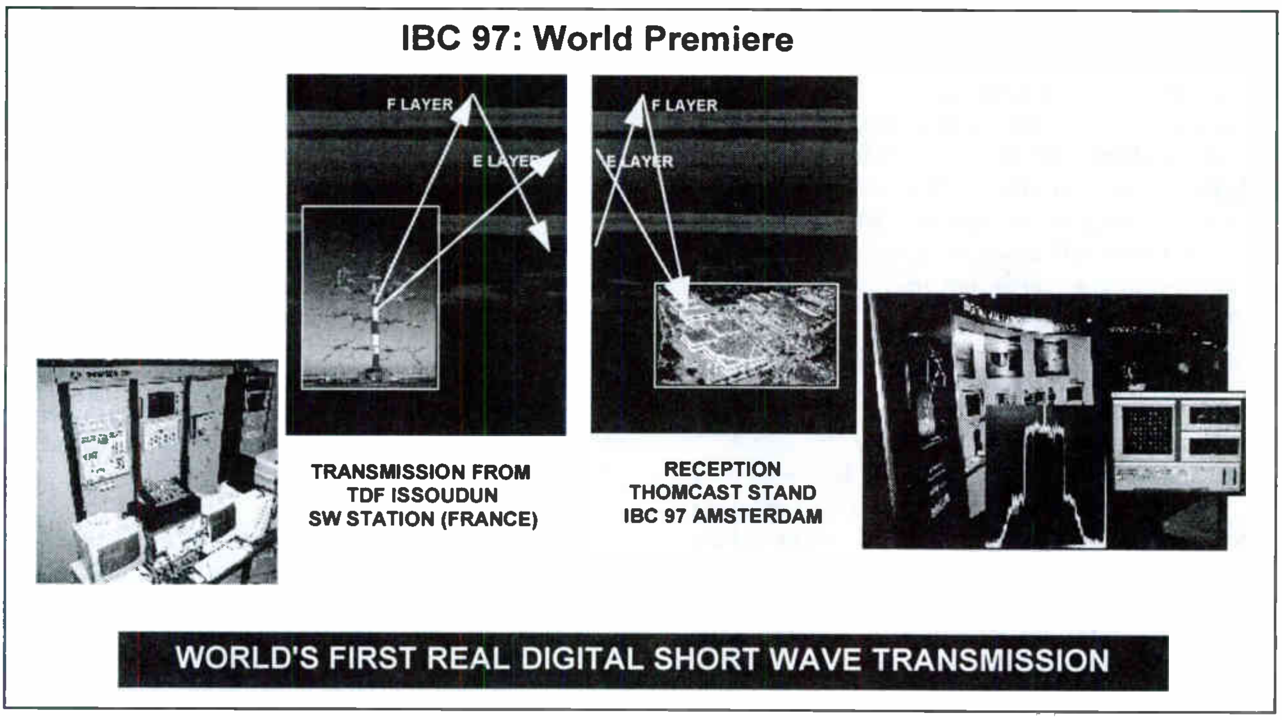
Since the requirements of Short Wave in terms of system robustness and propagation channel characteristics can be regarded as more severe than for Long Wave and Medium Wave transmissions, Skywave 2000 will also be perfectly **suitable for those lower frequency ranges.**

Skywave 2000 has shown itself to be a **valid proposal for a universal AM band Radiobroadcasting standard** which could bring valuable improvements when compared to current analog services as concerns:

- audio quality improvement,
- area coverage which will normally require much less power for digital transmission to cover the same area with better signal quality,
- compatibility with the existing services.

Today's AM and SW broadcasters and the public have a strong interest in conserving use of the unique characteristics of their propagation media well into the future.

Skywave 2000 represents a **valid system for all AM frequency ranges.** It constitutes the current Thomcast contribution for the two international on-going programmes (**Eureka 1559 NADIB** and **DRM Digital** Radio Mondiale).



CRITICAL ISSUES AND CONSIDERATIONS FOR AN ALL DIGITAL TRANSMISSION PATH

Frank Foti
Cutting Edge
Cleveland, Ohio

ABSTRACT

The trend in FM broadcasting is the all-digital transmission facility. Some believe that digital connectivity is merely as easy as joining together numerous AES/EBU signals and magically, the digital path appears. Real life experience indicates there is more to it than that. To date, there are known situations where a digital transmission path has caused added distortion, loss of loudness, overshoots, and/or excessive delay. A real world problem exists today! This paper examines the all-digital path and illustrates where benefits are realized, and where potential hazards can occur. An in-depth look at sample rate converters used in digital exciters will reveal the cause of modulation overshoots. Finally, a proposal is offered to illustrate the need for connectivity of a digital audio processor that incorporates a stereo generator with a digital exciter.

OVERVIEW

Digital. The technical *buzz* word of the 90's used as if it's a magical potion that makes everything perfect...well almost perfect. Truth is, under certain circumstances— or without a complete understanding— it can actually be degrading! Let's take a look at the digital broadcast facility from top to bottom. Sure, numerically, the signal path can now transverse from the microphone all the way to the FM exciter, but is that journey sonically as pure as analog? Some argue yes, and some argue no.

This presentation details where the strengths and weaknesses are in the digital path. Make no mistake: There is potential to create an outstanding performing broadcast facility using digital, but certain parameters must be observed and some guidelines understood. At issue are discussions about: sampling rate, AES/EBU, time delay, STL systems, codecs, and digital exciters. Each of these items play an important part in the all-digital facility. Before getting started, some history is in order.

BACK TO THE FUTURE...

If you're a broadcast veteran whose career spans back to the early/mid 1970s, the following reflection will hopefully be of amusing interest to you. If you're one of the younger "pups" to radio, please read on for some interesting revelation.

In the early/mid 70s, remember the first attempts at "competitive" audio processing in FM? You know, inserting a final limiter/clipper before a stereo generator, thinking that the hard limiting action of the clipper would stop over-modulation. Wow, rocket science you thought back then, right? Wrong! Remember the crashing and burning of that concept as the modulation would exhibit wild overshoots? Peaks that occasionally would reach up to 170%. Loudness was not gained, but lost! What happened?

As history shows, it was the non-linear response of the 15kHz low pass pilot protection filters in the stereo generator that were the culprits. Later, Robert Orban would research, design, and develop a complete processing system that eliminated the problem by integrating the hard limiter, final filter, and stereo generator together. From that point on, "competitive" audio processing was possible.

Why tell that story? We all know it. How does it relate to the topic at hand? Well, based upon performance thus far with certain digital transmission paths, some of those older 1970s problems have come back to haunt us. Even some of those veteran engineers of the 1970s are beginning to think that the digital path is revisiting history. To date, there are concerns that some digital systems exhibit overshoots, which can reduce loudness. Some generate a significant amount of time delay through the system. Enough in fact, that air talent can not monitor off-air in their headphones. Other paths are known to use a codec inside STL systems. That can have adverse effects on the audio, depending upon the coding algorithm and bitrate employed. Finally, the issue of sampling rate, and sample rate converters must be revisited. Is 32kHz enough for quality FM broadcasting that truly rivals analog? (I can hear the analog fanatics screaming already.)

How do we avoid these issues? Unfortunately, the answer can not be reduced to a single item as it was with the low pass filters in the stereo generator. It's a broader concern.

A SYSTEM-WIDE VIEW

As stated earlier, the total digital transmission path is more than a connection of linear AES/EBU signals that exist between program origination and the transmitter. While that is desirable and, hopefully, possible soon, the path actually exists as a number of cascaded digital signal types that must be made compatible with one another. For example, the digital signals may vary in sampling rate among various components in the system. Also, the use of a codec-based STL adds yet another dimension to the sonic aspects of the audio, as well as sampling rate.

Placement of various components that make up the system can have a dramatic effect on performance. Probably the most important is audio processing. It's location within a specified system, coupled with the remaining items that comprise that particular system, will have monumental performance benefits or drawbacks. Where pre-emphasis is inserted is yet another important concern.

The only exception to this, would be a digital facility where the entire plant is co-located. In that instance, it is possible to connect the path via linear AES/EBU between the studio and signal processor, and then between the processor and the exciter.

These are only the surface issues. Of equal importance are propagation time delay. Can the on-air talent continue to monitor themselves off-air? Sample rate conversion: Does it adversely affect the output of an audio processor when coupled to a digital exciter?

Each of these items are critical in a digital audio path. In some instances, the path may have to deal with one or all of them. Following are in-depth views of the important aspects that comprise the digital transmission system and recommendations that yield beneficial performance improvements.

PATH CONFIGURATIONS

The digital transmission path can be configured in a few ways. For discussion purposes, it will be assumed that some form of Studio-To-Transmitter (STL) link is involved.

First consider if the STL path will be linear, or coded. The goal is to be as linear as possible, but sometimes circumstances may make this possible.

If the path is coded, the next decision involves where to insert the audio processing. The preference is to place the processing at the transmitter for technical and sonic reasons, but it is much easier to adjust if installed in front of the STL. Figure 1 is an example of the processing located after the codec and at the transmitter sight.

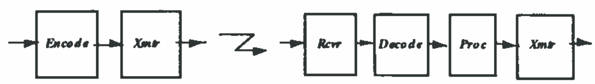


Figure 1

Figure 2 is an example of the processing inserted before the codec.

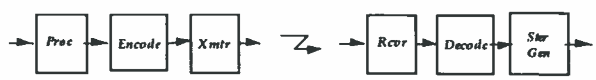


Figure 2

If a linear path is possible, then installation of the processing can be at either the studio or transmitter location. Good technical and sonic performance will be possible at either location.

THE COMPONENTS

Examining the transmission path is literally the sum of its parts. This section describes and details each of these components so that a better understanding of the complete system will result. The one commonality to the whole system is the AES/EBU interface. The author will assume that the reader has enough understanding of this standardized protocol, such that further discussion and review is not warranted. AES/EBU should be viewed as the "glue" that ties each of the pieces together.

Digital FM Exciters

Digital FM exciters are the latest entry to the digital path. Capable of incredible modulation performance, the digital exciter offers two forms of signal input. Analog composite (MPX) for the non-digital transmission sight, and AES/EBU.

The analog MPX input to be modulated digitally, requires a very high sampling rate. Consider for a moment that the audio spectrum of FM is 99kHz, the digital exciter must provide a sampling rate of at least 200kHz. That would provide a Nyquist frequency at 100kHz, which would cover the baseband spectrum.

The AES/EBU input accepts the Left/Right signal in the digital format. Because it still is in the discrete Left/Right state, the exciter must perform the stereo generator function. Here is where the story gets interesting.

Consider for a moment the signal at the AES/EBU input of the exciter. It might be a different sampling rate than the exciter is expecting. If so, a sample rate converter is employed to make the proper transition. This can pose problems, as the digital filter in the sample rate converter can generate overshoots to the already tight peak-controlled audio data that is being converted.

As mentioned, the audio will have already been peak-controlled and bandlimited by the audio processor. (The processor can be either analog or digital, it does not really matter for this discussion.) The processor also would have already applied the needed pre-emphasis. Hence, the Left/Right stereo signal only needs the matrixing and MPX encoding for stereo modulation to occur. That is all.

What is present in most digital exciters is yet another low pass filter, potential stereo limiter, and in some cases the addition of, yet again, pre-emphasis. The latter may occur, if the incoming Left/Right audio signal was de-emphasized earlier in the path.

In essence, the signal that only needed to be matrixed and MPX encoded has now had additional elements of conditioning applied to it. This can degrade the modulation efficiency and sonic performance. Let's have a look at why.

Sample Rate Converters (SRC): This device transforms one system sampling rate to another. This becomes necessary when interfacing digital equipment that uses different sampling rates, and thereby permitting compatibility among different systems.

In an example of changing 48kHz sampling to 32kHz sampling, the conversion is accomplished by scaling up, or interpolating the original sampling rate, usually by a factor of ten; then, at the 10x rate 480kHz, filtering the signal with a low pass filter that is set to the Nyquist of the new, *desired*, sampling rate. This filter is required to "'smooth out" the 10x rate. If it was not used, aliasing products would result. Finally, the signal is scaled down, or decimated by the factor needed, in this case ÷15, to the achieve the new rate of 32kHz. Figure 3 is a block diagram of a SRC. While this sounds quite simple—and basically it is—there are a few issues to consider. Of main interest is the interpolation filter.

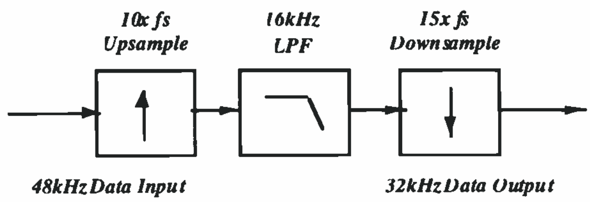


Figure 3

All audio processors, both analog and digital, apply some form of overshoot control in the output filtering section. In most designs, this function is performed by an integrated protection clipper working around the final low pass filter.

In each case, the overshoot component can be determined by the "Gibbs Phenomenon"[1], which states that an overshoot will occur at one-third the cut-off frequency of any low pass filter whenever a non-linear waveform is passed through it. In the case of broadcasting, the non-linear waveform would be that of a clipped signal.

Knowing that the audio bandwidth used in FM stereo is 15kHz, overshoot components will begin above 5kHz with any non-linear waveform.

Should the slope of the previously described up-sampled interpolation filter appear greater than the slope of the final filter in the audio processor, then output overshoots may result in the sample rate conversion process. Unfortunately, these overshoots are generated after the audio processor. To remove them would require another limiting device—thus the reason for the added limiter in the digital exciter.

Not all sample rate converters will cause overshoots. But in most cases, the filtering used in the sample rate converter will be of a large magnitude in the bandstop rejection area. In all probability, it will be an FIR filter with at least 96dB rejection in the stop-band.

Of interest is the direction of rate conversion. Should the host sampling rate be higher in value, than the incoming rate, chances of overshoot are small. This happens due to the up-sampled filter being set to a broader spectrum than the spectrum of the incoming signal. Potential problems may arise when transforming a larger sampling value to a lower rate, as the example of converting 48kHz to 32kHz sampling. Then the details of the above description apply.

Input Sampling Rate: It is unknown why 32kHz sampling rate is used in broadcast paths, when it has been discussed and demonstrated that 48kHz sampling is far superior in performance. The importance is not so much the added spectrum available with 48kHz sampling, but that 32kHz causes aliasing distortion in specific instances. This is clearly demonstrated by any DSP based audio processor designed before 1997, and extensively detailed by the author in a previous NAB presentation[2].

The use of 48kHz sampling as the AES/EBU input rate in the exciter ensures the best sonic performance. In addition, any input that needed to be converted up from 32kHz sampling would not create any overshoot component in the modulator. These factors ultimately benefit the broadcaster.

Integrated Limiter: Due to the above SRC scenario about SRC, most digital exciters provide a baseband limiter to eliminate the overshoot problem. Also, there are certain path configurations that can cause overshoots that do not relate to the sample rate conversion process. Those are usually situations that involve use of a coded STL system where the desire is to insert the main audio processor before the encoder of the STL. It has been shown that employing audio processing in front of a codec can have sonic and modulation performance penalties. The codec issue will be discussed in an upcoming section.

The integrated limiter used in the exciter is combined with part of the stereo generator. This is not a composite clipper, but a time-delay, feed-forward limiter that controls peaks with a zero attack time. Waveforms are controlled with little or no harmonic distortion (T.H.D.) components, but will produce a larger intermodulation (I.M.) level.

Technically, this style of limiter will operate sufficiently when controlling overshoot peaks or as an additional limiter to the audio processing. Sonically, this type of limiter will produce a "busier" sound. It will sound more like a limiter that is operating with "heavy" levels of compression. That is the result of the added I.M. In the audio processing realm, adding more I.M. to an already processed signal is undesirable

Of interest to the author is that a digital overshoot clipper is not employed. It has been proven that digital composite clipping is possible, while maintaining a clean spectrum. Composite clipping produces far less I.M. products, as does a delay limiter, and it will yield cleaner sound for the same amount of limiting/clipping used.

Pre-emphasis: The exciter has the option of adding the required pre-emphasis. Optimally, the transmission system would be set up so that the pre-emphasis is generated once in the audio processor. Then, the pre-emphasized, processed signal is coupled directly to the exciter.

Broadcast audio processors employ pre-emphasis within system architecture. Since emphasized audio must also fit within the imposed modulation limits, the processor employs specialized high frequency control sections that provide both the boost and control of the high frequency energy. In this manner, efficient high levels of modulation are easily obtained since the processor is designed and set to limit any tradeoffs resulting from pre-emphasis and high frequency limiting requirements. Basically, these two sections work in concert with one another to allow pre-emphasis to be employed, and yet control the emphasized energy content.

In situations where a codec STL system and audio processing are inserted before the encoder—the codec must pass "flat" (non pre-emphasized) audio. This requires adding de-emphasis to the output of the processor in order to send the restored "flat" signal to the codec. Figure 4 illustrates this. A flat signal is required by the coder because of the use of masking in the encoding process. Any significant change, or imbalance of the frequency spectrum, can cause the threshold curve of the coding system to possibly have a profound effect on the output of the coded audio[3].

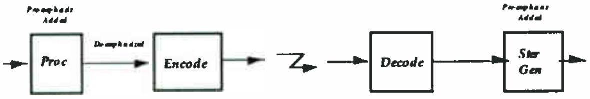


Figure-4

These additional de-emphasis and pre-emphasis steps will add modulation overshoot to the total transmission system. To eliminate the added overshoot, another limiter must be employed.

Unfortunately, tests have shown that operating a transmission processor with an emphasized output into a codec will generate audible high frequency distortion. This occurs because the spectral balance to the codec masking process is not spectrally flat, which is what the masker signal wishes to operate on.

Based upon the previous discussion, one can see that it is advantageous to install the audio processing system as close to the exciter as possible. As it allows employing the emphasis in the processing, and using a "flat" input on the exciter. Also, internal limiting in the exciter becomes unnecessary and allows the audio processing system to provide all of the required peak control.

STL Systems

There is a choice in digital STL systems: Linear or coded. The linear systems have mainly been available as "nailed-up" data links, (such as T-1) but recently there has been the introduction of the "uncompressed" radio links as well. In analyzing the STL link, two items are of critical interest: Time delay and coding algorithm.

The first item, time delay, is an issue with either the linear or coded STL system, as there is a propagation delay (in milliseconds) for the audio data to travel from the input to the output of the system. If the delay is excessive, then the on-air talent can not comfortably monitor themselves off-air. This occurs because the off-air audio in their headphones is delayed relative to the arrival time of their voice directly through bone conduction.

The following table, based on real-world tests, was created by a radio engineer to illustrates the effects of time delay for on-air talent:

- 1-3 ms: Undetectable delay.
- 3-10 ms: Shift in voice character audible to person speaking. (comb filter effect)
- 10-30 ms: A slight echo turning to obvious slap @ 25-30 ms.
- 30-50 ms: Disturbing echo, disorienting the announcer.
- >50 ms: Too much delay for live monitoring.

Thanks to Jeff Goode in Indianapolis for providing this information! The key break points seem to be that delays of 5-7 ms are not a problem; from 10-30 most announcers can work live, but anything above 25-30 is annoying. These are real world tests.

While the subject of time delay is being discussed, it must be pointed out that time delay is a cumulative item. Each device that exhibits any delay will add up to produce a total system delay. Should this system delay exceed some of the figures in the above chart, then off-air monitoring will not be possible. In addition to the STL system, audio processors, digital exciters, and even digital modulation monitors can generate delay.

A possible second item for consideration—coding algorithm—is only applicable if a non-linear STL is used. These devices make use of 'lossy' data reduction algorithms to compress the so it can fit within the existing bandwidth of the STL system. While there are a number of specific algorithms from which to choose, most STL manufacturers have made use of proprietary digital formats that are derivatives of prior development. Most common usage has been ISO/MPEG Layer-II, ISO/MPEG Layer-III, apt-*x*, and Dolby AC-2.

Detailed operation of the above-mentioned algorithms is not needed for this discussion, as each system possesses both strengths and weaknesses for this application.

What *is* of importance here is to understand that each audio coding algorithm will have a specific sonic effect on the audio. Along with the fact that the signal will also be dynamically controlled, the use of audio coding can have a negative effect on the performance of the audio processor. It has been demonstrated that any use of a coded STL should be done where the audio processor is inserted *after* the coding. There will be less degradation of the audio, and the processor will perform peak control more efficiently.

SAMPLING RATE

Stated earlier, the issue of sampling rate must be looked at again: 32kHz is simply too low. With the cost of DSP chips, and converter sets constantly falling in price, there is no reason why 48kHz can not be used. We have now reached a point in the professional audio domain where 96kHz is the desired sampling rate. How will that affect the broadcast community?

Transmission systems thus far have used 32kHz as a base sampling rate, which in turn sets the Nyquist freq. at 16kHz. Considering that conventional FM stereo broadcasting requires 15kHz of audio bandwidth, this leaves only 1kHz of guard band spectrum before the Nyquist point. To facilitate this, a filter of very large magnitude must be employed in order to suppress all energy by at least 96 dB at the Nyquist, or aliasing occurs. This can be accomplished digitally using a finite impulse response filter (FIR). The only drawback is that many "taps" are required within the filter to achieve this level of stopband rejection. The significance of the 'taps' is that for every two taps in the filter, it requires one sample to perform its duty. For a 15kHz FIR filter of this magnitude, 101 "taps" are needed. This in turn results in 50 required samples, equating to 1.56 milliseconds of propagation delay through the filter.

Add up the number of 15kHz filters employed in a 32kHz sampled transmission path, and those alone will create enough time delay to disorient on-air announcers.

A broad question is: Why the use of 32kHz as a base sampling rate? Tests and research show that a base of 48kHz would make all of the aforementioned problems much easier to deal with. The guard band to the Nyquist is much farther out, which in turn moves out the aliasing point. This would allow a final filter with less time restriction, and the propagation delay associated with 48kHz would be faster in itself and makes this rate more desirable.

Most likely, 32kHz sampling was chosen in the past, because there would be more machine cycles available to handle the workload. That would be the only reason to possibly support a lower sampling rate.

DIGITAL EXCITER PROPOSAL

In the discussion section about the digital exciter, numerous options were explained about the interfacing possibilities of the audio processor to the exciter. All of them revolve around the usage of the AES/EBU input protocol. In that configuration, the audio data arrives in Left/Right format and requires the exciter to perform the MPX generation.

Question: Why can't the digital audio processor, which has the MPX encoder built-in, connect it's digitally-generated baseband signal directly to the digital modulator of the exciter? This would be analogous to the analog composite input on any exciter.
Figure 5 is a block example of this.

Figure 5

Unfortunately, there is not a standard for transporting the MPX baseband signal, but an "official" standard is not needed. Consider the following: In either the audio processor or the digital exciter, a fast sampled section is needed to accommodate the MPX signal. (Remember we're dealing with a spectrum out to 100kHz.)

Knowing that most DSP families use some form of serial data stream, it is possible to "publish" the needed input data format so that an audio processor can provide that format as a digital composite output. The important factor is that both systems agree on sampling rate. Again, the use of a very fast rate is advantageous, say, 384kHz. That's 8x above 48kHz.

Naturally, this type of configuration would require installing the processing at the transmitter facility, since transporting a digital composite signal of this speed and size would be cost prohibitive. Current generation digital processors provide some form of computer control via modem, or network making processing at the transmitter somewhat less inconvenient.

It is curious that none of the digital exciters available provide—or even propose—an application like this. It provides the best possible coupling to the exciter, and the performance benefits are significant. Imagine having the power of a complete digital processing system and integrated stereo generator that is directly connected to a digital modulator. Now we're talking about super efficient modulation capability. Zero overshoots due to added emphasis, coding, or sample rate converters. That's real power.

CONCLUSION

The total digital transmission path is capable of providing some outstanding performance results. To achieve this, audio processing must be inserted at the transmitter site, and a "flat" input should be used on the digital exciter. If an STL system is employed, a linear system would be preferable, but a high bitrate coded system is acceptable as long as the dynamics processing occurs after the coding.

As long as the systems design engineer in a broadcast facility is aware of these critical issues, there is no reason why an all-digital broadcast facility can not exist today, providing broadcasting of exceptional quality.

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ADVANCED AUDIO CODING FOR DIGITAL SOUND BROADCASTING

KARLHEINZ BRANDENBURG, OLIVER KUNZ, SCHUYLER QUACKENBUSH
FRAUNHOFER GESELLSCHAFT AT&T LABS - RESEARCH
ERLANGEN, GERMANY FLORHAM PARK, NJ,
U.S.A.

ABSTRACT

Digital sound broadcasting systems are always in need for better bandwidth efficiency. The recently standardized MPEG-2 Advanced Audio Coding system delivers a factor of two improvement in compression efficiency over older technology like MPEG-Audio Layer-2. It has been designed for all applications needing high efficiency compression for high quality audio signals supporting mono, stereo and multi-channel material. Even more flexibility and coding modes will be available with the upcoming MPEG-4 audio coding standard. It contains MPEG-2 Advanced Audio Coding and adds modes for low bit-rate speech coding, audio coding at very low bit-rates (down to a few kb/s) and the seamless integration of audio synthesis.

OVERVIEW

The ISO/IEC MPEG-2 Advanced Audio Coding (AAC) technology [1] delivers unsurpassed audio quality at rates at or below 64 kb/s/channel. It has a very flexible bitstream syntax that supports multiple audio channels, subwoofer channels and embedded data channels. AAC combines the coding efficiencies of a high resolution filter bank, backward-adaptive prediction, joint channel coding, and Huffman coding with a flexible coding architecture to permit application-specific functionalities while still delivering excellent signal compression.

AAC supports a wide range of sampling frequencies (from 8 kHz to 96 kHz) which enables it to have an extremely wide range of bitrates. This permits it to support applications ranging from professional or home theater sound systems to internet music broadcast systems.

A block diagram of the AAC encoder is shown in Fig. 1, in which each coding *tool* in AAC is represented by a block. The blocks are:

- **Filterbank** AAC uses a resolution-switching filterbank which can switch between a high frequency resolution mode of 1024 (for maximum statistical gain during intervals of signal stationarity), and a high time resolution mode of 128 bands (for maximum time-domain coding error control during intervals of signal non-stationarity).
- **TNS** The Temporal Noise Shaping tool (TNS) effectively modifies the filterbank characteristics such that the combination of the two tools is even better able to adapt to the characteristics of the input signal.
- **Perceptual Model** A model of the human auditory system that sets the quantizer step sizes based on the input signal statistics.
- **Intensity and Coupling, M/S** These two blocks actually comprise three tools, all of which seek to achieve coding gain based on correlation between two or more channels of the in put signal
- **Prediction** A backward adaptive prediction tool that attempts to remove additional intra-channel redundancy beyond that achieved by the filterbank.
- **Scale Factors** The filterbank outputs are scaled to modify the quantization step size. This does the coloration of the quantization noise according to the output of the psychoacoustic model.
- **Quantization, Noiseless Coding** These two tools work together. The first quanitzes the spectral components and the second applies Huffman coding to vectors of quantized coefficients such that the entropy per token (here taken to mean the probability of the token multiplied by the length of the token in bits) is as nearly constant as possible
- **Rate/Distortion Control** This tool adjusts the scale factors such that more (or less) noise is permitted in the quantized representation of the signal which, in turn, requires fewer (or

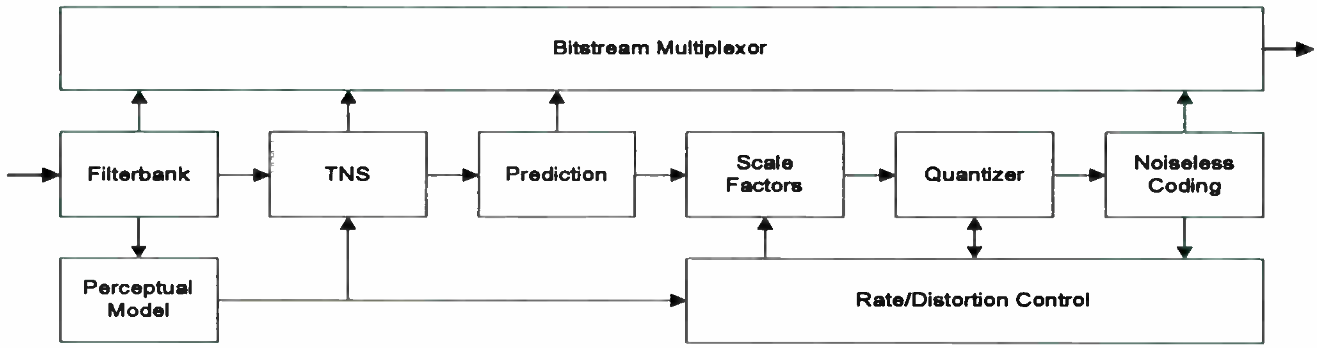

Figure 1: AAC Main Profile Encoder Block Diagram

more) bits. Using this mechanism the rate/distortion control tool can adjust the number of bits used to code each audio frame and the quantization noise in each scalefactor band.

- **Bitstream Multiplexor** Assembles the various tokens to form a bistream.

PROFILES

There are three profiles in the AAC standard: Main, Low Complexity (LC) and Scaleable Sampling Rate (SSR). Each profile specifies a different set of coding tools and a different set of parameters for those tools.

Main profile provides the greatest compression at a given quality, and is appropriate when processing power and storage are not of significant concern. With the exception of the gain control tool, all tools may be used in order to provide the best data compression possible.

Low Complexity profile provides nearly the same compression as Main profile for all but the most demanding signals, but requires significantly less processing power and storage than Main profile. The features in this profile are a subset of those in Main profile such that any LC profile bitstream can be decoded by a Main profile decoder. The most significant reduction in complexity comes from restricting the LC profile to not use the prediction tool.

The *SSR profile* is significantly different from the other two profiles in that it does not inter-operate with them, but rather is in itself a set of four inter-operable specifications that permit bandwidth scalability and hence sampling rate scalability. This is appropriate for applications that require a family of AAC decoders, each of which achieves a different price/performance tradeoff. Specifically, SSR profile is similar to LC profile but with a modified time to frequency mapping tool that splits the signal into four equal bands and applies gain control to each band prior to further processing. The bitstream has a matching embedded structure such that a fill-bandwidth SSR bitstream can be decoded by a 1-band, 2-band, 3-band or 4-band SSR decoder to yield 6 kHz, 12 kHz, 18 kHz or 24 kHz bandwidth output signals, respectively. As the number of bands supported by the decoder decreases, its complexity (and cost) also decreases.

AAC AUDIO QUALITY

Test Conditions: The AAC system has been tested for subjective audio quality in both 2-channel (stereo) and 5-channel (surround) presentation formats. It was tested in 5-channel format in the fall of 1996 at the BBC research and Development Department at Kingswood Warren, UK and at the NHK Science and Technical Research Labs, Tokyo, Japan [2]. In this test there were a total of 56 listeners, 32 at the BBC and 24 at NHK. It was tested in 2-channel format in the fall of 1997 at NHK Science and Technical Research Labs [3]. In this test there were a total of 31 listeners.

The 2-channel presentations tested seven coding systems: main profile AAC at both 96 kb/s and 128 kb/s, low complexity profile AAC at both 96 kb/s and 128 kb/s, scaleable sampling rate profile AAC at 128 kb/s, MPEG-1 Layer-2 at 192 kb/s and MPEG-1 Layer-3 at 128 kb/s. The 5-channel presentations tested four coding systems: main profile AAC at both 256 and 320 kb/s, low complexity profile AAC at 320 kb/s and MPEG-2 BC Layer II at 640 kb/s.

As much as possible, test methodology adhered to recommendation ITU-R BS 1116 [6]. The tests were conducted according to the triple-stimulus/hidden-reference/double-blind method, and the grading scale used was the ITU-R 5-point impairment scale. Descriptors assiciated with grades on this scale are as follows:

Grade	Degradation
5.0	Imperceptible
4.0	Perceptible but not annoying
3.0	Slightly annoying
2.0	Annoying
1.0	Very annoying

Test Results: The results from the tests are presented on an inverted scale of *difference grades* (diffscores), in which 5.0 is subtracted from all scores so that larger negative scores correspond to lower grades on the impairment scale. Test result mean scores and 95% confidence intervals on the diff grade scale are shown in Figs. 2 and 3.

The most significant outcome of the tests is that AAC is the first coding system that satisfies the requirements of ITU-R broadcast quality for both 2-channel presentation at 128 kb/s and 5-channel presentation at 320 kb/s. Specifically, in the 2-channel presentation AAC main profile at 128 kb/s and AAC low complexity profile at 128 kb/s qualify as ITU-R broadcast quality, with AAC scaleable sampling rate profile at 128 kb/s misses qualifying by a very slight margin. In the 5-channel presentation AAC main profile at 320 kb/s qualifies and AAC low complexity profile at 320 kb/s misses qualifying by a very slight margin.

In order for a coder to be considered ITU-R broadcast quality, decoded signals must be "indistinguishable" from the refrerence (uncoded) signal, as indicated by the fact that the 95% confidence intervals of the test and reference subjective assessments overlap. Ideally, all test items should be indistguishable from the reference. However, ITU-R broadcast quality requires only that at least 70% of the total number of test items be judged indistinguishable (in the case of both tests, 7 items), and that the remaining test items have the ratio of the upper confidence interval limits of test and reference greater than 0.85.

Another significant outcome is that in the 2-channel presentation both AAC main profile and AAC low complexity profile at the rates of both 128 kb/s and 96 kb/s score better than MPEG-2 Layer II at 192 kb/s for all program item scores combined. Similarly, in the 5-channel presentation both AAC main profile and AAC low complexity profile at the rate of 320 kb/s score better than MPEG-2 Layer II at 640 kb/s for all program item scores combined. This clearly indicates that AAC has a factor of 2 advantage in compression relative to the MPEG-2 Layer II coding system.

MPEG-4 AUDIO

MPEG-4 is the future „do-it-all" multimedia format. MPEG-4 audio goes along this paradigm by incorporating not only generic audio coding for a wide range of bit-rates, but coding formats for

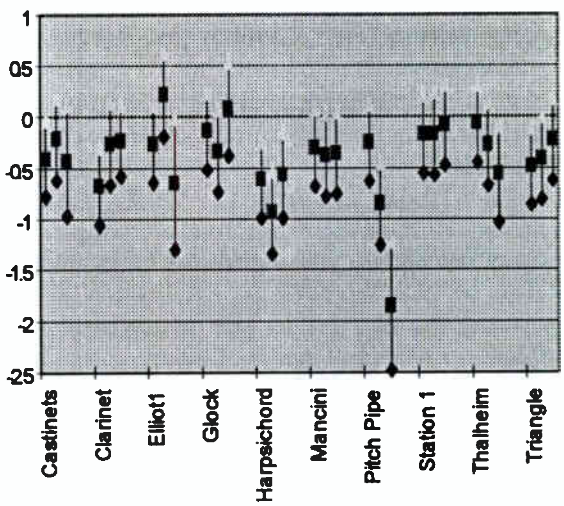

Figure 2: Five-Channel Test Results for each Codec for each Test Item, BBC results only. Shown are mean scores with 95% error bars. In each group of three results, the leftmost is AAC Main Profile at 320 kb/s, the middle is AAC Low Complexity Profile at 320 kb/s, and the rightmost is MPEG-2 Layer II BC at 640 kb/s.

very low bit-rate speech and audio (bit-rates 2 to 8

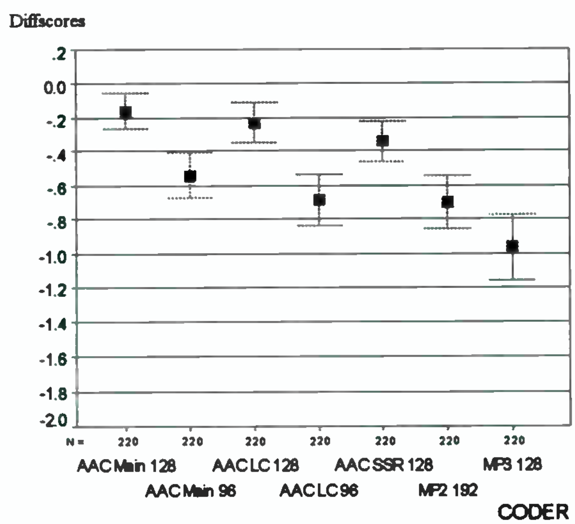

Figure 3: Two-Channel Test Results, scores averaged for all Test Items. Shown are mean scores with 95% error bars.

kb/s), more traditional speech coding (bit-rates 4 to 24 kb/s) and synthetic speech and audio. In addition, MPEG-4 audio supports the manipulation of audio content (speed and pitch control, mixing etc.).

The main features of MPEG-4 audio compared to earlier audio coding standards are:

- Content-based interactivity including the possibility to manipulate the presentation of audiovisual data (audiovisual objects).
- Extensibility and flexibility far beyond older standards.
- Compression efficiency which is comparable or better than any current audio coding standard.

To fulfill these requirements, MPEG-4 audio contains MPEG-2 Advanced Audio Coding as a subset. For high quality signals (bit-rates around 64 kb/s per channel) MPEG-2 AAC is used as is. For lower bit-rates (at 16 kb/s and above), the MPEG-2 Advanced Audio Coding tools are available (as is MPEG-2 AAC itself) and can be combined with other tools to construct MPEG-4 audio objects. One of the main technical features to reach the planned flexibility and other objectives is scalability. Scalability in the context of MPEG-4 is defined as the property that some part of a bit stream is sufficient for decoding and generating a meaningful audio signal with lower fidelity, bandwidth, or a selected content. The bit stream or audio object contains several layers of audio with a core layer which represents the core content of the signal and the higher layers adding quality and/or bandwidth. An example illustrates this concept:

For the „digital AM" broadcast system outlined below the scalable version could be defined as follows:

- The core layer is the MPEG-4 speech coder with additional possibility to synthesize jingles or similar signals at the decoder. This layer provides the basic speech service if only 6 kb/s can be recovered. This layer can be switched to 6 kb/s MPEG-4 flavor AAC or other core coders if music instead of speech is transmitted.
- The second layer is decoded if up 24 kb/s can be recovered. Higher bandwidth audio including music signals can be decoded. The difference layer is based on AAC tools.
- If better HF reception (or more bandwidth) is available, the full 48 kb/s can be recovered. The signal will sound better at a bandwidth up to 11 kHz. Again the difference layer is based on AAC tools.

This is an example of true graceful degradation that, of course, requires a modulation scheme that can do a more reliable decoding of a lower bit-rate if required.

AUDIO BROADCASTING USING MPEG-2 ADVANCED AUDIO CODING

For digital audio broadcasting systems, bandwidth is a very scarce resource. Each new proposed systems tries to use this resource as good as possible. Some systems, including Eureka 147 DAB, are already deployed and others, like the WorldSpace system (see [4]), are in the final preparation stage. If the broadcasting world follows earlier paradigms (keep full compatibility for all receivers), these systems will continue to use only the current generation of compression systems (MPEG-Audio Layer-2 or, with already better efficiency, MPEG-Audio Layer-3). Nonetheless there is the search for more channels for future systems and MPEG-2 Advanced Audio Coding is a prime candidate for such systems.

The following list contains some of the prime requirements for audio only broadcasting systems. Dependent on the target markets (primarily developed countries versus worldwide coverage) and the modulation schemes envisaged, the weighting between the requirements would vary.

- Good bandwidth efficiency = good compression: As mentioned above, this is number one priority for digital audio broadcasting systems. The target bit-rates vary between below 32 kb/s for mono or stereo signals up to 320 kb/s for 5.1-channel signals.
- Low complexity: For some of the planned systems, the possibility of cheap hardware decoder implementations within the next couple of years is vital. In any case, low decoding cost is an important requirement.
- Graceful degradation under channel errors: None of the current systems permits graceful degradation over a large range of channel errors. This will be a field of applications for future MPEG-4 audio systems (see above).
- Completed standardization phase: Since decoders in the field cannot be upgraded

(at least in the currently planned systems), the broadcasting standard definition has to completely freeze the audio format. The authors believe that this might change in the future with a more rapid succession of formats and more flexibility in the receivers.

Over time, complexity arguments become less important while for wireless systems bandwidth efficiency will stay to be of prime importance.

Possible Applications: There are three different types of digital audio broadcasting systems where AAC might be applied:

- Multichannel high quality services for terrestrial and satellite TV: Current and currently planned broadcasting services capable of stereo look for an upgrade path to multichannel sound. While MPEG-2 backwards compatible systems are one elegant way for an upgrade path, the aggregate bit-rate required for high quality (even then not broadcast quality for some signals) 5.1 multichannel signals is 640 kb/s. A simulcast AAC signal will add just 256 or 320 kb/s (for full broadcast quality) to the stereo signal delivering a better quality at a lower aggregate bit-rate. AAC has to be considered a prime candidate for future multichannel audio broadcasting.

- Digital AM: As described in ISO document [7], the NADIB group proposes a new „digital AM" format for broadcasting. The aim of the NADIB system is to improve the current broadcasting services within the AM bands (LW, MW and SW) by introducing digital programs in addition and/or substitution of the current analog programs. Improvements of the current services include:
  - Better perceptual audio quality
  - PAD (Program Associated Data)
  - External data services
  - User friendly receiver.

As much as possible, the NADIB system shall be compatible with existing modern transmitters and shall lead to low cost / lost consumption receivers. MPEG-2 Advanced Audio Coding (maybe in a scalable variation according to the upcoming MPEG-4 audio standard, see below) can deliver the necessary compression efficiency in the bit-rate range of 6 to 24 kb/s as required by NADIB. Since at these bit-rates the sampling frequency of the audio will be at 24 kHz and below, the computational complexity of AAC is small enough to allow battery operated receivers very soon.

- Next generation terrestrial and satellite broadcasting: The development of new digital audio broadcasting systems, both for terrestrial as for satellite based transmission, is far from complete. MPEG-2 Advanced Audio Coding will be a hot candidate the next time the specification of a new digital broadcasting system is undertaken. The advantage for this choice would be better compression efficiency and format consistency with upcoming (not the current) Internet audio applications. The disadvantage is higher complexity and the format inconsistency with current digital audio broadcasting systems.

Quality Comparison to Earlier Systems: MPEG-2 Advanced Audio Coding shifts the efficiency/complexity trade-off farther to somewhat more complex but more efficient coding systems. Compared to MPEG Layer 2, formal listening test results for stereo as well as for multichannel coding show an improvement factor of 2. This translates to double the number of channels compared to earlier systems. At very low bit-rates, MPEG-2 Layer 3 or MPEG-2 AAC are the possible alternatives with AAC being the newer but more complex system.

Of all audio coding systems tested so far, only AAC fulfills the broadcast requirements according to ITU-R (see [5]) at bit-rates of 128 kb/s for stereo or 320 kb/s for 5-channel audio with tests done according to the strictest recommendations on test methods.

For future multichannel systems there will be the choice between systems with a fixed downmix matrix at the encoder or decoder (maybe a couple of matrices to choose) or the simulcast of a good optimized stereo together with the multichannel signal. Bandwidth requirements up to now made the downmix solution unavoidable. In future audio broadcast systems, MPEG-2 Advanced Audio Coding will allow high quality simulcast at bit-rates below the ones necessary for downmix solutions.

AAC Configurations for Broadcasting: Depending on the application as defined above, the following configurations of Advanced Audio Coding are prime candidates:

- Multichannel high quality service: AAC 5.1 low complexity profile
- Digital AM: AAC 2 channel low or main complexity profile at up to 24 kHz sampling (probably in an MPEG-4 configuration as described above).
- Two channel stereo terrestrial and satellite broadcasting: AAC 2 channel low complexity or scalable sampling rate profile at sampling frequencies of 48 kHz and lower and bit-rates of 128 kb/s (probably 96 kb/s) and lower.

CONCLUSIONS

MPEG-2 Advanced Audio Coding is an universal audio coding system designed to augment and supersede MPEG-1 and MPEG-2 Layer 1, 2 and 3 for new applications. It will probably soon become the major format for computer based audio as well as for future digital audio broadcasting systems. MPEG-2 Advanced Audio Coding defines the state of the art in generic audio compression at all bit-rates where formal (or informal) testing has been done.

ACKNOWLEDGMENTS

The Advanced Audio Coding system described in this paper is the result of the collaborative work of numerous people at different companies. The most active contributors to the standard can be found at Fraunhofer Gesellschaft, AT&T, Dolby Laboratories, Sony and Hannover University. The authors want to express their gratitude to the past and present chairmen of MPEG-Audio, for their patient guidance in the MPEG work. Thanks to all the people who have contributed to this project, including Marina Bosi, Grant Davidson, Mark Davis, Charles Robinson, Louis Fielder and Steve Forshay at Dolby, James D. Johnston and Nikil Jayant at AT&T and Lucent Bell Laboratories. Thanks as well to the members of the Audio Group at FhG-IIS including Martin Dietz, Jürgen Herre, Uwe Gbur, Bodo Teichmann and many more.

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TECHNICAL DEVELOPMENT OF S-BAND SATELLITE MOBILE BROADCASTING AND COMMUNICATION SYSTEMS

Toshiaki Sato, Takashi Ito, Yoichi Kawakami, and Shigetoshi Yoshimoto
The Advanced Space Communication Research Laboratory
Tokyo, Japan

INTRODUCTION

The Advanced Space Communication Research Laboratory (ASC) was established on Mar. 31, 1993 to research and develop key technologies for future broadcasting and communication systems. ASC was established for 7 years and will complete its activities in 2000. The Japan Key Technology Center (KTC) funded 70% of ASC's the other 30% is funded by commercial companies such as NEC Corp., Mitsubishi Electric Corp., Toshiba Corp., and Pioneer Electronic Corp. ASC, employees come from various companies, such as the Japanese Broadcasting Corporation (NHK), National Space Development Agency of Japan (NASDA), Nippon Telegraph and Telephone Corporation (NTT), and Communication Research Laboratory (CRL) (Laboratory of Ministry of Post and Telecommunications of Japan).

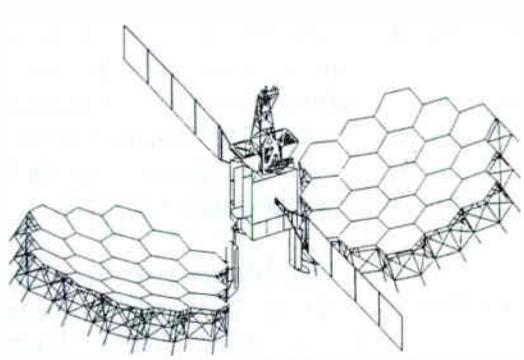

Fig. 1 Satellite for Mobile Broadcasting and Communication

More specifically, our company's objectives are to design an S-band Satellite Mobile Broadcasting and Communication System. We are currently developing the key technologies to realize such a system. [*1, *2, *3] This system will facilitate CD-quality mobile broadcasting audio services and other multimedia broadcastings. This will also enable anyone to access mobile communications anytime from anywhere and at very reasonable costs.

To realize the Satellite Mobile Broadcasting from a Geostaionary Earth Orbit (GEO) Satellite which is very far from the Earth, it is necessary to employ a large Effective Isotropic Radiated Power (EIRP) by using a high gain antenna and high output power transmitter. To realize the Satellite Mobile Communication from GEO satellite, it is necessary to have not only large gain-to-noise temperature ratio (G/T) but also high quality miniature mobile Earth terminals. There must also be switching equipment on the satellite to shorten the delay time in satellite communication and to be roust against terrestrial disasters.

In this paper, we first describe our concept of a satellite mobile broadcasting and communication system Second, we describe individual key technologies to realize it. We are now planning to perform many kinds of experiments using an engineering test satellite which adopts these key technologies. One of them is the Orthogonal Frequency Division Multiplexing (OFDM) Broadcasting Experiment. Third, we describe results of indoor experiments which have been done recently. Figure 1 shows a satellite for mobile broadcasting and communication

SATELLITE MOBILE BROADCASTING AND COMMUNICATION SYSTEM

Choice of Satellite Orbit and Frequency

Satellite Orbit GEO systems can facilitate high-quality audio broadcasts and enable access to mobile communications from anywhere in Japan. Such systems can be built more cheaply than terrestrial broadcasting systems and terrestrial mobile communications systems due to Japan's mountainous geographical features.

In addition, GEO satellite systems are easier to operate than Low Earth Orbit (LEO) or Medium Earth Orbit (MEO) satellite systems because of the constant satellite looking angles and stable communications' quality. For these reasons, we chose a GEO satellite system.

Frequency WARC-92 allotted 120MHz bandwidth for satellite broadcasting and 35MHz bandwidth for mobile satellite communications, both in the S-band (2.6/2.5GHz), to Asia and its periphery (called the third area in the ITU-R).

At the present time in Japan, a satellite mobile telephone system, the N-STAR system provides commercial service for portable and vehicle terminals.We chose a frequency adjacent to the frequency of that system.

Mobile Broadcasting and Communication System

We selected the Japanese islands and surrounding sea area as our mobile broadcast service area. The EIRP of the satellite is the most important factor in the satellite mobile broadcasting system specification. However, there are many more important factors in satellite mobile communication systems. For example, they have to receive a very weak radio signal and change transmitting powers in each beam depending on the distribution of users in space and time. For this reason, we decided to develop the satellite system specification from the necessary conditions of a satellite mobile communication system.

Estimating of Demand We estimated that there would be 30 million terrestrial mobile communication system subscribers at the beginning of the 21st century and 60 million at the saturation point. Of the 30 million subscribers, 6 to 7% or about 2 million subscribers would use both the satellite communication system and terrestrial systems. If these subscribers were equally divided among the following three systems, we would be able to expect 0.6 to 0.7 million subscribers in our satellite system.

- LEO or MEO systems
- Other Asia systems
- Our Japanese system

We also estimated that our Satellite Mobile Communication System would have the capacity of handling 9,000 communication channels under the following conditions.

- Average use: Once a day
- Average holding time: 3 minutes per call
- Maximum concentration probability: 20%
- Rate of mobile to mobile communication: 20%
- Unconnected probability: 1%

Voice Coding, Access, and Modulation The number of Japanese terrestrial digital mobile telephone (Personal Digital Cellular (PDC)) users was increasing rapidly. We expected that they would be potential users of our Japanese satellite system. We also considered it to be important that our satellite system be compatible with the PDC system. As a result, we thought our Japanese satellite communication system should adopt the following.

- Voice coding method: PDC half rate digital
- Multiple access method: Time Division Multiple Access (TDMA)
- Modulation method: $\pi/4$ shift Quadrature Phase Shift Keying (QPSK)

Frame Structure Considering future multimedia, we determined that the bit rate in one carrier is 32kbps which consists of five multiplexed voice bits ($5.6\text{kbps} \times 5 = 28\text{kbps}$) and additional bits. Adding error correction bits (convolutional code of 1/2 coding rate), the rate becomes 70kbps at the input and output points of the on-board processor. The carrier frequency interval is 50kHz.

Frequency System Bandwidth As we transmit five multiplexed voice signals on one carrier, we can handle five communication channels per 50kHz. Therefore, 90MHz spectrum width is necessary to handle the 9,000 communication channels estimated as the Demand.The N-STAR system, which provides satellite mobile communication service around the Japan area, occupies 30MHz band at 2.6/2.5 GHz. We can reuse this frequency.

band at three to four times. As a result we can cover all Japanese islands with 10 beams in tandem which have diameters of about 450km. Figure 2 shows the beam arrangement.

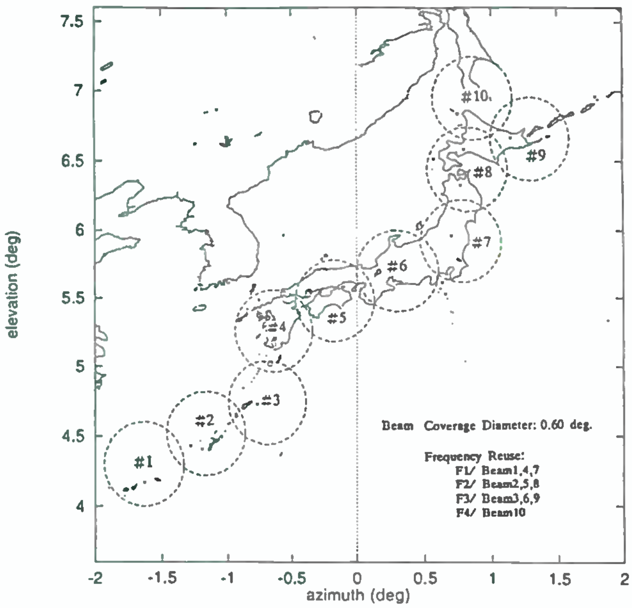

Fig. 2 Beam Arrangement

Table 1 Link Budget between Mobile Terminals via Satellite

	Uplink	Downlink
Frequency (MHz)	2657.5	2502.50
Tx peak power (W)	5.9	4.3
Feeder loss (dB)	1	2.00
Tx antenna gain (dBi)	2.5	42.50
EIRP (dBW)	9.26	46.85
Distance (km)	38000	38000
Free space attenuation (dB)	192.5	192.0
Polarization loss (dB)	1.1	1.1
Fading loss (dB)	4.9	4.9
Total loss (dB)	198.5	198.0
Rx antenna gain (dBi)	42.5	2.5
Rx feeder loss (dB)	2	1
Rx power (dBW)	-148.8	-149.6
Rx noise temperature (K)	500	450
Rx G/T (dB/K)	13.5	-25.0
Theoretical E_b/N_0 (dB) @5×10^{-5}	3.8	3.8
Hardware implementation margin (dB)	2.3	2.2
Interference allowance (dB)	1.3	1.0
Bit rate (bps)	35000	35000
Required C/N_0 (dB)	52.84	52.44

Note:
Transmitter average power (37%/5 multiplexing)
uplink (0.44W), downlink (0.32W)

Sattelite Transmitting Power For a 15m diameter parabolic antenna with 0.6 degree half beamwidth, we can expect about 42dBi for antenna gain after subtracting filter and other losses. Under this condition, we designed a link budget between mobile terminals via satellite.

For an average load rate 37% which is derived from voice activation, we can handle about 3,200 communication channels using 1,000W transmitting power. In the link budget, we expect a 6dB fading margin. In addition, if we reduce transmitting power on average of1.5dB, we can handle about 4,500 communication channels. From the above discussion, we will be able to handle the estimated demand of 9,000 communication channels using the two satellites with 1,000W transmitting power. In addition since we use two satellites, the system will not completely shut down by a problem in one satellite.

Realization of Satellite If we estimate the efficiency of a high power amplifier in this class of commercial satellites to be about 30%, the transmitter will consume about 3kW. Considering other communications' equipment such as the on-board processor and bus equipment, the satellite must generate a total of 5kW to 6kW. From the result of above considerations, we found that a 2.5- to 3- ton-class satellite will be needed to support a 10m or more diameter antenna and handle several thousand communication channels.

KEY TECHNOLOGY DEVELOPMENT

Figure 3 shows a satellite block diagram which is used for a mobile broadcasting and communication system. We found that the next items are key technologies for realizing this kind of satellite system.

- Reflector surface of large deployable antenna
- Feeding system of large deployable antenna
- High power amplifier
- On-board processor
- Miniature mobile Earth terminal

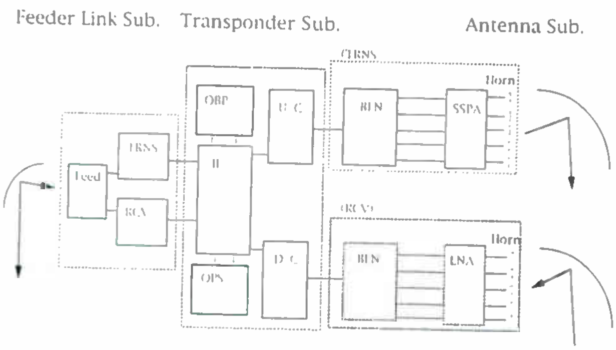

Fig. 3 Satellite Block Diagram

Reflector Surface of Large deployable antenna

To realize communications of small hand-held terminal with GEO satellites, at least a 10m diameter satellite antenna is necessary. To develop this large antenna, we focused on a deployable antenna which has a mesh reflector and truss structure because it is easy to accomodate and light weight and has sufficient strength. We obtained structural characteristics and electric characteristics such as the optimum tension force of one module mesh. We also established how to attach the antenna to the satellite by fixing one frame of the Hexa-Link and how to deploy it. In addition we developed a new test method which enabled us to simulate antenna deployment under no-gravity conditions. We made a partial model which consisted of seven 1.6m modules and performed deployment tests.*4 Figure 4 shows the deployment test of the 7m diameter partial model.

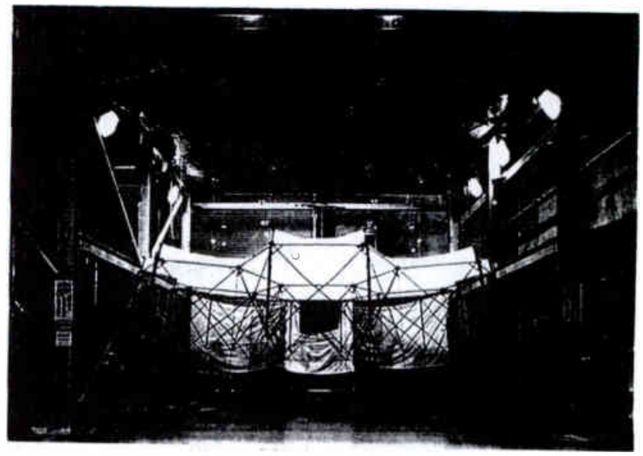

Fig. 4 Deployment Test of 7m Diameter Partial Model

Feed System of Large Deployable Antenna

We have been developing a phased-array-fed reflector antenna to achieve beam area changeability, beam tilt changeability, and flexibility, with respect to the traffic volume. With a 13m aperture offset parabolic antenna, and after optimizing parameters such as defocus length and element spacing, we were able to obtain an area gain of over 43dBi and an isolation between every three beams of over 26dB.*5 Figure 5 shows the beam transmission area.

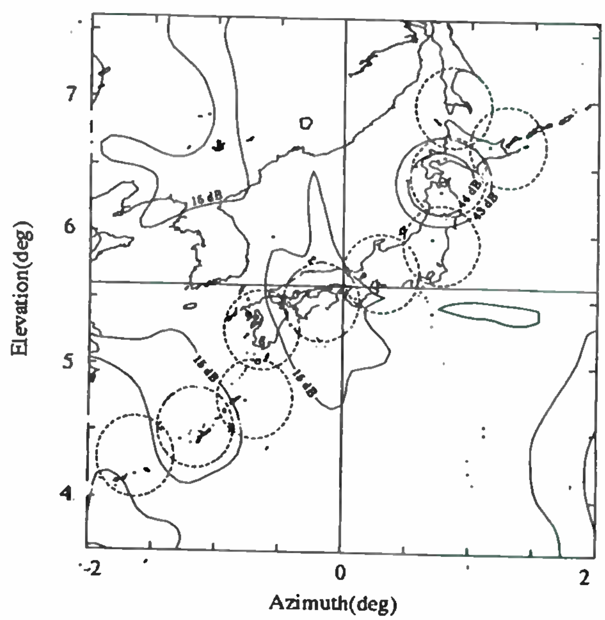

Fig. 5 Beam transmission area after optimization

We adopted a cupped microstrip antenna as the radiating element of the phased-array because it is small and light and has lowloss, lowcoupling with other elements. It also provides uniformity of characteristics.[*6] Figure 6 shows the radiating element array of the transmitting antenna which is divided into three groups by the amplitude weight.

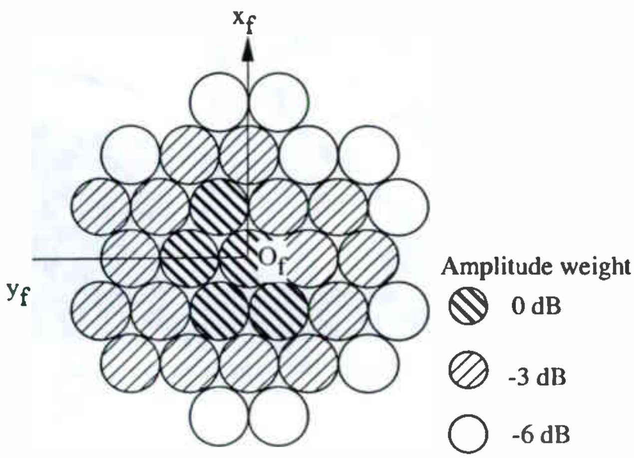

Fig. 6 Radiating Element Array of Transmitting Antenna

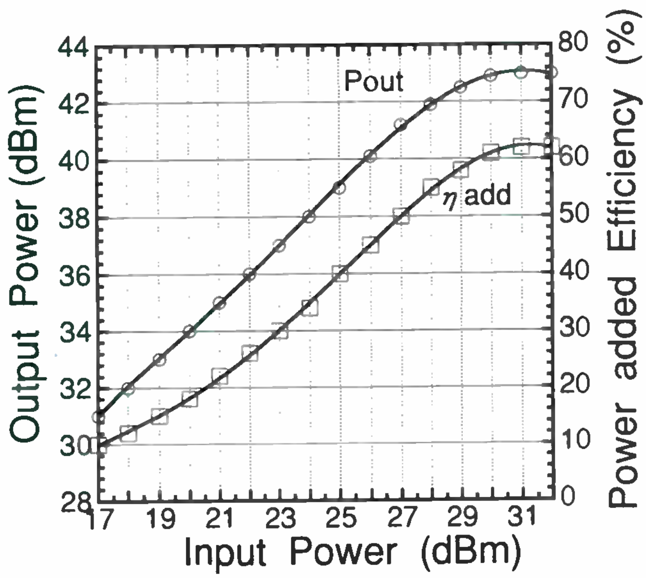

Fig. 7 Input-Output characteristics of FET module

High Power Amplifier

As the maximum power of one element is small in the phased-array-fed antenna, we adopted a solid state power amplifier (SSPA) as the high power amplifier (HPA). We developed a Field Effect Transistor (FET) Module for the last stage of the SSPA and it determined the SSPA efficiency. We could get a 45% efficiency at the point of Output Back Off (OBO) 3dB with a continuous wave (CW) signal.[*7] Figure 7 shows the input-output characteristics of the FET module. Figure 8 shows the SSPA.

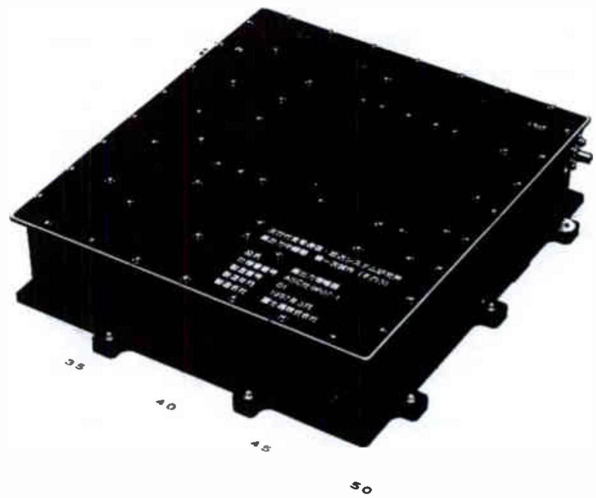

Fig. 8 SSPA

On-board Processor

We are also developing an on-board processor which can exchange signals in the mobile-to-mobile links. As the exchange function is usually located at the ground facilities, the signals must pass through a double hop connection link (two round trips to the satellite). This generates a large time delay in communication and makes conversations difficult.

In our system, however satellite has a processor with regeneration, filtering and switching functions. It also exchanges communication links one by one. This method will reduce the delay time by half. It can also be used for multi-media communications because of the 32kbps data exchange facility.[*8]

Miniature Mobile Earth Terminal

Satellite Pack Antenna We are researching and developing a small antenna for handheld Earth terminals with Arai Laboratory of Yokohama National University. The terminal must be easy to use and consider the influence of the human head. It must also consider dual mode operation with the terrestrial systems. From these points of view, we investigated the "Satellite Pack Antenna." The satellite pack consists of a battery, RF circuit for satellite phone, and stacked patch antenna. It replaces the cellular phone battery, and dual mode operation is easily obtained. The antenna and RF circuit are deployable as shown in Figure 9 and are fixed approximately in the horizontal plane for both left hand and right hand use. This design also gives about 10cm or more separation between the antenna and head tissue, which effectively suppresses irradiation power received by the head tissu from the antenna. *9

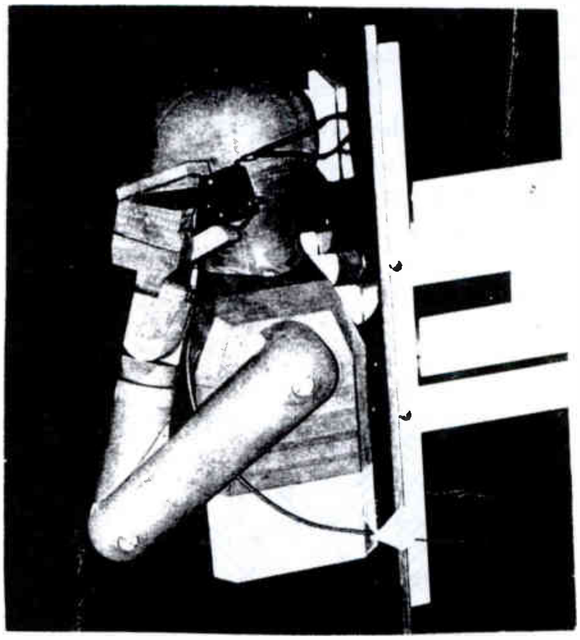

Fig. 10 Measuring antenna pattern

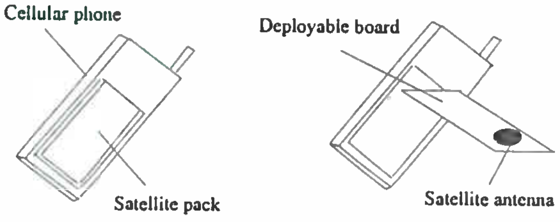

Fig. 9 Satellite Pack Antenna

We made a mockup of the miniature mobile Earth terminal equipped with a satellite pack antenna and we measured the antenna pattern when human body model held the miniature mobile Earth terminal .Figure 10 shows the antenna pattern measurement. Figure 11 shows the antenna pattern with a human body model.

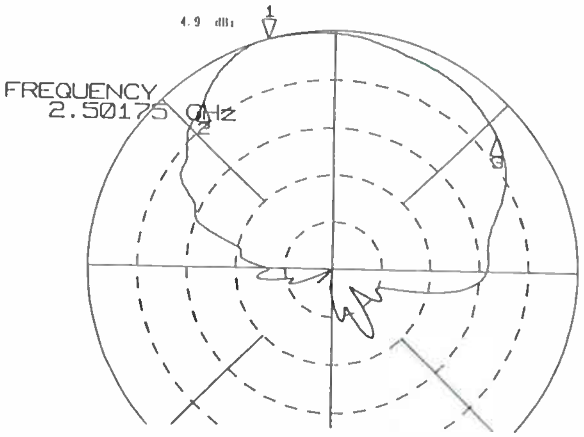

Fig. 11 Antenna Pattern with human body model

Another Element Technology We also manufactured for trial a helical circularly polarized antenna using double helices, a circular patch array antenna for vehicular mounting and slip phase control PLL/NRL frequency synthesizer. *10

SATELLITE MOBILE BROADCASTING EXPERIMENT

We are planning many kinds of broadcasting and communication experiments using a GEO satellite. OFDM is one of the most powerful methods in broadcasting to mobile receivers. For indoor experiments and propagation tests from a GEO satellite, we developed an OFDM signal generator and an OFDM signal performance receiver which are based on the DAB-Eureka147 (ETS300 401). Figure 12 shows the OFDM signal performance receiver.

Fig. 12 OFDM Signal Performance Receiver

As the OFDM signal consists of multi-carriers, non-linearities in the signal transmitting route degrade the bit-error-rate (BER) performance due to intermodulation. When we use an OFDM signal for the satellite broadcasting, we are thus forced to balance two conflicting requirements for on-board amplifiers: low degradation due to non-linearities and high output, high efficiency operation. Table 2 shows the link budget of an OFDM broadcasting system that we are planning to use as a test satellite system. The frequency is 2.5GHz. Figure 13 shows the beam transmitting area of the OFDM broadcasting system. It shows that it is possible for receivers with an antenna gain 6dBi to demodulate the coding rate of 0.51 DAB broadcasting over most of Japan.

Table 2 Link budget of OFDM Broadcasting

Uplink(30GHz band)						
Tx EIRP (dBW)	60.8					
Free space attenuation (dB)	-213.4					
Atmospheric attenuation(dB)	-0.3					
Rain attenuation (dB)	-4.8					
Polarization loss (dB)	-0.2					
Pointing loss (dB)	-1					
G/T (dB/K)	10					
Uplink C/N_0 (dB·Hz)	79.7					
Downlink(2.5GHz band)						
Stellite EIRP (dBW)	61.8					
Free space attenuation (dB)	-192.1					
Polarization loss (dB)	-0.5					
Antenna gain (dBi)	6			12		
Feeder loss (dB)	0.5					
G/T (dB/K)	-20.8			-14.8		
Downlink C/N_0 (dB·Hz)	77.0			83.0		
Overall C/N_0 (dB·Hz)	75.1			78.0		
Required C/N_0 (dB·Hz) (Code rate)	72.2 (0.34)	74.4 (0.51)	77.9 (0.75)	72.2 (0.34)	74.4 (0.51)	77.9 (0.75)
Margin (dB)	2.9	0.7	-2.8	5.8	3.6	0.1

Note: Hardware implementation margin 6dB

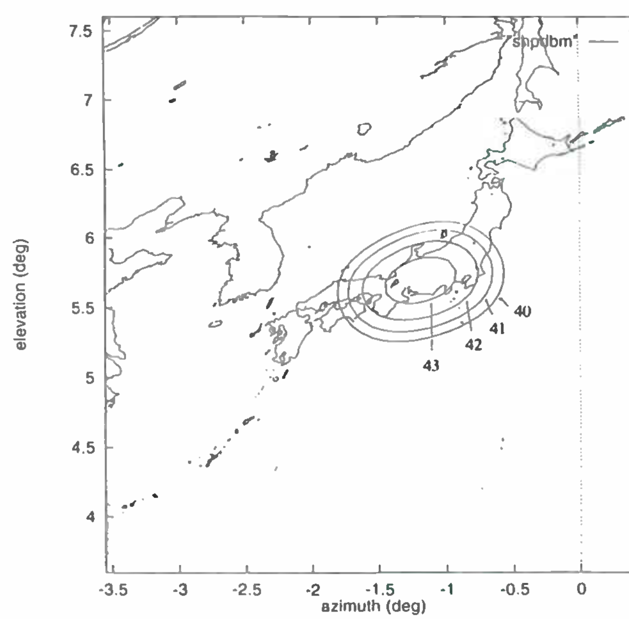

Fig. 13 Beam transmitting area of OFDM Broadcasting

Result of Indoor Experiment

We performed indoor experiments to confirm the influence of non-linearities in the SSPA described in this paper. Figure 14 shows one of the results. The C/No of the OFDM signal was equivalently degraded by 1.9dB at a BER of 10^{-2} for an Output Back Off (OBO) of 0dB.

For an OBO of 3dB and after error correcting, BER became less than 10^{-4}. Most listeners usually can not noticeable that rate. The result of this experiment shows that this SSPA has sufficient linearity for use as an HPA on the satellite for the OFDM broadcasting and will provide useful data for designing details of the OFDM satellite broadcasting system.

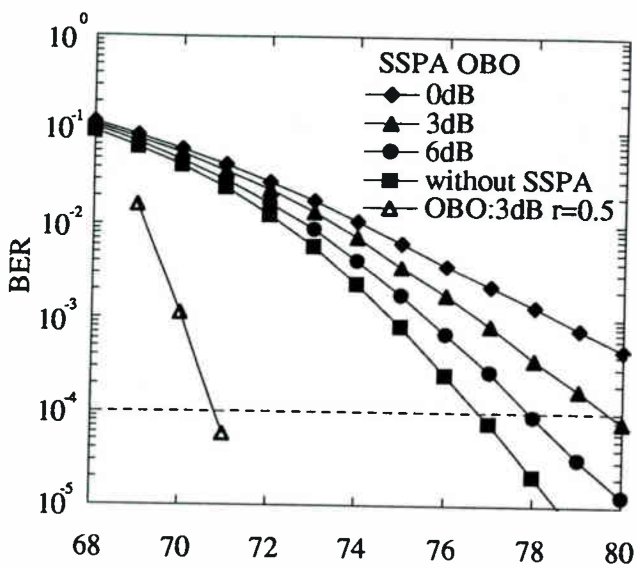

Fig. 14 Performance of OFDM signal with SSPA

CONCLUSION

We first presented the concept of satellite specifications necessary to construct a satellite mobile broadcasting and communication system.

Second, we introduced the key technologies for realizing such a system.We also showed their present development status. From these results, we found it was feasible to manufacture such a system. Third, we described the OFDM satellite mobile broadcasting experiment which we are planning to perform on an Engineering Test Satellite. We realized the possibility of satellite mobile broadcasting using the OFDM method by adopting a high gain antenna and good linearity, high output amplifiers. A practical propagation test in a real environment in which loads are changing in space and time, and with meteorological conditions will be significant.

The equipments manufactured using these key technologies will be installed on Japanese Engineering Test Satellite 8 (ETS-8) which will be launched in the summer of 2002.

Finally, we sincerely hope that these key technologies will be verified in space and will be applied to the future commercial satellites.

ACKNOWLEDGMENT

We would like to express our appreciation to ASC Executive Vice President Hideo Kishida, Hideo Mitsumoto, Reiji Matsumoto and Hiroyuki Arai Associate Professor at Yokohama National University and KENWOOD Corp. General Manager Toshiyuki Takegahara for their encouragement and support in this work.

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The Path to DTV: Issues of Global Importance

Monday, April 6, 1998
10:30 am - 12:00 pm

Chairperson:
Craig Tanner
Advanced Television Systems Committee,
Washington, DC

The Status of Development of Digital Terrestrial Television Broadcasting in Japan
Osamu Yamada
Association of Radio Industries and Businesses
Tokyo, Japan

International Standards for Digital Terrestrial Television Broadcasting - How the ITU Achieved a Single-Decoder World
David Wood
EBU
Geneva, Switzerland

VALIDATE Field Trials of Digital Terrestrial Television (DVB-T)
Chris Weck
Institut Fuer Rundfunktechnick GmbH
Munich, Germany

THE STATUS OF DEVELOPMENT OF THE DIGITAL TERRESTRIAL TELEVISION BROADCASTING IN JAPAN

Osamu Yamada*, Makoto Sasaki*, Masafumi Saito**
Association of Radio Industries and Businesses
Tokyo, Japan
(*NHK Science & Technical Research Laboratories, ** DTV-Lab.)

ABSTRACT

In Japan, the experimental transmission system called BST-OFDM(Band Segmented Transmission-Orthogonal Frequency Division Multiplexing),was fixed based on various kinds of transmission tests on OFDM in Association of Radio Industries and Businesses(ARIB) last year. The experimental equipment for BST-OFDM is now under manufacturing. The equipment provides a large amount of parameter combinations based on the BST-OFDM scheme, and these parameters are to be optimized by the results of the actual transmission tests. This paper describes the outline of the background and discussions concerning the experimental system-specifications. Then some results of field trials and simulations are shown.

INTRODUCTION

Digital broadcasting in Japan has undergone rapid development in two last years. The plan at this time is to begin digital broadcasting by broadcast satellites in 2000 and to decide the technical standards for terrestrial digital broadcasting before 2000.

Current terrestrial analog broadcasting in Japan provides abundant program services through an extensive broadcasting network consisting of more than 14,000 stations reflecting the mountainous terrain of the country. These stations consists of about 7,000 NHK stations and about 7,000 commercial broadcasting stations. In this network, the only unused channel throughout the country is channel 13. In particular, the Kanto area of Japan, which includes the Tokyo metropolitan area, has a large concentration of broadcasting stations resulting in a very tight frequency situation. There are therefore many issues to be addressed in digitizing the infrastructure of terrestrial broadcasting, the present core means of broadcasting.

The Telecommunications Technology Council (TTC) has specified requirements for a future transmission system taking into account the advantages of terrestrial digital broadcasting. The main requirements here are that the system be able to broadcast high-definition programs even in regional areas, provide mobile services, and achieve high frequency-spectrum efficiency in light of the scarce frequency situation described above. Based on these requirements, NHK and DTV Labs. prepared a joint proposal for terrestrial Integrated Services Digital Broadcasting (ISDB-T) transmission system and presented it to ARIB. After a series of discussions, this system was designated an ARIB experimental system in July 1997 as a forerunner to a terrestrial digital broadcasting transmission system, and the experimental equipment is currently manufactured based on the experimental system specifications.

ARIB

The R&D Group for the Digital Terrestrial Broadcasting System in ARIB, which consists of over 60 companies and institutions including broadcast operators, manufacturers, and research laboratories, has been holding technical discussions on the transmission system. Progress in such issues as coding of source information, multiplexing, conditional access, etc., has been made by establishing a liaison with the R&D Group for the Satellite Digital Broadcasting System.

Following discussions on the experimental system

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specifications, a Working Party (WP) for Experiments was established. This party is making preparations to hold basic-function verification tests and actual-scale system tests using test equipment. In the system tests, a transmission system will be set up on Tokyo Tower to transmit test signal. The objectives of these transmissions are to collect reception data to determine the service area for both fixed reception and mobile reception, and to know the technical conditions on the realization of SFN(Single Frequency Network).

REQUIREMENTS FOR TERRESTRIAL DIGITAL BROADCASTING

Present terrestrial broadcasting accommodates television, and FM and AM radio services. While analogue modulation schemes are used to transmit the main services, digital technology is employed for additional multiplex services such as teletext. Satellite broadcasting in Japan offers HDTV and conventional television services with digital sound broadcasting services.

Under these circumstances, it is important to clarify the merits provided by digital broadcasting. In addition to offering high-quality video and sound, and eliminating interference, digital broadcasting needs to provide attractive multimedia services. Because new services are greatly expected in the future, the transmission method has to be sufficiently flexible and extendible to accommodate these services.

ISDB-T should:
- have the capability to provide a variety of services composed of video, sound, and data,
- have sufficient ruggedness against multipath and fading interference to make portable and mobile reception possible,
- have receivers dedicated to television, sound, and data respectively, as well as fully integrated receivers,
- have the flexibility to accommodate different service configurations and ensure flexible use of transmission capacity,
- have service extendibility to ensure the future needs,
- be able to achieve effective use of frequencies using SFN technology,
- be able to use vacant frequencies effectively in congested broadcasting frequency situations.

TERRESTRIAL ISDB SYSTEM

Segmented OFDM Scheme

The OFDM method densely multiplexes many carriers, while maintaining orthogonality among them in a specific transmission band. This method can have a much longer symbol interval than that of a digital modulation method using a single carrier. It can suppress multipath interference by inserting a guard interval in the time domain. FFT (Fast Fourier Transform) schemes are used for modulation and demodulation of OFDM signals. Unlike the conventional modem procedure, OFDM can be modulated by a numerical operation. OFDM is already used in the Digital Audio Broadcasting (DAB) and Terrestrial Digital Video Broadcasting (DVB-T) systems developed in Europe and the fundamental effectiveness of the scheme has been demonstrated.

NHK has been studying the OFDM technology since 1986 and developed some OFDM modems. A lot of transmission tests have been carried out. Based on these experiments, NHK and DTV-Lab. have developed and proposed the Band Segmented Transmission OFDM scheme (BST-OFDM) to ensure flexibility and expandability for ISDB-T. A BST-OFDM channel consists of a set of frequency blocks called BST-Segments having common structure of carrier usage. All of the BST-Segments has a bandwidth of 432 kHz. Some of the carriers are assigned as control carriers called Transmission and Multiplexing Configuration Control (TMCC) carriers, which transmit information on the carrier modulation scheme and the coding rate for each BST-Segment.

Overview of ISDB-T

Figure 1 shows the image of ISDB-T transmission applications. Two transmission bandwidths, 5.6 MHz and 432 kHz, are defined for ISDB-T(1). The 5.6 MHz bandwidth is mainly used for digital broadcasts of television programs, while the 432 kHz bandwidth is mainly used for audio programs. These two modes share all other parameters such as the OFDM carrier interval and frame configuration. Table 1 lists the parameters of this transmission scheme.

ISDB-T provides hierarchical transmission capability by several combinations of carrier modulation schemes (DQPSK, QPSK, 16QAM, 64QAM) and coding ratios of the inner code (1/2, 2/3, 3/4, 6/5, 7/8) within bandwidth of 5.6 MHz. This enables mixed transmission for different receiving conditions, which means that audio and data broadcasts for mobile and portable receivers can be performed simultaneously with television broadcasts for home use. The bandwidth allocated to each hierarchical level can be set in 432 kHz units of BST-Segments. Information for property of the BST-segments can be sent to receivers by a TMCC signal allocated to particular carriers in the BST-Segments.

For an experimental Terrestrial ISDB system having bandwidths of 5.6 MHz and 432 kHz, three types of receiver are supposed as shown in Fig. 1. The three types of receivers are as follows:

(1) An integrated receiver with a 5.6 MHz OFDM demodulator and an HDTV display, which can receive all of services,

(2) A light-weight mobile receiver with a 5.6 MHz OFDM demodulator and a small SDTV display,

(3) A portable or pocketable receiver with a 432 kHz OFDM demodulator for sound and data services.

In the case of the 5.6 MHz channel, the signal is frequency-interleaved within the bandwidth of 5.6 MHz, however, the sound services can also be transmitted using one BST-Segment at the center of 5.6 MHz. In this case, interleaving range is divided to two parts for a center segment and the rest of segments. These sound services could be decoded by a 432 kHz receiver.

STUDY OF COVERAGE AREA FOR STATIONARY RECEPTION

In this section, the simulation of coverage area of SFN are described(2). Table 2 shows the transmitting and receiving condition used in this simulations. Figure 2 shows the BER performance under a single echo with a constant DU ratio as a function of delay time for the guard intervals of 31.25 μs and 62.5 μs. In these figures, the CN ratio is set at 30 dB for the noise bandwidth of 5.6 MHz. Degradation of BER performance is small and almost constant while the delay time is smaller than the guard interval of 31.25 μs or 62.5 μs. If the echo delay time exceeds the guard interval, the BER increases due to inter-symbol interference.

Figure 3 shows the simulation results of the required DU ratio as a function of echo delay time. The required DU ratio is almost constant within the guard interval, and increases in proportion to the increase of delay time outside the guard interval. The ISDB-T system employs a convolutional code as an inner code and a Reed-Solomon code as an outer code. The receivers can correctly decode data received with a BER of 10^{-2}. Figure 3 suggests receivers can correctly receive digital broadcasting signals if the DU ratio with a certain delay time is over the curve corresponding to a BER of 10^{-2}. The result in Fig. 3 can be used to design a SFN.

Simulations of coverage using SFN are carried out for the Kanto area including the Tokyo metropolitan area. In this simulation, every digital transmitter is located at the same site as the existing analogue transmitters, and every receiver is set up for fixed reception using directional antennas. The number of transmitters is 186 shown in Fig. 4.

Transmitters radiate the same signal in the same frequency at the same time in the UHF band. Transmission power is -14 dB compared to existing analogue transmission power. Required field strength is 56 dBμV/m. Directional antenna in accordance with ITU-R Rec. BT. 419 is chosen as the reception antenna. Required DU ratios follow the curve maintaining the BER of 10^{-2} in Fig. 3.

In this simulation, field strength from every transmitter is calculated, taking into account diffraction attenuation, using terrain height information from a topographical database. Coverage ratio is defined as follows:

$$\text{Coverage ratio} = \frac{\text{Area where the BER without error correction} < 10\text{-}2}{\text{Area where the field strength} > 56\ \text{dB}\mu\text{V/m}}$$

The coverage ratio is 97.3 % for the guard interval of 31.25 μs and 98.6 % for the guard interval of 62.5 μs. The damaged area for the guard interval of 31.25 μs is shown in Fig. 5. There are following two themes on SFN to be solved.

(1) Transmission means to distribute the same signal to each SFN station.

(2) The method to recover the damaged areas and to increase the coverage areas.

TRANSMISSION EXPERIMENTS FOR MOBILE RECEPTION

Laboratory Tests

Figure 6 shows the bit error rate performance of the DQPSK-OFDM without error correction when Gaussian noise and a single echo signal are added to the desired signal. The delay time of the echo signal was set at 5 μs, and the DU ratios were 0 dB and 10 dB. The experimental results are in accord with the results of the simulation. The difference between simulated and measured line is about 1 dB for 10-4.

Field Trials

Field experiments of SFN have been conducted using two experimental transmitters connected with an optical fiber in channel 19 of 509 MHz for the center frequency. The transmitter A is located at NHK laboratories in Setagaya, Tokyo, and the transmitter B is at the NHK broadcasting center in Shibuya, Tokyo. The distance between two transmitters is about 8 km. The diagram of the experimental system is shown in Fig. 7. Figure 8 shows the distribution of bit-error rates measured along the route between two transmitters. Bit-error rates shown in this figure indicate the average for one second before error correction for mobile reception. Bit-error rates under 10^{-2} can be corrected sufficiently using an error correcting code such as a convolutional code and a Reed-Solomon code. The results show that the SFN extends the coverage area.

Delay profiles are also measured every one-second and are shown in Fig. 9. The delay difference and DU ratio between direct waves from the two transmitters depend on the location of the receiving point.

To obtain more detailed data, a co-operative body composed of the Communications Research Laboratory, the Advanced Digital Television Broadcasting Laboratory, and NHK has been formed. The mission of this body is to study OFDM transmission characteristics through SFN experiments using three SFN stations.

CONCLUSION

From now on, a 3-station SFN basic experiment using existing test facilities and consecutive detailed system experiments using the Tokyo Tower experimental station will be carried out this year. On the basis of experimental results, transmission parameters as well as detailed parameters applicable to 7 and 8 MHz bandwidths will be optimized.

Making use of empty channels in current analog broadcasting is an effective means of introducing new digital broadcasting under extremely congested frequency conditions. In this case, however, appropriate protection ratios against interference must be maintained between analog and digital signals.

To realize digital terrestrial broadcasting that can provide HDTV, multimedia broadcasting for mobile terminals, and other services in conjunction with technical advances and the needs of the times, it is vitally important that development work places priority on achieving benefit for the viewing audience in the introduction and migration period, in addition to providing attractive services.

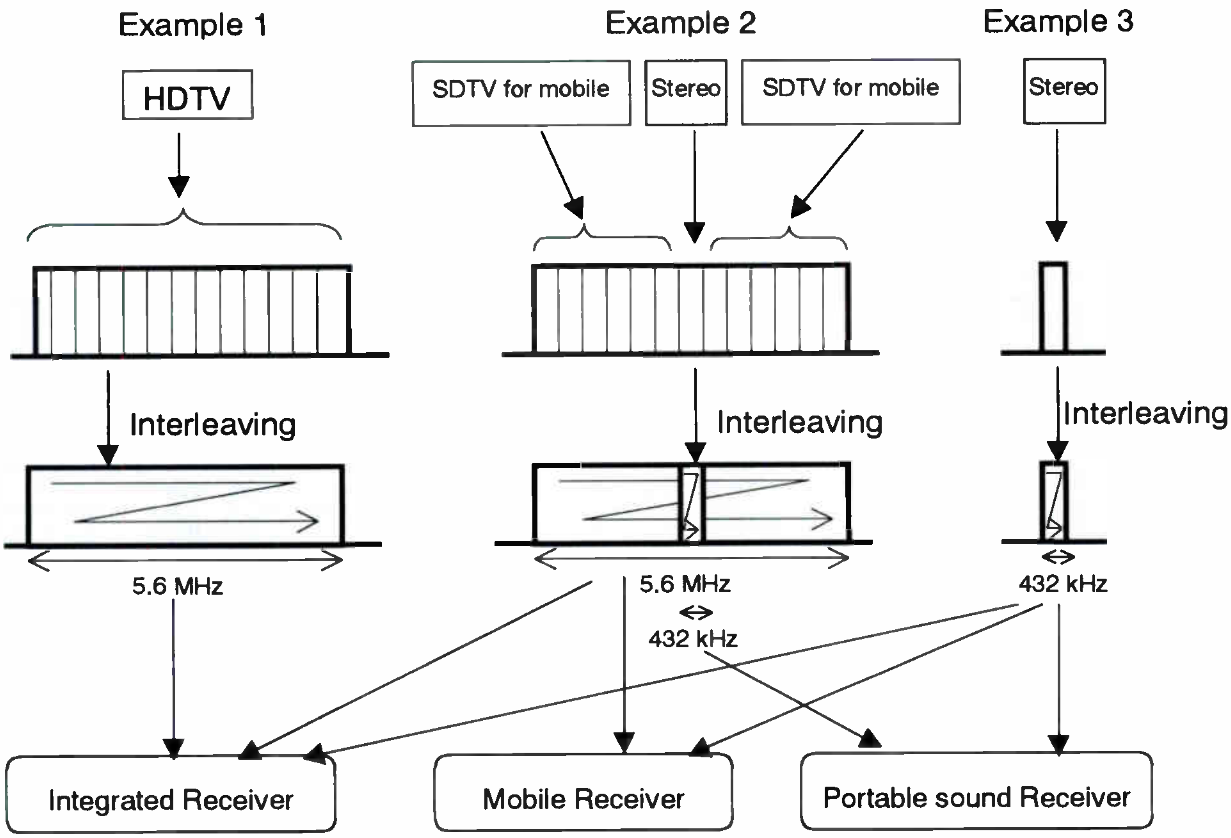

Fig. 1 Examples of application and transmission images of the Terrestrial ISDB system.

Table 1 Parameters of the experimental Terrestrial ISDB system

Mode		432 kHz ISDB-T (Mode 1)	5.6 MHz ISDB-T (Mode 1)	432 kHz ISDB-T (Mode 2)	5.6 MHz ISDB-T (Mode 2)
Number of BST-Segments (Ns)		1	13	1	13
Bandwidth		432(kHz)×Ns + 4(kHz)		432(kHz)×Ns + 1(kHz)	
Number of BST-Segments for DQPSK		nd			
Number of BST-Segments for QPSK, 16QAM and 64QAM		ns (ns+nd=Ns)			
Carrier spacing		4 kHz		1 kHz	
Number of carriers	Total	108×Ns + 1		432×Ns + 1	
	Data	96×Ns		384×Ns	
	SP	9×ns		36×ns	
	CP	2×ns + 7×nd + 1		8×ns + 28×nd + 1	
	TMCC	ns + 5×nd		4×ns + 20×nd	
Carrier modulation methods		QPSK, 16QAM, 64QAM, DQPSK			
Number of Symbols per frame		204			
Effective symbol duration		250 μs		1 ms	
Guard interval		31.25 μs (1/8), 62.5 μs (1/4), 15.625 μs (1/16), 7.8125 μs (1/32)		125 μs (1/8), 250 μs (1/4), 62.5 μs (1/16), 31.25 μs (1/32)	
Frame duration		57.375 ms (1/8), 63.75 ms (1/4), 54.1875 ms (1/16), 52.59375 ms (1/32)		229.5 ms (1/8), 255 ms (1/4), 216.75 ms (1/16), 210.375 ms (1/32)	
Inner code		Convolutional code (1/2, 2/3, 3/4, 5/6, 7/8)			
Outer code		RS (204,188)			
Information rates		283.1 kbps - 1486.3 kbps	3.680 Mbps - 23.420 Mbps	283.1 kbps - 1486.3 kbps	3.680 Mbps - 23.420 Mbps

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TABLE 2 Transmitting and Receiving Conditions.

Bandwidth	5.6 MHz
Carrier Spacing	4 kHz, 2 kHz
Carrier Modulation	64QAM
Guard Interval	1/8 (31.25 μs, 62.5 μs)
Site	Same as analogue transmitters
Number of transmitters	186
Radio frequency band	UHF band
Transmission power	- 14 dB (Compared with existing analogue transmission power)
Required field strength	56 dBμV/m
Reception antenna	Directional (ITU-R Rec. 419)

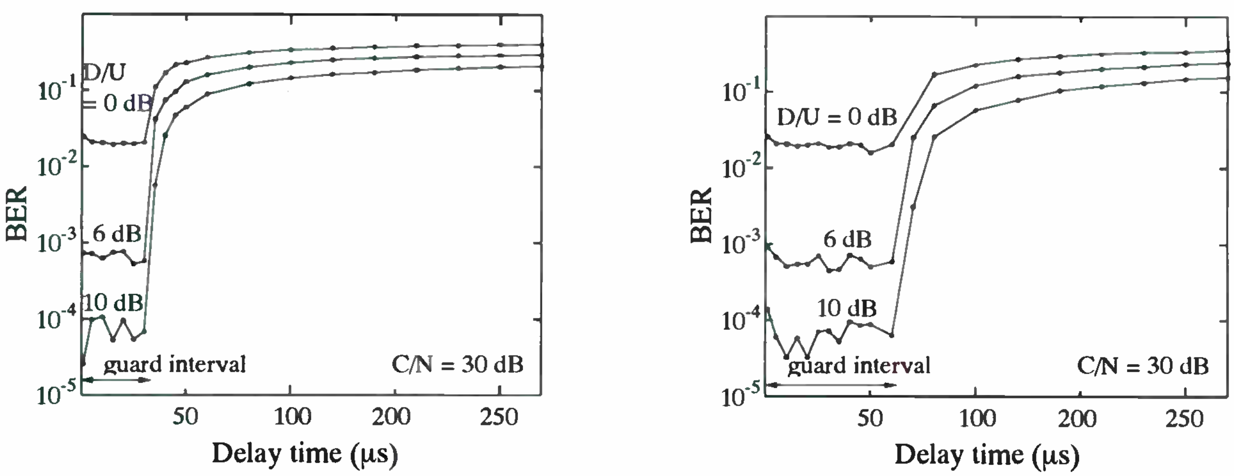

Fig. 2 BER Performance under a Single Echo as a Function of Delay Time.

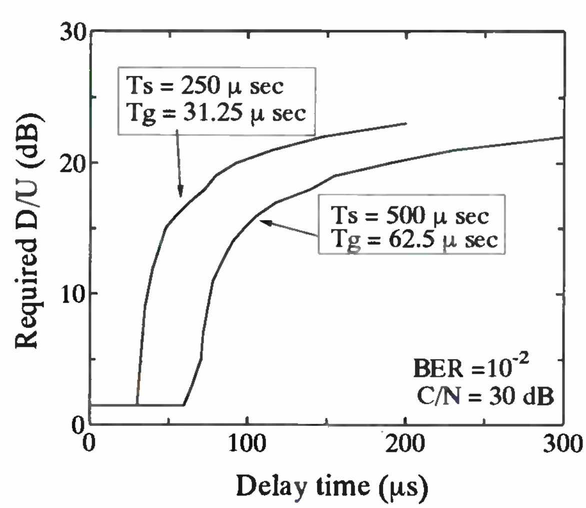

Fig. 3 Required DU Ratio for Given BER.

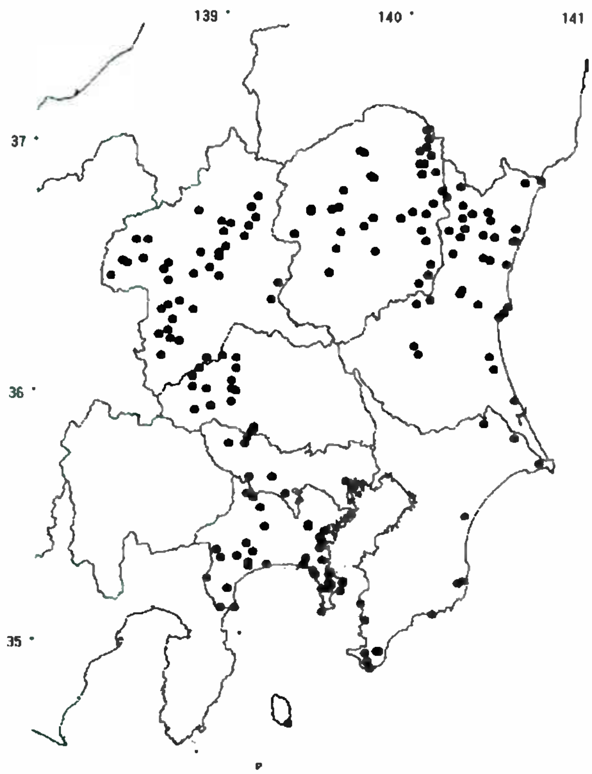

Fig. 4 **Location of Transmitters in the Kanto Area.**

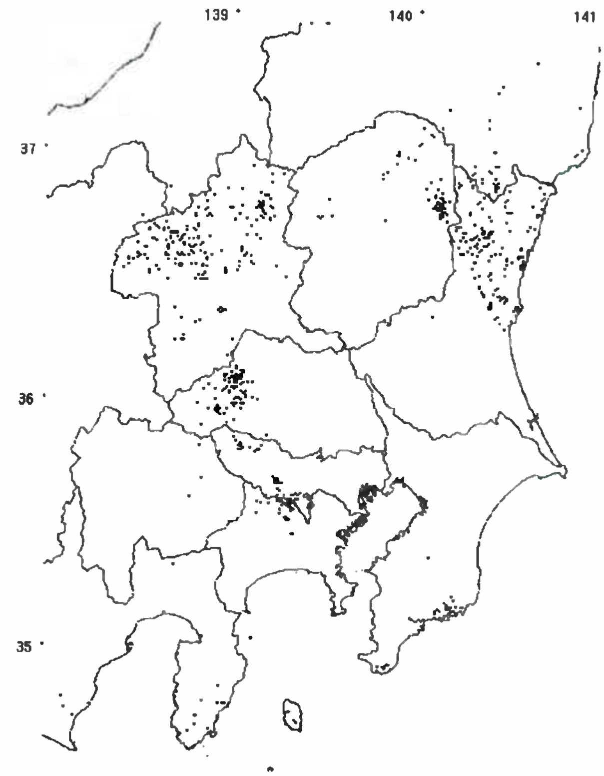

Fig. 5 Damaged Area of the SFN in the Kanto Area.

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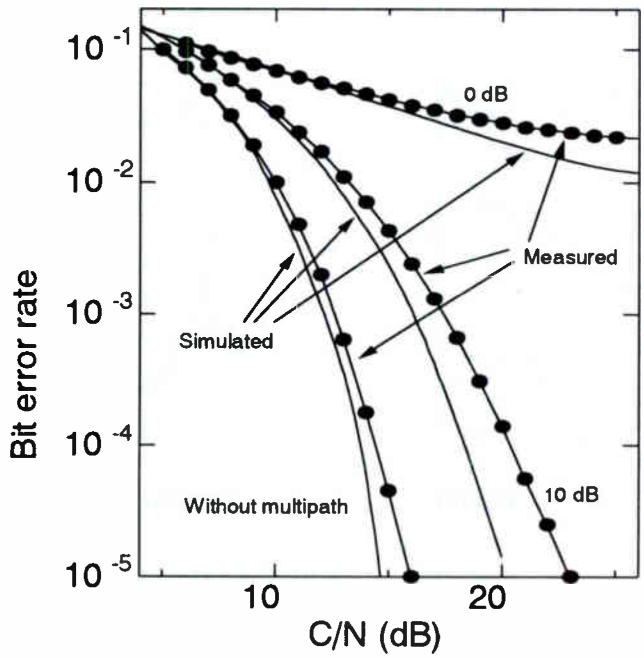


Fig. 6 **BER performance under multipath distortion.**

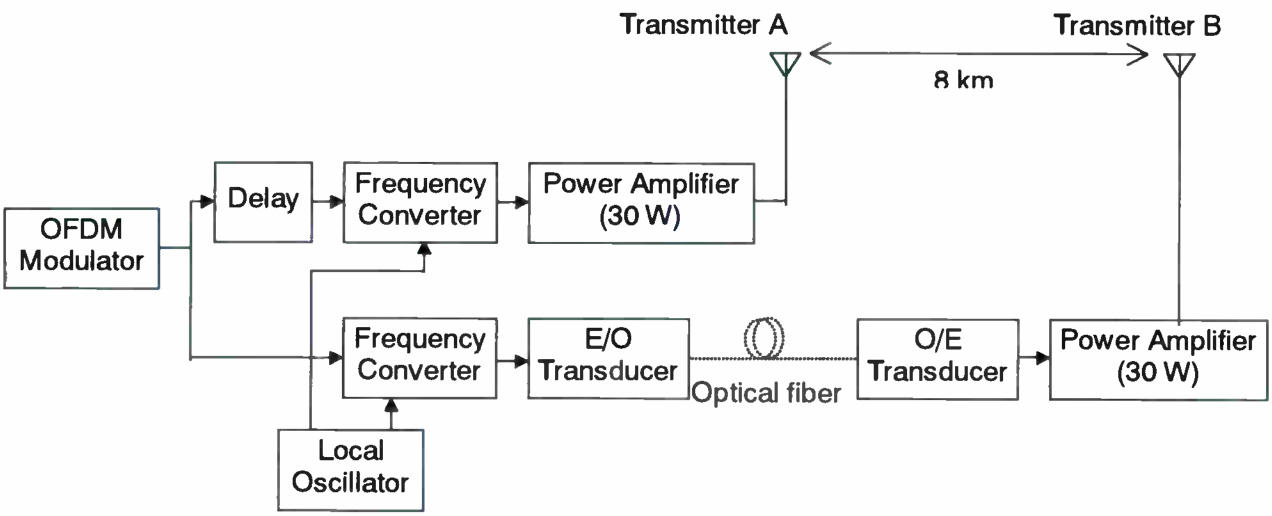
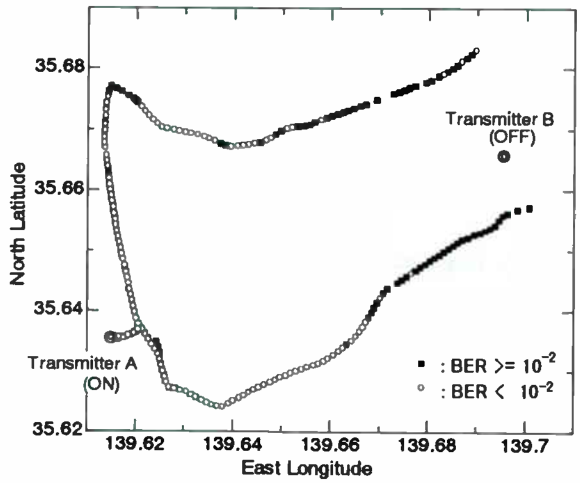
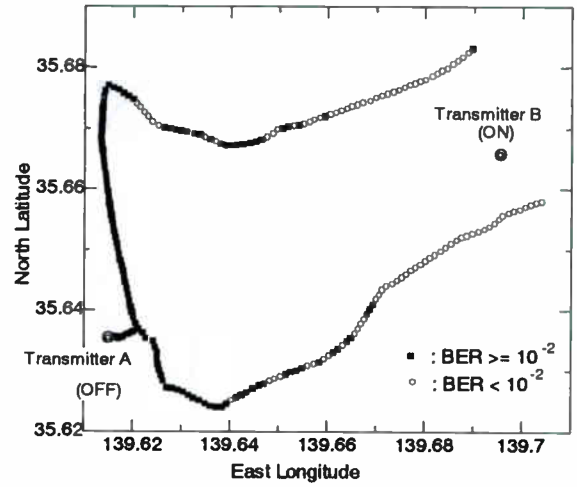


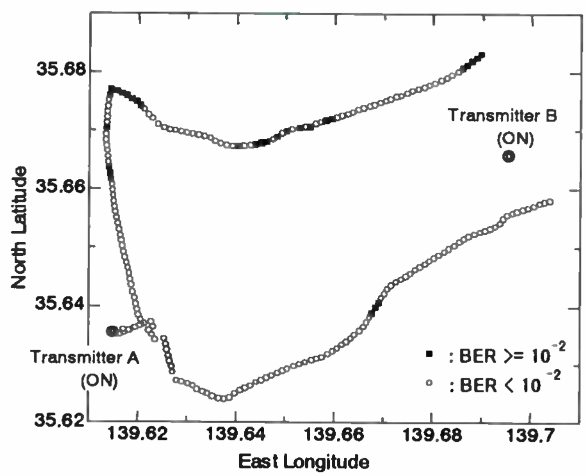
Fig. 7 **Diagram of Experimental Transmitters.**



(a) Transmitter A: ON, Transmitter B: OFF



(b) Transmitter A: OFF, Transmitter B: ON



(c) SFN operation

Fig. 8 Results of Measurements.

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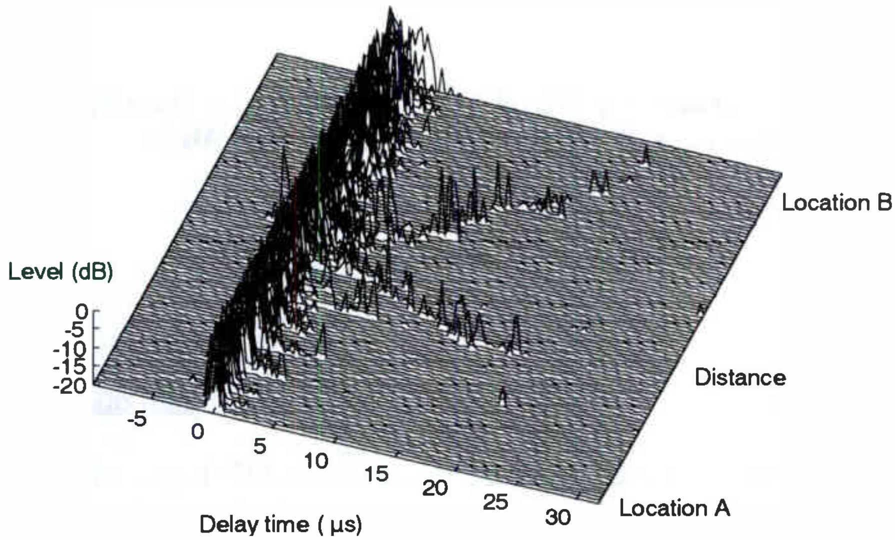

Fig. 9 Delay Profiles.

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International Standards for Digital Terrestrial Television Broadcasting - How the ITU Achieved a Single-Decoder World

Stanley Baron
NBC
New York, NY

Abstract

The International Telecommunications Union (ITU) has adopted a series of Recommendations (standards) defining a digital terrestrial television broadcasting (DTTB) system. The set of Recommendations and associated descriptive Reports represents a four-year effort by ITU Task Group 11/3.

The work is based in many ways on both the work of and the philosophy behind the MPEG-2 standard. The MPEG-2 standard provides a set of tools that may be used to describe a system. The set of Recommendations developed by Task Group 11/3 define a constrained set of tools that can be used to provide a DTTB service. The constrained the set of tools provides for a single, low cost-decoder that can deliver both ATSC and DVB coded images and sound.

TG11/3 established a harmonized subset of MPEG-2 that allows for a single decoder that can translate the service multiplex and transport layer and decode the audio and video compression and coding layers for any system that conforms to the DTTB set of Recommendations.

This set of harmonized Recommendations fully meets the request of the World Broadcasting Union for unique global broadcasting systems leading to single universal consumer appliances.

The paper describes the system.

Introduction

Task Group 11/3 began with the premise that they should establish an infrastructure that enables a communication environment in which a continuum of television and other data services could be brought to the consumer via wire, recorded media, and through the air. Terrestrial broadcasting presents the most challenging set of constraints of all of the forms of media. Therefore, a system that works well in the terrestrial broadcasting environment should suffice for other media. The first step in harmonizing the delivery of services to the consumer using a variety of media is the development of a suitable service model.

The Service Model

In addressing the harmonization of digital methods for delivery of services to the home,[1] the International Telecommunications Union (ITU) recognized that the technology involved in digital methods for the delivery of services was undergoing and would continue to undergo rapid development. The ITU also recognized that the methods of delivery over terrestrial broadcast channels, cable, satellite channels, via telephony services, and recorded media have different characteristics and require differing transmission (modulation) solutions. The ITU also addressed the need for services to be available at differing levels of quality appropriate to specific applications.

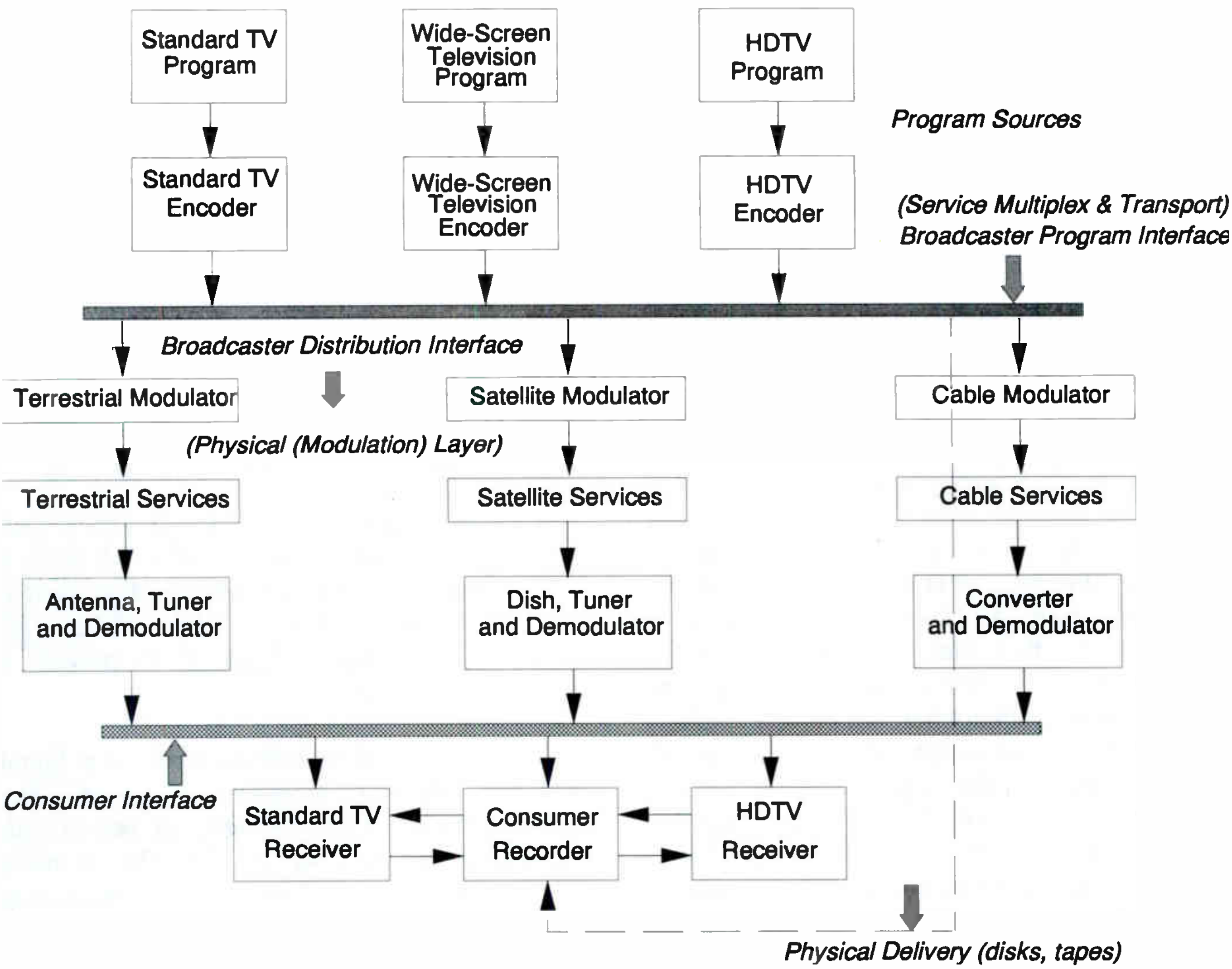

Figure 1
Digital Television System Model

In searching for a coordinated approach to the development of such delivery systems, Task Group 11/3 recommended[2] that:

• the terrestrial system should have maximum commonality with other delivery systems;

• the system should be designed as a "container", able to transport MPEG-2 and/or other data services in a transparent and flexible way;

• the base system should be a single layer system;

• a service information and header descriptor system should be implemented;

•source coding and transport mechanisms should be based on common processing algorithms and have a maximum of shared parameters;

• the mechanism employed allow use of receivers having differing levels of capability, and

• headers and descriptors be used to enable the receiver to identify and process a range of services having differing characteristics.

Figure 1 provides a service model for delivery of programming that accommodates the concepts of convergence and harmonization; providing the consumer with access to services over different forms of media. This model separates the digital communications system into a series of interconnected modules. In this model, services (Program Sources) at various levels of performance are digitally encoded, multiplexed into a digital data stream (Service Multiplex and Transport) and distributed through various transmission media (Distribution Interface) to the consumer.

The Service Platform[3]

Digital television services focus primarily on the quality of the imaging and sound characteristics of the new medium. However, digital television is a medium that transports moving images, sound, and data from one point to another, and consideration of appropriate audio and ancillary data services to accompany the service is a very important part of the implementation process. The implementation of digital television services meant development of new standards. Standards for new services emphasizing the transportation of information in a digital format offer opportunities to establish a new platform for services.

The primary premise for any platform is that the services provided in each instance should be appropriate to the needs of the individual consumers and should respect the realities of the marketplace. Therefore, the focus should be centered on providing the necessary flexibility to meet those individual marketplace requirements.

In order to meet those needs, these service platforms must have certain characteristics:

1. they must employ a basic set of standard features and capabilities that are well defined;

2. they must foster innovations that make the services easy to use;

3. they must allow the transport of video, audio and data across media boundaries;

4. they must be flexible (to allow for local options) and extensible (to provide for future improvements);

5. they must be ubiquitous (present everywhere at the same time); and most importantly,

6. they must be affordable.

The digital terrestrial television broadcasting (DTTB) model

Task Group 11/3 was charged by the ITU-R with developing a set of Recommendations defining a digital terrestrial television broadcasting system that met the above criteria.[4] Task Group 11/3 generated an Outline of Work[5] to provide a structure for its work.

The outline of work included a model of a digital terrestrial television broadcasting system. The model was divided into four areas of interest with subgroups assigned to develop the required Recommendations and Reports. The Task Group used the model as the basis of its investigations.

The four subsystems of the system model are as follows (reference Figure 2):

1) *Source coding and compression,*

2) *Service multiplex and transport,*

3) *The physical layer including the channel coding parameters and the modulation scheme, and*

4) *Planning factors (which includes consideration of both the transmission and receiver environments) and implementation strategies.*

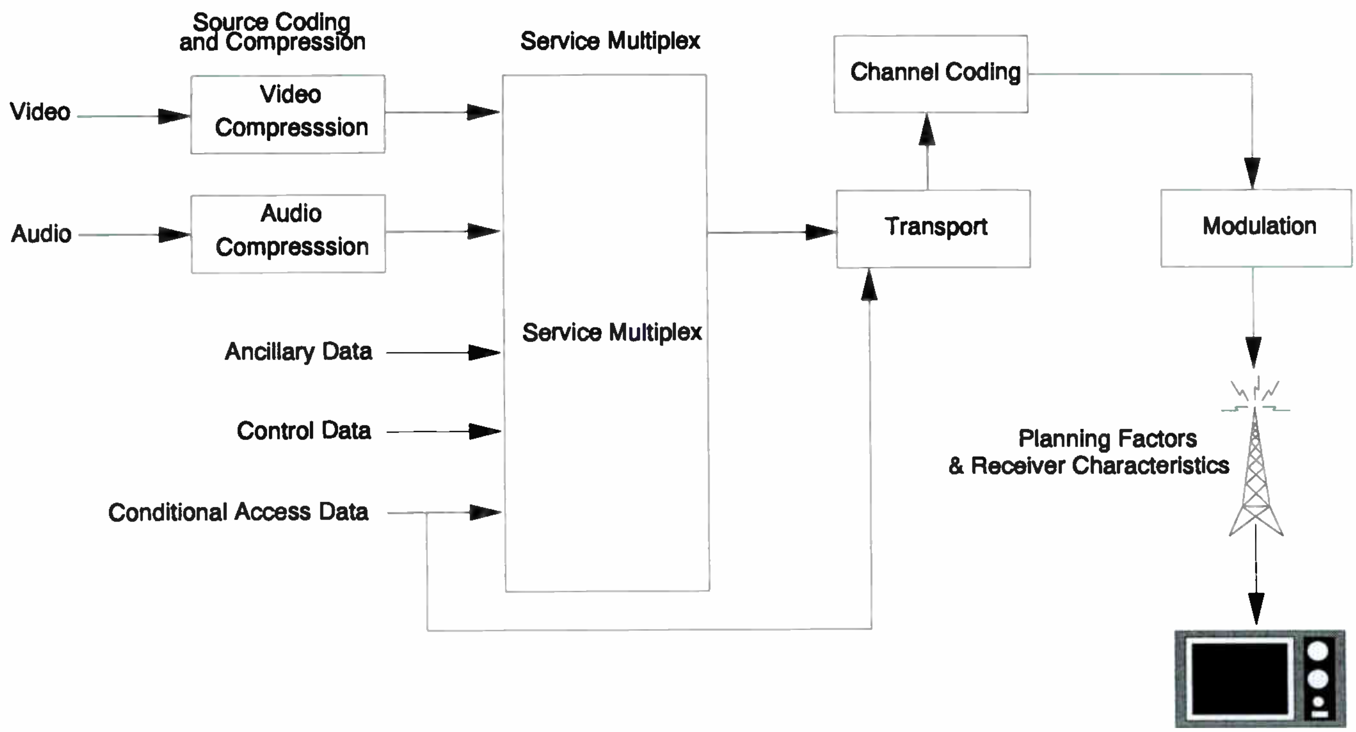

Figure 2
The DTTB System Model

Source coding refers to coding methods designed to reduce the very large data stream created when images are represented by a sequence of individual picture element (pel) values in the digital domain or when sound is represented by digital audio samples. Source coding involves processes that reduce the bit stream containing the image, sound or ancillary data information in such a way as to be able to recreate a representation of the original source at the receiving point without noticeable or unacceptable degradation. The purpose of the source coder is to convert the audio and video into data and minimize the number of bits needed to represent the information.

The "service multiplex and transport" refers to the means of dividing the digital data stream into "packets" of information, uniquely identifying each packet or packet type, and appropriate methods of multiplexing the video, audio, and ancillary data stream packets into a single program or service data stream. Multiplexing also provides the capability of combining different program data streams into a single broadcast channel for simultaneous delivery. In developing an appropriate transport mechanism, interoperability between digital media such as terrestrial broadcasting, cable distribution, satellite distribution, recording media, and computer interfaces must be a prime consideration.

The world community developed a standard, MPEG-2,[6] for the basic coding and multiplexing of video, audio, and data signals for digital broadcastings systems.[7] The MPEG-2 standard was developed for television applications in which channel bandwidth or recording media capacity is limited and the requirement for an efficient transport mechanism is paramount. The MPEG-2 standard syntax was also designed to be compatible with the Asynchronous Transfer Mode (ATM) transport mechanism. ATM was developed for use in the telephony environment where variable path transmission must be accommodated.

The physical layer includes the channel coding and modulation scheme. The channel coder takes the resulting compressed data bit stream and adds additional information that can be used by the receiver to recognize and reconstruct the images, sound, and ancillary data from the transmitted signal. This module includes mechanisms by which additional data are added to the multiplexed data stream to provide protection against loss of the signal. The characteristics of the channel coder are selected to support the modulation scheme adopted for the medium through which the data must be transported.

Modulation is a mechanism whereby the protected data stream is imposed on one or more carrier signals for transmission. These transmission systems are referred to as single carrier and multiple carrier schemes, respectively.

"Planning factors and implementation strategies" include consideration of the characteristics of the transmission media and the consumer appliance. In the case of terrestrial broadcasting, this includes strategies appropriate for the introduction and implementation of a digital terrestrial television broadcast service, taking into account existing broadcasting services.

Minimum Standards Set

To achieve exchange of program and data services across media boundaries, it is necessary to agree upon a basic set of standards. This set includes the following:

1. A common compression syntax for both video and audio.

2. A common program multiplex standard, providing for:

3. A common standard for identification (Headers/Descriptors).

4. A common recording standard for program interchange for the level of compression used.

5. A common electrical and mechanical interface standard at the data stream level.

In October 1994, S.Baron[8] and D.Wood[9] working together developed a Recommendation[10] that provided the basis for agreement on the minimum standards set for digital terrestrial television broadcasting system. The agreement provided for the following:

- A base system capable of conveying a single HDTV service or a number of conventional quality services;

- Coding video sources in conformance with the MPEG-2 standard at the Main Profile, Main Level (MP/ML) or higher;

- Coding audio sources in conformance either with the MPEG-2 standard Level II or the AC-3 standards. Manufacturers have developed single integrated circuits (ICs) which can decode both MPEG-2, level II and AC-3 audio coding.

- A service multiplex and transport that conforms to the MPEG-2 standard and is based on a common Service information and Header-descriptor system;

- For terrestrial broadcasting, a channel coding and modulation scheme using either the 8-VSB system where single carrier systems are appropriate or COFDM technology where multiple carrier systems are appropriate.

Summary of Task Group Recommendations

Task Group 11/3 completed its work in November 1996 and produced a set of Recommendations and Reports that defines a unique digital terrestrial television broadcasting system. A unique system provides benefits to two constituencies: consumers and those content producers who seek global markets for their products.

The work of the Task Group was approved by ITU-R Study Group 11 (Television) in April 1997 and by the ITU Radio Assembly in October 1997.

Specifically, the subset of MPEG-2 defined in the Recommendations of Task Group 11/3 allows for the design of a single decoder at the transport layer that can translate the service multiplex and transport layer and can decode the audio and video compression and coding layers for any system that conforms to the DTTB set of Recommendations. This establishes a "plug-and-play" environment for the consumer without the need to consider the specific coding subset used.

The next layer, the physical layer, provides the channel coding and modulation schemes appropriate for specific wired, wireless, and recorded media. The Recommendations concerning the physical layer for broadcasters take into consideration the existing 6, 7, and 8 MHz allocation of channel assignments and the need to accommodate differing environments and planning factors. The set of Recommendations and Reports can be viewed as providing a single, compatible system solution for DTTB within the practical physical limitations of the current worldwide channel assignment environment.

Two subsets of the standard's set of tools have been described in detail: System A (ATSC) and System B (DVB). The differences between the two subsets have been minimized and harmonized with respect to the video and audio coding and transport levels so that there are no conflicts and single, "plug-and-play" decoders are possible.

Subsystem capability summary

A list of the international Recommendations and Reports produced by the Task Group are found in **Table I.**

In summary, the system provides the following features:

Video coding: The MPEG video encoding profiles and levels being considered for inclusion in the international standards were reduced from more than 20 to one at the decoder level. Recommendation ITU-R BT. 1208 focuses on Main Profile at Main Level (MP@ML) encoding for SDTV source video and Main Profile at High Level for HDTV source video. The Recommendation goes on to state that in order for a television receiver to be able to decode various television services, it has to have functionality of the highest profile and the highest level proposed for these services. It further concludes: "This leads [--] the universal television receiver decoder to the choice of the Main Profile at the High Level as the conformance point in the MPEG-2 standard."[11]

The number of profiles and levels included has an impact on the cost of the consumer appliance (receiver) and the cost of providing programming content for the international market.

In the final stages of the discussion, the Task Group agreed to drop consideration of two profile/levels: the Main Profile at 1440 (MP@14) and the Spatially Scalable Profile at 1440 Level (SC@14).

The MP@14 profile/level had been considered for use in services that were not-quite HDTV (EDTV). It has been shown that the MP@HL profile/level described in the DTTB standard can accommodate those services efficiently.

The Spatially Scalable (SP@14) profile had been considered for accommodating spatially and temporally scaled services. After a thorough investigation, consideration of this profile was dropped as spatially and temporally scaled services have been shown to be extremely "bit-hungry", inefficient, and were considered inappropriate for use for terrestrial broadcasting, at this time.

Table I
Task Group 11/3 Reports and Recommendations

Area of Interest	Document
System:	
Report: "A Guide to DTTB in the VHF/UHF Bands" (11/4)	Handbook
Rec:"The Basic Elements of a Worldwide Family of Common Systems for DTTB"	ITU-R BT.1298
Video coding and compression:	
Rec: "Video Coding for DTTB"	ITU-R BT.1208
Audio coding and compression:	
Rec:"Audio Coding for DTTB"	ITU-R BT.1196
Service multiplex and transport:	
Rec:"Service Multiplex Methods for DTTB."	ITU-R BT.1209
Rec:"Service Multiplex, Transport,and Identification Methods for DTTB."	ITU-R BT.1299
Rec: "Data Services in DTTB"	ITU-R BT.1300
Rec: "Data Access Methods for DTTB"	ITU-R BT.1207
Rec: "Conditional Access Methods for DTTB"	[TG11/3-XXH]*
Physical layer:	
Rec: "Error Correction, Data Framing, Modulation and Emission Methods in DTTB"	ITU-R BT.1305
Rec: "Spectrum Shaping Limits for DTTB"	ITU-R BT.1206
Rec: "Performance of DTTB Emission Equipment"	[TG11/3-XXK]*
Rec: "Planning Factors Criteria"	[TG11/3-XYZ]*
Report: "DTTB Service Coverage and Field Trials" (11-3/R2)	Handbook
Report: "Planning Factors and Implementation Strategy" (11-3/R1)	Handbook

* Under further development within ITU-R, Working Parties 11A and/or 11C.

The set of DTTB video subsystem tools are defined in ITU-R BT.1208 and allow content producers to provide programming in conventional, widescreen, and HDTV formats. Single decoders capable of decoding the complete set of tools defined in ITU-R BT.1208 and capable of decoding the MPEG-2 Main Profile, High Level (MP@HL) have been developed, and chip sets are under development by multiple manufacturers. The existence of single decoders capable of decoding the entire set of tools defined in ITU-R BT.1208 fully meets the request of the World Broadcasting Union for unique global broadcasting systems leading to single universal consumer appliances.

Audio coding: The set of DTTB audio subsystem tools are defined in ITU-R BS.1196 and allow content producers to choose between both MPEG1 backward compatible and the highly efficient, AC-3 non-backward compatible compression and coding tools. Single decoders capable of decoding the complete set of tools defined in ITU-R BS.1196, already exist, and chip sets are provided by multiple manufacturers. The existence of single decoders capable of decoding the entire set of tools defined in ITU-R BS.1196 fully meets the request of the World Broadcasting Union for unique global broadcasting systems leading to single universal consumer appliances.

Transport level: The service multiplex and transport that provides the foundation for and defines the DTTB system is a constrained subset of the MPEG-2 standard tool set and is defined in a draft new Recommendation ITU-R BT. [1299] "Service Multiplex, Transport, and Identification Methods for Digital Terrestrial Television Broadcasting".[12]

The assignment of packet identification as described in the North American ATSC standards and the European DVB standards was harmonized to avoid the possibility of decoder errors. Systems which conform to the subset of the MPEG-2 transport defined in Rec. 1299 including the use of the Descriptor Tags and Table ID assignments, allows the development of single devices capable of decoding the entire set of tools defined in Rec. 1299. Such decoders do not yet exist. Development of such decoders was made possible by the decisions taken at the final meeting of Task Group 11/3. The existence of single decoders capable of decoding the entire set of tools defined in Rec. 1299 will meet the request of the World Broadcasting Union for unique global broadcasting systems leading to single universal "plug-and-play" consumer appliances.

One of the ATSC standards documents (A/57)[13] defines a unique content numbering system which is beneficial to content producers who seek global markets for their products. It is an extension to the MPEG syntax and corrects for a deficiency in the MPEG syntax. This addition to the MPEG syntax was adopted by Task Group 11-3 during the meeting in Sydney and incorporated in Rec. 1299.

Physical layer: The Recommendations concerning the DTTB physical layer (error-correction, data framing, channel coding and modulation and emission methods) are defined in Recommendation ITU-R BT. 1305[14] and take into consideration the existing 6, 7, and 8 MHz allocation of channel assignments and the need to accommodate differing environments and planning factors. The set of Recommendations and Reports can be viewed as providing a single, compatible system solution for DTTB within the practical physical limitations of the current worldwide channel assignment environment.

Draft new Recommendation 1305 provides for 8-VSB where a single-carrier modulation technique must be employed and COFDM where a multi-carrier modulation technique is found preferable. The difference in modulation technique employed depends greatly on local planning factors which are dependent on the bandwidth (6, 7, or 8 MHz) of the channels used in various parts of the world and on local environmental conditions.

Multi-program capability and interoperability with other media: The application of digital signal compression technology to coding television signals accommodates multi-program transmission in the existing channels. Compressed digital television systems offer the prospect of considerable improvement in service quality while appreciably improving spectrum utilization as compared with analog transmission methods. This capability can also be exploited to deliver multiple digitally compressed television programs instead of a single conventional, enhanced or high definition program. These digitally compressed television signals can be accompanied by digital high-quality sound, coded conditional access information and ancillary data channels.

Furthermore, the same approach could be implemented in the transmission of multi-program signals or stereoscopic television services over existing digital satellite or terrestrial links or cable television networks. Task Group 11/3 paid particular attention to constructing a digital architecture that could accommodate both high-definition television (HDTV) and conventional television services in the terrestrial broadcasting environment and was interoperable with cable delivery, satellite broadcasting, and recording media.

The approach taken provides harmonization between services by using a unified, common method of video and audio source coding and a unified, common service multiplex and transport. As noted above, two different subsets of this unified set are defined; System A and System B. The two subsets are compatible and single decoders can be provided that can extract either subset from the data stream.

This unified transport data stream is then provided with a framing structure, error protection mechanism, and modulation scheme appropriate to the distribution media. The common transport is seen as a "container" and facilitates the interoperability of the signal through different delivery media. This results in a common data stream after demodulation in the receiver which simplifies the complexity of the consumer receiver appliance.

HDTV: During the final meeting of the Task Group, several "50 Hz" nations indicated an interest in the 1080-line system as the basis of international agreement on an HDTV standard for program interchange. Such an agreement would be beneficial to content producers who have global market concerns. The current 1152-line/50 Hz (Europe) and 1035-line/60 Hz (Japan) standards were seen as specific to certain nations or regions. The 720-line (USA) standard was seen as too close in performance to the existing progressive-scan versions of the 625-line standard for consideration. Only the 1080-line standard was defined for both the 50 Hz and 60 Hz environments, provided a 24 frame-per-second film compatible mode, and offered sufficient improvement over existing formats to provide the basis for international agreement.[15] The 1080-line standard was approved for incorporation into Recommendation ITU-R BT.709 during 1997.[16]

The Consumer Appliance

A digital receiver designed to take full advantage of the DTTB set of Recommendations is a flexible, extensible consumer appliance. It provides the ability to process and display any television service up to including HDTV in a plug-and-play environment. This includes audio services from monaural through 5.1 channel surround sound. Service sources can be via terrestrial broadcasting, cable, satellite, recorded media, or computer based sources including the "net". All that is required is that the information "container" (the demodulated transport stream) comply with MPEG-2 Transport Stream rules.

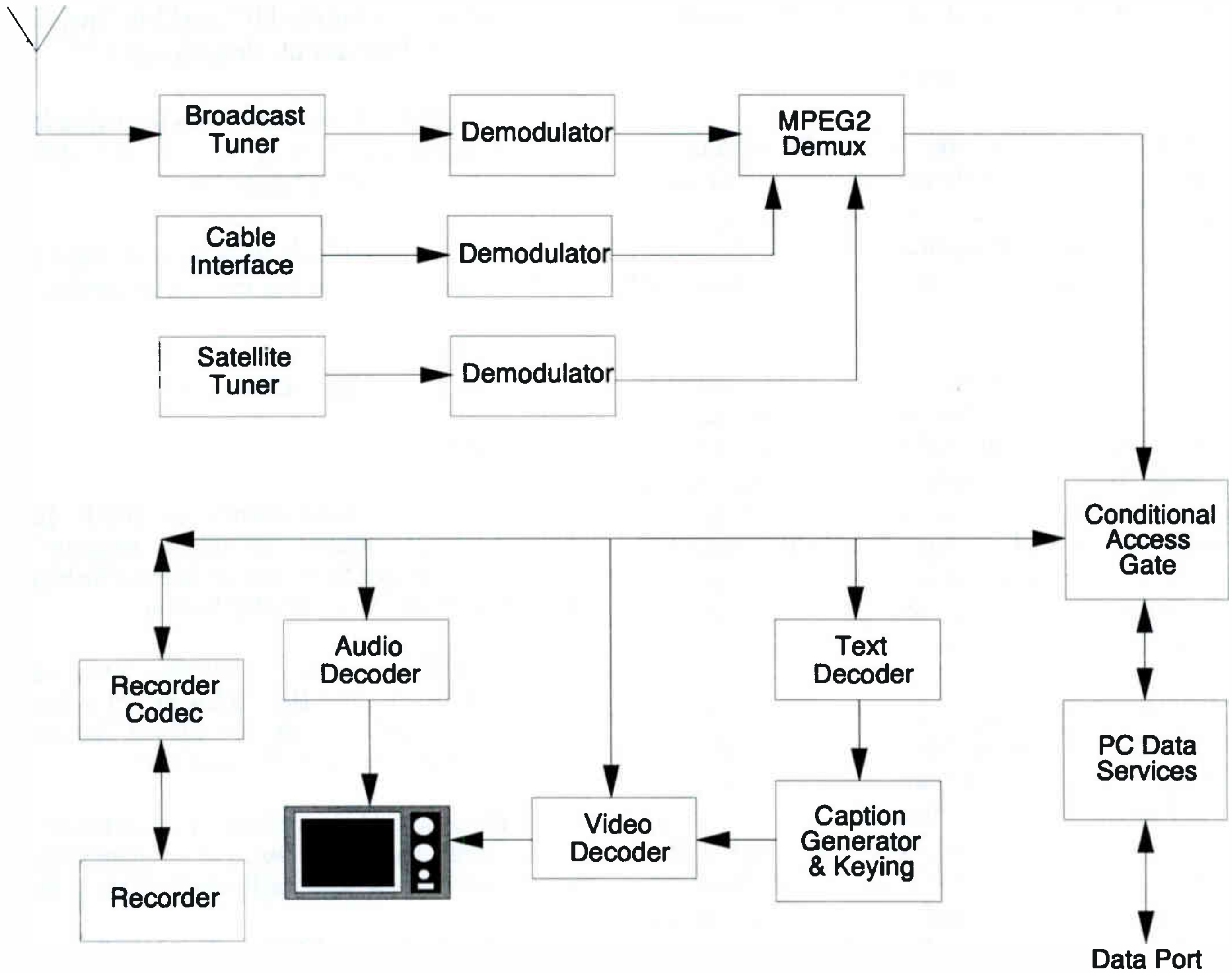

Figure 3
Receiver Block Diagram

Figure 3 provides a simplified block diagram of a possible consumer appliance. Various services cab be provided via multiple media sources using modulation schemes appropriate to the medium in question. Each service is demodulated and presented as a Transport Stream to the MPEG-2 demultiplexer. Packets within the Transport Stream are captured by the appropriate decoder module recreating the desired sound, image, and/or data service. This includes standard definition and high definition television services, monaural, stereo, or surround sound services, and associated or stand-alone data services.

Both the encoder and receiver models meet the desired goal established by the World Broadcasting Unions (WBU) to develop a system that could be used worldwide. It enables a compatible system to be developed on a global basis with all the economic advantages implied both for the content creators and their audiences.

Contributions

There were many individuals throughout the world who have made outstanding contributions to advancing digital television technology, and they number in the hundreds. I congratulate them for the excellence of their individual and collective efforts.

The work of Task Group 11/3 involved documenting a constrained and harmonized set of parameter values selected from the tools that were available. Agreement on a unique digital television system was achieved because the members of Task Group 11/3 for the most part shared a common goal and worked diligently to establish a set of Recommendations that supported that goal.

Within Task Group 11/3, a small group of individuals worked collectively to bring this vast body of work into harmony and to structure a set of Recommendations that defines the unique digital television system. They included the chair of Task Group 11/3; the two vice-chairs Mr. Terry Long (UK) and Mr. Osamu Yamada (Japan); and the Special Rapporteurs each of who accepted responsibility for one of the technical subsets of the total system: Mr. Richard Barton (Australia), Mr. Keith Malcolm (Australia), Mr. Thomas Ryden (Sweden), Mr. Brian Roberts (New Zealand), and Mr. David Wood (EBU). A special thank you must also be accorded Prof. Mark Krivocheev, chair of Study Group 11 and Mr. Giuliano Rossi (ITU) who guided the work of the Task Group.

Guidance was also provided in specific areas by the chairmen and vice-chairmen of Study Group 11, and in particular: Mr. Carmi Weinzweig (USA), vice-chairman of Working Party 11A, Mr. Stanko Perpar (Slovenia), chairman of Working Party 11C; Mr. Ken Davies (Canada), chairman of Task Group 11/1; Mr. Dominic Nasse (France), chairman of Task Group 11/2; and Mr. Troy Tepp (USA), vice-chairman of Task Group 11/5.

The work of the Task Group was also simplified by the foundation laid by the ISO/IEC MPEG chaired by Leonardo Chairligione.

To all of the individuals who gave of their time and expertise, we owe our collective thanks.

Stanley Baron,
Chairman, ITU-R Task Group 11/3.

END NOTES:

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[2] ITU/R Document 11/1008, Draft new Recommendation, "The Basic Elements of a World-Wide Family of Systems for Digital Terrestrial Television Broadcasting," 12 June 1997.

[3] S.Baron and M.Krivocheev, "Digital Image and Audio Communications: Toward a Global Information Infrastructure", Van Nostrand Reinhold, 1996, p.16-17.

[4]ITU-R, Question 121/11, "Digital Terrestrial Television Broadcasting".

[5] CCIR Document 11-3/2, "Outline of Work for Task Group 11/3, Digital Terrestrial Television Broadcasting," 30 June 1992.

[6] IEC/ISO 13818, "Generic Coding of Moving Pictures and Associated Audio Information", (MPEG-2)

[7] Chairman, ITU-R Task Group 11/3, "Report of the Second Meeting of ITU-R Task Group 11/3, Geneva, 13-19 October 1993," 5 January 1994, p.40.

[8] ITU-R Document 11-3/TEMP/1, "Chairman's Opening Remarks," October 1994.

[9] ITU-R Document 11-3/19, "Recommendation for

Main Elements of a Common Digital Terrestrial Television ITU-R Standard," 14 September 1994.

[10] ITU-R Document 11-37, Draft new Recommendation, "The Basic Elements of a World-Wide Family of Systems for Digital Terrestrial Television Broadcasting," 2 November 1994.

[11] ITU-R Document 11/1013, Draft Revision of Recommendation ITU-R BT.1208, "Video Coding for Digital Terrestrial Television Broadcasting," 12 June 1997.

[12] ITU-R Document 11/1010, Draft new Recommendation ITU-R BT.[11-3/XXE], "Service Multiplex, Transport, and Identification Methods for Digital Terrestrial Television Broadcasting," 12 June 1997.

[13] ATSC Doc. A/57, "Program/Episode/Version Identification", 30 Aug 96.

[14] ITU-R Document 11/1018, Draft new Recommendation ITU-R BT.[11-3/XXD and XXI], "Error-correction, Data Framing, Modulation, and Emission Methods for Digital Terrestrial Television Broadcasting," 12 June 1997.

[15] ITU-R Doc. 11-3/78, "Report on the Third and Final Meeting of Task Group 11/3," 1 Dec. 1996.

[16] ITU-R Doc. 11/95, "Draft Revision of Recommendation ITU-R BT.709-2, Parameter Values for the HDTV Standards for Production and International Programme Exchange, Part II - HDTV Systems with Square Pixel Common Image Format.", 23 May 1997.

VALIDATE FIELD TRIALS OF DIGITAL TERRESTRIAL TELEVISION (DVB-T)

Chris Weck
Institut fuer Rundfunktechnik GmbH
Broadcasting Systems Development
Munich, Germany

ABSTRACT

Since the completion of the European specification for digital terrestrial TV broadcasting, DVB-T [1][2], more equipment has become available and many field trials have been conducted in the framework of the ACTS project VALIDATE at different sites throughout Europe. This paper gives an overview on these trials and highlights some interesting results of the various test transmissions and the field work carried out so far.

Results from propagation measurements are summarised for signals from a single transmitter or from several transmitters, either by using gap-fillers, which work as on-channel repeaters, or explicitly within a single frequency network (SFN). Both outdoor and indoor reception are examined. Building penetration losses have been recorded and there is a brief report on the influence of a domestic gap-filler to improve the portable indoor reception within buildings.

Results on the system performance for different COFDM modes of the DVB-T system evaluated in the field are outlined for fixed as well as for portable reception and the possibilities for mobile reception are investigated, too. A few results are also available of the measured coverage probability in comparison to the service areas as predicted by computer simulations.

INTRODUCTION

VALIDATE stands for 'Verification And Launch of Integrated Digital Advanced Television in Europe'. It is one of the ACTS projects, sponsored by the European Commission in the fourth Framework Programme 'Advanced Communication Technologies and Services'. The project VALIDATE [3], led by the BBC as prime contractor, started work in late 1995 and aims to verify the European DVB-T standard for digital terrestrial television broadcasting and to prepare for the launch of services. There are 19 partners in nine European countries, representatives from several broadcasters including broadcasting research centres and the European Broadcasting Union (EBU) as well as telecom and network operators and professional and consumer manufacturers. A detailed list of partners can be found together with the bibliographical references at the end of his paper.

In VALIDATE, first computer simulations and laboratory tests and later field trials were performed in various laboratories and from several transmitter sites all over Europe. In two VALIDATE task forces, the measurement procedures for lab tests and field trials were agreed and the measurement results were exchanged and compared in order to verify the specification and to achieve reliable planning parameters for future DVB-T services.

This report gives an overview of the extensive field work and demonstrations carried out in the VALIDATE project so far. The following list shows the sites for DVB-T field trials already existing in the project:

- United Kingdom, BBC, London area and North East of England
- France, CCETT, Rennes
- France, TDF, Metz
- Germany, DTAG & TBerkom, Cologne and Berlin
- Germany, IRT, Munich
- Spain, Retevision, Madrid
- Italy, RAI, Torino
- Denmark, TeleDanmark
- Sweden, Teracom, Stockholm
- Ireland, RTE, Dublin
- The Netherlands, NOZEMA

Further transmitting sites and DVB-T pilot services are in preparation.

SCOPE OF FIELD TRIALS

The scope of field trials in VALIDATE is to confirm the extensive range of laboratory results and to investigate fully the overall performance of the DVB-T system for various transmission modes and various reception conditions, where the received signal is likely to suffer a combination of propagation and reception impairments that would be difficult to estimate in the laboratory.

Furthermore, field trials are required to answer questions of service planning, that is to evaluate the coverage probability for the different structures of transmitter networks and to investigate the different kinds of distortions of the transmission channel. Sufficient representative field-trial data has to be acquired to improve the accuracy of the values adopted for critical planning parameters (e.g. minimum field-strength and C/N, protection ratios, etc.) used within service prediction models. Investigation of the stability / reproducibility of measurements made at any given site over a period of time and under potentially different propagation and reception conditions (including the possible effects of periodic changes in climatic or atmospheric conditions) are still going on.

DVB-T SPECIFICATION

In the development and definition phase of the European DVB-T specification [4], the advantages of the multicarrier modulation outweighed those of the single-carrier methods in terms of broadcasting requirements (as it was likewise the case with EUREKA 147 DAB for digital sound broadcasting). The crucial factor in favour of the chosen COFDM method (Coded Orthogonal Frequency Division Multiplex) [5] is the ability to cope with strong echoes due to multipath propagation and the capability to set up single-frequency networks (SFNs), which offer network planners a higher network efficiency [6][7][8].

The specified DVB-T system offers a wide range of potential applications: conventional multi-frequency networks (MFNs), i.e. single transmitter applications, prohibited ("taboo") channel operation, etc. and single-frequency networks (SFNs). The service can be dedicated to stationary or portable reception or both using hierarchical transmission. The network operator can select technical parameters such as the number of OFDM carriers, the length of the guard interval (this is a cyclic continuation of the useful OFDM symbol), the degree of error protection and the modulation method. The last two parameters in particular allow the operator to reach an individual compromise between the number of programmes carried and their transmission reliability.

The transmitting system provides an MPEG-2 transport mechanism, a kind of data container whose size depends on the chosen transmission parameters (transmission mode). It allows for full flexibility with respect to the number of transmitted programmes, the content of any digital information or the kind of digital services (e.g. HDTV to LDTV, surround sound, data etc.).

The following transmission parameters can be selected:

- BANDWIDTH: 8 MHz, 7 MHz, 6 MHz
- MODULATION: 4-PSK, 16-QAM, 64-QAM
- HIERARCHY: 4-PSK in 16-QAM or in 64-QAM
- CARRIERS: 6817 (8K-FFT), 1705 (2K-FFT)
  SPACING: 1116 Hz (8K), 4464 Hz (2K)
- USEFUL SYMBOL DURATION: 896 µs (8K), 224 µs (2K)
- GUARD INTERVAL: 1/4, 1/8, 1/16, 1/32
  DURATION: 224, 112, 56, 28, 14, 7 µs
- INNER CODE RATE: 1/2, 2/3, 3/4, 5/6, 7/8

The latter refers to the error protection of the inner convolutional code. Together with the outer error-correction code (RS 204,188) and the overhead for pilot carriers a spectrum efficiency of 0.65 to 4.2 bit/$Hz_{bandwidth}$ can be achieved. For example in an 8 MHz TV channel, useful data rates from 4.98 to 31.67 Mbit/s can be transmitted.

PROPAGATION MEASUREMENTS

In comparison to analogue television signals, which behave like narrow-band signals because of the single video carrier, the DVB-T signal is a wide-band signal where the energy is distributed over a large number of carriers in the radio channel. The first issue for investigation was the received field strength at any reception point and the statistics of the location variation.

Some results for signals from a single transmitter are given in **Table 1**. The bulk of distribution functions of the received field strength for different test routes was following a log-normal law. Measurements of the CCETT showed

Location	Terrain Class	Standard Deviation / dB		
		fast fading	slow fading	combined
TERACOM, Stockholm, Sweden	open field	1.7	2.7	3.3
	forest	2.6	3.3	4.4
	suburban	2.3	3.2	4.1
	urban	2.2	2.5	3.3
CCETT, Rennes, France	rural			2.5
	suburban			3.3
	urban			3.9
BBC, London, UK	rural			2.5
	urban			4.0
DTAG, Berlin, D	suburban			3.3

Table 1: *Location variation of field strength for various locations and receiving conditions*

that the standard deviation of an analogue TV signal transmitted from the same tower is 0.1 to 0.4 dB higher than the one of the digital signal [16].

It is well known that, in contrast to analogue services, there is a very rapid degradation of a digital transmission system at the fringe of the coverage area. For DVB-T the margin in a Gaussian channel between the onset of impairment and the failure point is only about 1 – 2 dB, in a Rayleigh channel it can be much more, due to fading effects. Therefore, it is not sufficient to consider the median value of the field strength for coverage considerations, but to introduce a margin from e.g. 50 % to 99% of the covered locations. For analogue TV systems this margin was in the order of 16 dB. Latest results from CCETT [16] given in **Table 2** show that there is a distinct lower margin necessary for the digital system than for the analogue one. This result relaxes the power requirements for DVB-T.

Location	Terrain Class	Margin for 50 % to 99 % covered locations	
		analogue TV	digital TV
CCETT, Rennes, France	rural	11.7 dB	9.7 dB
	suburban	16.0 dB	12.0 dB
	urban	16.7 dB	11.8 dB

Table 2: Margin for the increase from 50 % to 99 % coverage

Single Frequency Network Gain

Conventional networks use individual radio frequencies for each transmitter (MFN: multi-frequency network) to avoid mutual interference. One programme transmitted within a network occupies therefore a set of radio frequencies. Now, the great advantage of the DVB-T transmission system is that a large area may be covered by transmitters working all on the same radio frequency, provided the relevant signals from various transmitters arrive at a reception point within the duration of the guard interval.

Such single-frequency network (SFN) has important advantages for network planning. The frequency efficiency of large SFNs can be up to 4 times higher than for MFNs. Furthermore, the power efficiency of an SFN is better, because the coverage probability at a reception location is increased owing to the signal diversity of the different propagation paths.

Measurements of the DTAG in Berlin in a SFN with 3 transmitters showed that the standard deviation of the received power is about 2.6 dB compared to 3.3 dB for the signal from a single transmitter. Especially, if the coverage probability in the middle between transmitters is considered, where the contribution of the main signal paths is usually in the same order and rather low, there is observed a high diversity gain in an SFN. For example, the SFN gain was found at such locations to be in the order of 4 – 6 dB (considering a value of 90 % for the coverage probability). Further measurements are still necessary to achieve more statistical results.

Professional gap-filler

One additional advantage of SFNs to be mentioned is, that every gap within a coverage area can be covered by using active deflectors working on the same single frequency, so-called professional gap-fillers. There will be in principal no physical, but may be a financial limitation to cover an area up to 100 % with such devices.

A prototype of a professional gap-filler for application in real SFNs was developed in the framework of VALIDATE by Mier Communicaciones, Spain. First field trials were performed in Berlin in co-operation with DTAG, where the gap-filler received the DVB-T signal from a tower at 21 km distance. The signal was then retransmitted with a power of 100 W ERP to cover the city of Potsdam, which is about 26 km away from the main transmitter and shadowed by hills. The trial was very successful. A very high antenna isolation of up to 105 dB was achieved between receiving and transmitting antenna. Further improvement is expected during the optimisation of the prototype.

Single Frequency Network Synchronisation

It is evident that the individual transmitters of a SFN have to transmit each single bit exactly in the same manner and at the same time. This is a minor problem if the broadcast signal is retransmitted directly or after frequency transformation. But in general the MPEG transport stream may be send through different digital links to each transmitter and it becomes essential to synchronise the modulation procedures of all individual transmitters.

VALIDATE investigated this problem in detail and supported the specification process for synchronising SFNs. A synchronisation device for DVB-T modems was developed by ITIS [8] and was successfully tested by RTE and ITIS in Dublin, Ireland, in November 1997 using two transmitters. It was the first SFN operation based on a real primary distribution network as described by the SFN-DS specification [17]. The synchronisation is based on a central SFN adapter inserting Mega-frame Initialisation Packets (MIP) into the MPEG2-TS with GPS time information. At each transmitter site the MPEG2-TS, e.g. from a 34 Mbps PDH link is modulated in a synchronous mega frame according to the local GPS time.

INDOOR PROPAGATION MEASUREMENTS

The field-strength distribution within buildings is very important for portable receivers using very small indoor antennas or poor whip antennas. To estimate the service coverage, measurements were made at a number of locations within office buildings and within a sample of domestic dwellings where set-top reception on portable receivers would typically be required (e.g. living room, kitchen, bedroom). The field strength was found to be log-normally distributed with standard deviations given in **Table 3**. The value of the standard deviations in rooms in direction to the transmitter was about 0.5 – 0.6 dB higher than the value for rooms at the opposite side of the house. For comparison, measurements of signals with lower bandwidth were performed, too. This was a DAB signal with 1.5 MHz bandwidth and an analogue carrier with 120 KHz bandwidth.

Location	Standard Deviation
BBC, London, UK (within one room)	2.8 dB 2.4 – 3.2 dB
CCETT, TDF, Rennes, France	2 dB
TERACOM, Stockholm, Sweden	3.2 dB
IRT (1.5 MHz bandwidth)* (120 KHz bandwith)*	3.5 dB 5.5 dB

Table 3: *Location variation within rooms*
*(* not DVB-T, for comparison only)*

Investigations of the time variance of the signal due to moving people in the room showed a log-normal distribution, too. The standard deviation was about 1 dB. However it must be pointed out that very deep fades corresponding to a total shadowing of the receiving antenna are not taken into account by such a standard deviation. Such fades are usually caused by somebody coming very close to the antenna and are not to be considered as a normal condition of reception.

Building penetration loss and building loss

The indoor measurements in conjunction with outdoor measurements at the same site allow accurate values for the building penetration losses to be determined for use in service planning models for portable reception.

Two different questions can be distinguished: What is the loss between outdoor and indoor reception of DVB-T based on the same antenna height (e.g. field strength outside the window), which is here referred to as the building penetration loss, and what is the loss, compared to the traditional roof-top antenna, which is here referred to as building loss. The results given in **Table 4** indicate that the height loss is a very significant part of the building loss, were the building penetration loss itself is in average lower than 10 dB.

Location	Average Building Penetration Loss
IRT, Munich, Germany	8.5 – 9.1 dB (VHF) 7.0 – 8.5 dB (UHF)
TERACOM, Stockholm, Sweden	6.4 dB (UHF) (standard deviation 3.6 dB)
CCETT, TDF, Metz, France	7 dB (3rd floor) 2 dB (5th floor) (standard deviation 4 dB)
Location	**Building Loss (incl. height loss)**
BBC, London, UK	21 – 23 dB (1st floor, UHF) 28 – 30 dB (ground floor, UHF)
CCETT, TDF, Metz, France	10 dB (5th floor flat) 12 – 17 dB (individual house)

Table 4: *Building penetration loss and building loss for various locations and receiving conditions*

Domestic gap-fillers

As portable reception is one of the chief advantages for DVB-T compared to satellite and cable transmission systems. VALIDATE investigated solutions to increase the field strength in buildings by using domestic gap-fillers, which work as on-channel repeaters inside buildings. Two different prototypes, differing in size and cost, were developed by Televés, Spain. One working as broadband device and the other amplifying three selected channels.

The first trials performed by Retevision and BBC of indoor reception of DVB-T using domestic gap-fillers validated this concept and were very encouraging. Real off-air signals received by a directional roof-top antenna or derived from a MATV system were rebroadcast in domestic houses and in a very hostile laboratory environment. An omnidirectional antenna or a small yagi antenna was used in the centre or in a corner of a house, respectively. An output of less than 250 µW from the domestic gap-filler was found to be sufficient for a set-top reception of DVB-T with 64-QAM in every room of a residential building. Detailed measurement are still going on at the BBC and have now been started at other VALIDATE test sites.

SERVICE COVERAGE MEASUREMENTS

Since propagation measurements merely consider the received power they do not take into account the properties of the DVB-T system. This means that the question whether one location is covered depends not only on the received power, but on the fact whether the receiver is really able to provide an error-free reception of the sound and picture or not. Therefore, the individual performance of the receiver, the chosen technical parameters like modulation and channel coding as well as the type of transmission channel have to be considered in detail.

The determination of the service coverage at any location can be based on the picture quality itself or on real bit-error ratio measurements (BER). A BER of $2\cdot10^{-4}$ after the Viterbi decoder (inner code) was adopted as one reference value corresponding to quasi error-free (QEF) operation after the outer Reed Solomon RS (204,188) decoder. The BER after Viterbi allows for more accuracy of the results than the BER after the Reed-Solomon error protection, which would be available on the MPEG transport-stream level. This QEF operation generally corresponds to a criterion based on an period of at least 30 seconds, where no picture or sound impairments occur. Concentrated errors usually are visible and audible if the signal-to-noise ratio *C/N* for $2\cdot10^{-4}$ is reduced further by about 1 dB, corresponding to a BER of more than 10^{-3} after Viterbi decoding.

In the DVB-T specification [1] there are figures available for the required *C/N* for all transmission modes to achieve a BER of $2\cdot10^{-4}$. These Figures are based on simulation results for three types of transmission channels, but without taking any implementation margin into account:

- Gaussian channel
  direct sight, no multipath, laboratory condition
- Rice channel (k = 10 dB)
  for stationary reception using directional antennas
- Rayleigh channel
  for portable reception using omnidirectional antennas

The transmission channel in the field is usually an intermediate type between a Gaussian and a Rayleigh channel. Therefore, the required *C/N* or the required signal power, respectively, to achieve BER $2\cdot10^{-4}$ will be different at different receiving locations. This fact is actually shown by the BER curves measured at different test locations in Munich [13]. Each of **Figure 1** to **Figure 6** shows the BER behaviour after Viterbi decoding of a chosen transmission mode at about 15 different receiving locations versus the receiver input power. Three modulation modes were examined in the field:

- 64 QAM, code rate R = 2/3,
  guard interval GI = 1/8, 8k-FFT
- 16 QAM, code rate R = 3/4,
  guard interval GI = 1/32, 8k-FFT
- QPSK, code rate R = 1/2,
  guard interval GI = 1/4, 8k-FFT.

The results are given for stationary reception (directional antenna at 10 m height) as well as for portable reception (omnidirectional antenna at 1.5 m height). The continual curves of the BER show that the results in the field are reliable and stable, which is important for the validation of the DVB-T specification.

Figure 1 to **Figure 6** can be used to determine the required receiver input power (abscissa) to achieve a quasi error-free reception, which is indicated in the figures by a horizontal broken line at BER of $2\cdot10^{-4}$. This value is of great interest for planning DVB-T services. But for the verification of the DVB system, the most important parameter is actually the required *C/N* value in comparison to the theoretical *C/N* values given in the DVB-T specification.

In the real transmission channel *C/N* can be determined based on of the equivalent noise figure of the receiver and the actual signal power received. The receiver noise figure in these tests was about 8 dB corresponding to a noise floor of –97 dBm with an accuracy of about ±1 dB (receivers may have noise figures as low as 5 dB). Based on this figure the actual *C/N* value can be estimated at each reception location. For comparison, the theoretical *C/N* values for the three types of transmission channel are indicated with a cross on the horizontal broken line in the figures (from left to right: Gauss, Rice and Rayleigh channel). The *C/N* evaluation, which should be considered as indicative only, allows nevertheless the conclusion that the implementation margin of the receiver in the field is rather low: 1.5 – 3 dB in addition to the theoretical value for a Gaussian or Rayleigh channel, respectively. This value was confirmed by measurements of DTAG in Berlin.

Comparison with service prediction

The planning of digital terrestrial services is based largely on computer predictions. Consequently, it is important to determine how well these predictions compare with the coverage obtained by measurements with real DVB-T equipment.

The tests in Munich considered above showed a good accordance with the prediction, but only few results of measurements were available for a statistical analysis.

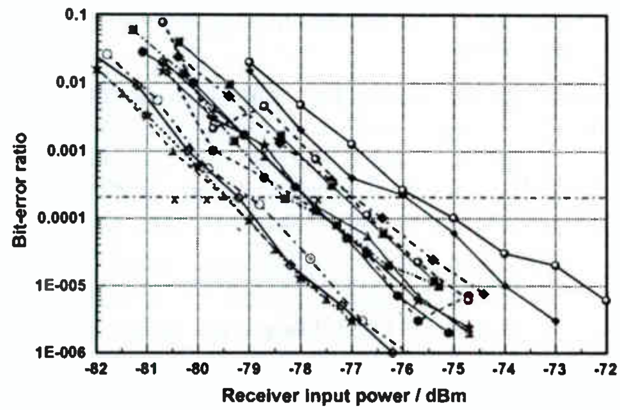

Figure 1: BER versus receiver input power
modulation: 64 QAM, R 2/3, GI 1/8, 8k FFT
directional antenna height 10 m above ground

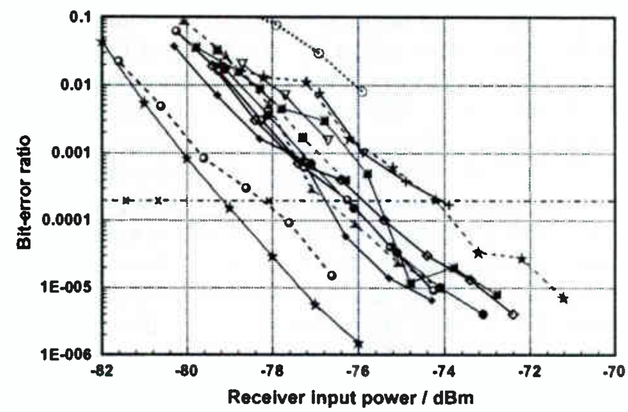

Figure 2: BER versus receiver input power
modulation: 64 QAM, R 2/3, GI 1/8, 8k FFT
omnidirectional antenna height 1.5 m above ground

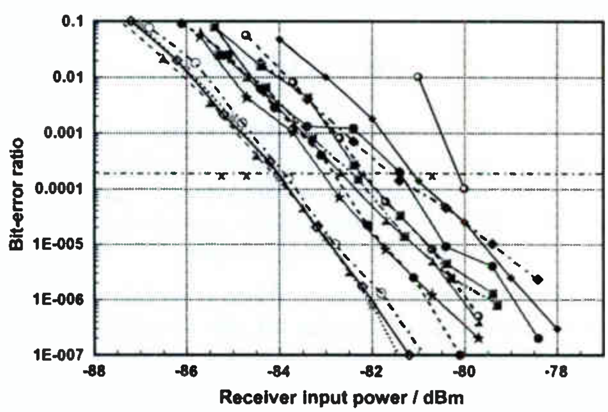

Figure 3: BER versus receiver input power
modulation: 16 QAM, R 3/4, GI 1/32, 8k FFT
directional antenna height 10 m above ground

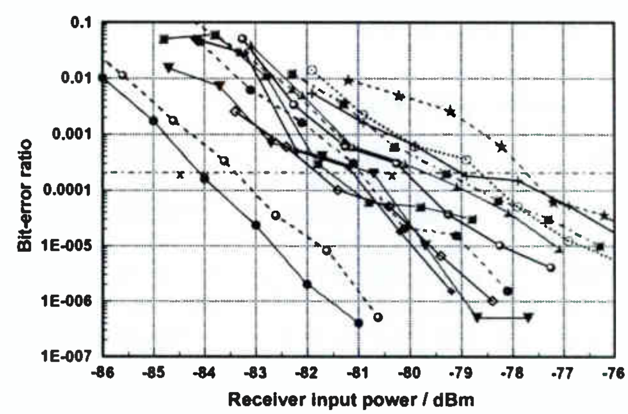

Figure 4: BER versus receiver input power
modulation: 16 QAM, R 3/4, GI 1/32, 8k FFT
omnidirectional antenna height 1.5 m above ground

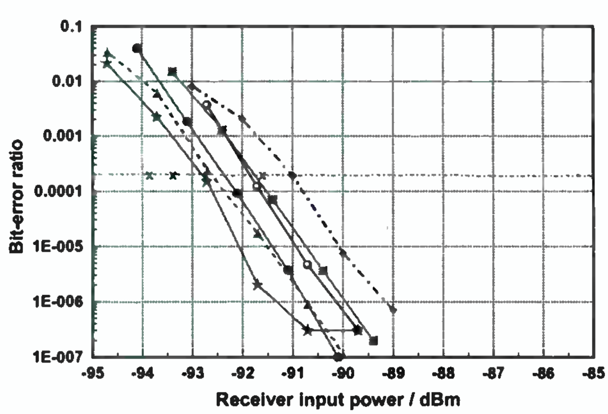

Figure 5: BER versus receiver input power
modulation: QPSK, R 1/2, GI ¼, 8k FFT
directional antenna height 10 m above ground

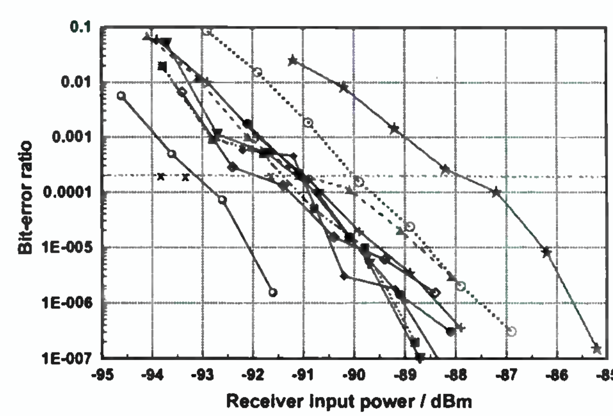

Figure 6: BER versus receiver input power
modulation: QPSK, R 1/2, GI ¼, 8k FFT
omnidirectional antenna height 1.5 m above ground

More extensive surveys have been conducted by the BBC in both the London area and in the North East of England to measure the percentage of coverage within a number of 1 km by 1 km squares. Only squares with a marginal level of coverage were considered. They were selected with respect to initial results of a coverage prediction using a computer propagation model. Measurements were made at 10 to 15 randomly chosen test points within each square to determine the percentage of these points at which reception was achieved. These results were compared with the predictions made using the methods adopted during the UK planning study. In 80% of the squares the coverage was better than predicted. Most of the areas, where the measured coverage was not as good as predicted, were known to suffer poor analogue reception. This was generally due to tall buildings which were not considered by the propagation model. Other areas which were predicted to be only marginally served were in fact completely covered. In conclusion, the measured coverage was found to be rather better than predicted.

Mobile reception of DVB-T

Even though the DVB-T standard was not developed for mobile reception there were very encouraging results of initial field trials performed by DTAG in Cologne. The tests in UHF channel 40 (626 MHz) showed, especially if the QPSK mode, R=1/2, 2K-FFT, is chosen, no reception loss due to the movement of the receiver. The tested speed was 170 km/h. It may have been advantageous reception conditions, because any failure of the system did occur at locations where the field strength was not sufficient at all. Tests with 16-QAM at a speed of 60 km/h were successful, too, but further investigation are required.

Especially, in the follow-on project of VALIDATE, which is named MOTIVATE, more tests (not only 2K-FFT) will be performed and the receiver and the channel estimation will be optimised for mobile reception of DVB-T.

DEMONSTRATIONS

Most of the VALIDATE trials conducted were combined with various professional and public demonstrations of the performance of DVB-T.

Since May 1996 the BBC performed public DVB-T demonstrations in the London area and the North of England using a fully DVB-T compliant 2K-FFT modem developed by them. This modem was also used to demonstrate DVB-T at IBC'96 and at DBC'96 in the Netherlands. The first public demonstration using the 8K-FFT modem developed in the European Race project dTTb (digital Terrestrial Television broadcasting) took place in Munich in January 1997.

Further demonstrations in 1997 with either 2K-FFT or 8K-FFT modems were performed or supported by VALIDATE partners, respectively, in Berlin, Cologne and Munich (D), Madrid (E), ITVS'97 Montreux (CH), Sutton Coldfield (UK) and at IBC'97 Amsterdamm (NL).

At the Internationale Funkausstellung Berlin (IFA'97) three UHF channels were used to transmit a total of 8 TV programmes and one DVB data service. Stationary reception from two different transmitters was demonstrated by DTAG. Portable reception of one programme was demonstrated by the IRT as well as mobile reception in a bus and a car. The high reception quality of DVB-T when using a whip antenna smaller than a pencil in a very hostile environment, especially in comparison to analogue TV reception, was highly convincing.

CONCLUSION

The VALIDATE project has verified the very complex European specification for digital terrestrial broadcasting DVB-T. A notable amount of field trials have been conducted so far and unambiguously proved the results of laboratory tests and computer simulations.

Propagation measurements performed outdoor as well as indoor showed clear benefits of the broadband DVB-T signal. The performance for stationary and portable reception in the field was as good as expected or even better in terms of C/N. Different modulation modes and channel coding rates were tested and the results confirmed predictions from simulation.

The field trials demonstrated the performance of DVB-T, the operation of a SFN, the concept of gap-fillers and the possibility of mobile reception. Under comparable poor receiving conditions the quality of DVB-T was found to be greatly superior to the analogue reception of TV.

ACKNOWLEDGEMENTS

The Author would like to thank all partners of the VALIDATE project for the contributions to the field work described in this paper.

The support of this project by the European Commission under AC106 is acknowledged as well. Furthermore, VALIDATE has relied heavily on the work done in earlier collaborative projects including the European RACE project dTTb, the Nordic HD-Divine project, and the German $^{H}DTV_{T}$ project.

VALIDATE PARTNERS

British Broadcasting Corporation (BBC)	UK
Robert Bosch GmbH (Bosch)	D
Centre Commun d'Etudes de Télédiffusion et Télécommunications (CCETT)	F
Deutsche Telekom AG (DTAG)	D
Deutsche Thomson Brandt GmbH (DTB)	D
Institut für Rundfunktechnik GmbH (IRT)	D
Innovations Télécommunications Image Son (ITIS)	F
Mier Comunicaciones (MIER)	E
Radio Telefis Éireann (RTE)	IRL
Rai Radiotelevisione Italiana (RAI)	I
Retevisión	E
Rohde & Schwarz GmbH & Co KG (R&S)	D
Télédiffusion de France (TDF)	F
Tele Danmark AS	DK
Televés SA	E
Teracom Svensk Rundradion AB (Teracom)	S
Thomcast	F
European Broadcasting Union (EBU)	
Nederlandsche Omroep Zendermaatschappij (NOZEMA)	NL

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Consolidation: Engineering Management Perspectives

Monday, April 6, 1998
10:30 am - 12:00 pm

Chairperson:
Tom McGinley
WPGC-FM/WPGC-AM, Greenbelt, MD

Panelists:
Al Kenyon; Jacor Communications Inc., Covington, KY
John Bisset; Multiphase Consulting, Springfield, VA
Jim Smith; CLEAR CHANNEL NETWORKS, Tulsa, OK
Chris Alexander; Crawford Broadcasting, Irving, TX
Jeff Littlejohn; WUBE-AM/WUBE-FM, Cincinnati, OH
Milford Smith; Greater Media Inc, East Brunswick, NJ
W. Simmons; Adventure Technology Inc., Hilton Head Island, SC; Glynn Walden; CBS Radio, Linthicum, MD

CONSOLIDATION OF RADIO FACILITIES
A MULTI-SYSTEM APPROACH

W. Lee Simmons and David L. Simmons
Adventure Technology, Inc. and W. Lee Simmons & Assoc., Inc.
Hilton Head Island, South Carolina

ABSTRACT

This paper describes how we have designed and implemented seven radio stations so that they can operate under one roof. This has allowed us to cut personnel costs by seventy percent. By doing this it has led be better efficiency and made them more scaleable with new technology and operating systems without the complete destruction of the existing operations. This system will allow consolidated stations to operate more efficiently than single stand-alone entities.

INTRODUCTION

One of the major problems with radio consolidation from an engineering stand point is no one company makes a system that can handle all the components it takes to make it work.

Mission Statement Per Market
Seven radio stations operating six different formats, twenty-four shifts a day, seven days a week... all in one location.

Three sales offices in two states servicing seven stations with three Sales Managers and fourteen salespersons.

Four transmitter sites in two states designed so that any of the sites can be backed up by the main transmitter site for operations under emergency conditions, i.e.; hurricanes, power outages.

The last but one of the most important, is to cut the staff by seventy percent and make it sound better than before.

Sounds like Buck Rogers, doesn't it? A few years ago, what I just described was unthinkable. Today, not only is it possible it's happening.

To make radio consolidation work, we must have a system that is usable, scaleable and in its implementation does not destroy the existing systems that are producing revenue for the new conglomerate. Coming up with a method that manages all of the different operating systems, and in addition, gives management the tools that are needed to fully understand how the stations are operating singularly and combined is imperative.

Who Will Be Left?
Along with consolidation comes another big question. After this is over who will be around to service these new systems?
A problem we are going to face as an industry is, as consolidation progresses there are going to be fewer service companies available. To protect ourselves as much as possible we need to design systems that will operate on standard software packages. I have a much better feeling that IBM and Microsoft will be around in five or ten years to service my needs than some computer companies that are on the market today.

Given this mission as the starting point we re-designed and integrated all of our existing systems with this goal in mind.

Multiple Use Studios

The first place to start is in the studio. In a multiple station operation, you do not have the luxury of a separate production studio for each station. We started by designing a studio that can be the primary control point for a specific station. Then we added, through digital production units, the ability to voice track, production or back-up any of the other stations. This gives a smaller staff the ability to stay in one studio and work on all six formats. Each studio is equipped with a ten channel analog board, a digital editor, two CD players, one DAT unit, a telephone interface, two mikes, a touch screen, and a full three channel digital production unit.

The Front End

We are using the Scott Studios System for the music playback and on-air control of the radio stations. There were limitations. The primary limitation was its operating system- DOS. DOS was limiting the storage space to 2 GB per partition which in turn required a separate drive letter for each partition. If the operating system was networked together, the maximum capacity was 25, 2 GB partitions. DOS also has no native way of sharing files with other DOS machines. There are several "Peer" network operating systems available but none are robust enough to handle the workloads we intended to impose on the computer. So, the problem became how to integrate existing technology with a standard, scaleable product that could link all of our stations together, keep cost down, maintain on-air quality and give management the new tools it needed to run the whole thing.

Scott Studios had the front end parts, but we had to find a different way to make it work. The first thing was to find out what system would replace DOS. We found out that IBM's Warp Server would run the existing Scott Studio's DOS software very well. We converted all our existing DOS based hard drives over to IBM's High Performance File System which in turn removed the 2 GB limit. The file system allowed us to utilize the maximum possible disk space available from the hard drives and monitor errors that occur on the drives over time. Warp Server is incredibly robust and is more than capable of providing access to the computers from other locations. Other computers can send, receive, modify, or delete files on the on-air machine while it is playing over the air. Special parameters were used in the server to insure that audio-quality will never suffer under any condition. Moving gigabytes of data to and from the on-air machine does not interrupt the on-air quality or stability of the system. Hard Drive replacement is also painless. Once the malfunctioning hard-drive is replaced, replacing the data is one click away. On-Air operations continue normally. Let me pause right here and interject that you will have hard drive failures. Plan on it and budget accordingly. If someone says that they have a system that the hard drives don't fail, run do not walk to the nearest exit.

Scott also has a new Windows based Voice Tracker system. This requires a "Peer" to "Peer" system. The standard 10 Mb/s Ethernet is fine for an AM-FM combo; however, with seven stations voice tracking and doing production at the same time, the problem multiplies quickly. You must install a high speed network to have any chance of making this work. We tested them all, 100 based T, Fast Ethernet, and 100 VG-Anylan. We found that the 100 VG-Anylan would sustain almost 97% throughput under loaded network conditions. The other two slowed to 37%.

When you are moving large audio files for seven stations all at the same time this becomes very important.

Through bench testing we found that each station needed its own server to play back digital audio. This gives better data throughput as well as many other advantages. We upgraded the units to Pentium 166 MMX with 32 Megs of RAM. These servers, or "On-Air Machines" as mentioned before, could play as many as three APT-X audio sources on-air while uploading five hundred megabytes of data, downloading one gigabyte of data, recording a new song and down-loading a commercial. This is about 500 percent more than you are required to do under normal conditions, but we have learned that our production people have ways of pushing the limit.

Stand Alone System?

We found that it was better to have each station be an integrated stand alone system. This may be contradictory, but follow this. Each station has its own audio file server, and its own touch screen controller. This gives us the high speed "Peer" to "Peer" system that is required for the Voice Tracker system, plus the ability to work as a file server for the integration of the rest of the components; i.e., traffic and scheduling. This also removes the single point of failure that is inherent in this type of system.

Voice Tracker System

For those of you who have never used a Voice Tracker system, this is quite a piece of software. First, the program director puts codes in the logs where the voice track should air; i.e., between two songs, between a song and commercial, or any other combination. Once the "virtual announcer" is ready to voice track, the voice track system will find the codes in the log and indicate to the "virtual announcer" the location of the voice track. The approximate time is also given to insure the show sounds live. Once the announcer has selected the particular voice track he or she wants to record, the system begins to play the last 5 seconds of the first event between the voice-track. The announcer then decides when they want to overlap the "outgoing" song by recording the voice track. After recording the voice track, the announcer can select at any time when the next event should begin to play. The announcer continues the same process for the rest of the voice tracking events. This voice track is then embedded into the log for that day. The announcer can voice track an entire four hour shift in about twenty minutes.

With this system we can voice track twenty-four different shifts on six stations with only ten people; thus achieving two of the mission goals: staff reduction by seventy percent, and a better "live local sound" for each station.

Touch Screen Controllers

Each station is controlled by a Touch Screen. We modified each "SS" controller to operate on its on station's audio file server. Each unit was modified from a 486 to a Pentium 166. This unit is also attached to the high speed net. This gives the station its stand alone ability. This design allows each station to run on its own system and at the same time each file server is on a "Peer" to "Peer" or server to server high speed net. Under normal operations, the logs are done by the Traffic Department on the master file server, then exported to the file server for each station. There are always two copies of the logs for any station at any given time.

Business System

We determined that when it comes to the business system, it was best to have a system that stood alone, but integrated with all the latest software packages that management needed to make good business decisions. Here we designed a RAID 5 twelve gigabyte file server with redundant power supplies and "hot swappable drive bays". The unit was equipped with a twenty-four gig DAT tape drive for back-up. We also installed an on-line, hot spare hard drive. This means that if the controller or operating system detects a drive in the process of failing, it will start building the hot spare as the replacement for the failing drive. Once a replacement drive is acquired, it can be installed in the file server while the system is running, zero "down time" This is much better than the familiar phone call from the business office, "What does cannot access drive mean?".

In our case the sales and management offices are located in a separate building from the studios. We connected these offices to the system by running category five wire in an underground conduit. We installed a separate hub for the office complex. This new operating system allows a print server to operate over tcp/ip, a huge improvement over the ipx system. This also allows us to put laser, line, color and terminal printers in addition to scanners on-line to the entire enterprise instead of just one work station.

New Backup System

Hear is the crux of the new system. How can you have a standby system for the enterprise capable of providing redundancy for any point of failure that even a weekend part-timer can use?

We designed a seventy-five gigabyte RAID 5 file server with dual power supplies and hot swappable bays. This unit is on the first hub of the entire system. proprietary back-up software is constantly looking at the entire network for changes in data (audio or business). Business back-up transactions are set to one minute. We have a ten minute wait to back-up all commercials and promos. Each studio's voice tracker computer is equipped with playback software identical to the on-air machine. If a particular station has audio problems, the operator pots up the voice tracker on the console, clicks the back-up icon for the station and instantly the on-air audio is produced from the voice tracker computer. The voice tracker computer receives all data from the back-up server which has an exact replica of the original computer. The transition is so fast we have done it while music was airing with no skips in the audio. This will let us take out the on-air file server and perform maintenance, replace a hard drive or replace the entire unit. Once the original unit is repaired, the backup system will synchronize all data. After the data is synchronized, simply activate the on-air machine and return to normal...no audio or commercials lost. This solves about ninety-five percent of your problems, but we were striving for one hundred percent. So we came up with a way to not only switch over the digital audio, but the touch screen, as well. This involves the same process, but instead of the voice track computer acting as an audio on-air machine it runs in a native "voice tracker" mode and plays the correct log. This allows the engineer to remove, update, re-install, change format or do maintenance on the station without fear of the world coming to an end. Voice tracks can still be done for the backed up station from one of the other studios as well as commercials and promos. The backup server is RAID 5 and will alert you if it is having any hard drive or internal problems. We loaded large memory banks in both the server and RAID controller. We have tested the Back-Up Server in the worse possible scenario, all station computers simultaneously dying. Even under those conditions you are only utilizing 30% of the entire backup system. In addition to the backup system, DAT tapes are made and are stored in a different location. This is in case of fire or other major disaster.

Let me stop and inject this very important point. Put these very large computer systems on true non-interruptible power supplies. We run all computers on their own individual "u.p.s.'". We also float each studio on its own u.p.s.. Please, do this before you start. You have no idea what the power company does to you, but it is not pretty. Also, please consider adding a full-blown automatic change-over generator to the studio building. Our people fought us over this, thinking the u.p.s. would hold us over long enough for the power company to fix the problem. We finally won out and put in the full back-up generator system. In November 1997, the unthinkable happened. A painter working on the sub-station fell into a transformer. The power was off for almost six hours. The u.p.s.' are designed for a twenty-seven minute back-up. The u.p.s.' held everything for the thirteen seconds it took for generator to kick in. No air time was lost on any of the seven signals. Please give this some consideration when you start designing new studio complexes. What once was a minor inconvenience can become a catastrophe when multiplied by seven.

Integration of New Services

With the installation of six air/production studios, three production studios, two newsrooms, three traffic workstations, and with one secretary, one business manager, six PD's, three announcers and one chief engineer in one building, plus connecting sales, promotion and management in the next building; we needed to design real world tools to make it all work.

Production E-mail

One small problem in a single station, becomes a large one in a multi-station operation. That is weather and news. How do you get weather and news up-dates to each station very quickly without making copies and running all over the building handing them out every fifteen minutes? Well, with these new systems you feed you ASCII news feed into one of the warp newsroom terminals. With this system you can create any story you wish, add weather, and any other information. With the click of a icon you can send it to all studios, or any particular station all without moving from the newsroom. It silently windows on the digital production unit so the air personality can see it without scaring him to death. It does not cover up the touch screen or any operating system. Its a small feature, but it saves hours of time and a lot of yelling down the hall.

The Internet Connection

The biggest problem, from an operational point of view, is how to keep an internet connection fresh without tying up personnel and time. This new system lets us attach a Lotus Domino server to the high speed network. It can be programmed to update itself daily, or hourly. Changing links, updating features, adding weather and news from the e-mail system is all automatic. Plus it has a very heavy security firewall system built-in. It works as a proxy server to the outside world. Management has full control of who can get on the internet and even what sites they can visit.

FAX Services

We say services because this is starting to be a viable option to communications management. This will let a designated workstation, or any workstation send and receive faxes directly from a word processor to the outside world. The new development of faxing over the internet is being looked at, although it will not become completely viable until the world has a permanent I.P. internet connection.

Notes Database System

With staff dispersed in two separate buildings, comes the problem of getting messages and documents instantly. Lotus Notes allows the receptionist to get messages to sales and management people very quickly. We also have put in a remote access terminal for our two remote offices. We simply have the offices dial in three times a day get the notes update. We are in the process of setting up Notes messaging that will interface to our local paging terminal. This way we insure the sales person will get the message.

If you wish to send a priority message the system will call each office and deliver the mail if it is urgent enough. Documents can also be sent through Notes.

Sales Department

Through the Notes system we have set up sales templates for the sales staff. This can include a color cover page with scanning client logo capabilities. We are also in the process of implementing a digital still camera into sales presentations for more visual impact.

All they have to do is call up the template for a particular type of proposal or promotion and fill in the blanks. The rates for this system can be altered by the sales manager, or updated by the traffic system. This is especially good if you are on the grid system with the rates being affected by inventory. This will produce a proposal that can be read by the sales manager at her terminal, make changes, send it over to the promotion department if needed, and back to the sales person for them to print and deliver to the client.

Traffic System

The original station in this group started with CBSI. We have found this system scaleable and easy to use. As we have added stations, CBSI has proven it can handle the work load, but as stated in the mission statement the system needs to be more flexible. Luckily CBSI provides a DB-4 database output. This can be integrated through the Notes system to provide management any type of information it wants or needs. For example, it is a simple matter to ask the Notes database system to find out how many car dealers spent over one thousand dollars per month on truck advertising during the first quarter of a particular year. This can also be correlated with the RAB database, Arbitron and any other reference you wish. At the same time this data can be loaded into DB-2 or any SQL database engines that you desire. Corporate Headquarters can connect remotely to the management computers extracting any report that they need, even down to the second a order is entered, if need be. Financial data can be customized and up-loaded at any time. This gives management the control it needs to make informed decisions.

Notes Database

The Notes Database can show department heads when and where problems are starting. Through the Notes system an employee attendance database can detect a problem before it starts. With remote access, a system, or group personnel director can watch for absenteeism in a remote location that the local manager may be missing. The group engineer can monitor individual equipment parts orders to determine whether it should be repaired or replaced. You can even set it to watch transmitters through data entry, or have the units auto-log their reading directly to the data base engine. Departments can put their budgets into the system and be alerted if they are exceeding any or all parts of it The system can be programmed to do just about anything. Remember, this is a standard scaleable product that will be around after you are gone.

CONCLUSION

The objective has been to save the existing system by combining with a industry standard operating system that will accommodate the consolidation of several radio stations. We believe we are doing just that. By using IBM and Microsoft software products, we believe we are enabling Management and all personnel to work more efficiently and at the same time are producing a quality product. With technology growing at such a rapid rate, we believe that we are in a favorable position to expand into the next millennium and utilize the future advancements that are ahead.

Broadcast Towers: Managing Your Vertical Real Estate

Monday, April 6, 1998
1:00 pm - 5:00 pm

Chairperson:
Robert Hess
WBZ -TV, Boston, MA

***Implementation of Digital Television**
Ronald Gibbs
Lodestar Towers, Inc.
Tequesta, FL

***A New Broadband Waveguide Transmission Line Structure For UHF High Power Transmission**
Derek Small and Spencer Smith
Passive Power Products
Gray, ME

Evaluating Existing TV Towers for DTV Antenna Installation
Madison Batt, P.E. and Tim Wolden, P.E.
Tower Engineering Consultants
Seattle, WA

Tower and Antenna - Holistic Solutions for Digital TV
Anthony Magris
Radio Frequency Systems Pty Limited
Lonsdale, Australia

***Waveguide or Coax for 1 Megawatt DTV**
James Stenberg
Dielectric Communications
Raymond, ME

Methods and Costs of Installing Initial and Interim DTV Transmission Facilities on Existing Towers
Sidney E. Shumate
BIA Media, Inc.
Chantilly, VA

***Radical Structural Solutions for New and Existing Towers**
Henry McGinnis
Landmark Tower Corp
Ft. Worth, TX

***Tower Management Panel** (Joint session with Broadcasters Law and Regulation Conference)
Moderator:
Barry Umansky, NAB, Washington, DC
Panelists:
David Bronston, Wolf Block Schorr & Solis-Cohen, New York, NY
Tom Hutton, Holland & Knight, Washington, DC
Charlie Morgan, Susquehanna Radio Corporation, York, PA
Jon Sinton, Sinton, Barnes & Associates, Atlanta, GA
Marcus Trathen, Brooks Pierce McLendon Humphrey & Leonard, Raleigh, NC
Jeff Horton, Royal Insurance, Charlotte, NC

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*Papers not available at the time of publication

EVALUATING EXISTING TV TOWERS FOR DTV ANTENNA INSTALLATION

Madison J. Batt, P.E., and Timothy A. Wolden, P.E.
Tower Engineering Consultants, Inc.
11065 Fifth Avenue NE, Suite A, Seattle, WA 98125

ABSTRACT

With the advent and implementation of DTV, broadcasters will need to consider whether their existing broadcast tower is capable of supporting the new DTV antenna in conjunction with the existing NTSC antenna and all other existing antennas. In many cases, existing guyed and self-supporting towers are capable of supporting these new antennas.

Broadcast towers should be evaluated by qualified tower specialists, preferably engineers experienced with tower analysis and design. The engineers should be directly involved in a field assessment of the tower, the computer analysis, and implementation of any tower upgrade work. The failure of several of the country's tallest towers in the last few years highlights the need for professional engineering involvement in the tower industry.

Information and checklists are included in this paper to aid the chief engineer in evaluating proposals and ultimately selecting an engineer.

IS EVALUATION NECESSARY?

The new DTV antennas currently under development are either stacked top-mounted antennas able to simulcast both the NTSC and DTV signals, side-mounted panel antennas, or side-mounted mast antennas that are used in conjunction with the existing TV antenna. These antennas, along with associated transmission lines and new STL microwave antennas, can significantly increase the wind loading on the tower, possibly causing member overstresses. Towers that will support the new DTV antennas should be evaluated for these antennas. Evaluation also includes assessment of the electronic characteristics of the new antenna in conjunction with its location on the tower. The contents of this paper will focus primarily on the structural aspects of DTV implementation.

In most cases, DTV antennas and large-diameter transmission lines will add significant loading to towers that may already be replete with antennas. Many of the towers in use today have been in place for a long time and may not have been designed for their current antenna configuration. Furthermore, most towers installed prior to the 1980s were designed to outdated design standards and may not meet the more stringent requirements of the current design standards.

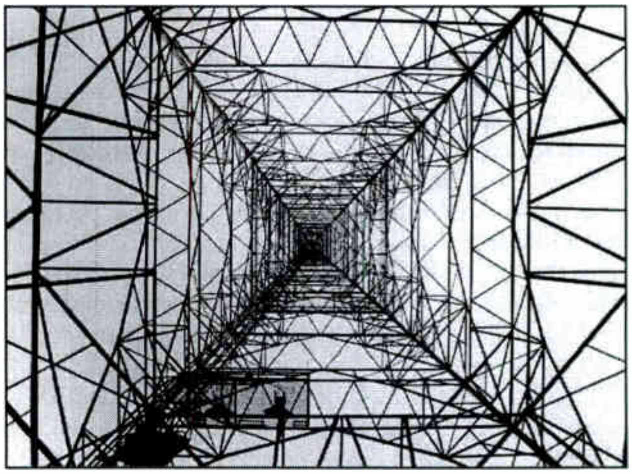

Figure 1. Looking up the 554' WTVH tower in Syracuse, New York

Wind modeling has evolved over the years to incorporate advances in modern wind engineering. The current design standard, TIA/EIA 222-F, "Structural Standards for Steel Antenna Towers and Antenna Supporting Structures," incorporates the latest technology in wind loading formulas. Computer programs used to evaluate the tower should apply the current standard in the analysis.

There may also be requirements by the broadcaster's insurance provider for updating the tower to the current standards. In addition, most insurance companies require an assessment of the tower whenever significant changes or alterations are made to it.

SELECTING AN ENGINEER

Perhaps the most important step in the evaluation of a tower for DTV antennas is to retain a firm that has experience with broadcast antenna characteristics, tower inspection and assessment, computer analysis, and tower design. A number of broadcasters across the nation have begun to prepare for the addition of DTV antennas to their towers and have had to locate tower engineers for this task.

Many engineering firms are able to analyze structures for wind loading; however, only a few are actually qualified to render an analysis utilizing state-of-the-art software intended for analyzing towers. The software employed should be capable of utilizing non-linear analysis techniques on guyed towers applying the current design standards. Even fewer firms have the ability to perform a field assessment with an engineer climbing the tower, observing (inspecting) the tower, and evaluating all appurtenances.

It is recommended that the engineer providing the structural analysis also conduct the field assessment; however, this is not imperative. Field observation, document review, interviews with the station engineers, and computer analyses are best coordinated if one person or team is involved throughout the entire process. On towers where there are no existing documents, it is even more important that the structural engineers providing the analysis also climb the tower to measure all member sizes and document and understand the framing system.

There are several resources available for locating a qualified engineering firm specializing in the field of tower engineering. These include tower manufacturers, the Society of Broadcast Engineers, various broadcast publications, and the Internet. Contacting various local stations or some of the larger broadcasting corporations that have had an analysis performed may result in a referral to a qualified firm.

FIELD ASSESSMENT

A field assessment of the tower is essential to guarantee that the computer analysis properly represents the existing condition of the tower. The assessment includes reviewing any documents kept at the station, interviewing the station engineers to learn the tower history, climbing the tower to visually assess the conditions, documenting the antenna configuration and orientation, and verifying the member sizes and dimensions. When documentation on member sizes is not available, the field measurements become especially important. Any information about the manufacturer, date of construction, and foundation should also be obtained.

As engineers, we have been asked to perform analyses without visiting the site. This limits our understanding of the structural integrity of the tower. When engineers do not or cannot conduct a field investigation of the tower, they must limit their liability. When an analysis is performed by an office that does not visit the tower site, inaccurate assumptions can be made that are only answerable by a field assessment and observation of the tower framing. Assumptions that include inaccurate antenna configuration data, wrong member sizes and dimensions, and ignorance of member conditions can lead to incorrect recommendations. On towers with members that are corroding the potential for reduced capacity is real.

When the engineers conducting the field visit also perform the analysis, little information is left unknown. Any unsatisfactory conditions noted by the field observation can be implemented into the final recommendations, allowing the owner to achieve a complete solution to integrating the new DTV antenna system into the whole tower analysis. This total package should also be acceptable to insurance providers because of the thorough approach.

During the field assessment, photographs should be taken to document both typical conditions and any areas that may require maintenance or further observation. The written and photographic information can then become the basis for a detailed drawing of the tower, an accurate computer analysis, and a comprehensive narrative discussion of the tower.

The field inspection by the engineer should include, at a minimum, the following:

a. The tower is climbed by a team of two people to visually assess the condition of the framing, to measure or verify the member sizes, and to document the antennas and transmission lines on the tower.

b. For guyed towers, the verticality of the towers is determined with a transit, and the tensions in the guy wires are checked. The tensions are checked by either measuring the guy sags and comparing them to the design sag requirements, or having a tower contractor measure the tensions (a tension measurement report within the year is generally acceptable).

Guy tension measurements on taller guyed towers are more difficult and costly. Therefore, the sag measurements can be a cost-effective way to approximate guy tensions.

COMPUTER ANALYSIS RESULTS

In evaluating TV towers for DTV antennas and addition of other antennas, we have found that there are cost-effective methods for upgrading towers. Many of these methods are proven and have been used by the tower manufacturers to upgrade towers from the original design.

Every TV tower in service today that is scheduled for broadcast of DTV programming will need to be evaluated.

For towers that have been evaluated, the results show that many of these towers can be upgraded. However, some analyses will show that certain towers cannot be economically upgraded. Station owners and managers need to be alerted to the need for a replacement tower so they can prepare their operating budgets accordingly. In some cases, stations may want to get a second opinion.

Older tall self-supporting towers (30+ years old) comprise the group with the greatest number of towers requiring replacement. Most other towers designed to more current standards fall into the retrofit categories.

Self-Supporting Towers

Analyses of self-supporting TV towers with new top-mounted stacked antennas show that these towers have overstressed members. The extent of overstress varies. Most of these towers have top sections that are narrowed for maximizing the omni-directional characteristics of antennas. These reduced sections are generally undersized for the weight of new stacked or side-mounted panel-type antennas. Side-mounted slotted antennas in some cases do not overstress the tower members due to better aerodynamic characteristics.

Adding stacked antennas to the top of a tower increases load throughout the tower, with the largest force increase at the top of the tower. When local FAA administrators do not allow the increased height for the replacement antennas, one alternative that has been investigated is the removal of equivalent sections of the tower so that the top elevation stays the same. This may keep the bending forces at a more acceptable level. Other ways to upgrade the tower involve increasing the capacity of the legs and bracing. See the details under 'Upgrades' below.

The foundations for self-supporting towers should also be reviewed. The increased overturning and shear forces generated when taller and heavier antennas are used can overload the original foundation that may also be inadequate for the current design standards.

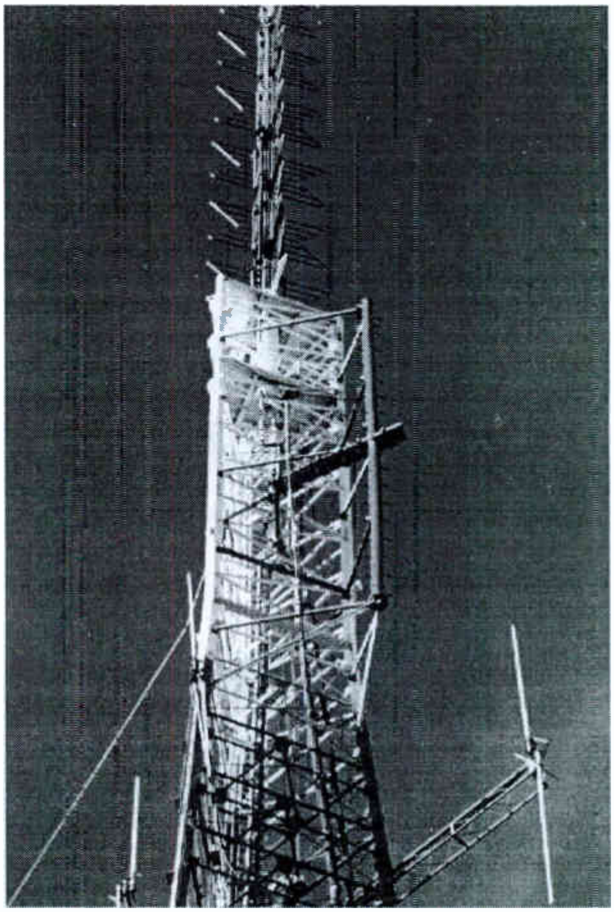

Figure 2. Side-mounted DTV antenna, KOMO-TV Tower

Guyed Towers

Some guyed towers in use today have been in service for up to 50 years. The difference between these older guyed towers and self-supporting towers of the same age is that the older guyed towers are typically more capable of supporting additional antennas with minimal upgrade required. Replacement of the guy wires for these towers may be an issue if the original guys are still in place.

Newer guyed towers, especially towers built during the previous television expansion era in the 1960s, may not be as fortunate. Depending on their condition, these towers may not be capable of supporting new DTV antennas without major upgrades.

The failure rate of guyed towers should also be considered. These towers fail at an alarming rate compared to other structures. It is estimated that 1 in 10 guyed towers fail before their useful life is over. The recent failure of four of the tallest guyed towers in the last year confirms this observation.

Figure 3. WLBT tower collapse, October 1997

In general, information on guyed tower failures is sketchy. No government agency or group in this country keeps track of these failures. However, in talking with chief engineers around the country, we have learned that many stations have experienced the collapse of a guyed tower. The causes of these failures fall into two major categories: human error and weather.

Weather factors may be the single largest cause of guyed tower failures. The primary culprit here is ice. Many of these failures occur when heavy ice storms hit a region. Earlier this year, several towers collapsed due to the severe ice storms in New England and eastern Canada.

Very few towers have been designed for the 100-year ice storms that seem to be occurring every 5 to 10 years somewhere in this country. Reducing these tower failures is expensive due to the member sizes required in building guyed towers that are capable of withstanding large ice loads. In ice-prone areas, however, larger ice loads are recommended.

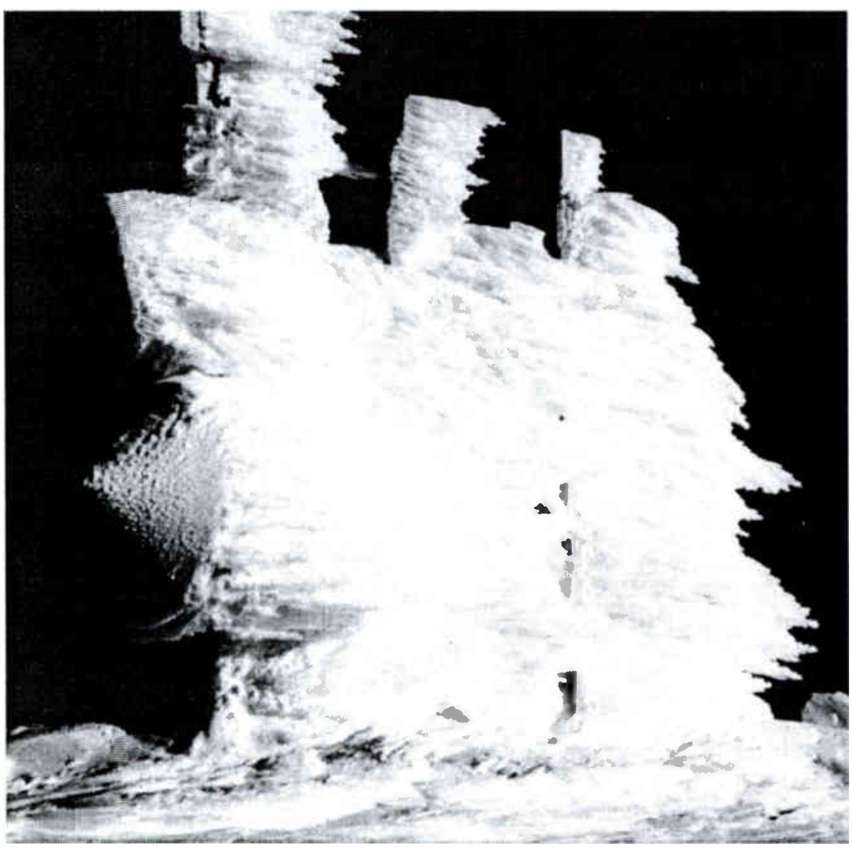

Figure 4. Three ice-covered towers in Alaska

Human-error types of failures come in several forms. Some of these failures occur when tower members such as bracing, bolts, or guys are being replaced. In these cases, safety frames or temporary guys were not in use by the workers. Removal of tower members without a safety frame has led to many tower failures, resulting in the deaths of tower workers. Another major documented type of human-error failure occurred while top-mounted antennas were being changed. Replacement of these antennas is difficult. Proper experience, coordination, and appropriate equipment play a major role in successfully completing this work.

The human-error factor, with regard to tower failure, is preventable. Education of tower workers, safety meetings, involvement of an experienced tower engineer, and creating a safe working environment will reduce guyed tower failures.

Upgrades

Upgrades for guyed towers include leg strengthening, bracing additions or replacement, guy wire replacement, or addition of new guy wires and anchors. The guy wires can be modified in several ways. They can be increased in size or the spacing of the guys can be changed. Also, new anchors installed further out from the tower can increasthe capacity of the same size wire without increasing the downward load on the legs. Some typical tower upgrade details are included below.

Legs. Overstressed legs are often the most expensive and most difficult part of upgrading a tower. Legs are generally in compression due to tower self weight, antennas, coax cables, ladder, wind load, and ice loading. Based on engineering principles and the buckled-column formulas, the capacity of a column decreases nonlinearly for longer unbraced lengths. Thus, reducing the unbraced length to half will more than double the compression capacity on members with long unbraced lengths. For example, in Figure 5, a 4-inch round solid rod leg with 20-foot long unbraced length has an allowable axial load of 32 kips (32,000 lbs.). For the same member with an unbraced length reduced to 10 feet, the capacity increases to 129 kips, and at 5 feet the capacity increases to 219 kips.

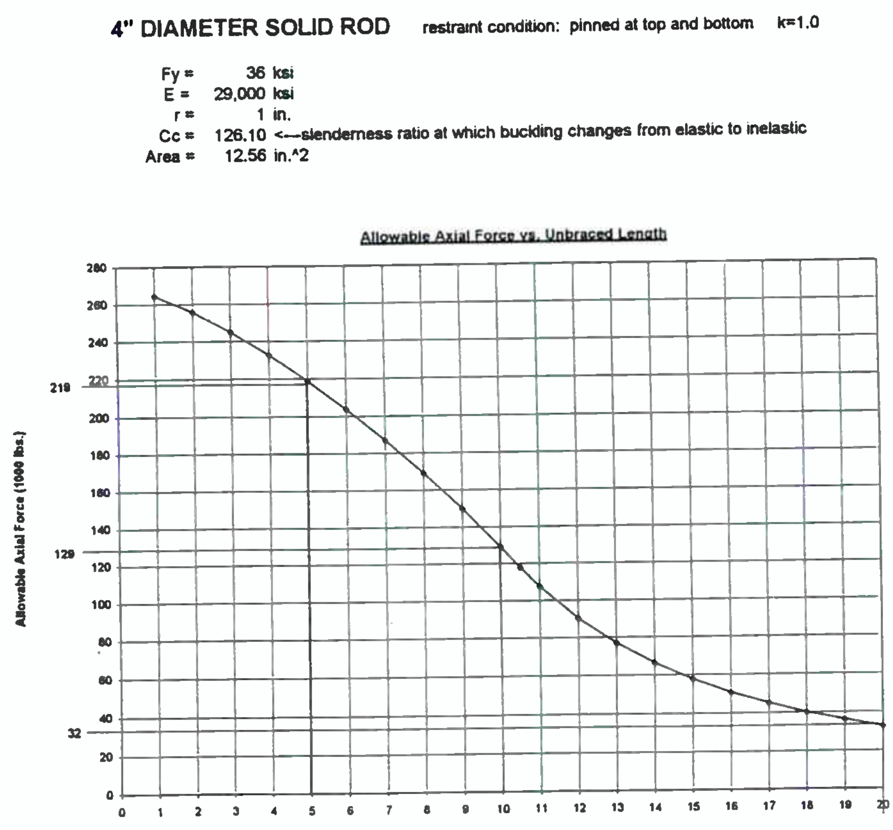

Figure 5. Allowable Axial Force

The best ways to upgrade the capacity of solid round tower legs are:

1. Reduce the unbraced length of the legs by installing additional horizontal bracing (see Figures 6 and 7), or installing additional diagonal bracing in Z braced sections to create an X bracing system.

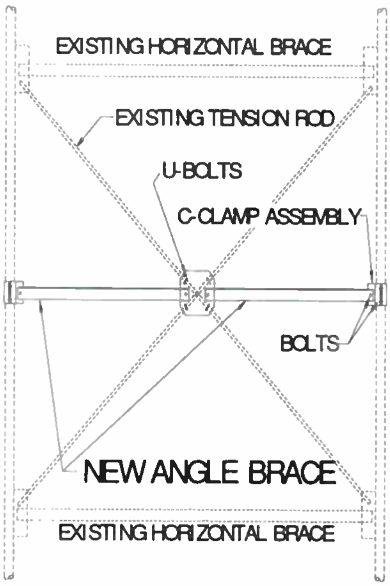

Figure 6. Added Leg Bracing Elevation

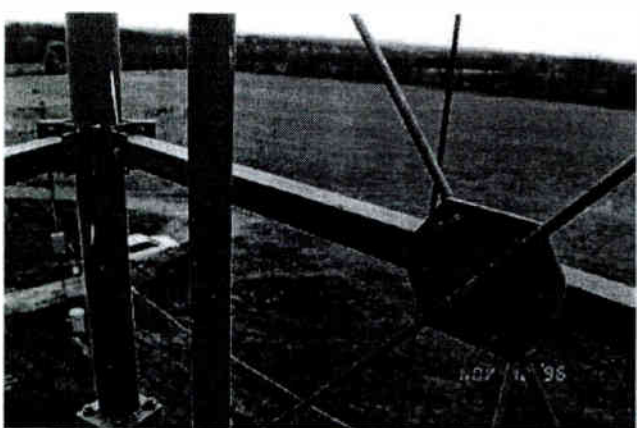

Figure 7. KMIZ-TV tower, Columbia, Missouri, with added bracing

2. Weld half-round sections of pipe to the legs to increase their capacity. This is usually the most expensive method for upgrading legs (see Figure 8).

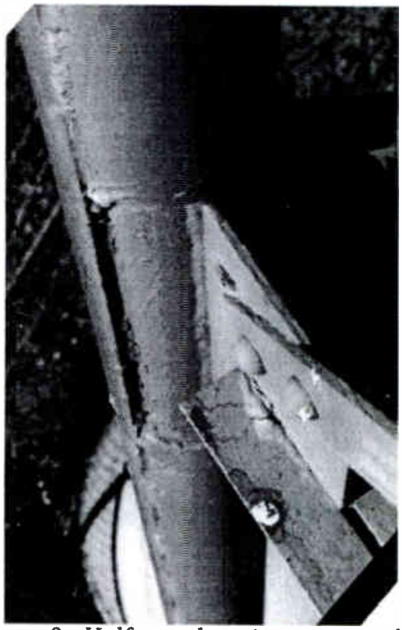

Figure 8. Half-round section on tower leg

For pipe leg towers, the same upgrades are suggested as above. In addition, pipe legs can be filled with high-strength concrete or grout. This adds capacity and stiffness to the legs and can dramatically add capacity to the tower. This is especially true for large-diameter legs.

For towers with angle member legs or unusual bent-plate legs, upgrades by welding or bolting plates, angles, or channels can increase capacity. This can be very expensive work. Reducing the unbraced lengths with added horizontal bracing is usually a more cost-effective option.

Bracing. Diagonal bracing that is overstressed will need to be replaced or changed from tension-only rod or undersized angle bracing to heavier angular bracing. Horizontal bracing that is overstressed or too long can be strengthened with the addition of internal bracing.

Antenna Radomes. Another option on most towers is to reduce the wind loading contribution of other antennas mounted on the tower. TV towers in most cities have large microwave antennas, many of which do not have aerodynamic radomes. By adding aerodynamic radomes to these antennas, the wind loading contribution of the antennas can be reduced by as much as 45 percent.

Foundations. Foundations should also be investigated and upgrades should be implemented on those that are overloaded. This will usually consist of adding weight or bearing area to the foundation system.

Cables. Bundling cables behind a leg or moving cables so that other lines shield them will reduce the wind load on the tower. Designing antenna systems to "share" or multiplex a transmission line should also be investigated.

CONCLUSIONS

The results of analyses on towers with new DTV antennas suggest that many towers will require structural upgrades. With the proper design of new members, the capacity of a tower can be increased significantly.

Most self-supporting TV towers will require upgrading. The majority of these towers can be brought up to requirements with minor or major retrofitting. For some towers it will not be cost effective to upgrade. This is generally due to the age and the condition of the tower. Guyed towers will generally be capable of supporting the new DTV antennas. Most guyed towers will require only minor, if any, retrofitting.

It is prudent for TV station owners to have their broadcast towers evaluated for the new DTV antennas. Installation of these antennas involves replacement of the existing TV antenna with a new stacked antenna or addition of a side-mounted antenna system. The cost to do this analysis will be well worth the investment if the tower is to remain in service.

Stations that require such an analysis should retain the services of an independent professional engineering firm experienced in inspecting and evaluating towers. These firms should demonstrate familiarity with the design standards and have state-of-the-art analysis software.

Many of the larger tower manufacturers are capable of performing the analysis but may be too busy or may not provide a field inspection service. The field inspection has proven to be an important component in the overall evaluation of the tower. It is beneficial if the engineer that performs or supervises the analysis is also present during the field assessment. Fewer assumptions are required if the whole tower evaluation is coordinated properly. This should result in the most efficient solution for adding DTV antennas.

The checklists on the following page may be helpful to chief engineers.

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CHIEF ENGINEER'S INTERVIEW CHECKLIST

An Aid for Selecting a Consulting Engineer to Evaluate the Tower

- ❑ Does the firm have experience evaluating towers?
- ❑ Will the firm perform a field visit and climb the tower to evaluate it?
- ❑ Has the firm designed towers or upgrades to towers?
- ❑ Is the firm familiar with the EIA/TIA and ACSE wind standards?
- ❑ What computer software is used for the analysis?
- ❑ Is the analysis for guyed towers a non-linear analysis?
- ❑ Does the software incorporate the latest EIA/TIA - 222 Standard?
- ❑ Does the firm have experience with broadcast antennas, in particular analysis for wind loading?

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CHIEF ENGINEER'S "TO DO" LIST

Assemble and provide as much information as possible to ensure a complete understanding of the tower and make copies for the engineer. This information should include:

- ❑ Tower drawings.
- ❑ Inspection reports.
- ❑ Previous analyses.
- ❑ Upgrade or modification reports and drawings.
- ❑ Guy tension measurement reports (guyed towers). Have tensions measured if they haven't been measured within the last two years (guyed towers).
- ❑ History of tower, when it was built, and who manufactured it.

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CHIEF ENGINEER'S CHECKLIST FOR EVALUATING PROPOSALS

A list of tasks that should be addressed in the Tower Evaluation Proposal from the professional tower engineer or company

- ❑ Observe the entire tower by climbing it to determine condition of members and connections.
- ❑ Determine the tower orientation with respect to north.
- ❑ Verify and/or measure the face width of the tower, member sizes, spacing, bolt sizes, visible foundation pad, and anchor bolt sizes.
- ❑ Obtain all guy sizes, connection elevations, and anchor locations (x, y, z).
- ❑ Dig out anchor rods to check for corrosion, where practical
- ❑ Record the guy hardware types, condition, and sufficiency for loads.
- ❑ Survey the tower for verticality.
- ❑ Measure guy intercepts and review tension measurement reports.
- ❑ Document location, orientation, type, and size of all antennas, lighting fixtures, ladders, and other appurtenances.
- ❑ Document sizes and location of all coax cables, waveguides, and transmission lines.
- ❑ Observe the condition and connections of all antennas, mounts, and appurtenances.
- ❑ Review any design documents, drawings, maintenance reports, and previous inspections.
- ❑ Interview any station personnel to obtain historical data on maintenance performed (routine and emergency) and any upgrades implemented.

TOWER AND ANTENNA - HOLISTIC SOLUTIONS FOR DIGITAL TV

Anthony Magris and Robin Blair
Radio Frequency Systems Ltd,
Melbourne, Australia.

ABSTRACT

This paper discusses the options available to existing NTSC UHF transmitter operators who must upgrade their tower and antenna systems to carry their newly allocated DTV channel. Amongst the options considered is that of upgrading the antenna to a multi-channel panel array. The paper reports on the outcomes of design studies in typical situations, comparing this solution to the more traditional approaches of adopting side mounted or candelabra assemblies of low wind load slot antennas. In every case it is shown that the panel array gives a significant advantage in terms of tower loading, promising simpler and cheaper tower upgrading costs. Taken together with their known electrical advantages, this result suggests that panel arrays may well be the system of choice for most situations.

INTRODUCTION

As the change over from analog to digital TV gathers momentum, the operators of existing TV transmitters must find some way of transmitting the new digital service in parallel with their NTSC service until the latter is phased out. Many may look to what seems the simplest solution; that of placing a side-mounted slot antenna on a side of their existing tower. The solution may seem simple, but it brings with it many deficiencies. In particular, the radiation pattern is unpredictable and may fall far short of replicating the NTSC coverage.

Being aware of this, some operators are considering using two slots in a candelabra arrangement, but this also has disadvantages. The candelabra support and the second feeder cable introduce a significant wind load which may well require an extensive tower upgrade. An alternative to both these traditional approaches is to consider using a multi-channel broad-band panel antenna.

Modern broad band panel arrays have quite low wind loads and can readily accommodate several services. They require only one feed cable or relatively small feeders in a split system, which may well be critical for tower loading in many situations. However, there is a perception that the size of a panel array, by itself, leads to greater wind load than do the other solutions. This paper demonstrates that this is generally not true.

The paper reports on a design study carried out on typical situations with each of the possible antenna configurations. It is seen that candelabra or side-mounted slot antennas do not automatically lead to a simpler or cheaper structural solution as is often supposed. In most instances the panel array is clearly the best choice from both structural and electrical points of view.

ANTENNA CONFIGURATIONS

Operators seeking to add a DTV capability to an existing NTSC transmitting facility have a limited number of choices. In short, these are:

(a) Replace the existing tower and antenna.

(b) Use side mounted slots below the existing antenna

(c) Replace the existing antenna with a candelabra array

(d) Replace the existing antenna with a broadband panel array

The first will generally be impractical for many reasons including environmental objections. The second is simple to implement, but suffers from the requirement for additional feeders, unpredictable pattern degradation, and with two or more operators, disputes over who should occupy the highest position on the tower. The use of candelabra systems at least partly overcomes these last two objections but introduces even more wind load in the form of the candelabra platform. All of this makes the use of the panel array attractive, particularly when it can be shown to have an overall advantage in tower loading, which is the object of this paper.

DESIGN PARAMETERS

The parameters chosen for the design studies in this paper are intended to reflect those that will apply to most of the current UHF NTSC operators upgrading to carry their new DTV service. The NTSC service is considered to be operating with a peak ERP of something like 1 to 2 MW, and this would be preserved. It is assumed that the existing and new antenna systems will be chosen to have a high intrinsic gain. With very high gains the narrowing of the vertical radiation pattern and the consequent loss of near-in service becomes a controlling factor, so that the maximum peak gain employed rarely exceeds 17dB. This corresponds to an antenna aperture of around 50 to 65 feet, and these values were used in the design studies.

The tower height is taken to be around 1000 feet and located in open country. Calculations on wind loading factors were based on TIA/EIA 222 Rev. F[1,2], with the load figures quoted herein applying to a wind speed of 100MPH.

ANTENNA WIND LOAD

Based on the literature from well known suppliers, a number of slot antennas, for both top and side mounting, were analysed for their "bare" wind loading and compared to that applying to large panel arrays within the gain and power handling ranges mentioned above. Not unexpectedly, the wind loads of the slots were significantly lower than that of the panel array, with the ratios covering the approximate range of 0.22 to 0.28. On the face of it, this would seem to give the slots a real advantage, but as we shall see, this is largely illusory when real situations are considered.

CASE1 - SOLUS OPERATOR

Consider the case of a single NTSC operator who occupies his own site and desires to add capacity for one DTV channel. If he elects not to use a panel array, then his choices are either a candelabra array with two top mounted slots or a side mounted slot below his existing antenna. The former may well be preferred because of the superior radiation pattern performance.

For this candelabra array, the wind load of the two slot antennas and the candelabra support alone, is found to be approximately twice that of the panel array. Hence, even without considering the extra loading down the tower of the additional feed cable, the advantage is clearly with the panel array.

Coming to the sidemounted slot, we find the combined wind load of the new and existing antennas to be about half that of the panel array. This seems a clear advantage until we consider the wind loading of the additional feed cable, which will surely be of large diameter to keep the signal attenuation low.

A generalised analysis of the effect of this cable over the whole height of the tower would be difficult to compare quantitatively with the effects of the antenna loads. However, consider this :- the length of $6^1/_8$ inch cable which would make equal the wind loads of the two systems is only 250 feet; about only one quarter of that actually required. Clearly, the effect of this cable cannot be neglected.

In this last analysis of course, the operator could choose to use a side-mounted slot of much lower gain, and accept the consequences of pattern degradation and higher transmitter power costs. It is the classic dilemma of weighing cost against quality which many, unfortunately, are soon to encounter.

CASE 2 - MULTIPLE OPERATORS

Consider the case of two or more NTSC operators sharing the one site and tower. Almost certainly they will be using a candelabra antenna array, or it may happen that a "secondary" LP service could occupy a side-mounted slot.

As the discussion above implies, the combined wind load of the existing antennas and candelabra platform will probably already exceed that of a panel array of equivalent performance, and this will be made much worse by the need to effectively double the number of supported antennas.

The operators may choose to use side-mounted slots, assuming they can resolve the argument of "Who goes on top?" Then, however, the tower loading imposed by the necessary number of new feeders becomes even more significant than that described above.

It seems obvious that in most cases the requirement would be better served by a single high powered panel array. This point is further emphasised if we look further at the feeder requirements in such a case.

A panel array can be fed with a single feeder, although the more normal arrangement is to use twin feeders to secure redundancy, and sometimes, to achieve a higher power handling capacity. Thus, in the case under discussion, even twin feeders will be equal in number to or fewer than those existing for the NTSC services alone. They will certainly be fewer than those required for a joint NTSC and DTV capability. Hence, this would also seem to be a major advantage in favour of the panel array.

Some panel arrays have been built with four $6^1/_8$ inch feeders, carrying typically five or six services with ERP's between one and two Megawatts, or a host of lower power services. Thus, in most practical circumstances, the use of a panel array implies that the number of feeders required will not exceed four, and will more generally be one or two, almost regardless of the number of services being carried. This cannot be said of solutions involving a multiple of single channel antennas.

ANTENNA WEIGHT

A four sided 12 level panel array weighs approximately 3.8 tons. A survey of suppliers' literature suggests that the weights of self supporting slot arrays for top mounting tend to range from 70% to 100% of this value. Hence, it is immediately evident that, when compared to candelabra arrays of equivalent performance, the weight advantage will always lie with the broad band panel array.

Side mounted slots generally tend to weigh about one ton. Hence, if we compare a system of one top mounted and one side mounted slot with a broad band panel array, there is no clear weight advantage in favour of either. In considering, however, that any additional cable required for the side mounted antenna weighs one ton for each 250 feet, it must be said that the advantage will again favour the panel array in almost all circumstances.

PANEL ARRAY CAPABILITIES

It is true that with current technology, panel arrays cannot be built with the same power handling capabilities as those single channel slots found at the upper end of the power range and normally fed through waveguide. Their capabilities are still impressive, and as mentioned above many in operation carry several services with megawatt ERP's. The available bandwidth also ranges over virtually the whole UHF spectrum in the one antenna. As an illustration, we quote below from a specification for one of these antennas:

Antenna Specification:
12 bays of 4 panels
Panel rating 5kW
Antenna rating 240kW (average power)
Feeders: 4 X $6^1/_8$inch
Channelling:
NTSC Ch 54 ERP = 2.5MW
DTV Ch 42 ERP = 1.0MW
DTV Ch 51 ERP = 62kW
DTV Ch 31 ERP = 69kW
Peak Gain 17.1dB

CONCLUSIONS

In looking at tower and antenna upgrades, one must be aware of how the choice of the antenna effects the total cost, the practicality and the ultimate overall quality of service. That is what the phrase " Holistic Solutions" in the title of the paper implies.

The paper has shown that, viewed in an overall environment of multiple high powered services, broad band panel arrays have many advantages over other competing arrangements. Their electrical advantages are well known, and it can now be seen that they offer also the potential of significantly easing the problem of tower upgrading. They must be serious contenders for most applications if the outcome is to be truly holistic.

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Methods and Costs of Installing Initial and Interim DTV Transmission Facilities on Existing Towers

Sidney E. Shumate
BIA Media, Inc.
Chantilly, VA

ABSTRACT

This paper will address the possible methods, design considerations, and calculated risks of refitting lightweight, low windload, relatively temporary antennas and transmission lines onto existing, already well-loaded transmission towers. Budgetary costs for each approach will be shown. Two UHF power levels will be considered; 100 kilowatt ERP for existing UHF broadcasters, and 1 megawatt ERP for existing VHF broadcasters. Design considerations will include methods for minimizing windload by minimizing the size of, and positioning of, transmission lines on the tower. Also discussed will be the relative merits and operating costs of using larger transmitters to compensate for the high losses of minimum-sized transmission line.

INTRODUCTION

Many stations that are preparing to build DTV transmission facilities should consider using initial or interim DTV installations utilizing minimum-cost, relatively temporary designs. These stations, and their reasons to consider these options, include:

1. Stations planning on temporary installations:
a. The station has a temporary DTV assignment at Ch. 60 or above.
b. The station will want to relocate it's DTV signal to it's high VHF, or more attractive present UHF, NTSC channel when NTSC broadcasting ceases.
c. The station plans to apply to change channel or power level after NTSC broadcasting ceases.
d. The station wants to delay refitting for a permanent installation until the old NTSC main antenna can be removed to make way for the permanent DTV antenna.
e. The station has a DTV assignment above Ch. 46 and does not want to commit to any more expense than necessary to implement DTV until the issue of possible channel reclamation above Ch. 46 is answered by the Federal Communications Commission.

2. Stations with a need for speed:
a. The station wants the promotional advantage of being the first to broadcast DTV in it's market.
b. The station is facing years of cutting red tape to obtain a new, permanent site suitable for a new tower, and the permission to build the new tower on that site.
c. The stations' competitor just signed on it's DTV transmitter; and the owner (or Group Manager) is on the phone wanting to know what the station is going to do about it.

3. Stations facing a financial squeeze:
a. Suitable land on which to build a new tower within 3 miles of the present site is prohibitively expensive.
b. The station simply cannot afford to initiate DTV transmission if it requires a new tower.

DESIGNING THE INSTALLATION

The first example will be for a UHF DTV assignment for an existing UHF NTSC station; we will design for a DTV Effective Radiated Power of 100,000 watts. For these examples, we will start the design with the antenna and work backwards thru the signal chain. The most important factors to consider on a DTV antenna are:

1. the vertical gain component of the antenna's

horizontal polarization; i.e., the gain of the antenna in a omnidirectional configuration. The present industry wisdom is to design for a maximum vertical gain component of 20 (13.1 dBd) at UHF. This usually translates into an slotted antenna design using up to 18 vertical slots.

The reason for this is to minimize the effects of what the late Richard Bogner of Bogner Broadcast Equipment Corp. (now a part of RFS Broadcast) referred to as "beam steering", or what Kerry Cozad of Andrew Corporation prefers to describe as "antenna differential gain"[1]. These are variations in the transmitted vertical pattern of the antenna as measured at the low end of the transmitted channel's 6 MHZ. wide frequency band, and then compared to the pattern produced at the high end. Their apparent effect is to vary (tilt) the received frequency response across the signal bandwidth depending upon the receive location.

In DTV, this will cause the ghost-canceling and frequency-response correcting circuitry in the receiver to run out of range sooner than expected, causing the receiver to "lock up" or "go dark". This results in effectively reducing the coverage area of the station. Even a strong DTV signal cannot be received if it becomes too distorted for the receiver's auto-correction circuitry to fix.

End fed, high power antenna designs are more prone to have this problem than are antennas with a center-feed or multiple-feed harness. Most side mount antennas use a center-feed or multiple-feed harness design.

2. A low windload design on the antenna, and placement of the transmission line for minimum increase of windloading during icing conditions. Most towers can handle more un-iced weight easier than they can handle increased windload. Also, where ice is a problem, a heater-equipped antenna has a lower windload, and melts off the ice; a radome-protected antenna has a higher windload at all times due to the radome. When ice gathers on the radome, the windload and the weight load on the tower both increase, pushing the tower closer to the failure point. Five tons of ice built up on WLNH-FM's tower in Laconia, NH, during the ice storm on January 8, 1998. Operations Manager Warren Bailey described it as "a giant popsicle in the sky" shortly before it collapsed.[2]

The following worksheet, Table 1, shows the calculations that will allow us to determine the minimum sized air-dielectric coaxial transmission line that can be used for this DTV installation.

file: ADFTXL02.wk1		Table 1
Loss and Power Handling at UHF frequencies		
for 2.25" air dielectric 50 ohm		
Transmission Line		
Loss, dB/100 ft.:	0.4	at
	500	MHZ
Maximum Average Power:	9.74	kilowatts
Rated at:		
VSWR of 1.0		
ambient temperature of 104 deg. F		
Assuming worst case		
reflected power not to exceed		
10%, and loss distributed on line,		
maximum allowable input to		
transmission line is:	8.85	kilowatts
ANTENNA INPUT POWER REQUIRED:		
ERP, DTV=	100	kilowatts
Antenna Gain	13.01	dBd
or times:	20.00	
ANTENNA INPUT POWER		
REQUIRED=	5.0	kilowatts
Note: assumes a 50 ft. run		
tower base to transmitter:		
TOWER	TRANSMITTER	
TX LINE	OUTPUT	
VERTICAL RUN,	POWER, in kW	
in feet:	REQUIRED:	
0	5.24	
100	5.74	
200	6.30	
300	6.90	
400	7.57	
500	8.30	
600	OVER	

This worksheet calculates the power required, and the amount of power required at the transmitter output, for various lengths of vertical transmission line run. It also indicates the maximum vertical run possible, based on the maximum average power rating limitation of the transmission line.

Note that in this example, at the low end of the UHF channels, this combination of light weight, side-mounted slotted UHF antenna and transmission line could be used up to a height of 500 feet above ground, and be driven by a UHF transmitter with an output power amplifier similar in size and type to a conventional 30 kW peak power output UHF transmitter now used for NTSC broadcasting. The transmitter output amplifier stage has to be designed to transmit without distortion the peak power output of the DTV signal, which is four times the average power. The transmission line, however, only has to be designed based on the average power level of the DTV signal.

While this initial feasibility study worksheet does include some compensation for the maximum return loss allowed by representative transmitters before shutdown, it is primarily intended to determine the minimum theoretical transmission line that can be used for this particular example. It should be noted that the power ratings specified for semi-flexible transmission line are often stated at a lower ambient temperature than those for rigid copper transmission line. Additional derating for effects of solar heating and maximum local air temperatures should also be considered in the final design. The recommendations of the transmission line manufacturer should be sought out before finalizing the design. A useful rule of thumb for initial planning is to consider using the next larger size if 80% of the maximum rating of the transmission line will be exceeded in the design.

As we look at the higher UHF channels, the reduction in power rating of the transmission line, and the increased losses that occur as the frequency increases, limit the usefulness of 2.25" transmission line in this example. The calculations in Table 2 show a maximum useable 200 ft. vertical run at 800 MHZ, at the top of the UHF TV spectrum, before the maximum power handling capabilities of this transmission line are exceeded.

file: ADFTXL1A.wk4		Table 2
Loss and Power Handling at UHF		
for 2.25" air dielectric 50 ohm		
Transmission Line		
Loss, dB/100 ft.:	0.519	at
	800	MHZ
Maximum Power:	7.5	kilowatts
Rated at:		
VSWR of 1.0		
ambient temperature of 104 deg. F		
Assuming worst case		
reflected power not to exceed		
10%, and loss distributed on line,		
maximum allowable input to		
transmission line is:	6.82	kilowatts
ANTENNA INPUT POWER REQUIRED:		
ERP, DTV=	100	kilowatts
Antenna Gain	13.01	dBd
or times:	20.00	
ANTENNA INPUT POWER		
REQUIRED=	5.0	kilowatts
Note: assumes a 50 ft. run		
tower base to transmitter:		
TOWER TX LINE RUN, in feet:	TRANSMITTER OUTPUT POWER, in kW REQUIRED:	
0	5.31	
100	5.98	
200	6.74	
300	OVER	

By changing the loss and maximum power specifications in this worksheet, it can be determined that for this example, one particular brand and model of Low VSWR 3" semi-flexible transmission line could theoretically be used up to a vertical tower run of 1300 feet at 500 MHZ, requiring a transmitter DTV power output of up to 13 kilowatts average power. At 800 MHZ, this transmission line would be limited to a vertical run of 600 feet, and would require a transmitter output of 9.5 kilowatts average power.

For a 4" semi-flexible air dielectric low VSWR coaxial cable, the worksheet gives a result at 500 MHZ of a possible 2300 foot vertical run, requiring a DTV transmitter output of up to 23.6 kilowatts average power. At 800 MHZ, the 4" cable would be limited to a 1400 foot run, requiring a maximum of 17.7 kilowatts of average power from the DTV transmitter.

For a 5" semi-flexible air dielectric low VSWR coaxial cable, the worksheet gives a result of more than a 2500 foot vertical run possible, even at 800 MHZ. It calculates a requirement for 12.3 kilowatts of transmitter power for a 2000 foot vertical run at 500 MHZ, and a requirement for 16.3 kilowatts of power at 800 MHZ.

This presents several design opportunities for UHF stations with a 100 kilowatt ERP limitation, to determine if they can install an interim or initial DTV antenna and transmission line on an existing tower, without exceeding the design limitations of the tower.

A less-attractive option is to proceed by accepting the risk of a time-limited, determined-by-tower-analysis reduction in wind and ice-load rating of an existing tower, during the time that the temporary installation will be in place.

These determinations may be calculated based upon the latest design standard for towers, TIA/EIA-222-F; or by using the design standard in use when the tower was originally designed and installed. For planning considerations, it would be preferable to assess the situation using both standards.

It is often possible that the tower would support the additional load using the design standards that were originally used when the tower was built. After all, the fact that the tower is still standing makes a case for the adequacy of the original design standard.

TRANSMISSION LINE SELECTION CRITERIA:

There are several reasons for using semi-flexible cable for these calculations. One is the ease and speed of installation. This minimizes the cost of installation, reduces the amount of time that the station(s) with antennas on the tower must operate at reduced power or shut down to allow tower crews to operate, and allows installation by crews that are adequately experienced and equipped to install FM and TV UHF translator-sized antennas and transmission line.

A second reason is's flexibility to take maximum advantage of placement to minimize wind and ice-loading. One example of this is the opportunity that exists on some old, rectangular, self-support TV towers whose main legs are designed using large, L-shaped steel members. Plan to mount the temporary small transmission line within the corner of the leg, significantly reducing the effective additional windloading and the ice-loading exposure. Make sure that the engineer doing the tower analysis is aware of such opportunities, and that he considers these opportunities in his analysis.

If you are dealing with a guyed tower with large tubular or solid legs, plan on placing the transmission line inside the tower. A most advantageous location would be on the back side of the south-west leg if possible, especially on the lower part of the tower, in order to take advantage of any afternoon shade provided by the tower leg. This is to minimize solar heat loading of the transmission line.

A third reason is the range of size choices available in semi-flexible transmission line, allowing multiple choices of tradeoffs in weight and windload versus loss and transmitter power required.

DESIGNING FOR A 1 MEGAWATT INSTALLATION

An antenna with a gain of 20 will require an input of 50 kilowatts to produce a 1 MW ERP. While side-mount antennas are available to handle this power level, the power limitations of semi-flexible transmission line will limit our range of design options.

Calculating for an antenna input of 50 kilowatts and using specifications for a 5" air dielectric semi-flexible standard high power cable, we find that at the bottom of the UHF band, the 5" cable is limited to a maximum 300 foot vertical run, and would require a transmitter output of 59 kilowatts average power. At 800 MHZ, the power limitation of the transmission line drops below the 50 kilowatts required at the antenna input.

Calculating for an antenna input of 50 kilowatts and using specifications for a 6-1/8" rigid copper 50 ohm transmission line gives only a small improvement; the rigid line can operate only up to 400 feet. at 471.25 MHZ (Ch. 14), requiring a 56.3 kilowatt transmitter output, and with only a 100 foot vertical run at Ch. 21, where 52.2 kilowatts would be required from the transmitter.

At this power level, a single-antenna and single-transmission line design will require use of a 8-3/16 inch or 9-3/16 inch rigid transmission line, or use of waveguide, depending on the height of the vertical run. These transmission lines, and waveguide, are relatively high-windload items, however, and significantly reduce the chances of being able to mount an interim antenna on an existing tower. It also reduces some financial advantages of a temporary installation.

There are, however, some interesting options yet available. Low VHF channel TV broadcasters are already familiar with splitting the RF signal into two transmission lines to send up the tower. Using two transmission lines cuts the effective power limitation of the transmission line in half. There are UHF slotted antenna designs that can take advantage of this technique; for example, the Bogner DUI series of tower-leg-mounted, low cost, low wind load medium-power antennas, utilizes an externally-mounted power divider mounted behind the antenna to split the power into the two halves of the antenna. Consider moving this splitter to the output of the transmitter.

Another design option is to produce a near-omnidirectional pattern by mounting a narrow-beamwidth, side-mount antenna at the same height on each of the tower legs, with the antennas aimed 120 degrees apart. This assumes a triangular tower. The directivity of the antenna can be increased to reduce the input power required to produce 1 MW peak to 16.7 kilowatts. Each antenna could be driven by a separate transmitter output section, with all three output driver sections fed by a common exciter.

By splitting the power into a three antenna array, three 5" air dielectric high power 50 ohm transmission lines could handle a vertical run of over 2000 feet at 500 MHZ, and up to 1500 feet at 800 MHZ. However, at 500 MHZ, 27 kilowatts would be required at the transmitter for each transmission line for a 1000 foot vertical run, and 42.9 kilowatts for a 2000 foot vertical run. At 800 MHZ a 1000 foot vertical run would require 32 kilowatts per line, and a 1500 foot vertical run would require 44 kilowatts of average power per transmission line.

A UHF translator installation, utilizing three such UHF slot antennas aimed north, east, and south from the same tower, is now on the air. The signal from each antenna is separately licensed as a 1 kilowatt UHF translator. All are on Ch. 30 and driven by a common exciter. It's effectiveness can be observed along Interstate 81 from Lexington, VA to near Harrisonburg, VA. The translator is located on Elliot Knob, 10 miles west of Staunton, VA.

Here is an additional alternative: a peanut-pattern signal could be generated by two separate antennas, with one each mounted on two of the tower legs, fed by two, or four, transmission lines.

Be sure to consider and test-design all options that may make possible a less-expensive interim installation. The conversion to DTV will be expensive at the studio, as well.

COMPARISON OF CONSTRUCTION AND OPERATING COSTS

To start with, if you are a UHF broadcaster with a old back-up transmitter, don't even consider converting that ancient RCA, GE or Townsend klystron-based transmitter by changing out the exciter and "seeing what happens." Several transmitter manufacturers spent more time than they expected in the late 1980's and early 1990's perfecting new-technology high efficiency klystrode, IOT, and tetrode-based UHF transmitters. Many bugs have now been worked out, and the technologies proven. The electrical cost savings are significant. IOT-based transmitters can produce almost three times as much RF output power as a klystron-technology based transmitter for the same size electric bill when transmitting NTSC.

A klystron-based transmitter (without pulser, and not a depressed-collector type) will draw the same amount of current whether sending out a high APL black picture or a low APL white picture. A high efficiency transmitter, by comparison, draws less current when the average power level of the signal goes down. The average power level of an NTSC signal is approximately 67 % of the peak; the average power level of a DTV signal will be about 25% of the peak. Therefore, it is not accurate to say that a 172 kilowatt visual, 17.2 kilowatt aural- rated IOT NTSC transmitter with a $7,500 per month electric bill when refitted for DTV should produce a 47.3 kilowatt average-power DTV signal for the same electric bill. It will not be more; it will probably be $1000 to $2000 less per month depending upon the formula used by the power company to determine the electric bill.

To summarize, this means that a 50 kilowatt average-power DTV transmitter should have an electric bill, including transmitter building air conditioning and powering of terminal equipment, of approximately $6300 per month for budgeting purposes.

Return to the 100 kilowatt ERP example. Use an (extreme example) 2000 foot tower, which would require 18 kilowatts of transmitter output into a single 5" coaxial cable to obtain 5 kilowatts of 800 MHZ input power to the antenna. By comparison, 8 kilowatts of transmitter power would be required if you were using 8-3/16" 75 ohm rigid transmission line, which, we would assume for this example, would require you to build a new tower.

This would save 10 kilowatts of RF output power.

Now, let's compare the costs to build and operate both installations for 10 years. We will use the same antenna for both for comparison purposes.

For the 5" coax transmission line on a existing tower:

Antenna:	$48,000
Transmission Line:	$75,000
Transmitter Building	$55,000
Transmitter	$700,000
Emergency Gen.	$40,000
Installation:	$40,000
Electrical Wiring:	$15,000

Cost to Build:	$ 973,000
Electricity:	$ 378,000
2 new IOT tubes:	80,000
	--------------
10 year operating cost:	$458,000
10 year total:	$1,511,000

For the 8 3-16" coax transmission line on a new tower:

Antenna:	$48,000
Transmission Line:	$180,000
Transmitter Building	$45,000
Transmitter	$480,000
Emergency Gen.	$30,000
Installation:	$60,000
Electrical Wiring:	$10,000

Comparison subtotal:	$ 853,000
New Tower, inst	$2,500,000

Cost to Build:	$3,363,000
Electricity:	$ 125,000
3 new tetrodes:	30,000
	--------------
10 year operating cost:	$155,000
10 year total:	$3,518,000

An important consideration in designing interim facilities will be to discuss critical design areas of your plans with your suppliers; at least one antenna and transmission line manufacturer is offering a generous trade-credit on antennas purchased for interim DTV use when the time comes to refit your tower with a permanent DTV installation. This should be factored into your budgetary planning for DTV implementation.

CONCLUSION

An individual comparison of alternatives, including construction and operating costs, should be calculated for each station, based upon the many variables and possible approaches at each site. This paper presented a overview of methods, including examples, of budgetary designs for interim DTV transmission facilities. The generic financial comparison example shows that any approach that allows interim operation of DTV on an existing tower will usually produce a significant savings. This comparison also shows that the larger transmission line would save a half-million dollars over 10 years if given the choice of using either size transmission line to refit an existing tower.

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[2] Operations Manager Warren Bailey of WLNH-FM, Laconia, N.H., as quoted by Matt Spangler, in "Radio Helps New England Brave Winter's Big Chill", January 16, 1998 issue of *R&R, The Industry's Newspaper*, Radio & Records, Inc. pp. 1&34.

The DTV Studio: Design and Equipment Issues for Digital Television

Monday, April 6, 1998
1:00 pm - 5:30 pm

Chairperson:
John Swanson
Cox Broadcasting, Atlanta, GA

***Showtime Networks: A Case Study of a Fully Digital Production Pipeline**
Dirk Van Dall
Showtime Networks
New York, NY

Cost Versus Flexibility in DTV Broadcast Stations
Paul Harr
Scientific-Atlanta
Atlanta, GA

Resolution Comparison Of SDTV and HDTV Formats
Henry Mahler
CBS Inc.
New York, NY

Becoming Digital: The Facility Migration into Digital & Multi-channel Broadcast Television Operations
Donald P. Archiable
The Austin Company
Cleveland, OH

***Digital Video and Audio Storage and Transport A Complete Production/Post Production Facility Server**
Christopher Romine
Sierra Design Labs
Incline Village, NV

Emerging Trends in the Architectural Design of Broadcast Facilities
Kevin Schaeffer
Gensler
San Francisco, CA
John Aalto
National TeleConsultants
Los Angeles, CA

***Post Production For HD: What Are the Requirements?**
Jon Pannaman
Quantel, Inc.
Darien, CT

The Impact of Digital Video Servers on Broadcast Studio Efficiency, Profitability and Growth

Ernesto G. Leon
Concurrent Computer Corporation
Ft. Lauderdale, FL

Comparative Analysis of Full Bandwidth Versus Compressed HDTV Routing

Paul Berger
CBS Inc.
New York, NY

*Papers not available at the time of publication

COST VERSUS FLEXIBILITY IN DTV BROADCAST STATIONS

Paul Harr
Scientific-Atlanta, Inc.
Atlanta, GA

ABSTRACT

Broadcasters are faced with a multitude of challenges as they move to digital. Maintaining the flexibility of their existing analog systems today must be weighed against the cost of the equipment that it will take to provide that same flexibility in a digital architecture.

One DTV architecture that allows the broadcasters the same flexibility as analog is one that incorporates high definition (HD) and standard definition (SD) encoders. In this architecture the network and contribution signals are decoded to uncompressed serial digital which allows editing video in the same manner as in analog. This option however, will be initially expensive as the first HD digital production equipment becomes available on the market.

On the other hand, lower cost options will exist that include processing the incoming network and contribution signals at the ATSC transport stream level. This option, however, provides less flexibility than decoding and re-encoding. Video in a transport stream cannot be edited in the same manner as analog because editing of MPEG-2 video streams is currently prohibitively expensive because it requires decompressing and recompressing. Technologies are coming available which provide switching between multiple signals at the transport layer.

This paper will compare technologies which are used in high cost / high flexibility DTV broadcast stations with lower cost / lower flexibility DTV stations.

INTRODUCTION

Digital TV promises to change the complexion of television. It will surely change the appearance of the broadcast infrastructure. From simply transmitting a single digital signal to managing a dynamic digital multiplex with data broadcasting, broadcast facilities will have many different looks.

The deployment of Digital TV offers the broadcaster many options to convert his station to digital. Some broadcasters have already begun the transition and have installed new digital component facilities in anticipation of DTV and to improve operational efficiencies in analog stations. These stations are typically located in top twenty markets and are affiliates of major networks. As a result, they can best afford to be early adopters of digital technology.

What about stations that are not in the top twenty markets and not network affiliated? Can these stations afford to convert their facilities to digital? In small to medium markets stations simply do not generate cash flows that can afford the $7 to $10M dollars to install a full digital component facility.

This paper has no plans to address return on investment of DTV, although the author believes DTV represents a tremendous opportunity for the broadcaster. The basis for this can be examined by considering whether the digital channel is used for a single HDTV service or a multi-channel service coupled with data broadcasting. Broadcasters' success has always been tied to the quality of content. If it is popular, then the ratings reflect this and drive advertising revenue up. Broadcasters will learn to use multi-channel service and data broadcasting to enhance their primary offering and to develop unique niches where possible.

However, architectures that provide this flexibility are expensive today and will not exist initially in DTV. They will develop over time. In addition, new digital services will have to prove themselves financially.

TRADITIONAL STATION'S ARCHITECTURE

Compared to the complexity of new DTV architectures, today's analog broadcast architecture appears refreshingly simple. Today's network is built around transmitting just one video channel at a time. Granted, the complexity of delivering one channel has significantly increased over time. Nevertheless, technology and worker skill sets have matured, providing a high degree of experience and comfort level in operating NTSC stations. Also, one of the most important factors that is overlooked but permeates throughout an analog station is the common denominator of NTSC. Every piece of equipment in a station knows it, handles it, and is inter-operable because of it.

Figure 1 illustrates a simplified NTSC station. Each signal, whether it is received off satellite or from a local camera, is input into the routing switcher. From there, outputs to be mixed are routed to the production switcher. Video effects and editing are accomplished using the production switcher. It allows fades, wipes, and pushes with no problem. The program director uses the production switcher to do live on-air pushes and combine local news advertisement with the credits of the network feed.

Broadcasters have grown to appreciate and take advantage of this flexibility, and so have the viewers. Adding video effects allows the broadcaster to potentially capture a viewer for the upcoming program before he/she switches the channel. Other critical uses of the production switcher are local live news or events in progress that can go on air without switching away from the network or syndicated feed. These effects and overlays give the broadcaster a competitive advantage over his competition and create market identity.

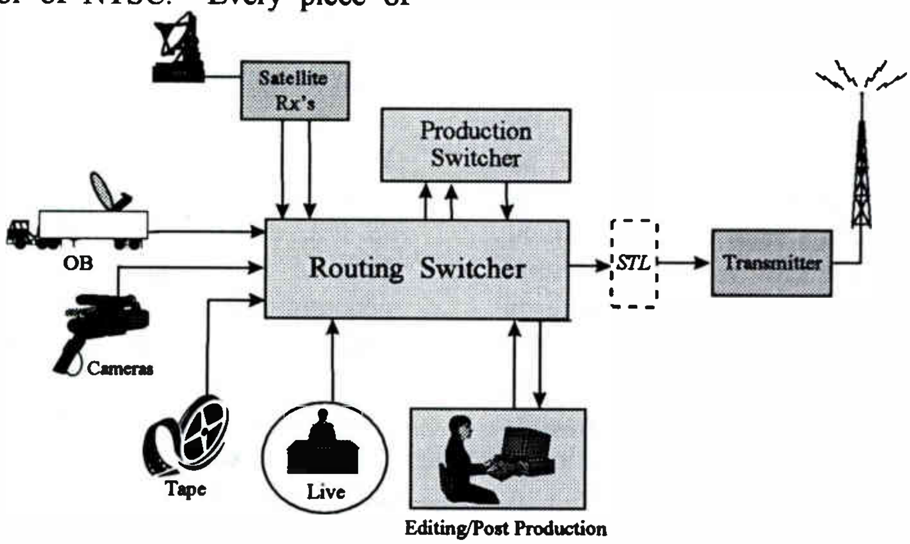

Fig. 1 Analog Broadcast Architecture Simplified

DTV CHALLENGES

DTV, on the other hand, does not offer broadcasters the flexibility of adding video effects and editing simply, without a component digital infrastructure. Moreover, it is not a simple matter of purchasing a digital infrastructure and being instantly ready for DTV. Digital equipment available today does not operate with all digital video formats, cannot handle them, and is not inter-operable. As a result, several challenges must be addressed in the production facility.

The broadcaster must examine his options for a digital station. For example, he must have an idea about the format of programming he plans to deliver. The FCC has given the broadcaster 18 options in the now infamous Table 3 from the ATSC specification A/53. However, from a broadcaster's point of view, Table 3 should not be looked at as 18 different formats, but whether it's HD or SD and whether it's progressive or interlaced scanning.

	Interlaced	Progressive	Aspect Ratio
HDTV	1080 @60Hz	1080@30/24 720@60/30/24	16:9 16:9
SDTV	704x480@60 640x480@60	704x480@60/30/24 640x480@60/30/24	16:9/4:3 4:3

Table 1

Table 1 illustrates this. By looking at the digital infrastructure in this manner it segments the technology and simplifies where to invest. The single biggest issue when upgrading to a digital infrastructure is knowing whether you will primarily be editing and adding video effects to HD or SD format material.

The decision between HD and SD is focused primarily on the data rates of the uncompressed data stream. Component 1920x1080I HD SDI runs at 1.3Gb/s (SMPTE 274M) and SD runs at serial digital interface (SDI) or 270/360Mb/s (SMPTE 259M). For a digital infrastructure to operate with the same flexibility as an NTSC facility, the router and production switcher, camera, and storage media must operate at the component data rates of the material format. Each of these architectures are furthered examined later.

Another big decision is whether to operate your facility in progressive or interlaced scan. The point here is not to debate the issue, rather identify that the broadcaster must currently choose between progressive or interlace production equipment, at least during the early days of DTV. Multi-scan equipment is just not available yet, affordably. This will change, however, as technology improves and costs decline.

Although noted in Table 1, aspect ratio does not factor in the cost of new production equipment when upgrading to digital. Most digital cameras today provide switching between both 16:9 and 4:3. The issue becomes more a matter of handling existing content in 4:3 and how it will be presented on a 16:9 widescreen television. Two options exist and are shown in Figure 2. One option is overscanning and chopping the top and bottom portions of the picture which can be thought of as the opposite of 16:9 "letter box" picture on a 4:3 screen. The other option which is less objectionable is transmitting the picture with black "curtains" on the left and right side.

Figure 2

DTV ARCHITECTURES

Transmission Costs

The transmission system upgrade to digital in a broadcast station is expected to have a wide range of cost variability. Minor upgrades such lashing a dipole antenna to the existing tower will save much expense. However, if major upgrades to the transmission system are required, it will significantly impact the cost. For example, some stations will have to add another tower to support the DTV antenna. This can drive transmission element costs from $500k to $1.5M-$2.0M. Thus, regardless of the digital implementation, cost of the transmission system will be the same.

Pass-Through Implementation

Pass-through is the least expensive way to implement digital in a broadcast TV station. The simplest method is to take the network output of a digital integrated receiver decoder (IRD) and input it into the 8VSB modulator at the transmitter. The network will deliver an ATSC 19.39Mb/s transport stream with all appropriate program and system information protocol (PSIP) information included. PSIP is what was originally called SI (system information) tables for DTV television navigation and EPG (electronic program guide).

Pass-through is the process of receiving and transmitting the exact signal from the network. It allows no provision for commercial insertion, video editing or effects, and no provision for switching to other feeds. Figure 3 illustrates the equipment needed for pass-through implementation. In addition to the transmission system components needed, you need only an integrated receiver decoder (IRD) with the decrypted ATSC transport stream output. The IRD must output no more than a 19.39Mb/s signal with DTV-Ready PSIP information. The IRD output is routed directly into the 8VSB modulator. The output of the IRD must conform to the modulator input specification or a conversion box is needed. This interface is a porposed SMPTE standard PT 20.04/010 which a Synchronous Serial Interface for MPEG-2 digital transport stream.

When the satellite IRD is not co-located with the 8VSB transmitter the pass-through becomes more complicated. With most studio-to-transmitter links (STL's) limited to the current frequency and path authorization, adding the DTV signal will require using the STL link for both signals. Figure 3 shows this in the dotted line area. A digital video encoder is added to encode the NTSC signal and then multiplex it with the incoming digital network feed. An IRD is located at the transmit site to recover the analog signal and ATSC transport stream. Including redundancy, the incremental cost of not having the satellite receive system co-located with the transmit site could be as much as $250k. An alternative is to install a satellite antenna at the transmitter site, receive the digital feed and output the ATSC transport stream directly to the 8VSB modulator. The cost of this option could be well under $20k dollars (does not include transmission system).

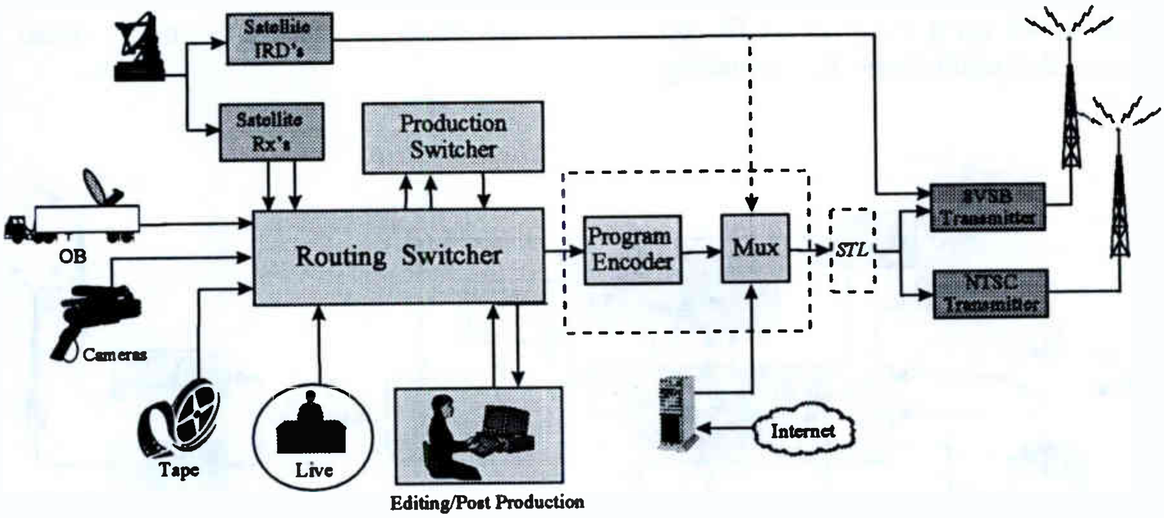

Fig. 3 Analog with Digital Pass-through Broadcast Architecture Simplified

While it does not offer the flexibility of video editing and effects, pass-through does minimize the initial capital outlay to support DTV so that cost can be spread over a longer period of time. Pass-through architectures can be easily upgraded to add local content. By adding an encoder and multiplexer broadcasters can enhance their service offering by switching in local programs and advertising. In addition, data broadcasting can be added at the multiplexer for Internet-like services.

SDI Digital Architecture

Serial digital architectures operate at SDI rates (270/360Mb/s) and are common today in new digital broadcast infrastructures. SDI infrastructures provide flexibility in video editing and effects for standard definition digital television that greatly surpass that of analog. Digital production equipment allows much faster and easier editing while providing capacity to store script effects that can be recalled exactly as originally created.

A serial digital broadcast facility looks similar to an analog facility. Its main element is the routing switcher and production switcher. In addition, it includes IRDs, video encoders and multiplexers. Figure 4 illustrates this. The input and output transmission elements remain basically the same as analog but, are DTV-ready. IRD's receive the network's digital feeds while the 8VSB modulator prepares the ATSC transmission. Because SDI data rates are 270/360Mb/s, this architecture is best suited for delivering multi-channel standard definition (SD) streams. At data rates of 4-5Mb/s per video stream the station can transmit four or more channels. Broadcasters will pass-through some SD feeds and add effects and local commercials to the network primary feed. Cost of SDI infrastructures vary according to the number of streams that are edited. Estimates for a full SDI infrastructure excluding transmission costs range from $3M to $10M.

New components in a digital facility will include decoders, encoders and multiplexers as well as server-based compressed storage. A data broadcast server is likely to also be included to cache Internet-like services. Signals received from contribution or distribution links will be decoded to SDI for production routing. Some signals will remain compressed to reduce concatenation (decode/re-code) noise and simplify operation. These compressed signals will be re-combined with the primary signal at the multiplexer. Other compressed signals will be stored on a compressed file server and routed into the multiplexer for switching to advertiser based or local content.

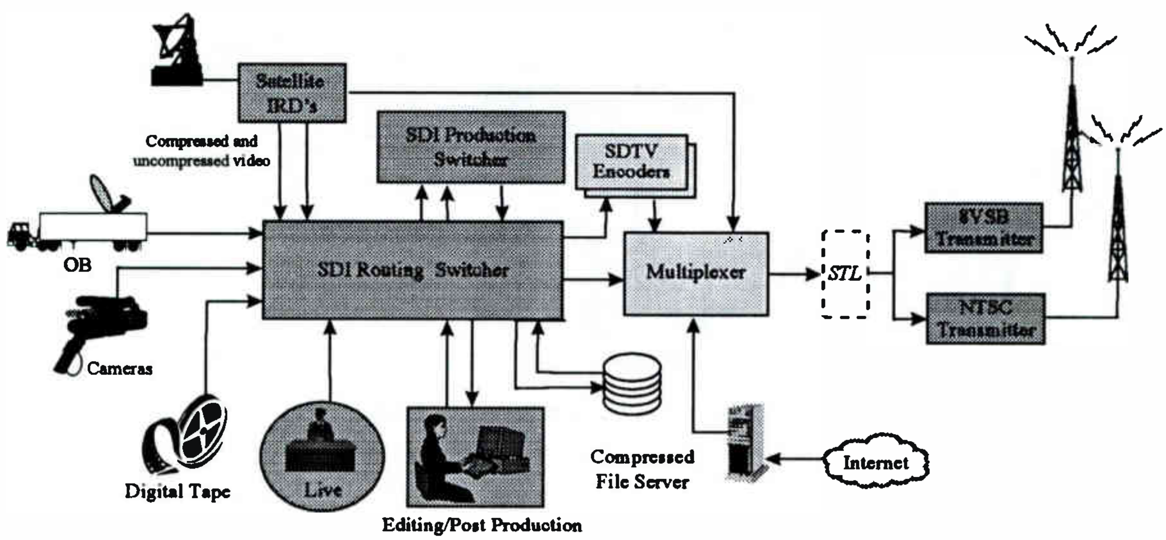

Fig. 4 SDI Digital Broadcast Architecture Simplified

Full Bandwidth HD Architecture

TV stations that launch HD services will have to deploy HD SDI in order to have the flexibility of today's analog stations. This may change in time as technology improves. In the meantime, the only viable approach to edit HD material is an HD serial digital production facility.

Figure 5 illustrates what an HD serial digital station might look like. This diagram actually illustrates a combined SDI and HD serial digital infrastructure. SDI hardware elements are used primarily for routing of local and standard definition video. The HD hardware production elements are isolated in the station to keep the number of HD components and therefore, the costs low. HD content is routed between HD hardware elements at 1.3Gb/s. Distances between equipment at these data rates must be very short. Before it can be routed into SDI interfaces or the multiplexer for transmission, the HD signals must be lightly compressed or encoded to transmission data rates. Light compression of the HD signal has received a lot of interest as it allows HD to be routed in 270/360 Mb/s SDI infrastructures. Conceptually, light compression takes the 1.3Gb/s HD signal compresses it 4 to 1 to achieve rates within 270/360Mb/s SDI. These new products called compressors and de-compressors would be used on the inputs and outputs of the HD equipment. Cost estimates of HD production equipment range from 3 to 6 times that of equivalent SDI serial digital equipment. These costs will decline provided sufficient numbers of broadcasters adopt and transmit HDTV.

To support proposed schedule splits of HD during prime time hours and SD during other viewing hours, stations will likely support multi-channel SD as well as HD. The multiplexer will be used switch between SD and HD signals.

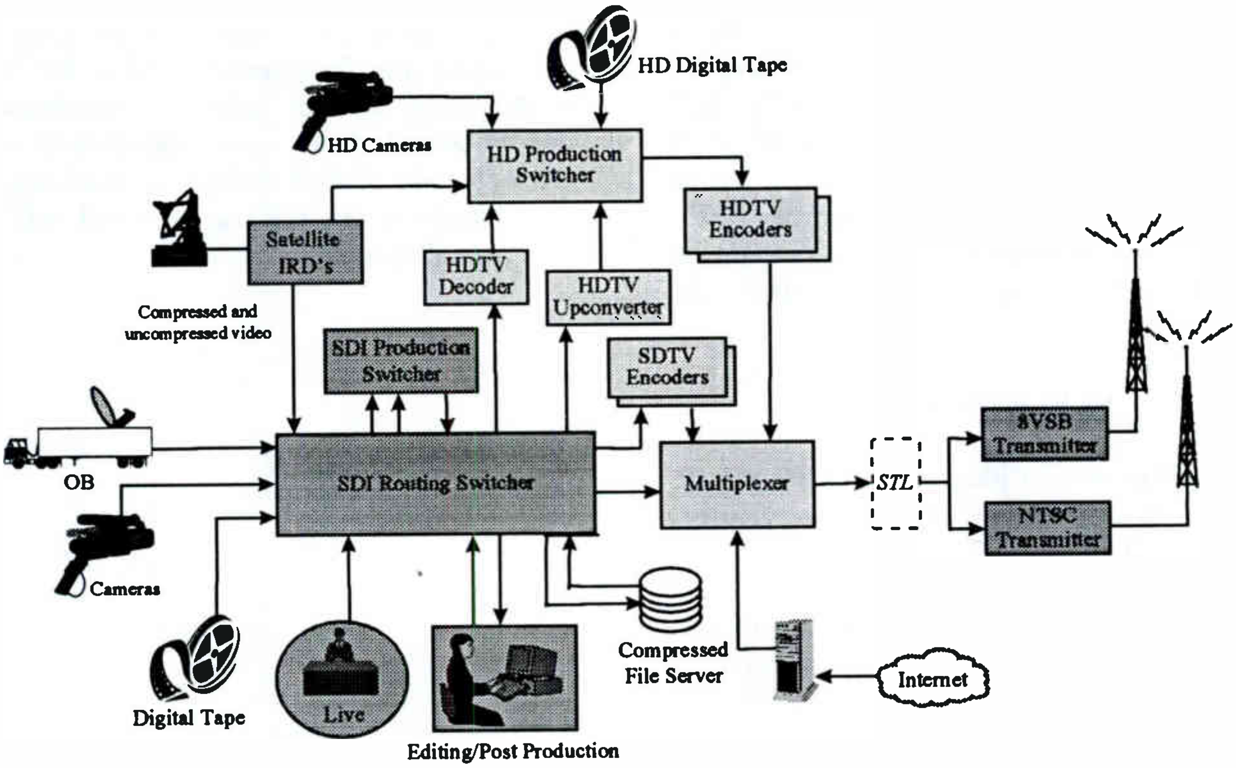

Fig. 5 Digital HD Broadcast Architecture Simplified

CONCLUSION

In summary, Broadcasters will have to weigh the costs of converting to digital versus the flexibility they currently enjoy in analog. The uncertainty of how fast DTV rolls out plays a large part in answering the question of how a broadcaster will invest in digital.

Described here have been three general DTV models which can be used for digital broadcast deployment. Pass-through provides the lowest cost upgrade to DTV but also, offers the least flexibility. In this configuration, the broadcaster must rely on the network signal for all its content. The big disadvantage is the inability to add local advertising and branding. However, this architecture can be easily upgraded to allow the broadcaster to switch in local content through the addition of encoders and multiplexers. Video effects and editing will still not be possible. But, technologies such as bit stream splicing will allow switching in local encoded streams seamlessly without a picture glitch. These technologies represent a simple upgrade path from a pass-through configuration without investing in a complete digital facility.

SDI infrastructures are today's state of the art production equipment that is capable of handling standard definition digital video streams. SDI infrastructures provide for full video editing and effects at 270/360Mb/s data rates. This architecture affords more production flexibility for SD but, does not solve post production for HD signals.

HD requires 1.3Gb/s infrastructures, if video editing and effects are to be added to HD material. The cost for HD production equipment is estimated at 3 to 6 times as much as SDI equipment. These costs are likely to remain high until technology improves and significant volume increases.

Broadcasters must ultimately decide which architecture is best for them and what is affordable. Is it a low cost / low flexibility plant or is it a high cost / high flexibility plant? Regardless of which DTV technology broadcasters adopt first, it will be important to be part of early deployment. DTV represents an opportunity for broadcasters to create or modify their existing business model. By deploying early broadcasters get an early start on developing the knowledge needed for operating digital stations. In addition, broadcasters will greatly benefit from early experimentation to find the right mix of high definition, multi-channel and data broadcast services that is hoped will generate incremental revenue.

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[2] Digital production equipment manufacturers providing cost information who respectfully wish to remain anonymous.

[3] A/53, *ATSC Digital Television Standard,* December 1996

[4] SMPTE 259M-1993: 10-Bit 4:2:2 Component and 4fsc NTSC
Component Digital Signals - Serial Digital Interface

[5] SMPTE 274M: Television - 1920 X 1080 Scanning and Interface

[6] Proposed SMPTE Standard: PT20.04/010
Synchronous Serial Interface for MPEG-2 Digital Transport Stream

Resolution Comparison of SDTV and HDTV Formats

Henry W. Mahler
CBS, Inc.
New York, New York

ABSTRACT

The resolution capability of a particular television format is dependent upon parameters of that format; field/frame rate, number of active image lines, active line time and bandwidth (for analog systems), the number of active horizontal pixels (for digital systems), and whether the format is interlaced or progressive. CBS Engineering has analyzed and compared the resolution capability of several current and proposed formats to be broadcast utilizing the FCC approved, ATSC digital transmission system.

1.0 INTRODUCTION

The resolution of a television image is a function of the television system used to create the image. When that image is delivered to the home, it may be affected by many elements in the delivery path between the original image source and the home receiver. However, the resolution of the image can never be better than the resolution of the original source material, whether it is digitally generated, imaged electronically, or captured on film. For that reason, it is extremely important to utilize the source format and media which are appropriate to a particular television delivery system.

The FCC has approved a system for transmission of HDTV images to the home which can deliver a 30 MHz bandwidth, 1125 line signal with a horizontal resolution capability of 873 TVL/PH.

A series of tests by CBS Engineering has shown that to properly match images generated by HDTV cameras and other sources; 35mm film, rather than 16 or super 16, should be used for programming intended for the HDTV system. This was determined subjectively as well as objectively. Images from both 16mm sources were noticeably "softer" than those from 35mm and an HDTV camera.

The same reasoning may be extended to the proposed use of wide screen 525 line sources up-converted for HDTV transmissions. The basic resolution limitations of this format, both vertically and horizontally, will cause this source material to appear subjectively softer and of lower quality than original HDTV programming.

However, since there will be the necessity to up convert 525 line material for HDTV, an analysis of the basic NTSC resolution capability is performed. The horizontal resolution limits of the composite and component digital standard are examined and also the inherent reduction of resolution within these formats if widescreen 525 sources are employed.

2.0 Resolution Definition

In designing a television system, a number of parameters need to be considered including field rate, frame rate, vertical resolution (number of TV lines) and horizontal resolution (system bandwidth). Once the first three parameters have been determined, then the system bandwidth is defined by the desired horizontal resolution.

With the field rate, frame rate and number of TV lines held constant, the vertical resolution is in-dependent of system bandwidth and horizontal resolution is directly related to system bandwidth.
Vertical resolution determines how well horizontal lines in a picture are resolved (See Figure 1). The maximum vertical

VERTICAL RESOLUTION

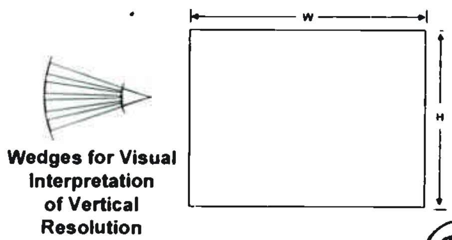


FIGURE 1

resolution is fixed by the number of active scanning lines. This is the number of system scanning lines (525 for NTSC) less those utilized for blanking. Although one might conclude that all the active horizontal lines in the image could be resolved, this is not the case since the television vertical display is not continuous, but is a series of spaced scanning lines. In practice a reduction of this value based upon the Kell[1] factor is usually assumed. A value of 0.7 is commonly applied to an interlaced picture, producing a vertical resolution of approximately 0.7 times the active lines and expressed in TV lines per-picture-height (TVL/PH). Similarly, a factor of 0.9 is applied to a progressive scan picture to calculate vertical resolution.

Horizontal resolution is the ability to define vertical lines in the image (See Figure 2).

HORIZONTAL RESOLUTION

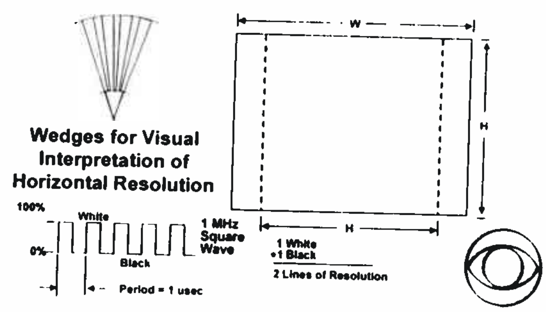


FIGURE 2

The number of pixels which comprise the imaging sensor and the system bandwidth place a limitation on the ability of the camera to reproduce black to white and white to black transitions.

To correctly relate the two forms of re-solution, horizontal resolution is defined as the number of vertical black and white transitions which will fit within the vertical height of the image and is expressed in TV lines-per-picture-height (TVL/PH). Thus, the total horizontal resolution is divided by the aspect ratio, the ratio of the width to the height, to produce the horizontal resolution in TVL/PH.

The relationship between horizontal re-solution and the required bandwidth for any system may be calculated using the horizontal active line duration and the aspect ratio, per the following calculation:

$$\text{Horizontal Resolution (TVL/PH)} = \frac{\text{Active Line(usec)} \times 2 \times \text{Bandwidth (MHz)}}{\text{Aspect Ratio}}$$

3.0 NTSC Resolution

3.1 Analog Resolution

The parameters for 525 line, 59.94 field, 2:1 interlace, 4:3 aspect ratio, NTSC composite analog television are described in SMPTE 170M. This standard specifies the interface for analog interconnection and serves as the basis for the digital coding necessary for digital interconnection of NTSC equipment.

[1] Kell Factor: The Kell factor is the number obtained by dividing the raster pitch distance by the width of the picture resolution element. Or to restate, it is dimension from the top of one line to the top of the next line divided by the dimension of the picture resolution element which is perpendicular to the scanning path. It is an experimentally determined value for which 0.7 is the generally accepted number. REFERENCE: R.D. Kell, A.V. Bedford, and G.L. Fredendall, "A Determination of Optimum Number of Lines in a Television System." *RCA Review*, 5.1 (July 1940).

Vertical resolution of the NTSC signal is determined by the 525 total lines less the 42 lines of vertical blanking (See Figure 3). The 483 active

NTSC ACTIVE PICTURE AREA

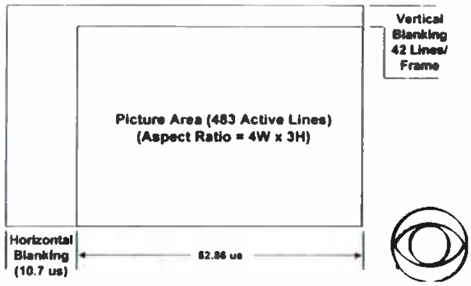


FIGURE 3

lines with a Kell factor of 0.7 yields a vertical resolution of 338 TVL/PH, (483 x 0.7 = 338). Since vertical resolution is independent of bandwidth, this is the resolution transmitted to the viewer.

The horizontal line frequency (f_h) is defined for NTSC as the color subcarrier frequency divided by 455/2. This yields f_h = 15734.27 Hz producing a total horizontal line time of 63.56 micro seconds(μsec.). Using the SMPTE 170M nominal horizontal blanking time of 10.7 μsec., this yields an active line time of approximately 52.86 μsec (63.56 - 10.7) (See Figure 3).

NTSC ACTIVE PICTURE AREA

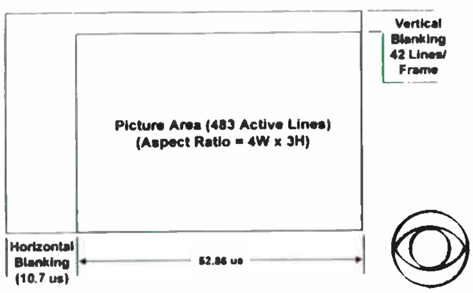


FIGURE 3

In the analog domain the maximum horizontal resolution is determined by the video bandwidth and the active line time. Since a 1 MHz signal has a period of 1 usec it will produce 52.86 cycles in this line time, and since TV lines of resolution are defined as both black and white transitions, 105.72 lines of resolution will be generated by this 1 MHz signal across the total active line. Thus, the number of TVL/PH produced by a 1 MHz signal is 105.72 divided by the 4:3 aspect ratio, or 79.30 TVL/PH. This is commonly rounded to 80 TVL/PH for calculations of the NTSC System resolution capability. Therefore a 10 MHz bandwidth system can reproduce 800 TVL/PH, 5 MHz will produce 400 TVL/PH, etc. Since transmission of NTSC signals requires the video bandwidth to be limited to 4.18 MHz, no horizontal resolution beyond 331 TVL/PH will reach the home viewer. Notice how this is very close to the transmitted vertical resolution of 338 TVL/PH providing essentially equal limiting resolution in both the vertical and horizontal direction.

$$\text{Horizontal Resolution (NTSC)} = \frac{52.86\text{usec} \times 2 \times 4.18\text{ MHz}}{4/3} = 331\text{ TVL/PH}$$

3.2 Digital Resolution

When a video signal is digitized, the maximum bandwidth and therefore horizontal resolution, is determined by the sampling frequency of the digital system. Theoretically, the highest video frequency which may be reproduced is one half of the sample frequency, which for the two currently employed digital standards are composite digital at 14.3 MHz (SMPTE 244M) and component digital at 13.5 MHz (ITU-R BT.601).

Thus, the composite standard utilizing 14.3 MHz has a maximum theoretical horizontal resolution capability of 567 TVL/PH, and the component standard utilizing 13.5 MHz provides lower resolution, a maximum of 535 TVL/PH for a 4:3 aspect ratio system.

$$\text{Horizontal Resolution (244)} = \frac{52.86\text{usec} \times 2 \times (14.3\text{MHz}/2)}{4/3} = 567\text{ TVL/PH}$$

$$\text{Horizontal Resolution(601)} = \frac{52.86\text{usec} \times 2 \times (13.5\text{MHz}/2)}{4/3} = 535\text{ TVL/PH}$$

In practice this limiting resolution will be greatly attenuated due to the properties of the anti-alias filters required at the input to the digitizer, which further reduces the resolution. The 601 luminance filters are specified to be flat within 0.1 dB at 5.75 MHz (456 TVL/PH) and at least -12 dB at 6.75 MHz (535 TVL/PH). A typical filter has been measured at -4.3 dB @ 6.3 MHz (500 TVL/PH) and -6.6 dB @ 6.5 MHz (515 TVL/PH).

4.0 HDTV Resolution

4.1 Analog Resolution

4.1.1 1125I Analog Resolution

The standards which define the video signal characteristics of the 1125/60 high definition production system (SMPTE 240M and 274M) specify a total line time of 29.63 usec, an active line time of 25.86 usec, and a nominal bandwidth of 30 MHz. Therefore, a true high definition 16:9 image has a horizontal resolution capability of 873 TVL/PH.

$$\text{Horizontal Resolution(1125I)} = \frac{25.86\text{usec} \times 2 \times 30\text{MHz}}{16/9} = 873\ \text{TVL/PH}$$

Normalizing this value for a 1 MHz signal yields a resolution of 29.09 TVL/PH. This value may be rounded to 29 TVL/PH per 1 MHz for the calculation of HDTV System Resolution capability.

The vertical resolution based upon a kell factor of 0.7 and SMPTE 240M which specifies 1035 active lines will be 725 TVL/PH. The later SMPTE 274M specifies 1080 active lines producing a vertical resolution capability of 756 TVL/PH.

4.1.2 720P Analog Resolution

The standard which defines the video signal characteristics of the 720P/60 high definition production system (SMPTE 296M) specifies a total line time of 22.22 usec, an active line time of 17.24 usec, and a nominal bandwidth of 30 MHz. Therefore, this 16:9 image has a horizontal resolution capability of 582 TVL/PH.

$$\text{Horizontal Resolution(720P)} = \frac{17.24\text{usec} \times 2 \times 30\text{MHz}}{16/9} = 582\ \text{TVL/PH}$$

Normalizing this value for a 1 MHz signal yields a resolution of 19.39 TVL/PH. This value may be rounded to 19 TVL/PH per 1 MHz for the calculation of the progressive HDTV System Resolution capability.

The vertical resolution based upon a Kell factor of 0.9 and SMPTE 296M which specifies 720 active lines produces a vertical resolution capability of 648 TVL/PH.

4.2 HDTV Digital Resolution

4.2.1 1125I Digital Resolution

The sampling frequency for the 1125 line high definition production standard is 74.25 MHz permitting a maximum of 1080 TVL/PH for its 16:9 image. This affects the square pixel configuration, which is inherent in this format.

$$\text{Digital Horizontal Resolution (1125I)} = \frac{25.86\text{usec} \times 2 \times 74.25\text{MHz}/2}{16/9} = 1080\ \text{TVL/PH}$$

4.2.2 720P Digital Resolution

The sampling frequency for the720P high definition production standard is also 74.25 MHz permitting a maximum of 720 TVL/PH for its 16:9 image reflecting the square pixel configuration of this format.

$$\text{Digital Horizontal Resolution (720P)} = \frac{17.24\text{usec} \times 2 \times 74.25\text{MHz}/2}{16/9} = 720\ \text{TVL/PH}$$

5.0 CCD Camera Resolution

4 x 3 CCD Camera Resolution

The maximum resolution obtainable from a CCD camera is determined by the number of horizontal active, or effective, pixels in the CCD array. To achieve the 535 TVL/PH reproducible in the component digital standard at a 4:3 aspect ratio, the minimum number of effective pixels required is 535 x 4/3 = 713.

However, to prevent excessive aliasing an optical low pass filter is placed in front of the CCD imager. Since these filters cannot produce the sharp cut-off obtainable with electrical filters, using a low pixel count CCD results in a camera which produces very low MTF at higher resolutions.

Thus, in an effort to achieve increased levels at these frequencies, camera manufacturers have been driven to design cameras with higher and higher pixel count. For example, the Ikegami HK 377 and Hitachi SK-2600 cameras currently employed by CBS have 1135 effective horizontal pixels which is a total of 548,205 picture elements for the 4:3 image area. This is the equivalent of a maximum horizontal resolution of 852 TVL/PH which provides a sufficient margin for optimum optical filtering.

The CCD's employed for switchable 4:3/16:9 cameras from Ikegami and Hitachi utilize 891 effective horizontal pixels in the 4:3 mode. This yields only 430,353 picture elements reducing the maximum unfiltered resolution to 668 TVL/PH, thus compromising the 4:3 performance of the camera. In the 16:9 switchable mode there are 1188 horizontal pixels (573,804 elements), producing the same resolution of 668 TVL/PH.

16 x 9 HD CCD Camera Resolution

High definition cameras for 1920 x 1080 I employ CCDs with 1920 effective horizontal pixels which therefore can provide a maximum resolution of 1080 TVL/PH and a total of 2,073,600 pixels in the 16:9 image. High definition cameras for 1280 x 720P employ sensors with 1280 effective horizontal pixels, which produce the maximum digital resolution of 720 TVL/PH. Progressive scanning utilizes a lower pixel count than interlace and this has only 921,600 pixels in the 16 x 9 image. 525 line video down converted from either high definition source will produce a 4:3 or 16:9 image with minimal fall-off at the maximum resolution permitted by either of the 525 line digitizing standards.

6.0 Conversion of 4:3 525 Line Cameras to 16:9 525 Line Aspect Ratio

6.1 Analog Resolution

When using an NTSC switchable camera, the NTSC parameters remain constant while the aspect ratio can be switched from 4:3 to 16:9 or vice versa. Although the 105.72 total horizontal lines of resolution for a 1 MHz signal is unaffected by aspect ratio changes, when using the wide screen mode it must be divided by the 16:9 ratio to obtain TVL/PH. Thus, the resolution capability for the 16:9 image will be 59.5 TVL/PH per 1 MHz rather than 79.3 TVL/PH for a 4:3 image, a reduction of 25%. Rounding this to 60, a 10 MHz system will be reduced to 600 TVL/PH, and 5 MHz to 300 TVL/PH, etc. If a 4:3 image is extracted from this 16:9 source it will have the same TVL/PH resolution limitation as the 16:9 image. Therefore, its limiting resolution is reduced by 25% compared to standard 4:3 source material.

6.2 CCD Camera Resolution

The current cameras employed by CBS utilize CCD's with 1135 effective horizontal pixels providing 852 TVL/PH resolution. As noted above, the same camera modified for wide screen switchable operation has only 891 effective pixels in the 4:3 mode, reducing its resolution to 668 TVL/PH.

6.3 Digital Resolution

If a 16:9 aspect ratio signal from an NTSC camera is digitized by either SMPTE 244M or the ITU-R BT.601 standard, the maximum resolution capability is reduced from 567 TVL/PH to 425 TVL/PH for the composite system, and from 535 TVL/PH to 402 TVL/PH for the component system. Once again this loss of resolution will be present in the 4:3 image derived from the 16:9 signal.

A comparison of CCD camera resolution performance is given below. Note that due to the 30 MHz. bandwidth limitation, the analog resolution of the 720P system is only slightly better than that of digital NTSC.

CCD CAMERA RESOLUTION COMPARISON

Camera System	Effective Pixels/Line	10MHz Analog Resolution TVL/PH	Composite Digital Res. TVL/PH	Component Digital Res. TVL/PH
HDTV				
1125I/16:9	1920	873	n.a.	1080
720P/16:9	1280	582	n.a.	720
SDTV				
Fixed				
525I/4:3	1135	852	567	535
Switchable				
525I/4:3	891	668	567	535
525I/16:9	1188	668	425	402

7.0 Conversion of Signal Formats

7.1 Up conversion of 525 to HDTV

Up conversion of current programming to HDTV will be a necessity during the transition period from NTSC to HDTV production. The up conversion process from 4:3 525 line sources will generate 720 or 1125 line 4:3 images with side panels to complete the 16:9 image. The resolution both vertically and horizontally will be limited to the original 525 resolution of 338 TVL/PH vertically and 535 TVL/PH horizontally.

If up conversions are performed from 16:9 525 line sources, the horizontal resolution will be 402 TVL/PH. This is less than 50% of the transmitted HDTV horizontal resolution for 1125 line images and approximately 69% of the horizontal resolution for 720P images.. However the up conversion will provide a 16:9 image which can fill the entire HDTV display area. It should be noted that in addition to the resolution limitations of the 16:9 and 4:3 material, the up converted images may contain some visible artifacts introduced by the conversion process.

7.2 Down conversion of HDTV to 4:3 NTSC

HDTV cameras are now being supplied with built-in down converters to generate simultaneous NTSC signals and stand alone converters are available to produce NTSC signals from prerecorded HDTV programming. Such down conversions will produce NTSC images superior to our present NTSC cameras since they originate from images with much higher vertical and horizontal resolution. There is less potential for visible artifacts to be introduced by the conversion process due to over sampling of the image.

8.0 Conclusion

There are many factors which determine the ultimate resolution of television pictures . Primary among these are film formats, optical sensors, analog bandwidth of recording and processing devices and the sampling rates used in digital systems. The tables below and Figures 6 through 9 show the limitation of the formats and conversions discussed.

HORIZONTAL RESOLUTION LIMIT COMPARISON, TVL/PH

System	Camera	30 MHz Bandwidth	Digitized	Relative Limit	Horizontal Resolution
1080 I 16:9	1080	873	1080	100%	Highest
720 P 16:9	720	583	720	66.7%	Medium
480 I 4:3 Upconv.	852	535	535	49.5%	Lowest

FIGURE 4

HORIZONTAL RESOLUTION LIMIT COMPARISON, TVL/PH

System	Camera	Bandwidth	Digitized	Relative Limit	Horizontal Resolution
1080I 16:9	1080	873 30 MHz	1080	100%	Highest
720P 16:9	720	582 30 MHz	720	66.7%	Good
480I 4:3	852	793 10 MHz	567-Composite 535-Component	52.5% 49.5%	Medium
480I 16:9	668	595 10 MHz	425-Composite 402-Component	39.4% 37.2%	Lowest

FIGURE 5

The results of this study demonstrate that the resolution capabilities of 525 line systems have been optimized for use with analog NTSC transmission. Both the 16:9 and 4:3 aspect ratio signals produce the maximum possible horizontal resolution (331 TVL/PH) when subjected to the 4.2 MHz bandwidth restriction of NTSC transmitters. However, before broadcast, the various production forms of 525 line signals provide varying horizontal resolution capabilities ranging from the lowest value of 402 TVL/PH for the Component digital 16:9 to a high of 567 TVL/PH for Composite digital 4:3. In general the 525 component digital signals and 16:9 aspect ratio signals produce lower resolution than 4:3 composite digital signals.

To produce the highest possible resolution , source material for the new HDTV broadcast system must originate from 35mm film or HDTV cameras.

Optical sensors for 1920 x 1080 HDTV cameras contain over two million elements as compared to only 573,804 for the 16:9 525 cameras. Use of program material originating from 4:3 or 16:9 "525 line" systems results in pictures with about one-half the horizontal and vertical resolution of 1125 line HDTV originated pictures. Although there are resolution differences between the various 525 line systems shown in the table, "Horizontal Resolution Limit Comparison, TVL/PH", use of the wide screen 16:9 component digital format reduces horizontal resolution capability from 535 TVL/PH to 402 TVL/PH and is detrimental to the resulting image quality. Up conversion of this material to HDTV (See Figure 9) will produce images with limiting resolution far inferior to the original HDTV video and will be noticeably softer with a limiting resolution 25% lower than standard 4:3 material.

The predicted resolution limitations (400 TVL/PH) of 16:9 525 component digital images that have been upconverted to HDTV were subsequently verified in the Engineering Laboratory using switchable cameras from several manufacturers.

RESOLUTION OF 4:3 ANALOG NTSC 525-LINE FLOW DIAGRAM

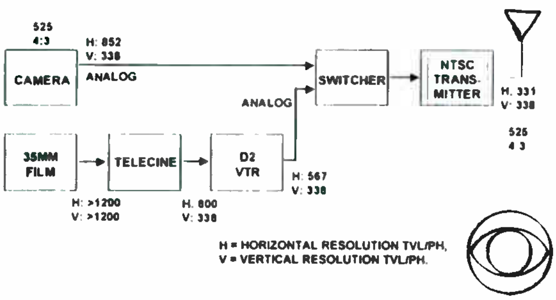


FIGURE 6

RESOLUTION OF 4:3 COMPOSITE DIGITAL 525-LINE FLOW DIAGRAM

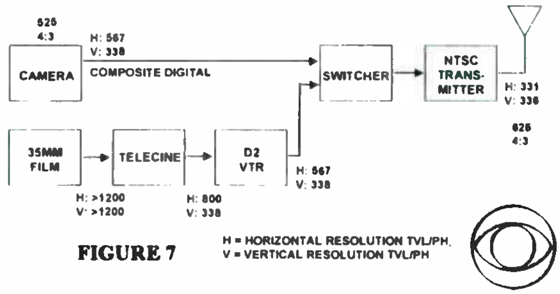


FIGURE 7

RESOLUTION OF 4:3 COMPONENT DIGITAL 525-LINE UPCONVERTED TO HDTV

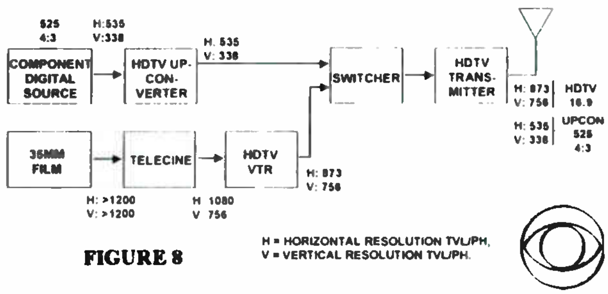


FIGURE 8

RESOLUTION OF 16:9 COMPONENT DIGITAL 525-LINE FLOW DIAGRAM

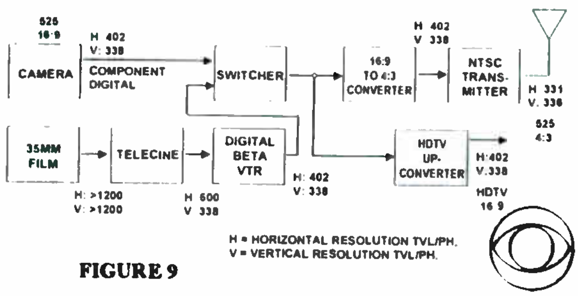


FIGURE 9

BECOMING DIGITAL: THE FACILITY MIGRATION INTO DIGITAL & MULTI-CHANNEL BROADCAST TELEVISION OPERATIONS

Donald P. Archiable
The Austin Company
Cleveland, Ohio

ABSTRACT

This article addresses the transition from analog single-channel television facilities to digital multi-channel broadcast operations. For the most part, traditional design and engineering methods will not apply in supporting electronic and operational requirements of the turn-of-the-millennium television facility. "Becoming Digital" will require sophisticated multi-discipline engineering, planning and design, including a careful process of evaluation and selection of technologies. Broadcasters must carefully define the specific impact of new technologies on operations, business plan and multiple revenue stream functions of the station, all of which are driven by the local management, group ownership, and/or the network.

INTRODUCTION

To know what will make sense in the future demands an understanding of what has worked in the past. The two basic principles that seem to set the benchmark for the future are master planning and preparing the "highways" for flexibility and change. It appears to most engineering executives that one thing is for certain: change is inevitable. Not only are broadcasters facing a most certain revolution in technical equipment and its function, they are also facing constant change in terms of the multiple-revenue stream aspect of the multi-channel business opportunity. This change in the broadcast industry is even more radical than the introduction of color in circa-1953 in Chicago. The major difference today is that a great deal of information can be assigned to this flexible signal. To sum up the situation: we are no longer embracing a technical revolution, *we are in it!*

This concept of digital television affects not only broadcast operations, the signal and what broadcasters do with it, the concept greatly impacts facility infrastructure and the safe and efficient functioning of the broadcast building. One big question broadcasters must ask themselves is: Do we have a business plan designed to meet the increasing demands of a radically changing broadcast industry over the next five years? Answering this question requires a critical understanding of the industry and its growing technology, and what it takes to build a strong foundation for the future. Migration "move plans" are extremely critical in this swing process. The process of transferring "X" from point "A" to point "B" is more crucial than ever. The move plan must account for new equipment, old to-be-transferred equipment, old to-be-archived equipment, and (many times forgotten) related present remote location equipment. In most cases, once the budget parameters are set, returning for more funds creates serious problems. At the heart of the digital revolution are new technological "toys." These new "toys" are often unforgiving in nature, require new facility services, create new design and operational challenges, and may refuse to work together if they don't get their way. And, they represent a significant electrical fire hazard which can threaten the entire building infrastructure if not properly planned for in the facility design process.

The creative department manager and connected broadcast executive alike will be challenged by the changes that this new sophisticated equipment will require. Studios and related control rooms for news, transmission, master control/day-of-air, production and sales/traffic will all be affected in terms of their operational, technical and staffing needs. In addition, throughout the strategic planning process there is a vital need to consider business opportunities to generate added revenue. Networks, corporate owners, and local stations alike face a brave new world in the broadcasting revolution.

MASTER PLANNING

In the professions of architecture and engineering, it is often said that paper is cheap compared to moving around bricks and mortar. Today, planning is by far the most crucial element of any A/E effort. In regard to the technical planning process, there are several issues that impact the architecture of the facility. First, the screen size format ~ 4:3 vs. 16:9 ratio ~ requires new formulas in studio cavity designs and related control rooms. Second, the process is affected by the NTSC interlace method vs. the digital, progressive method of operation which impacts electronic "fibre net" distribution and server placement. Finally, line scan selection determines signal quality and resolution for ergonomic and related parameters.

In addition to the architectural issues at hand, there are four primary changes associated with this revolution that alter classical television operating philosophy and translate it into an issue of adjacency and engineering support. This first change involves the second-channel digital television signal, which includes bi-directional/interactive/multi-faceted transmission. It is likely to transmit two redundant, yet possibly different, signals. This change greatly affects ergonomics, power, and cooling. Second, this nonlinear "tapeless" technology paves the way for server disc-based technology. Tape, for the most part, is eliminated, and this new technology elicits instant response and related changed applications. Third, this revolution emphasizes the "news room of the future," an electronically-controlled journalism operation with related editing and packaging capabilities. Finally, classic television philosophy is altered by the studio complement of cyclorama, grid to floor relationships and innovative design parameters in lighting systems and related control room applications.

DEVELOPMENTAL PROGRAMMING

Adjacency requirements and relative sizing involved in the transition to the world of digital technology raise several important questions. How much personnel will it take to operate all of this new technology? Will this new operation add staff, space, or staff functions? Will this new facility result in downsized operations or transferred functions? Swing phasing plays a significant role in most long-range programs; this phasing determines the architectural overlap. The primary question which remains is: Where are these swings and when do they occur?

Adjacencies are significantly different with the new operating and managing environment. The news operation will quite possibly be the function most severely affected by these adjacency changes in the grand scheme, since radical changes to the assignment desk and production desk take place. With digital technology, news room computers are tied to servers in creative ways that affect cable management and bulk routing. Editing, electronic journalism, graphics and the news management process have a new twist with regard to Electronic News Gathering (ENG)/Satellite News Gathering (SNG) feed locations. Additionally, the concept of field acquisition still leaves unanswered questions for most. Whenever this issue is resolved, it will have to fit into the overall master plan.

Other operational areas significantly affected by adjacency are sales/traffic and master control. Depending on the selected system, the relationship between these can be very different. The 16:9 aspect ratio also impacts cavity and adjacency configurations. In addition, many corporate managers for group owners are interested in combined newspaper and broadcast operations. This combined-business strategy makes excellent sense and is becoming more prevalent in planning processes throughout the United States today.

FACILITY ARCHITECTURE

The formulas, components, and execution of activities involved in facility architecture are affected by this digital revolution in many ways. First of all, design criteria and the rules for height and square footage requirements are quite different from that required for a traditional facility. Distances from sub-servers and satellite disc-based system areas determine the adjacency layouts for now inter- and intra-related spaces. The idea of centralized vs. decentralized design becomes an issue from the master plan intent. Sound-sensitive architecture, such as wall thickness and paralleled elements, greatly impacts acoustics. In addition, radio frequency (RF) screening, faraday cage design and "tuned rooms" for sensitivity aspects surrounding electromechanical interference must be built into wall, floor, and ceiling structures. These calculations must fit frequency and other related criteria.

New electronics drive different adjacency relationships among the components of facility architecture. These conditions change the classic program questionnaire, replacing older methods in architectural programming with the process of "scan card data gathering." As stated earlier, the 16:9 aspect ratio changes cavity formulas in studios, control rooms, news-air areas and master control spaces. Structure-related elements impact a number of operational aspects, including: uninterrupted power supply (UPS) loading, roof alterations and additions, floor leveling at 1/8" per 10 linear feet, server group loading, added duct banks for future cable management, and slab awareness for electromagnetic interference (EMI). Innovative studio transmitter link (STL) and related microwave tower multi-utilization for cosmetic concealment represent still more architectural change elicited by the digital revolution.

SITE APPLICATIONS

Utilities at the core of the broadcast industry - electrical, telecommunications, technical "duct banks" for broadcast "interconnect" ties - are also now a major issue. In addition, in broadcast-related civil engineering, questions remain concerning security, specialized technical underground raceways, and remote microwave runs. The digital revolution affects satellite dish applications such as uplink, downlink, simulsat, KuBand, C-Band and related codes, permits and foundations. The digital revolution elicits the need for towers for STL, inner-city microwave, weather radar for WX dishes, ENG links, and new transmission requirements for second-channel digital television (DTV), standard-definition television (SDTV), and high-definition television (HDTV). Also, the ENG/SNG garage port with all-function shore power and connectivity is necessary; however, dual distribution may be required for this application.

The revolution also elicits new requirements in generator set design criteria and related components, such as foundations, housings, duct banks for mufflers, day tanks and controls, and code-approved fuel storage facilities. Cathodic protection is of paramount importance due to new digital electronics sensitivity requirements, particularly with regard to large tanks near tech core areas.

Digital technology also raises questions about environmental impact. Asbestos and underground storage and other concerns of the Environmental Protection Agency (EPA), Federal Aviation Administration (FAA), and Federal Communications Commission (FCC), must be addressed. Future parking requirements in accordance with new business and revenue stream plan considerations extend far beyond present business plans. Infrastructure protection issues influenced by new on-site technology include: fire protection capabilities, hydrants, standpipes, additional cooling tower loads, and secondary service feeds from the television station to the Telco central office.

ACOUSTICS

The new digital landscape introduces innovative design criteria for acoustic requirements. Noise criteria (NC) ratings and sound transmission class (STC) ratings embrace new sensitivity levels, particularly with regard to NC in digital audio pickup spaces. Encoded audio, multi-channel "surround sound", and compressed vs. non compressed signal applications, acoustically impact rooms in terms of size, shape, structure, isolated raceways, electrical cable routing, and wall/ceiling finishes. Speaker placement ~ with two in front (right and left) and two center-rear, at designed bandwidth pass ~ plays a significant role in the operator ergonomics of the console placement, viewing windows, and live-end/dead-end treatment.

In addition, transfer into other non-acoustically treated adjacent spaces must be addressed in the subject of cable distribution. EMI 60 hertz radiation from floor slab inverse-square rise can also impact tape heads and related electronically-sensitive components.

INTERIOR DESIGN AND ERGONOMICS

Ergonomics, the engineering science concerned with the physical and psychological relationship between equipment and the people who use it, takes on a whole new role in the digital revolution. In the areas of master control, day-of-air transmission and studio control rooms, it is crucial to provide room for future expansion while revising the ergonomics for operating technicians. These revised and often expanded ergonomics certainly impact room architecture and the growth element for future increased revenue stream activity, which is usually unknown at the time of programming.

Planning to operate with a 16:9 aspect ratio greatly impacts interior architecture as well. Ceiling heights, wall widths, floor layering, tilted wall components and room finishes must all be addressed with the operator, producer, and director of program sessions. Interior design budgets for on-air look areas, such as flash cam or control room shots, must be covered for cost control, as well as lighting, finishes, and casework consideration. In addition, news rooms, weather centers, sports centers, etc., are often impacted not only by finishes and set packages, but also by significant electrical and mechanical additions in both cost and loading. This is most likely altered or added with the incorporation of 16:9 shot blocking and a change in cyclorama format. It is crucial to consider the interior of the news room in the shot, particularly the assignment desk. Bulkheads, eyebrows and monitor bridges should also be modeled and even shot in 16:9 35mm film.

ELECTRICAL ENGINEERING

Electrical engineering is the engineering discipline most affected by the digital revolution. The learning curve in terms of requirements for UPS systems has been exhausted. Engineers now have a better understanding of actual loading, non-glitch vs. regulations requirements, and primary, secondary, and tertiary distribution. These requirements have a strong impact on structural loading.

Generator set and related automatic transfer switch (ATS) priority staging takes on entirely new standards. With the new DTV multi-revenue base, new applications for power generation arise. Complexity in grounding and bonding with three-phase, five-wire WYE system distribution has risen beyond what anyone would have expected. Design in the counterpoise system, the ground cluster panels, ground risers, tech core bussing and the service neutral electrode tie has become almost its own discipline.

Switching power supply applications has created the now "not-so-new" third harmonic, nonlinear electrical loading which heats up unbalanced phase conductors and, most of all, neutrals. This problem is overcome with the addition of engineered dual neutrals from branch circuitry to the service feeders and by adding K-rated transformers in the nonlinear load areas. Application of the appropriate K-rating is the most effective engineered solution to these problems. The issue of cable code CL-2 toxic fume requirements is more of a concern in large market cities, such as New York, Los Angeles, and Chicago. If overlooked, these requirements may elicit an enormous expense.

MECHANICAL ENGINEERING

The most significant impact to mechanical engineering with regard to the digital revolution is simply stated: more cooling. As with electrical engineering, if critical mechanical requirements are not met, the consequences can be detrimental. Liebert-style environmental control in the specific tech core is mandatory for operation. Sensitive humidification and dehumidification controls must also be incorporated.

In acoustically-sensitive areas, "tuned" ductworks with traps and lined noise criteria (NC) rating insulation must be matched and in concert with the end rating result. Zoned efficiencies in all matched area requirements must balance with their servers, remote electronics, and operating points. Server chains and "fibre net" hub areas should all be compatible. In addition, vibration supports and vane axial fan style for blade clearance are recommended for acoustic integrity.

Digital technology requires dry-type and preaction sprinkler systems for tech core areas, with the virtual elimination of Halon and related products. Specialized drain design is critical for these areas. In addition, it is crucial that Building Management Systems (BMS) be a part of the controls for mechanical heating, ventilating and air-conditioning (HVAC), fire alarm, security, and mechanical/electrical/plumbing (MEP) equipment. All of these systems should be monitored from a command control operating center.

ELECTRICAL BUILDING SYSTEMS

With digital distribution, there is a strong impact on what would otherwise be more traditional in analog or commercial applications. Master Antenna Television (MATV) and Cable Antenna Television (CATV) systems have an opportunity to replace broadcast router installations in order for signal in-house RF systems to deliver air product to news and other related departments. Head-end, splitter hubs, rack amps, and cable distribution through corridors and operating points all have new design requirements. This is an entirely new approach to in-house signal routing at a greatly reduced cost.

Fire alarm, coding, supervision, and zoning requirements change with the application of new digital technology. The use of dual-clock systems is another change brought about by the digital revolution. These dual clock systems include a digital version tied to a WWV/Rubidium drive-equivalent broadcast version and a classic "master clock" with a standard analog face on a twisted cable for non-timed requirements. This combination of two different systems with two different functions is a necessity. In addition, more sophisticated intercom, interrupt feed back (IFB) and intercommunications systems products require separate cable management distribution.

TELECOMMUNICATIONS

There are so many program questions relative to the revolution in technology that did not exist as recent as one year ago. One of the larger surprises and a "sleeper" in the new digital broadcasting facility is the telephone "switch" and related telecommunications package. This includes all aspects of the telecommunications program – from purchase of components to subsequent installation, with special attention to the unique broadcast component requirements.

The digital revolution elicits a number of technical changes regarding the telecommunications aspect of the broadcast industry. Dual-service feeders should be installed via separate routing to the central office for safety backup. In addition, not only is Category (CAT) 3 telephone cabling needed for office/business communication, but CAT 5 and CAT 7 applications are also needed for data, broadcast services, and transfer and information technology distribution.

It is recommended that broadcasters commission a reputable broadcast telecommunications consultant for this endeavor.

BROADCAST SYSTEMS INTEGRATION

Coordination of the broadcast systems integration package is critical for survival in the digital age. The bonding of architect, MEP engineer, and integrator from the very first "schematic" step is of primary importance. The three must work together to ascertain that ergonomics in control point areas are in concert with one another.

Prior to digital requirements, power, cooling and cable methods could be estimated with forgiving results. Analog program methods are not appropriate for digital; and for the most part, will not work. Central rack rooms are different for digital. It's not about 19" traditional racks anymore; it's about decentralization and servers with new adjacencies

In the digital age, broadcast service panels (BSPs) are more important than ever. They distribute the "highways" of cable routing, such as triaxial, XLR TSP audio, and communication cable for intercom/IFB. Regardless of the state-of-the-art equipment at the connect point, the "highways" remain the same and will pass signals of any nature. While making excellent sense, this also works for any system patch panel throughout the facility. The only impact on this cable arrangement is bandwidth.

The broadcast systems integrator faces new functions relative to architecture as well. Tape archiving, phasing out of analog equipment systems and even training on the new systems all play a part in the scope of the schematic design architecture and engineering. The integrator must be aware of the often-missed roof plan, old to-be-retained equipment, and equipment purchased by the owner on a separate contract. All of these can have a major impact on systems integration.

CABLE MANAGEMENT

More than ever before, a sophisticated cable management program must be incorporated into all aspects of facility design. The use of cable trays with barrier separators for next-phase installation and all-discipline-area color coding is a necessity. Each area should be defined by color: e.g. news=yellow, RF=gray, master control=red, studios=green.

New cable in connection with the digital revolution could include CAT 5 and new CAT 7 cable, new versions of fiber and glass, new CATV aluminum sheath trunk cable, and new (to be determined) higher bandwidth cable products. The use of these new products will have an effect on traditional cables, including: coaxial, triaxial, TSP, multi-conductor audio, and standard fiber types.

A crucial issue regarding cable management is the creation of the "highways" and the "path," and being in a position to transmit any changing product. In providing the final "swing," it is imperative to consider everything from duct bank connector ties from old operations to new function areas with bypass cable systems. These issues should be addressed in the up-front design process long before the move plan is complete.

STUDIOS AND CONTROL ROOMS

Design requirements for the studio cavity and required control rooms for the 16:9 format areas are very different from those for the old 4:3 aspect ratio. Many of the camera shots zoom at a logarithmic function rather than a linear function, and this concept complicates the calculations of cyclorama wall and grid height as it relates to the station's studio product. Floor flatness must be maintained throughout the studio floor at 1/8" per 10 linear feet radius. This applies not only to past control room floor level requirements; it is mandatory for operations with the sensitivity of "homing" in on the robotic process. The new specifications add "flatness" and "levelness" as well.

There are many alternatives in studio lighting availability. In the past we relied on standard 3200-degree Kelvin color temperature lighting with about four manufacturers. Now we have at least eight options ~ all quite different from one another ~ including 5600-degree Kelvin HMI, fluorescent systems, and "Intelligent Power Systems." Each of these has different power and cooling specifications.

We are actually returning to many studio specs of the past that were associated with the image orthicon days. This is particularly true with lighting ratios. BSPs are of special design and are customized to the actual studio need, and these should be fed from a perimeter cable tray with the same future provision as tech core distribution. Sound locks, ingress/egress, audience holding tank, and property storage all have an impact on the new digital studio as well. In addition, structural considerations - rigging, grid support, flown floors/catwalks, lighting connector strips, and lighting instrument heights - have new formulas.

TRANSMISSION

It is crucial to address the transmission component of DTV and the master control tie to its function. If the second-channel digital transmitter is to be at the facility location, many of the same design parameters remain. These include structural loading ultra-high frequency (UHF) beam current supplies, high-voltage power (for transmitter and antenna de-icing), high-end mechanical servicing, specialized grounding and bonding, lightning protection, tower partnering aspects, and the integrity of the transmitter structure. The operation that will emanate from master control as a single affiliate/O&O station with dual (analog and digital) signals or multi-revenue stream programming will require the appropriate ergonomic operating position and the MEP/integration service support.

MIGRATION MOVE PLAN

For the first time, a move plan is a requirement rather than an option. To invest all of the station's facility capital in a DTV operation and not have a plan to pull it all together would be ludicrous. The "sign-on" is critical: old to new, old to addition, old to old, old within new functions.

The move plan is more than a logistical transfer of "X" from point "A" to point "B." It encompasses shutdown, phased bypass, downtime, and return to new air location, among other issues. This occurs for both network and local live signals. The plan emphasizes responsibility and accountability with an assigned team leader and point person for every task in the move operation. Every task of the operation is plotted on a critical path milestone schedule (CPMS).

The move process, as with the master plan process, is programmed with a scan-card system on a 40-foot wall, with time going from left to right, and function from top to bottom. Training must be in concert with the move and be addressed in a separate plan. Constant monitoring is critical to the success of the move plan.

FUTURE AND EXPANSION CAPABILITIES

"Futureproofing," as it has been called, has become an immense challenge. Without knowing what technology will bring over the next five or even three years, let alone the bandwidth, how can anyone design for the requirements of the television facility at the turn-of-the-millenium and beyond? One approach is to make a prorated, educated estimate of power cooling requirements. Another provision that can be made now is allowance for enough cable tray capacity to "highway" the routing of unknown cable requirements (most probably fiber) into a space based on server, not 19" rack, technology.

As long as the tech core facility is designed with these provisions, with accommodations for increased loads for UPS, generator provision, and grounding and bonding, the general manager should be able to transmit a clean, strong signal in the format of his choice. The key to spending capital wisely is to provide the above MEP service and create an award-winning cable and patching network that can implement any application. Planning is everything!

CONCLUSION

The information and data in this paper is only a small part of the learned condition that we know to date. The formulas, calculations, charts, and field studies are too numerous for this report; however, all are available through various research vehicles and consultants. With the new opportunities in multiple-revenue stream operations, and digital functions out of these new master control centers, there are so many "air products" that can serve the American, if not the global viewer. Based on the fact that this new technology and its on-air signal is so new to the world, and the profit track record is so uncertain, it becomes very clear that flexibility and capacity for change are critical elements for inclusion in a broadcaster's business plan.

The future holds a challenging endeavor which necessitates an adherence to the appropriate steps in order to ensure a strong, effective plan for dealing with the digital revolution at hand. First, broadcasters must determine actual station goals and equipment acquisitions as part of a long-term plan (10-year recommended). Second, from this plan, they must prepare a specific business "vision" master plan for short- and long-range goals, based on their corporate mission statement. Third, broadcasters must select a logical broadcast systems integrator and develop a related equipment plan for their technical core and signal operating method. Fourth, an itemized capital budget needs to be developed based on actual estimates from the master plan document. A pros and cons analysis is necessary in order to project business opportunities and revenue stream. For a smooth transition from analog to digital, a CPMS administrator, together with a swing-phase director, should be assigned to coordinate the move plan.

Finally, in order to "Become Digital," broadcasters must take great care in selecting an architectural and engineering professional who can plan and implement a broadcast center capable of facilitating multiple future technologies in the face of rapidly changing digital technology.

EMERGING TRENDS IN THE DESIGN OF BROADCAST FACILITIES

Kevin Schaeffer
Vice President
Gensler
San Francisco, California

John Aalto
Vice President
National TeleConsultants
Los Angeles, California

INTRODUCTION

Major changes in the business and technology of broadcasting have placed new demands on the design of broadcast facilities. This paper provides a brief overview of the forces of change and examples of how emerging trends in the industry are impacting the design of buildings that are intended to last well into the next century.

The mandate for broadcasters to transmit digital television signals brings with it new technology requirements and new business opportunities. As an example, new technology will allow broadcasters to transmit additional outgoing channels with new potential revenue streams. Embracing this technology will have a significant impact on the design of studios, control rooms, and related support areas. Strategies for migrating to this technology must be clearly identified. Work environments and building infrastructure must be designed flexibly to accomodate changes.

The new broadcast hardware model is one of more computer networks, fewer matrix routers, more magnetic and optical disks, fewer video tape machines, more on-screen virtual control panels, and fewer dedicated control surfaces. The new operational model is one of providing more services with fewer people. These and other trends also have an impact on the design of new broadcast facilities. Examples are provided to illustrate these changes and their impact on broadcast facilities.

THE FORCES OF CHANGE

With the advent of digital technology, HDTV, corporate buyouts of networks, and increased competition for audience, the broadcast industry is being bombarded by change.

Clearly, technology advancements and market demands are the primary forces driving change for broadcasters. Competition for market share has resulted in the rapid expansion of cable channels, premium and pay-per-view cable, direct satellite broadcasting, wireless cable, digital video disc technology, and information access via the Internet.

Not only is competition heating up in terms of the TV "product", but there is a significant demand for qualified personnel to provide technical support for production and post-production facilities. This, coupled with client demand for amenities, is changing the shape of many broadcast facilities at both the local and network levels.

Increased competition has also strengthened the networks desire to boost audience loyalty. Many broadcasters are concerned now with the image projected by their facilities. They are starting to use the different elements of their buildings to reinforce their "brand" and identity. Components of a building's design are sometimes used to reinforce a level of quality or "brand" associated with a particular broadcaster.

The rate at which change is occurring in the broadcast industry is unprecedented and is only expected to increase. Broadcasters and their production facilities must be capable of rapidly adapting to changes in technology and/or business plans.

IMPACT OF DTV & NEW TECHNOLOGIES

Digital television legislation presents broadcasters with opportunities and challenges. All analog stations will have to migrate to one or more forms of digital transmission. Yet the matrix of services offered under the digital umbrella will remain in flux for a number of years. Other new technology, some of it unrelated to the requirements of DTV, is available and needed by broadcasters to stay competitive. These technological changes have a direct impact on the planning of new broadcast facilities. For example:

Multi-channel Broadcasting

With the reality of multi-channel broadcasting, larger, more flexible monitor walls are being planned in new facilities, for a number of reasons. The number of channels which a master control room (for example) needs to monitor, including return and break-away feeds, is likely to change over time, requiring increased monitoring flexibility and capacity. In addition, most broadcasters expect the transition from 4:3 to 16:9 monitors to be gradual, given the cost of wholesale replacement. Some monitors may not fit EIA 19" rack mount standards, so a more flexible mounting scheme will be needed.

Additional Master Control rooms are being planned by many broadcasters to accomodate multiple operators. The cable networks have proliferated at tremendous speed in recent years. Companies in the business of launching these new channels (e.g. Turner, TCI, Rainbow Network Communications) are very familiar with the requirements of multi-channel control and monitoring. Now that broadcasters are allowed to transmit up to four channels instead of one, they have to decide how many channels one operator can effectively control and monitor.

With automation and the probability that some of the new channels may provide very little revenue, one operator per channel may not be justified. Some plan should be incorporated in any new station project, however, to support one or more control positions beyond the traditional single master control.

Additional sales, traffic, and support areas may also be required as new channels are developed. Requirements for conditional access, program guides, subtitling, and data-cast support services should all be considered, even if these are not part of the business model on day one.

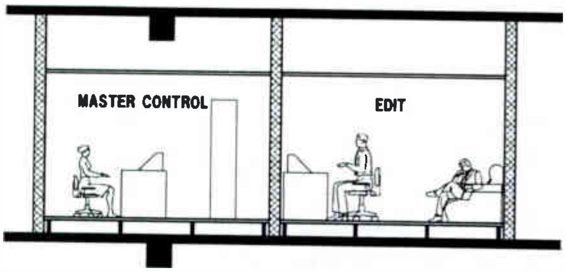
5.1 Surround Sound

The implications of Dolby Digital (AC-3) surround sound include acoustical issues of design significance. The high energy, low frequency sub-woofer sound will typically bleed into adjacent spaces unless exceptional methods are used to isolate one room from another. (Figure 1) In spite of tremendous advances in the technology of television sound, the laws of physics still prevail. Mass is needed for airborne sound isolation and floating slabs and ceilings for structure born sound isolation. Such techniques common in recording studio and announce booth construction, may now be required for every room using 5.1 surround sound if adjacent room isolation is required. These could include master control, edit, sound sweetening, and production control rooms.

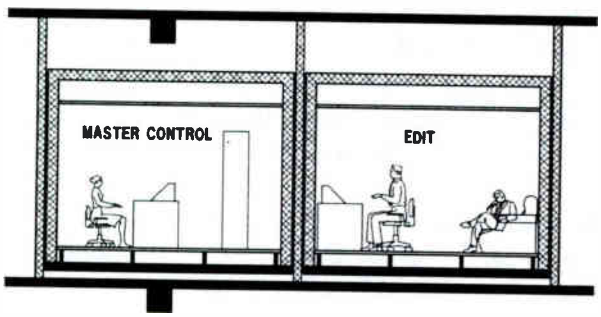
The likely need for slab-to-slab walls having substantial mass and isolated floor construction will make renovating or relocating these rooms more costly than in the past.

Color Judgment and Control Room Lighting

Critical viewing of television picture monitors for the evaluation of analog color demands well controlled viewing conditions. Digital television will yield better color reproduction with excellent repeatability. This places greater demands on video origination and control facilities such as camera and



Conventional Control Rooms



Digital 5.1 Surround Sound Control Rooms

Figure 1:The use of digital 5.1 surround sound will require increased sound attenuation, thereby increasing construction complexity and cost.

production control rooms. It also suggests the need for greater conformance to standard practices such as SMPTE RP 166-1995, which recommends color-correct lighting (D_{65}), defined screen luminance, viewer distance, room color, reflectance and a neutral surround. These issues have been addressed in the past in the design of most sophisticated color correction rooms. These same techniques may now have to be applied to what were previously seen as somewhat less critical spaces such as master control, edit and production control rooms.

Fileservers and Automation

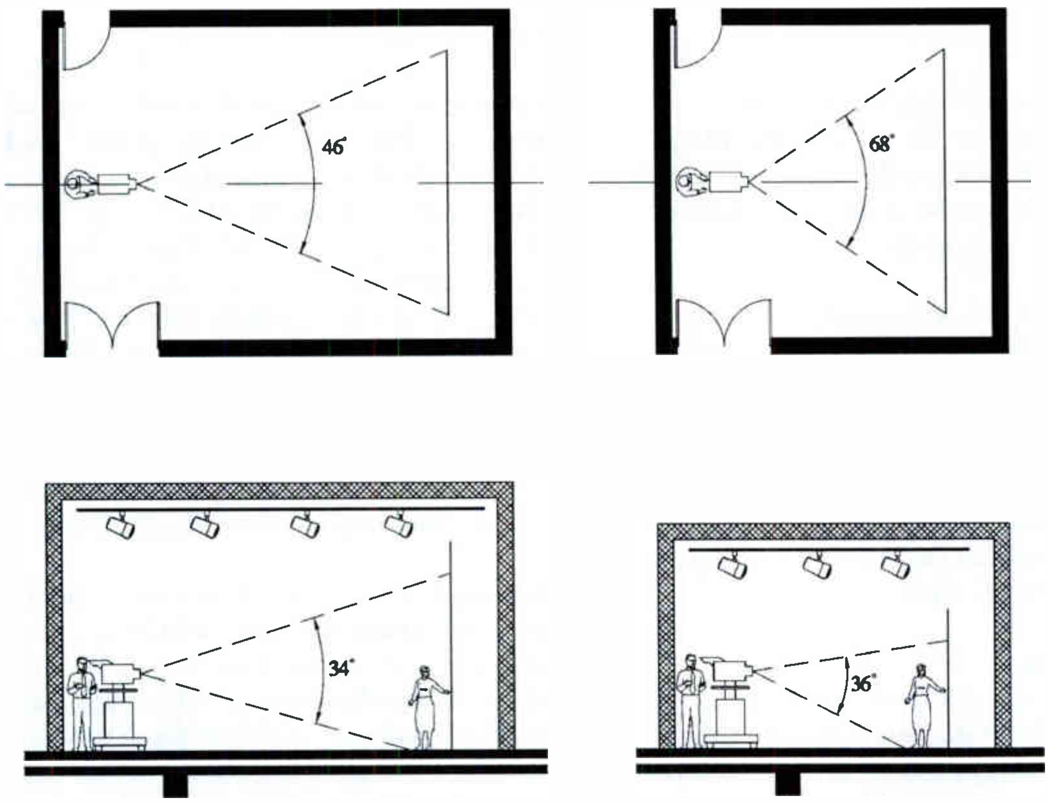
Fileservers and automation have transformed the way programs and commercials get on the air in the contemporary television plant. Bulky videotape machines and their associated tape storage requirements are quickly being replaced by fileservers and automated playout. The need for video tape operators is diminishing and large tape libraries will be replaced gradually by mass optical or other digital storage. It may prove feasible to expand mass digital storage into the library space which video tape now occupies, replacing shelving with additional server frames.

As the cost and performance of data communications improve, off-site mass storage of television assets is becoming an option. This alternative can reduce or eliminate the space, power, and environmental costs associated with large fileservers and data tape robotics. Speer Communications has developed one model for this and others are reviewing similar options.

Studio Camera Aspect Ratio & Studio Dimensions

While a number of rules of thumb have existed regarding the studio floor dimension ratio, the use of the studio, not the taking lens, has typically dictated its dimensions. As an example, a local news, weather and sports studio typically accommodates only these three sets. The studio length to width ratio needs to be sufficiently regular that it can also serve as a general purpose studio if it is not needed for news production in the future. In most local markets with multiple studios, commercial and other program production is typically supported by a second studio. Additional news sets may be built in a third studio to allow for back to back newscasts using alternate on-camera news teams. With 16:9 cameras, wider sets will be needed and dimensions of these types of studios are likely to be impacted. (Figure 2)

For network productions, meduim-size stages built primarily for daytime television drama tend to be relatively low and large enough for six or more sets. Game shows, situation comedies, and specials typically include audiences and have other unique requirements that often call for larger, general purpose stages. In these cases, the impact of 16:9 is less clear, as studio dimensions are usually 75 feet or more in each direction. Most stage sets for television are not this wide. For a larger stage, total floor space is often more important than the exact ratio



S.D.T.V. Digital T.V.

Figure 2: Aspect ratios of digital cameras may impact the proportions for production studios.

of dimensions, assuming a reasonable ratio is provided (i.e., 2:3).

Grid height and resulting overall structure height have a considerable impact on the construction cost of a studio. Simple calculations and experience can be used to establish ideal and preferred minimum heights. (Figure 2)

Studio Floors

HDTV demands higher quality studio sets, lighting and overall production values. The studio floors for high quality television have traditionally required "super flat" concrete floors (F_{F100}/F_{L50}) to maximize image stability for on-air cameras as they are dollied across the floor. If studio-wide productions are desired without limitations on camera movement, HDTV equipment will require even tighter tolerances for flatness of studio floors. American Concrete Institute specifications need to be identified early in the design process to avoid costly surprises at the end of studio construction. Elastomeric coatings also need to be specified early (if used) to insure that slab heights are coordinated with acoustical door seals and other details.

Desktop Editing & Graphics

Changes in television equipment and infrastructure are making it possible for producers and writers to rough cut and even final edit video material from their desktop. The implications for the technical areas in a facility floor plan are numerous, including fewer "cuts-only" edit booths, fewer CG and paint stations, and the ability to de-couple these and similar activities from the core technical rooms.[1,2]

Virtual Sets

The technology for multi-camera virtual set applications has been proven in live broadcasting (Turner Sports, ABC-TV, et. al.) and is expected to migrate to local TV stations as the market for this equipment grows and costs drop.[3]

The real promise of this technology for facility designers lies in its potential to reduce the need for studio space. With new studio construction costing $200 per square foot and up, the ability to pare hundreds or even thousands of square feet of space from these expensive rooms can offer a substantial savings. Conventional studios can also accomodate virtual sets, of course. In fact, most new studio projects include hard cycloramas.

Power Reliability

With fewer videotape machines and more file servers, more of the broadcaster's product is vested in fewer places, making program and commercial assets potentially more vulnerable to power outages. In the past, if power was disrupted briefly while a tape cart machine was playing back a commercial, the commercial (and all others in the machine) would still be available for playback once power was restored and controllers rebooted. With fileservers, erratic power could cause a less than graceful shutdown, possibly requiring a partial or full rebuild of the commercial or programs stored on the fileserver. Given this risk, and all the other usual ones, virtually all new broadcast projects include plant-wide uninterruptible power systems in addition to generators.

Signal Topology and Cable Management

Conventional television systems topology includes one or more matrix-type routing switchers, with their associated DAs, patch fields, and high cable density. Networked broadcast equipment, including fileservers and storage systems, with very high speed networks, promise an addressable distribution system with vastly less cable.[1,2] While existing routers and new gigabit routers will both continue to serve valuable functions, very high speed LANs will be increasingly relied on to meet broadcasters' infrastructure needs.

What does this mean for facility architecture and engineering? Networked devices typically have few restrictions on their physical placement within a facility. Gigabit data circuits on very high speed STP and fiber optic cables will likely serve the broadcast signals through structured building wiring. Adding a new graphics workstation or editing workstation to a distant office within the building will require patching and a plug-in wall connection, not today's coax and equalizing amplifier solution (to say nothing of twisted audio and data pairs). Structured wiring will prevail over costly and inflexible point to point conduit and cabling.

While not practiced today, future access flooring depths may be reduced from 18 to 24 inches to 7 or 8 inches based on using increased LAN rather than matrix signal topologies. Where high density racks may require 31" to 36" in depth to accommodate today's coaxial cabling, future designs might be supported by 25" deep racks, saving floor space in technical areas. Some of the 24" and 36" wide cable trays might be replaced with 12" and 18" trays, also saving cost and space.

Technology Migration

New facilities are often a mix of old and new equipment. Five years ago, new TV stations had hybrid 270 Mb SDI and analog infrastructures. Today, they have hybrids of analog and digital coax, fiber, and LAN circuits. They are migrating to the latter as the technology matures in broadcast applications. This type of migration benefits from extra cable management capacity to allow adding the new before removing the old (certainly a requirement of 24 hour/day on-air operations). Good facility planning will provide extra wireways and equipment racks to support this migration.

NEED FOR FLEXIBILITY & GROWTH

Flexibility is often expensive. Broadcasters must

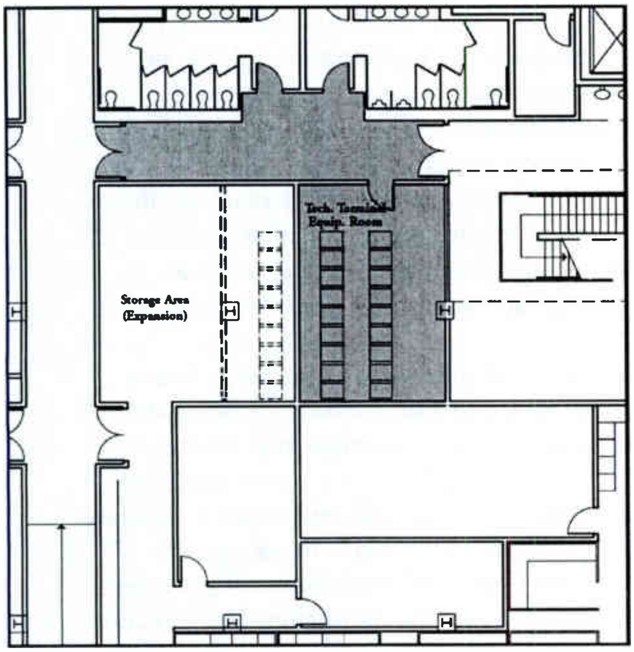


Figure 3: Plan illustrates how strategically placed "soft spaces" in technical suites can accomodate future expansion.

weigh the initial capital outlay associated with flexibility against their business plan, likely technological advancements, and available budget. Typical considerations for flexibility include the use of oversized risers for cable and power distribution and provision of access floor in non-technical areas to allow an easier conversion of these areas to technical environments. Increasing the depth of ceiling plenums can also make the transition from non-technical areas to sound-sensitive environments easier by offering more vertical clearance for larger duct-work and sound-attenuating construction.

Thoughtful, forward-thinking planning at the outset of a project can reduce the cost of change and minimize growing pains. Initial planning phases for new facilities should include strategies for expansion, particularly when assessing locations for studios and technical facilities. If these areas are "landlocked" by inflexible building elements or site constraints, options for growth will be stifled. Technical cores should be centralized and stacked within a building to provide a hub for cable and power distribution. Proper placement of terminal rooms and risers helps to minimize cable runs and allows expanding technical areas to radiate out and rise vertically through a building in a compact, cost-effective manner.

Operational work flows as well as vehicular and pedestrian traffic patterns should be analyzed to meet both current and future expansion needs. The arrangement of rooms within technical suites can allow for growth by incorporating non-technical or "soft" spaces like storage rooms or offices into the plan. (Figure 3) If additional technical space is needed, these rooms can be taken over for that purpose.

Broadcasters cannot afford to operate under-utilized facilities. Rooms that serve a dual purpose can increase a facility's flexibility and increase productivity. Control rooms that can double as edit rooms, reception or conference areas that can also be used as audience holding areas, and studios that can be controlled from multiple locations are all examples of ways to maximize a building's efficiency. (Figure 4) Creative planning can achieve this without the multi-purpose aspect of a room being apparent to its users.

COMPETITION FOR PERSONNEL & CLIENTS

Just as competition for market share has increased for broadcasters, so has the competition for finding and keeping qualified personnel. The quality of the work environment can be an important factor in this regard. Everything from the quality of the equipment and technology being utilized to the comfort and attractiveness of the facilities come into play.

In new and remodeled broadcast facilities more attention is being given today to the design of the work environment in terms of acoustics, lighting, and aesthetics. Eliminating "break out" noise from adjoining rooms and improving interior room acoustics have added to staff comfort and effectiveness. This will only grow in importance as sound

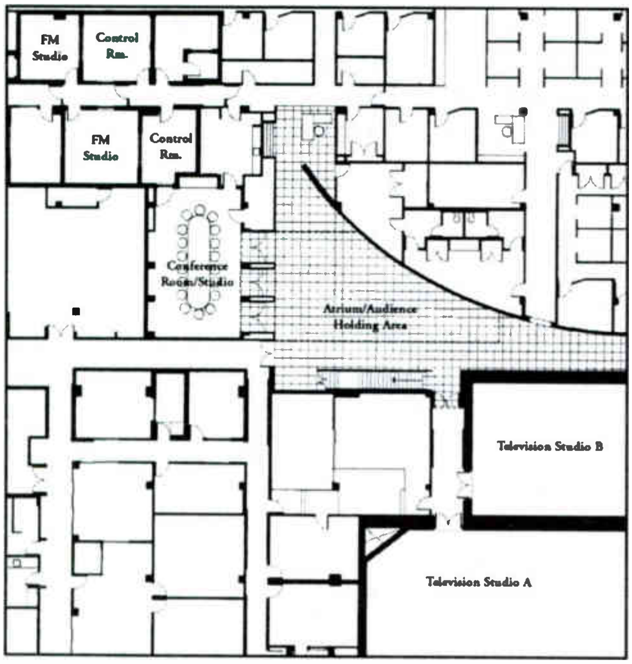
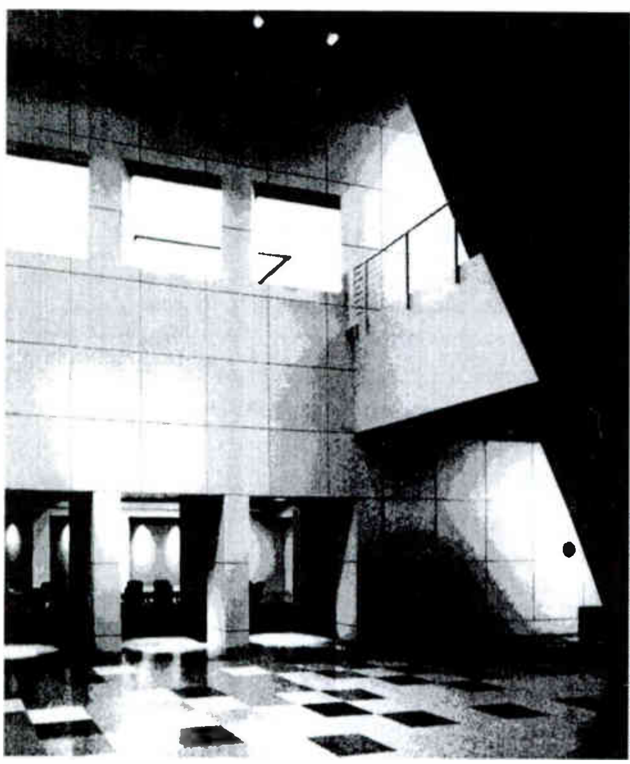


Figure 4: Plan illustrates rooms with dual purpose and studios that can be controlled from multiple locations.



Photographed by Chas McGrath

Figure 5: Central atruim serves as a gathering space for staff and community functions while doubling as an audience holding area for adjacent studios (KQED, San Francisco)

quality improves and monitoring becomes more critical. The quality and quantity of lighting in technical suites also has a major impact on the comfort and effectiveness of staff. Good lighting not only impacts critical viewing of monitors, but reduces problems associated with glare or eye strain for operators. Effective use of materials, finishes, and natural light (when appropriate) increases employee satisfaction with the work environment.

To attract and hold on to staff and clients, many broadcasters and post-production companies are including amenities such as larger and more comfortable viewing and edit rooms, "break out" and informal conference areas, teleconferencing facilities and access to work areas where clients can temporarily "hang their hats" while projects are underway. Other improvements can include lounges and game rooms, shower facilities, and sleeping quarters.

BRANDING/IMAGE

Another result of the increased competition faced by broadcasters is their need to establish name recognition and a sense of loyalty in their viewers. Broadcasters are constantly looking for new ways to differentiate themselves from their competition. The identity or "brand quality" of a network or station is primarily established through the product it delivers. Many broadcasters find it advantageous to carry this branding through to other aspects of their operation, including their facilities. Greater attention is now given to the location and appearance of new and remodeled structures. With a prominent location, a building can serve as a billboard to the surrounding community through signage and expressive features. Whether it is a state-of-the-art, high-tech appearance or a more traditional, established character, key design features can generate the desired perception of quality.

Many networks reinforce their brand identity through the sale of retail products or by promoting themselves through public tours. The Discovery Channel, Disney and QVC are examples of this phenomenon. As the general public's fascination with television and how it is produced has increased, studio tours have become popular as tourist destinations. This is an ideal opportunity for broadcasters to promote their products and gain allegiance from their viewers. The impact of these issues on the design of a network facility can be significant. Retail operations and tours that bring the public into and through these facilities must not interrupt work flow. Tours must be interactive and woven into the fabric of the facility without being disruptive.

DESIGN FOR CHANGE

Whether driven by marketplace factors or technological advances, it is inevitable that competitive forces will bring continual and rapid change to the broadcast industry. Facilities must be designed to allow broadcasters to embrace change. Broadcasters have to adapt to new technology and market trends in a constant effort to maintain their competitive edge. A flexible, well-planned facility will allow them to position themselves appropriately for whatever the future may bring.

[1] K. Gnzik, "Architecture of the Virtual Broadcast Studio", SMPTE Journal, December 1997

[2] C. Woollord, "Distributed Digital Post Production", SMPTE Journal, 705:709, October 1997

[3] D. Drew, "Practical Studio Productions Using High Quality, Low-Cost Virtual Sets", SMPTE Journal 618:623, September 1997

The Impact of Digital Video Servers on Broadcast Studio Efficiency, Profitability and Growth

Ernie G. Leon

Concurrent Computer Corporation

Ft. Lauderdale, Florida

ABSTRACT

This paper will explain the functions and impact of digital video servers and how they can help a broadcasting or production studio become more efficient, profitable and achieve low-cost growth – without having to reconfigure an entire studio system. The general effects of digital technology on the television industry will be discussed. Emerging technologies will also be addressed, noting how the digital video server has evolved into each new technology sector as a provider of efficient digital video delivery. Technical advantages will be discussed specifically how certain aspects of different operating system environments such as NT and UNIX® affect the delivery of digital video via a digital video server.

DIGITAL TELEVISION AND THE EMERGENCE OF THE DIGITAL VIDEO SERVER

Television as it is known today has remained virtually the same for almost 50 years – with no change in how a signal is transmitted. As the new millenium approaches, major changes in the television and broadcast industry are occurring. In essence, the digital revolution has begun! In the early 1960s, television was combined with early computer technology to create simulated environments for aircraft pilots. As simulation technology expanded, it provided new advances in computer graphics. This period in time, when television and computer technology merged, could be considered the advent of digital television. In addition, this early technological combination provided the seeds of virtual reality and advanced multimedia [1]. In aircraft simulation applications, computerized terrain databases were created to offer different simulation options for training. Terrain databases were nothing more than large program files that created video scenes of particular places on very large television screens. This, in turn, created large databases that required large and expensive storage space. As time and technology advanced, multiple simulators shared the same database thus creating the basis of multiple streams. Even though it was not known at the time, simulation technology was creating the first-ever digital video server.

To effectively make the television and the computer function as one component for the broadcast industry much work had to be done. As training simulation grew, so did the advancements in television, video and computer technology. Today, that pioneering work has allowed for television, cable, telephone companies and the Internet to be interlaced to provide entertaining and interactive digital video information to the consumer at a very fast pace and in real time. One might say that the simulation industry "pushed the envelope" in digital technology.

As digital technology advanced, the question for the broadcast industry then became "How will all the video information be transferred and how will it be managed?" Similarly, in the 1980s the PC computer industry was trying to determine how to transfer files effectively between two computers; thus, with some networking knowledge, the creation of the file server. In the video domain, the development of the digital video server has solved many problems of sharing and delivering video data simultaneously between two networked systems. However, the file server and the video server differ in the CPU processing power required for each to perform its job. While a file server can be derived from a standard PC platform, the video server needs to incorporate a multi-processor platform that can handle interrupts and high input/output (I/O) throughput to manage large amounts of digital video data.

By working with digital video and video servers, broadcast and production studios can easily manipulate any video data in a tapeless environment, thereby eliminating redundant and unnecessary production tasks. With the advent of more powerful video servers, the standard tape-based systems will eventually become a thing of the past and be completely phased out. With the advancements of High-Definition Television (HDTV) for the new millenium, it makes better sense to include a video server as part of the studio function in order to adapt to future capabilities. A digital video server will ultimately serve as the epicenter between all the functions of a broadcast studio serving the newsrooms, post-production facilities, cable companies, Internet services and satellite downlinks and uplinks. These functions are very similar to the first databases delivering terrain information to the different simulators. Figure 1 illustrates how a digital video server will be the central component to all digital media being distributed and produced by all digital technologies.

The Digital Language

The advancement of digital technology has brought new terms and buzzwords to the digital television industry. To understand the video server market and the industry as a whole, this new terminology needs to be understood. First and foremost are the compression schemes of the Motion Picture Experts Group standard or simply MPEG. The standard is officially known as the Generic Coding of Moving Pictures and Associated Audio [1]. MPEG has several standards that include MPEG-1, MPEG-2 and MPEG-4. MPEG-1 and MPEG-2 are the more common standards in the digital video server industry. MPEG addresses the compression and decompression issues associated with video and audio. For video production, digital video servers should be able to handle these compression schemes. MPEG-1 is mostly used for lower quality compression that is superior to VHS quality. MPEG-1 video files are generally encoded at 1.5 Megabits per second (Mbps) rates. MPEG-2 is a compression scheme that is used for higher quality video compression and the data rate can vary between 3 Mbps to 6 or even 15 Mbps. MPEG-4 is a new standard that will officially be introduced in 1998 and will deal with low bit-rate compression (i.e., 10kbps, 20kbps) with very good quality [2].

Other compression schemes exist such as the newer standard for digital productions simply called Digital Video or DV. Another compression algorithm standard is the Joint Pictures Experts Group standard, which is called JPEG. This standard generally deals with the compression algorithms for still pictures. M-JPEG or Motion JPEG uses the JPEG algorithm to compress multiple still pictures together to provide a moving output. M-JPEG is typically used by non-linear systems due to the frame-by-frame accuracy of each independent compressed frame needed for quality and exactness in editing. (Linear systems are analog tape editing systems while non-linear are digital computer-operated

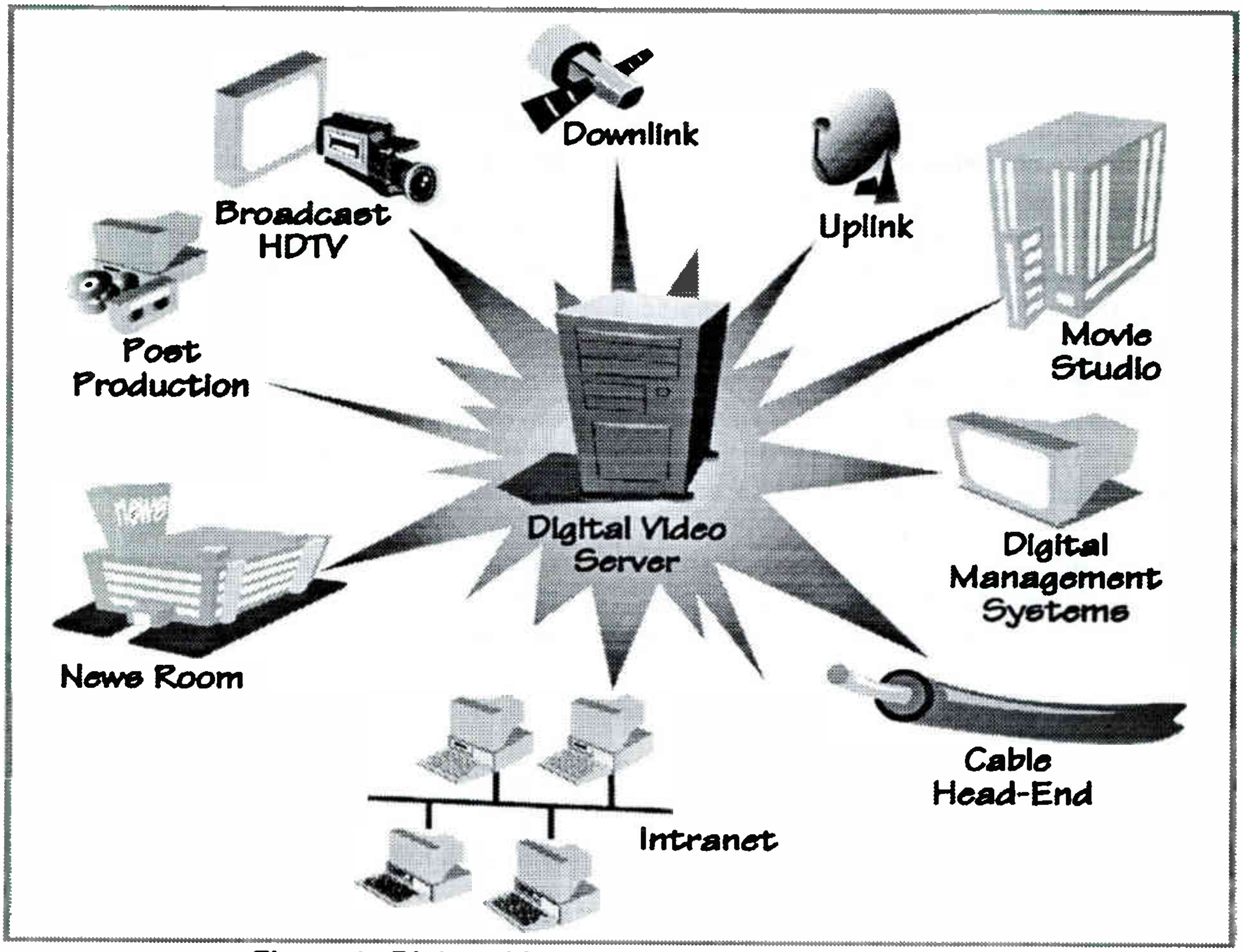


Figure 1. Digital video servers will be the epicenter of all produced and distributed digital media.

editing systems. Non-linear editing systems have become the standard in most production studios).

Of course, the compression scheme is a function of the type of encoder used. Encoders are components that take an analog video input and compress it to an MPEG, JPEG, M-JPEG, or DV data file. Encoders are currently extremely expensive components, but their price will decrease as technology advances. Similarly, decoders are components that take an encoded data file and decompress it to an NTSC (National Television Standards Committee) standard, PAL (Phase Alternation by Line) standard or SECAM (Systeme Electronique Couleur Avec Memoire) standard analog signal. Both encoders and decoders are available as stand-alone components or as software solutions.

For less compression quality and less cost, Wavelet technology exists. Wavelet technology does not break down an image into discrete areas during the encoding process -- as compared to MPEG or JPEG which compresses the video input to the pixel level. Wavelet technology is mostly used for medical imaging, but can also be used well for video encoding. Besides compressed data, uncompressed data can also be used. This will provide the highest quality since no compression ever takes place. However, the data is maneuvered digitally. Table 1 shows the different compression schemes along with their respective typical application.

Compression Scheme	Type of Compression	Picture Quality	Typical Application
AVI	Still Pictures/Motion	Low	PC Applications
Wavelet	Still Pictures/Motion	Low	Medical Imaging
JPEG	Still Pictures	High (Stills Only)	Photography Scanning
M-JPEG	Motion of Still Pictures	Low to Medium	Post-Production/Editing
MPEG-1	Motion Compression	Medium to High	Kiosks/Intranet/Security
MPEG-2	Motion Compression	High	Movie Delivery
DV	Motion Compression	High	On-Line Editing/Production

Table 1. Compression schemes used for typical applications.

HDTV is the new High-Definition Television standard that was adopted by the Federal Communications Commission (FCC) in 1997. The FCC will require that all television networks and transmission facilities transmit their signal in digital format by the year 2006. Digital transmission will eventually be the ***only*** transmission means for all media.

The digital video server will serve as the new business machine for all broadcast and production studio facilities. Video servers are inherently digital; HDTV is digital! Digital video servers are simply a perfect match for HDTV for storage, retrieval and delivery of digital data. MPEG and other compression standards will also evolve, as HDTV standards are formalized and new components are brought and introduced into the market. The digital video server will be able to handle any compression scheme and accommodate any HDTV standard change eliminating the need to change components whenever there is an industry change.

BUSINESS IMPACT OF A DIGITAL VIDEO SERVER

The broadcasting and production industries including the motion picture industry are currently in a state of change. These industries have literally been transformed over night with digital technology. In addition to simulation technology, the change is also due to the merging of computer talent with television gurus and the fascination with computer graphics, interactive games, video entertainment and the Internet. Interestingly, the broadcast and production studios of tomorrow will need to implement digital video servers in their facility to eliminate the continuous support of multiple analog video systems and avoid the purchase of new television equipment at every turn just to keep up with the new digital competition.

New HDTV standards will add to the option of implementing a video server due to the major digital systems that will need to be supported by broadcasting and production studios. Consequently, one main advantage of a video server is that the digital format is transparent. In other words, multiple standards will not affect a video server since it does not matter whether it deals with NTSC, PAL or SECAM television signals once the signal is compressed. The video server is also unaffected by whether a digital data video file is compressed (i.e., MPEG) or uncompressed. In addition, the type of output is transparent to a video server. For example, if the output is a Fast Ethernet network such as 100 BaseT or Asynchronous Transfer Mode (ATM), the digital video server will support it as long as it has the correct software drivers or integration boards. Even input or output devices such as video cameras or television monitors will be supporting HDTV standards by the year 2006.

After the year 2006 many of the analog standards will be put in a closet at the FCC in Washington, D.C. with the coming of the HDTV standard. Video servers then will eliminate countless tape systems to support any standard. It is true that there are a number of tape decks and television monitors that support multiple television standards today but a video server will eliminate any thought of purchasing options due to television standards. Well, almost any thought! A digital video server will provide the efficiency, profitability and growth capability that will eliminate expensive analog tape systems, including frequent system upgrades.

Digital Efficiency

Efficiency can be measured in different ways depending on the focus of a particular broadcasting or production studio. A post-production suite that is dedicated to only duplication services will use a video server differently than a post-production suite that is dedicated to providing editing services. Broadcasting stations will use the video server in still a different manner. If the focus is duplication services for a production suite, a video server will only be used to store a pre-determined amount of digital video clips for digital transmission to a rack of VHS or Betacam tape decks. Digital video servers eliminate repetitive tasks allowing more time for creative work. If the focus of the post-production suite is to provide editing services, a video server will be used to support multiple editing systems and different compression schemes to store, retrieve and transmit a digitally compressed data stream especially if multiple suites and/or studios are involved. Digital video severs will minimize editing and post-production work.

Efficiency can be broken down further by the use of a digital video server. Video servers will ultimately eliminate the countless tape systems that support a post-production facility, thus converting the studio from a seamlessly tapeless environment to a digitally integrated studio. Access to digitally stored data will be fast and will require no search time -- completely eliminating the need to search a tape library in the next room. Multiple users can also access a video server supporting multiple streams without the need to wait for someone to finish with an editing or multimedia system.

Ad-Insertion. In a broadcasting studio, a digital video server may be used in most instances as an ad-insertion system that contains many digital clips that will be used in the course of a daily or weekly transmission with many frequent updates of its digital content. Compressed digital data is available to multiple users regardless of the simultaneous use of the same data stream without ever degrading video quality or performance of the video stream. A video server can efficiently serve multiple and redundant video streams with limited hands-on operation.

Video Server Profitability

Efficiency in the studio allows for managers to creatively produce more and generate a wider revenue stream for the studio. As an example, different video files can be stored on the same networked video server by different tapeless non-liner systems. This eliminates the need to purchase tape-based systems leaving more money in the bank for creative needs. Time is also maximized with a video server. Working with digital data eliminates the time spent searching physical tapes for a video clip or worrying about a tape-head getting humid. Just the availability of having money and time to create new programming and features can account for profit with a new client or an existing customer. Obviously, these are intangible and unforeseen capabilities of a video server but once implemented the efficiency factor turns into profit. At the same time, if the need arises that the same video file is needed for two different reports or programs a video server allows the user to access the video file without having to wait for it to become available. Individuals in the business of managing or directing a broadcast or production studio are accustomed to seeing a new system on the market every 12 to 18 months. Yesterday it was U-Matic ¾" based systems and later came S-VHS, BetaCam and now DVCPRO. Imagine having to retrofit a production studio every time a standard was introduced. Unfortunately, new equipment standards have deliberately closed shop for many producers and broadcasters because of the expense incurred in keeping up with television technology. Literally, millions of dollars can be spent on new hardware just to be looked at by a client. Digital video servers change the entire way a broadcast or production studio changes formats. Simply, it does not matter what formats are introduced in the market because video servers deal only with one format -- digital video, whether compressed or uncompressed. In the long run it saves money on upgrades of new equipment again allowing for more profits to return to the studio. Digital video servers concentrate the entire production studio around one central production system.

Studio Growth

Video servers can also add value to a post-production suite or broadcasting studio by growing with the needs of the suite or studio. Simple software or hardware upgrades are easily installed thus eliminating the need to purchase new equipment every 12 to 18 months. Initial stream delivery for a video server in a production environment will be small but will grow with the expansion of new business opportunities. As efficiency and profits increase by implementing a digital video server in a broadcast or production studio the need to grow will be more relevant. The task to grow becomes easier and will not be as expensive as a conventional tape-based or linear system studio. As new standards are introduced into the industry the only change in a digital video server will be the need to upgrade existing hardware or software to comply with those new standards. For example, if your current video server only deals with MPEG compression schemes and you would like to upgrade to DV standards, it is easy to add software that will deal with DV compression schemes without having to purchase new equipment to support those standards. In addition, the video server will still deal with previous compression formats. Not only are you upgrading the digital video server to serve different compression schemes but also are allowing the server to deal with older standards without spending too much money for the upgrade. With a couple of hundred dollars you have just upgraded a digital video server to handle growth and new industry standards. The amount of money saved, when compared to traditional ways of upgrading complete broadcast systems, can be quite significant.

In simple terms, it is very hard to keep current with the new technologies in the world of broadcasting and television. This is also true for the personal computer marketplace. However, the cost of retrofitting broadcasting stations and post-production suites is very expensive when compared to upgrading your computer at home. Cost for new television equipment can run in the millions of dollars for high-end post-production suites. Imagine spending that money every 12 to 18 months. It can simply make or break a post-production suite. Video servers add a new dimension by allowing the cost to be concentrated on a single digital system that will allow many different standards to be implemented. Because a video server is nothing more than a computer, it only needs software upgrades and, at times, hardware upgrades. Video servers grow with the different needs of the broadcast or production studio, without the need to spend millions of dollars on new retrofits.

MERGING DIGITAL TECHNOLOGIES

Digital video server technology can cross many boundaries. High demand for video servers will come from the broadcast industry, but there are other industries to consider as well. It seems the broadcast industry is merging with the Cable, Hospitality, Intranet and Digital Video Management industries in terms of similar and synergistic technologies.

Cable Television Industry

For example, video servers can serve the multiple stream requirements of the cable industry (CATV). The residential entertainment marketplace is increasing its demands for better on-demand video services. A high number of stream outputs is required to serve a wide cluster of home users with a variety of choices. These services must be fully interactive and provide responsive and quality video via many distribution and networking options such as the Digital Video Broadcast-Asynchronous Serial Interface (DVB-ASI). High-performance, multiprocessor video servers that meet real-time interactive requirements can deliver the absolute optimum quality of video-on-demand. Multiple requests to the same video file must also be an important function of the video server. At the home, set top boxes (STB) are needed to receive the encoded signal to be displayed on the television. Interactive control that includes select, play, pause, fast-forward and stop are also required.

Hospitality Industry

Another very suitable application for a video server is the hospitality industry. This primarily includes the hotel industry but also includes the healthcare and cruise ship industries. The requirements in this industry also include multiple stream requests. Today, a video server must be able to deliver exceptional video quality to hotel or hospital guests via existing broadband RF coaxial (COAX) cabling. The use of high-end decoders is needed as part of a video server system. Decoded signals must be sent from the video server to modulators that distribute them to the guestrooms. As in the cable industry, STBs are required at the users end for reception of the signal and for interactive control. Video servers would replace the limited viewing choices and racks of VHS systems in a hotel or hospital basement. Digital video technology will eventually change current COAX structures in the hospitality industry.

Intranet Training Industry

This market extends itself more to the corporate, governmental and educational institutions. Training is a major part of any industry. Video training has become increasingly important to transmit messages to employees, government officials and students. Video servers provide a means to deliver fast and responsive video data to many people at different times directly to their personal computers. The ease of training from the office computer or receiving yesterday's lecture from the dorm room provides convenient and necessary access to information. Video servers provide the necessary components for video delivery interactively. Requirements for this market include a user-friendly graphical user interface (GUI) at the PC client or user terminal.

Digital Video Management Systems

As can be taken from this paper, digital video transmission will be the way of the future – with applications ranging from home entertainment to university distance education. Digital video technology provides multiple industries with complete access and control for digital video content management. For example, advertising agencies will use video server technology to catalog their video and multimedia library for internal and client use. The surveillance and security industry will use digital video to monitor criminal activity at a place of business or home. Easy interactive digital access will be available without the need to search through VHS tapes for a certain spot on the tape. Video servers will also allow manufacturing plants to monitor automatic machines. In this way, if a machine ever malfunctions, the problem can be identified through easily stored digital video files. In another application, Digital Video Management Systems (DVMS) will allow training procedures to be recorded and stored for interactive training purposes, such as in the military. Easy access to digital video will only be a click away. One of the major requirements for the DVMS market will be the integration of high-end encoders for the digital capturing of real-time video data.

TECHNICAL CONSIDERATIONS

Digital video servers -- or video-on-demand servers as they have become known -- need to provide real-time capabilities for true high-end performance in a broadcast or production studio. A real-time computer system can be defined as a system that performs its functions and responds to external, asynchronous events within a predictable or deterministic amount of time. In addition, real-time systems must provide efficient interrupt capabilities to handle asynchronous events and high I/O throughput to meet the data handling requirements of time-critical applications. [3]

Digital video servers need to react predictably to a high number of system interrupts predictably from different users requesting different video files. Those requests need to be processed quickly and delivered through an output that can handle a large amount of high I/O throughput, such as MPEG and DV data along with interactive commands.

Is it NT or UNIX?

There have been many debates among industry analysts concerning the future of UNIX vs. general-purpose operating systems such as Windows NT®. The simple difference between these operating systems is the value that each OS puts into meeting system requirements of determinism, performance and resiliency. If unusual delays or schedule mishaps in the kernel are tolerable then an NT OS can be suitable for the application. This is unacceptable in a broadcast or production studio. As an example, if a user opens a window while a process is running and this process is interrupted, then a delay or a process with less priority in the OS has occurred. In other words, the kernel process was interrupted by a low priority task such as opening a window and thus the true data was either delayed while being processed or "dropped" from being delivered on the network. In a properly configured timesharing real-time environment, no task should wait indefinitely for a system resource [3]. In digital video transmission, data needs to be delivered with optimum predictability. A user cannot wait for a video frame to appear just because another user on the network opened a window frame or clicked on a mouse. Conversely, a user cannot afford to have "packets" of video information dropped.

In digital video delivery, as it is in current analog video delivery, drop-outs or paused satellite frames are unacceptable for the expected quality and performance of video systems and transmissions. UNIX-based real-time operating systems offer solutions for the aforementioned problems that are specific to digital video delivery in video servers. A real-time operating system such as the UNIX environment must handle critical tasks that must receive the system resources they need when they need them, without concern for the effect this may have on other executing tasks [3]. Tasks that require deterministic responses must be serviced at all times, such as the request for a digital video file from the video server without ever interrupting its process. A real-time computer system is typically controlling a process by: [3]

1. Recognizing and responding to events within predictable time intervals, and
2. Processing and storing large amounts of data.

The two descriptions above should define what a digital video server requires for predictable delivery of digital video files. This will ensure predictable delivery of video files to users or other network servers that request the digital video file from the video server including storage management. User requests for stored files for program use or interactive functions for video-on-demand applications must be received by the video server and serviced deterministically (meeting the delivery schedule on time without interruptions) to the requestor without being interrupted by any other process. Specifically, operating systems that offer in-kernel buffer management will offer the best direct input/output response of any video server system without interrupt delay from low priority tasks. Performance will be optimized for true digital video delivery with optimum quality. A real-time operating system will also offer the resiliency of an OS system, which will continuously guarantee the integrity of the file system regardless of any system crashes. File systems in a real-time OS are unlikely to be damaged or lost in case of true havoc in a system shutdown. An unexpected shutdown can destroy video files and many hours of production work in a general-purpose operating system. This, in turn, can result in major revenue loss for any firm in the Broadcast, Cable, Hospitality, Intranet, or Digital Video Management markets.

The technical differences between NT and UNIX make it clear that a real-time operating system is the choice for a true digital video server. With a real-time operating system, stored digital video files will always be available for access without delay, delivered deterministically with quality and performance never degraded. General purpose operating systems can be used at a lower level of expected performance and will act well with certain faults while a real-time kernel can be used for high-end response processes such as digital video delivery, without ever having to worry about dropped frames or unrecoverable system crashes. Predictable system responses and video availability and quality are very important for the broadcast professional. Video server operating systems that include video-specific enhancements to handle the unique characteristics of video data are desirable.

Capacity, Content Storage, and Networking Requirements [5]

In establishing buying or implementation criteria for a video server the user must define video server capacity, content storage and networking requirements.

Video Server Capacity. In defining video server capacity requirements, the user needs to determine the number of video streams to be concurrently delivered or processed via a digital video server. Equally as important, the specific type of video streams to be delivered needs to be specified. For example, a sports short-subject content file with MPEG2 content encoded at 6 Mbits per second requires more video server capacity than a documentary content file with MPEG1 content encoded at 1.5 Mbits per second.

Content Storage. A critical component of a video server, content storage is often the largest single cost factor and sometimes can exceed 50 percent of the total cost of the video server system. The video server buyer must define the number of content titles with the average length of each content title – or, alternatively, the total number of hours of stored content. The type of storage interface is also important (i.e., IDE, SCSI, Fibre Channel). In broadcast or production applications, Fibre Channel-Arbitrated Loop (FC-AL) storage interfaces are the most common since multiple systems can access the same central storage device without degrading delivery of the content.

As with video server capacity, content storage is highly dependent upon the type of content. MPEG2 and DV compression and higher encoding data rates will provide higher video quality than MPEG1 or Wavelet compression and lower encoding data rates, but will also require higher content storage capacity. Variable bit rate encoding can significantly decrease content storage requirements by an average of 40 percent when compared to constant bit rate encoding, depending upon the specific video file. Also, variable bit rate encoding delivers consistent and generally higher video quality when compared with constant bit rate encoding.

Networking. By definition, a video server must deliver video streams through a distribution network. Accordingly, network functionality and performance requirements are essential buying criteria. Obviously, the video server's network throughput must be able to support the number of video streams to be delivered.

However, networking requirements are considerably more complex than merely network throughput. If client devices with decoding capabilities are to be used and digitally encoded video streams are to be distributed, the network requirements must include the type of network (i.e., ATM or 100 BaseT), the type of physical connection (i.e., SCSI), and the network protocol (i.e., TCP/IP).

System Architecture Features [5]

To select the optimum video server the desirable system architecture features for specific applications should be determined. For example, independence of video streams is a desirable architectural feature for video servers supporting multiple video streams – particularly in a multiple user broadcast or production environment. Independence of video streams allows video content to be stored and delivered in the most cost-effective manner.

For acceptable levels of video quality, action and sports video titles might require MPEG2 data encoded at 6 Mbits per second or higher, while a documentary might only require MPEG1 data encoded at 1.5 Mbits per second. If a video server does not support independent video streams, the user must choose between either sacrificing video quality or needlessly increasing content storage costs.

Input/Output performance is another system architecture feature to consider. The task performed by a video server is input/output-intensive rather than computationally intensive. While the computer processing activity is minimal, video content must be continuously accessed from content storage and distributed to the network. Multiple, independent, and high-performance input/output channels are much more important than the CPU microprocessor.

General-purpose file systems are optimized for short and random data access rather than large sequential data access. A video file system that can support the large file sizes associated with video content files and optimize access to video content files is preferable. The buffered input/output of general-purpose operating systems actually degrades video data throughput. A video server that includes direct input/output as a complement to buffered input/output optimizes video data throughput.

Finally, digital video server users should look for a scalable system with an open architecture. These last features protect the buyer's investment in hardware, system and networking software, and application software. In the event of system growth, a video server with a scalable system architecture will accommodate expansion. Furthermore, a video server with an open system complying with agency and accepted industry standards will ensure supplier independence.

Client/Server Considerations [5]

Digital video server applications are client/server applications. That is, a video-on-demand application consists of a video server program (or programs) that cooperates with a client program (or programs) executing on the client device. The client device is generally responsible for the user interface and video display.

For example, personal computers and intelligent set-top boxes with television monitors are client devices. If the server and client program are to cooperate, they must share a common application program interface (API). That is, the program interface to the server program must be the same as the program interface to the client program. As an example: The Digital Audio/Visual Council (DAVIC) has defined a Digital Store Media-Command & Control (DSM-CC) API. When establishing the video server requirements, the application program interface must be specified.

Special Considerations for Unique Applications

Digital video servers should support dynamic content management. In dynamic content management, the video server monitors in real-time the number of video streams supported by any single video content title. When the limit is approached, the video server dynamically replicates the video content title. Once the dynamically replicated content is no longer required, the video server deletes the replicated copy.

A final special consideration is fault-tolerance. In many applications, specifically in a broadcast and production studio failure of the content storage subsystems is an obvious concern. If the storage subsystem fails, content cannot be accessed and delivered. One approach is to implement content storage with Redundant Array of Independent Disks (RAID) technology, which will provide protection against failure of any single storage subsystem component. However RAID disk array technology is costly.

Another more cost-effective approach is to implement a hierarchical storage subsystem with dynamic content management. First-level devices would be implemented as a JBOD ("just a bunch of disks") array. If dynamic content management encountered a failure in the JBOD array, the afff F!"ected video content title can be dynamically re-created from the second-level content storage devices. Fibre Channel Arbitrated Loop (FCAL) can be implemented for both a JBOD and RAID storage configuration to support multiple requests from users using the same storage device in a broadcast or production studio. Figure 2 shows a checklist for selecting the optimum digital video server that will serve the many different functions of a broadcast and production studio including the Motion Picture, Cable, Hospitality, Intranet and Digital Video Management industries.

CONCLUSION

Ideally, a digital video server is the most complete solution to the massive storage and reproduction requirements of the broadcasting, production and motion picture studios beyond the year 2000. Video servers implement a cost savings solution over current analog tape systems and static storage devices. Video servers provide countless new options to many decision-makers in the business of broadcasting and production studios, including all the emerging technology markets such as the Cable and Hospitality industries and the Intranet and Digital Video Management Systems markets. The video server company that offers the most open standard system with an operating system

Optimal Digital Video Server Features and Requirements

- ✓ Supports multiple compression schemes
- ✓ True real-time operating system
- ✓ High I/O throughput capabilities
- ✓ Multiple stream capability
- ✓ Compliant with open standards
- ✓ Client device support (API)
- ✓ Encoder/decoder device support
- ✓ ATM/Fast Ethernet support
- ✓ Fibre channel interface
- ✓ Large storage availability

Figure 2. Checklist for digital video server selection.

that delivers true real-time performance with the features outlined in this paper will be the true winner. This will help managers and executives in the broadcast, production and other industries prepare for the new digital millenium while making their studio efficient, profitable and expandable – without overspending on new components. Only those broadcasters who implement video servers will achieve the levels of efficiency and profitability needed to compete in the marketplace of the new millennium.

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Comparative Analysis of Full Bandwidth versus Compressed HDTV Routing

Paul Berger
CBS Inc.
New York, New York

Abstract

Broadcasters are now facing the dilemma of how to transition to digital distribution of HDTV signals. Central to this decision process is the need for modification or adaptation of an existing plant to handle the HDTV signal. For many installations a key consideration of the plant design is the possibility of employing an existing central router and the resultant need for use of signal compression. This study compares various routing scenarios in an effort to identify issues involved and determine the benefits and limitations of compressed versus full bandwidth routing configurations.

Introduction

Broadcasters are now facing the dilemma of how to transition to digital transmission of HDTV signals. Central to this decision process is the need for modification or adaptation of an existing plant to handle the HDTV signal. This issue is extremely complex because system concepts and equipment designs are still in the formative stages, there is limited availability of hardware, costs are high and the FCC timetable for conversion to DTV is very aggressive. For many installations a key consideration of the plant design is the integration of an existing central router and the resultant need for use of signal compression. This is a fact of life since most if not all existing routers do not support the full bandwidth data rate required for HDTV. This study compares various routing scenarios in an effort to identify the issues involved and determine the benefits and limitations of compressed versus full bandwidth routing configurations.

System Configurations

Typical full bandwidth and compressed routing systems are shown in simplified block diagram form in Figures 1 and 2. Figure 1, showing the uncompressed routing system is conceptually the simplest of the proposed schemes. It illustrates the required routing for a single re-entry control room. In this typical example, it is a sports re-entry control room where origination of the on-air signal switches between the highlights control room and the sports coordination control room. This re-entry process is generic and applicable to many other on-air activities such as news and entertainment. Since control room switching takes place during program, a frame delay must be inserted in the incoming signal path to the sports control room to avoid timing disturbances. From a signal handling point of view it is a very attractive arrangement since signal processing is minimal, one decompressor at the incoming receive point and one compressor at the outgoing distribution point. However, the system employs a full bandwidth router capable of handling the 1.5 Gbps SMPTE 292M signal connections between the router and its sources and destinations. Although system latency is minimal, it is finite and cannot be ignored since delays are introduced by decoders in the remote input paths, encoders at the signal origination points and frame delay units needed for timing router re-entries. As shown the "plant" delay in the signal path consists mainly of two frame delay units plus decoder and encoder delays and represents a "minimum delay scenario".

Figure 2 illustrates the same studio configuration and signal routing as shown in Figure 1, with the only significant changes being use of a "medium" data bandwidth router and compressed signal distribution. It is assumed that to avoid the necessity of rewiring the plant with fiber interconnections the encoders and decoders will be located at the source and destination terminal points of the system. This approach has the obvious advantage that an existing medium bandwidth digital router can be used for signal distribution and routing. Also, there is no need for recabling since existing "copper" cables can be employed for signal interconnections. Although at first glance this scheme appears to be very attractive and cost effective, it does introduce a number of significant technical problems which the systems designer must cope with. In this system each destination must be equipped with a decoder which, based on currently available products, introduces a one or more frame delay of the signal. Similarly, the studio re-entries and other router inputs must be equipped with an encoder which also introduces a one or more frame delay of the signal. Thus, a single pass from the router through a studio back into the router is delayed by two or more frames plus any studio processing delays.

Issues

Comparing the system architecture options described above, raises a number of significant issues for the system designer. How does the cost of the two systems compare? Is full bandwidth routing with fiber optic interconnections more cost effective than a "medium" bandwidth router with encoders and decoders at each terminal point? How does the delay in the signal path affect On-Air operations and what operational limitations does the delay impose? If compressed routing is employed, how does concatenation of codecs affect video quality? Does this impose any limitations on further processing of the video signal? Does the format of the compressed bit stream permit seamless splicing of the signal? Can video effects, that are used today, such as fades, keys, wipes and DVE moves be accomplished in the compressed domain? If it is planned to use an existing router for HDTV signal distribution, can it support the additional I/O required for the HDTV signals or must it be expanded? Can it be partitioned and handle the dual bit rates needed for parallel routing of HDTV and SDTV signals?

Compression

Although use of video compression appears to offer an opportunity to reduce system costs, this apparent benefit brings with it some significant concerns for the ability to further process the video while maintaining adequate signal quality. Naturally, the best quality video is derived from the highest quality video input. This means that applying compression to previously compressed signals and/or noisy signals may result in unacceptable signal quality. As shown in Figure 2, there are at least three encode/decode compression cycles in the signal path from the remote input to the plant output. Another compression cycle must be added to account for the compression at the source end and decompression at the station end. Given the fact that the signal path passes through four cascaded compressors, steps must be taken to minimize the loss of quality from one generation to another. First, it is important to insure that the process begins with a high quality low noise signal. If necessary, to prepare for encoding, the signal should be filtered and/or pre-processed to remove unneeded information. To preserve signal quality, the number of different compression schemes (i.e. JPEG, MPEG-1, MPEG-2, Sony HDCAM, DVC-PRO-HD, Panasonic AJD-2000) in the path must be minimized and in general high bit rates should be used to allow for use of short Group Of Pictures (GOP's) needed for editing and signal manipulation. Since, signal processing often includes effects which introduce some spatial and temporal shifts, signal to noise may be reduced by as much as 3db.[1]

In the uncompressed routing system shown in Figure 1, the signal passes through only one compression cycle. This inevitably results in higher signal quality and fewer concerns relative to the concatenation of multiple compression schemes. Preservation of signal quality is reduced to an issue of performance of the contribution and distribution codecs and the amount of intra-plant processing. For best performance, the same codecs should be used in the incoming and outgoing paths. Although this may result in some compromises in the selection codec parameters, in general the needs of the network and the affiliate stations are similar. Each requires use of high bit rates and short GOP's to maintain signal quality, minimize latency, permit easy signal manipulation and bitstream splicing. In both cases, compressed and uncompressed routing, one must consider the interface at the distribution point to the affiliate stations. The choice of compression scheme for this application must be made with careful consideration given to the upstream compression systems (i.e. contribution link, compressed recording and where appropriate compressed intra-plant distribution). Also data bandwidth of the distribution link must be considered from both an economic and practical point of view. To avoid concatenation of multiple compression schemes, the appropriate choice of "Distribution Compression System" would be to select the same system as that used for contribution links. This minimizes the number of different compression schemes in the program distribution chain and will lend support to establishment of an industry standard for distribution/contribution links. Similarly, if a common compression method is used for recording and intra-plant distribution the concatenation of differing compression schemes within the plant will be minimized. However, this requires the cooperation and agreement among recorder manufacturers on a common compression standard. At the present time achieving this end may not be possible. However, as systems evolve and users and manufacturers begin to work on solutions to HDTV system problems it is not unreasonable to expect standardization on the choice of an intra-plant/recording compression algorithm.

Signal Latency

Systems which employ compression processing introduce signal delays that are directly dependent upon the GOP structure of the compressed signal. Use of long GOP structures creates a dependency of some frames on other frames which results in encoding/decoding delays that translate into signal latency. There may also be differential delay between the video and the audio which can change with variations in the signal path in compressed systems. Therefore, careful attention must be paid to maintenance of video/audio synchronization with introduction of additional delay units to account for timing differences. In practical systems the total delay must be of limited duration to allow for live news, sports interviews and two way conversations between field reporters and studio anchors. This can only be achieved by minimizing the number of codecs in the signal path and careful selection of the compression parameters.

In the diagrams shown, the delays through the systems are one compression cycle and two frame delay units for Figure 1 and three compression cycles and one frame delay unit for Figure 2. Using one frame of delay (0.033 sec) for each compression cycle, the system latency is three frames for Figure 1 and four frames for figure 2. This translates to a delay of 0.1 seconds without compression and 0.13 seconds with compression. Adding an additional .25 second delay for satellite transmission (see Figure 3) yields a total path delay in the range of 0.35 seconds to 0.38 seconds. This overall system delay is on the borderline of acceptability and may interfere with the spontaneity of live interview segments. To avoid additional complications due to latency of program feeds, separate telco circuits may be used to provide return audio feeds to the originating source (see Figure 3). This minimizes the latency of the program audio at the originating venue and improves spontaneity of interview or conversational responses.

If three frames of delay (0.1 sec.) are used for each compression cycle the system latency is five frames for Figure 1 and ten frames for Figure 2. This translates to a delay of 0.17 seconds for figure 1 and 0.33 seconds for Figure 2. . Adding an additional .25 second delay for satellite transmission (see Figure 3) yields a total path delay in the range of 0.42 seconds to 0.58 seconds. This overall system delay is beyond the acceptable range and will very likely interfere with the spontaneity of live interview segments.

However, achieving even this level of system latency requires very carefully chosen codec parameters which may not yield the best video quality obtainable at a given data rate. Since it is unlikely that the codec delay can be reduced below one frame, alternative transmission systems may have to be employed to reduce latency. With the exception of ENG feeds, where landlines may not be readily available or accessible, fiber optic transmission may be used in place of satellite circuits to reduce latency of live feeds.

Cost
Costs for the two routing systems have been compared using the following simplifying assumptions:

- Router size is 32 x 32 square
- Cost per router signal path equals router cost ÷ number of busses
- Full bandwidth system employs fiber optic signal connections
- All router connections are 500 feet long
- The full bandwidth per buss cost is based on the list price of an existing first generation product built by a leading manufacturer

Original (97)
COST COMPARISON
($)

System	Router Buss	+	Interconnect Costs: Fiber Interface	+	500' Cable	+	Audio/Video Muxing	+	Codec	=	Path Cost	Interconnect Cost
Full BW	8,125		6,000		60		6,000		-		20,185	12,060
Comp.	530		-		127		-		50,000		50,657	50,127
Diff. B/(W)	7,595		6,000		(67)		6,000		(50,000)		(30,472)	(38,067)

Revised (98)
COST COMPARISON
($)

System	Router Buss	+	Interconnect Costs: Fiber Interface	+	500' Cable	+	Audio/Video Muxing	+	Codec	=	Path Cost	Interconnect Cost
Full BW	2,084		1,675		60		6,000		-		9,819	7,735
Comp.	530		-		127		-		50,000		50,657	50,127
Diff. B/(W)	1,554		1,675		(67)		6,000		(50,000)		(40,838)	(42,392)

Both cost analyses clearly show the codec to be the most significant cost element of the compressed system. Consequently, even if an existing digital router or any portion of it is used for compressed signal distribution, system costs will still be driven by the codec cost. From the above analysis it is clearly seen that system cost varies directly with codec cost and system size.

There is much less significance and very little uncertainty in the other cost components of the system. However, with improving technology and mass production of codecs, system costs are very likely to decline over time. If agreement can be reached on a set of compression standards for contribution, production and distribution; there will be increased competition among codec vendors which will also help to drive prices down even more quickly.

Uncompressed system costs are most sensitive to the cost of the router path and fiber optic interfaces. In 1997 router technology was shown that can handle the 1.5Mb/sec. data rates required for HDTV. With the growing need for HDTV equipment and increased development activity on the part of manufacturers, costs for uncompressed systems have already begun to decline as shown in the revised cost comparison. However, costs for compressed systems have not changed materially since a suitable codec has not yet been introduced. Fiber optic interfacing is the other major cost component of uncompressed systems. Most signal distribution systems currently in use employ SDI coaxial interconnections which are best suited for medium data rate short run interconnections. Use of fiber optic signal interconnections is required to support longer cable lengths and/or high data rates. Since intra plant fiber optic transmission systems are not widely used at present, costs have remained relatively high. However, because of its ability to transport high data rates over long distances, adoption of fiber optics for signal distribution would appear to be a natural outgrowth of the transition to HDTV. This projected increased use of fiber connections has already drawn the attention of equipment manufacturers and has resulted in lower projected costs for the optical interface equipment used in the 1998 cost analysis.

Surprisingly in the original 1997 analysis, for the 32 x 32 system, per buss costs are lower for the full bandwidth implementation than the compressed system. However, the number of crosspoints in the router increases as the square of the router size. Thus, there are four times as many crosspoints in a 64 x 64 than a 32 x 32. Consequently, one can expect that at a minimum, per buss costs will at least double when the router size is doubled (four times the number of crosspoints divided by two times the number of busses). Excluding the router costs, the cost difference between the compressed and uncompressed systems is approximately $38,000 per buss, in favor of the full bandwidth router. The following table shows the estimated increase in buss cost as router size is increased.

ROUTER COST COMPARISON (97)
$/BUSS

Router Size	Full BW	Compressed	Difference
32 x 32	8,125	530	7,595
64 x 64	16,250	1,060	15,190
128 x 128	32,500	2,120	30,380
256 x 256	65,000	4,240	60,760

From this table it can be seen that as router size increases from 128 x 128 to 256 x 256, the per buss cost difference increases from approximately $30,000 to more than $60,000. The Cost Difference/Buss for the 256 x 256 exceeds the $38,000 difference obtained from the non-router related differences of the earlier cost comparison above. Therefore, one may conclude that, based on 1997 cost projections, for systems sizes up to 128 x 128 a full bandwidth system is more cost effective than a compressed system.

Reexamining the router cost comparison using the latest cost projection (below) shows a significant reduction in the differential between the compressed and full bandwidth router. Given the reduced router costs with no comparable reduction in codec cost it is difficult to justify the use of compressed routing in any configuration. Based on the latest cost projection the codec cost must be less than $6,000 to justify consideration of compressed HDTV routing.

ROUTER COST COMPARISON (98)
$/BUSS

Router Size	Full BW	Compressed	Difference
32 x 32	2,084	530	1,554
64 x 64	4,168	1,060	3,108
128 x 128	8,336	2,120	6,216
256 x 256	16,672	4,240	12,432

Recognizing that costs for the full bandwidth router are based upon the initial implementation of new technology, as one might expect the per buss cost has declined significantly due to competion and further development. Since the per buss cost for the full bandwidth router has dropped by seventy-five percent without a comparable reduction in codec costs, the full bandwidth solution has become the preferred solution in all cases.

Up to this point in the cost analysis, the cost of compression/decompression units have remained fixed at $25,000 for each function for a total of $50,000. In the future, as a result of competition and use of VSLI circuitry, codec prices are expected to drop. Following similar reasoning, the table of Codec price points vs router size below, shows the approximate codec cost for which compressed distribution is more cost effective than full bandwidth distribution for a various router sizes. Given the significant reduction in projected router costs for 1998 combined with the technical concerns of compressed routing it becomes extremely difficult to justify use of compression in any case.

CODEC COST SENSITIVITY ANALYSIS (97)

Router Size	Cost Difference/Buss (Full BW- Comp.) ($)	Codec Cost ($)
32 x 32	7,595	7,500
64 x 64	15,190	15,000
128 x 128	30,380	30,000
256 x 256	60,760	60,000

CODEC COST SENSITIVITY ANALYSIS (98)

Router Size	Cost Difference/Buss (Full BW- Comp.) ($)	Codec Cost ($)
32 x 32	1,554	1,500
64 x 64	3,108	3,100
128 x 128	6,216	6,200
256 x 256	12,432	12,400

Thus, based on the current 1998 projection for a 128 x 128 configuration, compressed routing is preferred when codec costs are less than $6,200. Given this result, in the absence of a very low cost compressor, it would appear that most system designers will be inclined to employ full bandwidth routing.

Recorders

At the present time there are only two full bandwidth HDTV recorders available. The cost of these recorders ranges from $370,000 to $500,000 per machine. Obviously, this extremely high cost presents a significant barrier to their use in a practical broadcast plant. However, there are now at least two compressed HDTV recorders on the market which could be used for broadcast operations. As compared to the cost of a current high end digital broadcast recorder ($83k - $85k), these compressed DTV recorders sell for a modest premium and offer a reasonable option to the system designer. But, since these machines employ an intermediate 360 Mb/sec. compression scheme, they create the possibility for introduction of concatenation related compression artifacts in the video signal. Also, at the present time specific recorder compression schemes are proprietary to specific machine manufacturers. Thus, if the machine is used in a system which employs compressed distribution that differs from the machines native compression, a separate codec is needed to handle its I/O requirements. This increases system costs and adds latency to the I/O path of the machine which must be compensated for in the system design. The need for codecs for machine interfacing becomes extremely critical when the machine is used in an application such as editing or real time playback of recorded signals. For best results with minimum complications, the same compression algorithm should be used in both the system and the tape machine. However, with the exception of one manufacturer, there is no commercially available hardware i.e. free standing codecs, which match the tape machine compression algorithms and there is no intermediate compression standard defining the interface between the tape machine and the plant signal sources and destinations.

In a baseband 1.5 Gb/sec. distribution system, the problems relating to the use of compressed recorders are greatly simplified. Signal impairments due to concatenation of compression schemes is limited to concatenation of the decoder and encoder at the input and output of the plant with the codec employed in the recorder.

This can be further complicated by the use of varying recorder types with differing compression algorithms. However, with careful control, the number and variety of compression cycles applied to recorded video can be minimized and thereby optimum video quality is maintained. Naturally, the best results are obtained with the fewest compression cycles and use of a single "standardized" compression algorithm. In the case of real time playback systems or editing the output of the recorder can go directly to air or the editor since there is no signal latency due to use of compression in signal routing.

Processing and Manipulation

Uncompressed video can be processed and manipulated without concern for GOP structure, chroma sub-sampling and introduction of signal latency. On the other hand **compressed video cannot be operated on unless it is decompressed.** This translates to addition of delay for the decompression/compression cycle needed to perform the simplest key, cut, fade or other effect. Although means of splicing compressed bitstreams has been demonstrated, it is very cumbersome and restrictive since splicing can only be done on specific frames and temporal alignment of the bitstreams to be spliced is required. Thus, the granularity of splices is limited by the GOP and may not match production requirements for a switch at a given point in time. Additionally, if the compressed bitstream incorporates chroma sub-sampling, i.e. 3:1:1 or 4:2:0 sampling as employed in the ATSC 19.3 Mb/sec video, **the reduced video quality significantly limits the ability to further process this signal** even after decompression. Resolution of vertical chroma transitions is significantly reduced and is observed on a monitor as a softening of the vertical transitions in the color bar signal. This reduced resolution limits the ability to perform high quality chroma keying, color correction, and digital video effects. On the other hand, manipulation of uncompressed video with 4:2:2 chroma sampling is not limited by these concerns.

Summary

There are many issues relating to the design of an HDTV production facility of which the use of signal compression for transmission, routing and recording is among the most important. Listed below in tabular form is comparison of the major concerns that must be addressed when considering whether or not to employ compressed routing in the system design.

Issue	Compressed	Full Bandwidth
Signal Latency	Increases with each pass through the router, studio, or codec	Minimizes signal latency
Signal Quality	Reduced by each code/decode cycle and concatenation of compression algorithms	Unaffected by signal path
Cost	High, due to codec costs	Router is largest cost element
Signal Paths	Traditional Co-axial Cable	New Technology fiber optic cables
Interoperability	Devices may not interface directly because of variations in compression algorithms	SMPTE 292M standardized interface
Data Handling	Requires special provisions for handling data such as time code, captioning, program ratings, etc.	Data can be carried in the uncompressed bit stream
Audio/Video Synchronization	Requires careful housekeeping to maintain synch-ronization	Both signals embedded in one bit stream in real minimizes differential delays
Recording	Easy to record. However, currently there is no standard for recorder com-pression. Requires codec to match I/O unless system compression matches tape machine	Only two, costly full bandwidth recorders are available. Matches I/O on compressed machines
Processing & Manipulation	Must be de-compressed to manipulate or process	Easy to process

Conclusion

Based on both the cost and complexity of a compressed routing system, the uncompressed system appears to be the most attractive option. It avoids the latency, signal impairment and codec cost issues, which are so very difficult to cope with. Problems with equipment interfacing are minimized because the signal interface is standardized. The bit stream is easily processed i.e. cuts, fades, wipes, keys, edited or logos inserted without the need for decompression. Other vertical interval data services such as captioning, rating codes or proprietary data can be easily inserted and removed from the bit stream. The system is also future proof, in that as compared to the compressed system it is far less likely that the uncompressed standard will change than a given manufacturers proprietary compression algorithm. Given the FCC timetable and the potential difficulty of achieving consensus among manufacturers it is extremely unlikely that a standard could be passed and equipment built in time for use at the start of HDTV broadcasting. Consequently, for the foreseeable future, an uncompressed facility should serve the broadcaster extremely well. The only potential for change may result from the need to expand or improve, but this is not due to changes in standards or obsolescence. However, construction of the uncompressed system may require the use of fiber optic transmission equipment. Up to now fiber technology has not seen extensive use in broadcast plants. But, since it is stable, well developed and widely used by the telco and data communications industry it should not prove to be a problem for the broadcaster. Also, the widespread use of fiber optics will help to insure reasonable costs for the fiber hardware.

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FIGURE 1

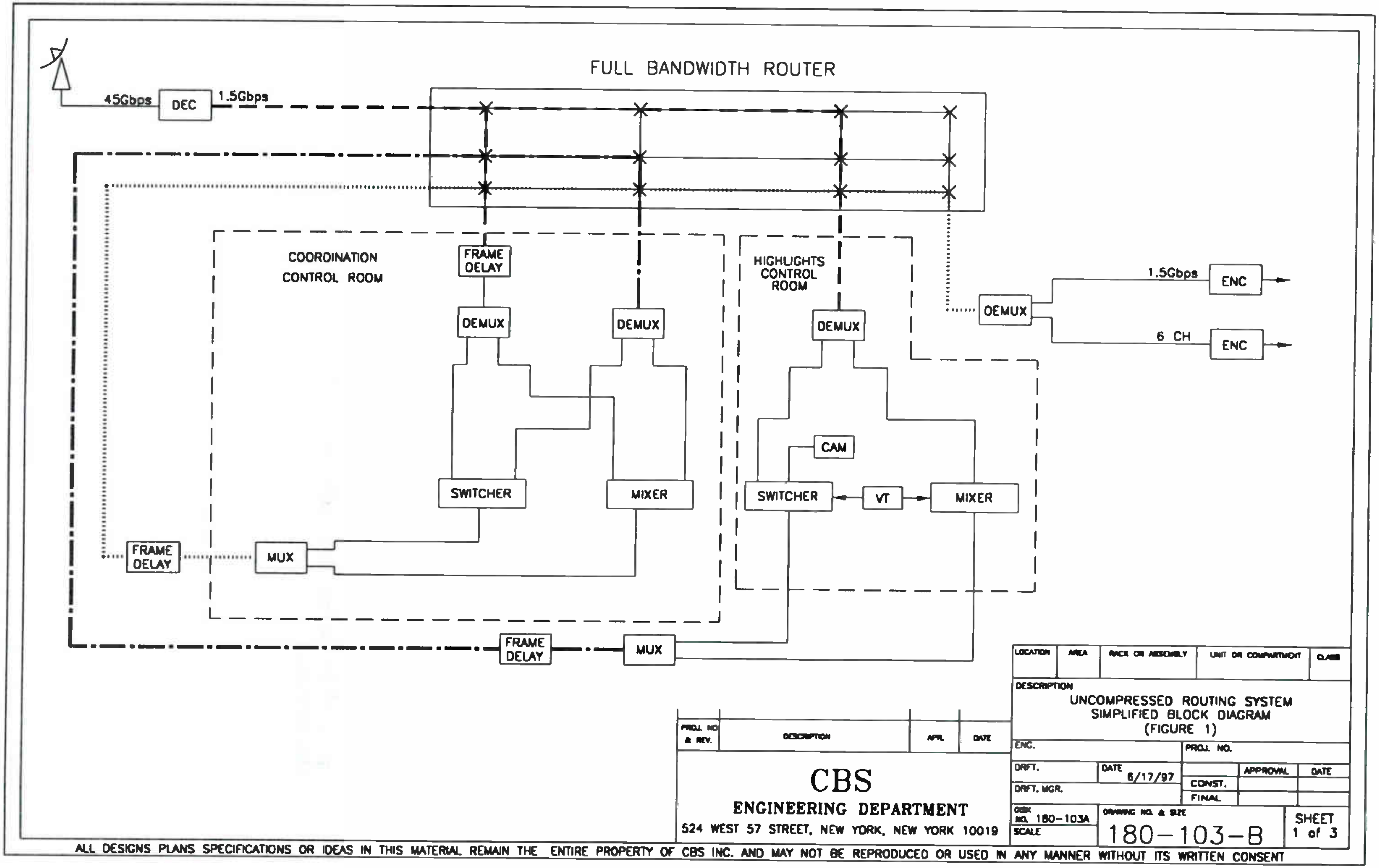
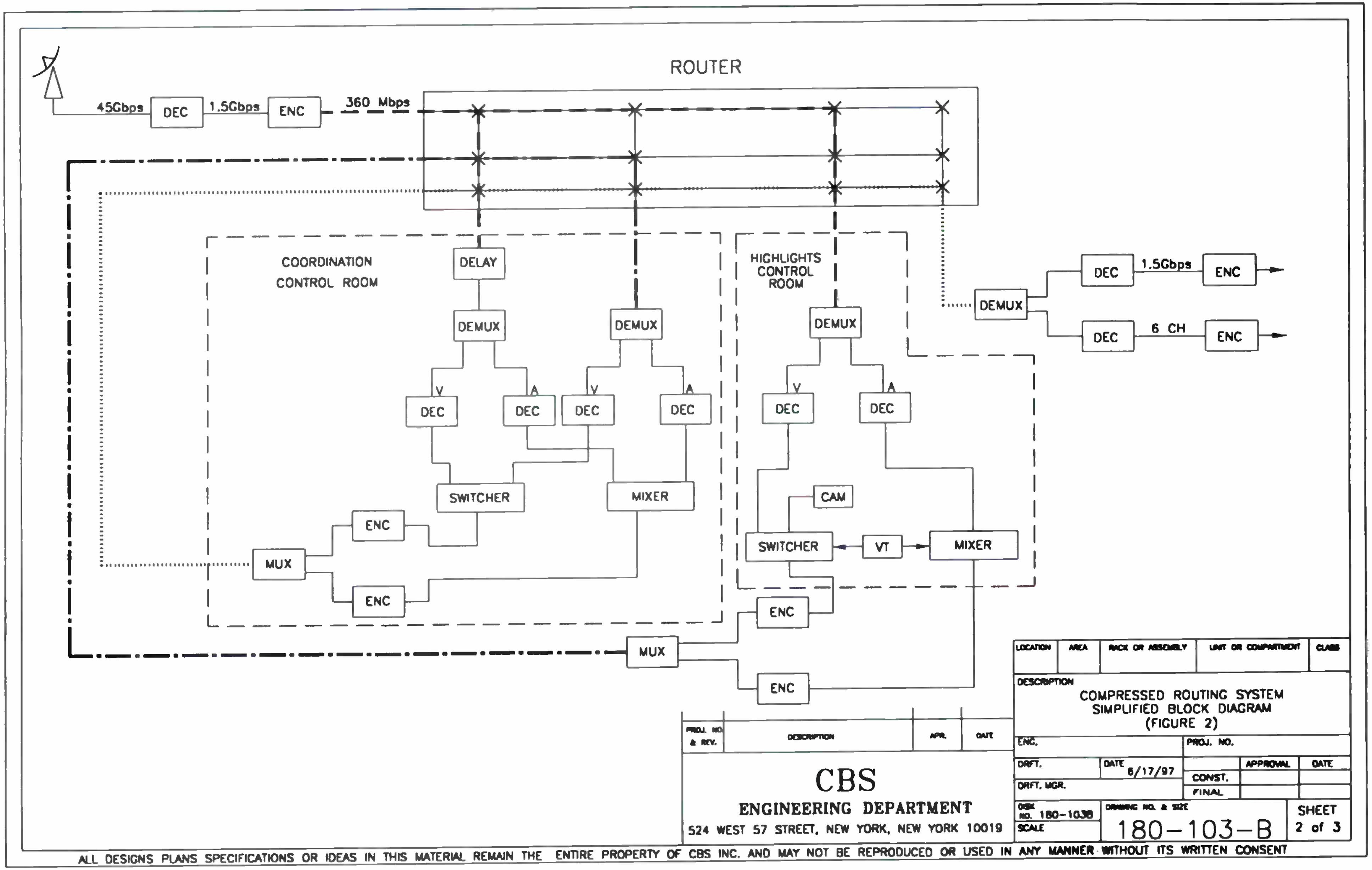
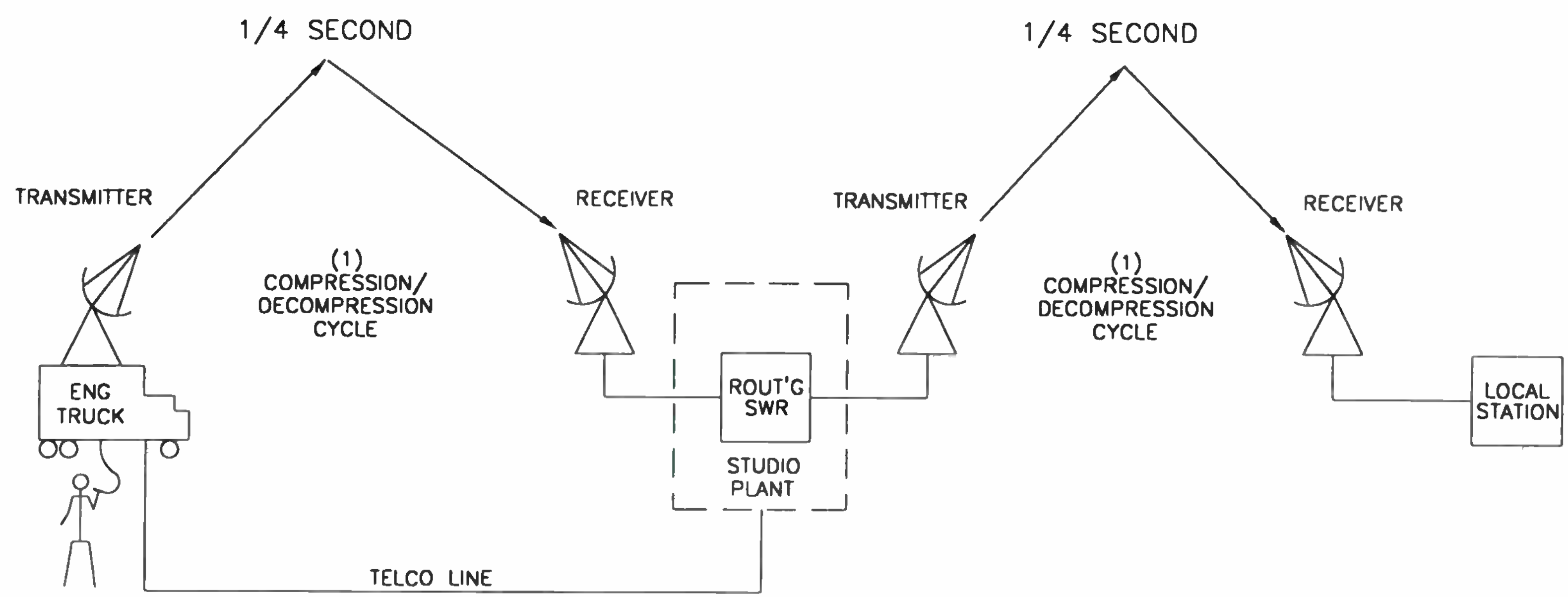

FULL BANDWIDTH ROUTER
45Gbps
DEC
1.5Gbps
COORDINATION CONTROL ROOM
FRAME DELAY
DEMUX
DEMUX
SWITCHER
MIXER
FRAME DELAY
MUX
HIGHLIGHTS CONTROL ROOM
DEMUX
CAM
SWITCHER
VT
MIXER
DEMUX
1.5Gbps
ENC
6 CH
ENC
FRAME DELAY
MUX
LOCATION
AREA
RACK OR ASSEMBLY
UNIT OR COMPARTMENT
CLASS
DESCRIPTION
UNCOMPRESSED ROUTING SYSTEM
SIMPLIFIED BLOCK DIAGRAM
(FIGURE 1)
PROJ. NO. & REV.
DESCRIPTION
APR.
DATE
ENG.
PROJ. NO.
DRFT.
DATE 6/17/97
APPROVAL
DATE
CONST.
DRFT. MGR.
FINAL
CBS
ENGINEERING DEPARTMENT
524 WEST 57 STREET, NEW YORK, NEW YORK 10019
DISK NO. 180-103A
SCALE
DRAWING NO. & SIZE
180-103-B
SHEET 1 of 3
ALL DESIGNS PLANS SPECIFICATIONS OR IDEAS IN THIS MATERIAL REMAIN THE ENTIRE PROPERTY OF CBS INC. AND MAY NOT BE REPRODUCED OR USED IN ANY MANNER WITHOUT ITS WRITTEN CONSENT

FIGURE 2



ROUTER
45Gbps
DEC
1.5Gbps
ENC
360 Mbps
COORDINATION CONTROL ROOM
DELAY
DEMUX
DEMUX
V
A
DEC
DEC
DEC
DEC
SWITCHER
MIXER
ENC
MUX
ENC
HIGHLIGHTS CONTROL ROOM
DEMUX
DEC
DEC
CAM
SWITCHER
VT
MIXER
ENC
MUX
ENC
DEMUX
DEC
1.5Gbps
ENC
DEC
6 CH
ENC
LOCATION
AREA
RACK OR ASSEMBLY
UNIT OR COMPARTMENT
CLASS
DESCRIPTION
COMPRESSED ROUTING SYSTEM
SIMPLIFIED BLOCK DIAGRAM
(FIGURE 2)
ENG.
PROJ. NO.
DRFT.
DATE 6/17/97
APPROVAL
DATE
CONST.
DRFT. MGR.
FINAL
DISK NO. 180-1038
DRAWING NO. & SIZE
SHEET 2 of 3
SCALE
180-103-B
PROJ. NO. & REV.
DESCRIPTION
APR.
DATE
CBS
ENGINEERING DEPARTMENT
524 WEST 57 STREET, NEW YORK, NEW YORK 10019
ALL DESIGNS PLANS SPECIFICATIONS OR IDEAS IN THIS MATERIAL REMAIN THE ENTIRE PROPERTY OF CBS INC. AND MAY NOT BE REPRODUCED OR USED IN ANY MANNER WITHOUT ITS WRITTEN CONSENT

FIGURE 3



NOTES:

USING ONE FRAME OF DELAY FOR EACH CODEC YIELDS:
UNCOMPRESSED STUDIO PLANT PATH = 1/4-SEC +0.10 SEC PLANT = 0.35 SEC.
COMPRESSED STUDIO PATH = 1/4-SEC +0.13 SEC PLANT = 0.38 SEC.
AND WITH THREE FRAMES OF DELAY FOR EACH CODEC:
UNCOMPRESSED STUDIO PLANT PATH = 0.42 SEC.
COMPRESSED STUDIO PATH = 0.58 SEC.
(REFER TO PAGE 6, PARAGRAPHS 2 & 3, FOR EXPLANATION).

PROJ. NO. & REV. | DESCRIPTION | APR. | DATE

CBS
ENGINEERING DEPARTMENT
524 WEST 57 STREET, NEW YORK, NEW YORK 10019

LOCATION | AREA | RACK OR ASSEMBLY | UNIT OR COMPARTMENT | CLASS

DESCRIPTION
SIMPLIFIED TWO-WAY INTERVIEW PATH
[FIGURE 3]

ENG. | PROJ. NO.
DRFT. | DATE 6/17/97 | APPROVAL | DATE
DRFT. MGR. | CONST.
FINAL
DISK NO. 180-103C | DRAWING NO. & SIZE 180-103-B | SHEET 3 OF 3
SCALE

Interactivity and Data Broadcasting for Television

Tuesday, April 7, 1998
9:00 am -12:00 pm

Chairperson:
Andy Butler
PBS, Alexandria, VA

***The Broadcast Web on DTV - A New Reach Medium?**
Rob Glidden
Quadramix, Inc.
Moroga, CA

Would You Like Data With Your Video?
Louise Wasilewski
Scientific Atlanta
Norcross, GA

A New Model for Broadcasting - The Interactive Container
David Wood
Chairman ITU-R SG 11/Working Party 11A
Geneva, Switzerland
Mark Krivocheev
Chairman ITU-R Study Group 11
NIIR
Moscow, Russia

***Convergence of the TV, PC, and World Wide Web: Future Applications**
Joey Hougham and John Kirby
Intel Corporation
Hillsboro, OR

Advanced Use of TV Set Top Boxes
Raphael Nave
NDS Technologies
Jerusalem, Israel

***Data Broadcasting Creates New Revenue Business Models for Digital Broadcasters**
Abe Peled
NDS Ltd.
West Drayton, United Kingdom

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*Papers not available at the time of publication

Session: Interactivity and Data Broadcasting for Television
Would You Like Data With Your Video?

Louise Wasilewski
Scientific-Atlanta, Inc.
Norcross, GA

ABSTRACT

Scientific-Atlanta and one of its customers, a major European service operator, have integrated a PC decoder card into the PowerVu MPEG-2/DVB digital video compression system. The PC card enables the PowerVu system to support real-time video and audio to the desktop and includes data capabilities such as IP multicast and high-speed Internet access. Initial applications will be in a business environment, but the technology applies equally well to the digital broadcast marketplace where providing data related to the programming, such as sports statistics, is seen as a growth opportunity and a means to engage the audience further in the viewing experience. The paper examines the challenges of the technology involved as well as overarching business considerations, and aspects of network security, installation and maintenance.

BACKGROUND

For the past several months, Scientific Atlanta has been working together with one if its customers, to produce a DVB (Digital Video Broadcasting) data standard[1] compliant means of transmitting data direct to the PC, or LAN (local area network). The target market is the business rather than the consumer community, and a primary goal was to support data delivery simultaneous with video. Market research determined the product line profile. It is a new and exciting product line, exploiting contemporary technologies, yet from an applications standpoint, it is not revolutionary.

A BRIEF HISTORY OF DATA WITH VIDEO

Contrary to what the Multimedia hype of the last year or two suggests, data with video as a concept is not new. When Service Providers are asked the question, "Do you want data with your video?", the answer has been a resounding "Yes!" for many years. The network operations manager running satellite distribution networks for a major programmer wants data with his video to control ad insertion and other switching equipment at cable headends. In the past this was achieved by analog subcarriers above video, such as operated on DBS (direct broadcast system) systems around the world, or it has been achieved by multiplexed components in the MAC (multiplexed analog component) environment, such as in the secure video transmission system B-MAC offered by Scientific-Atlanta. The carriage of Nielsen and Gemstar information in the analog VBI (vertical blanking interval) lines are other good examples of data with video, but this data was "invisible". The viewer did not know it was there, in fact sometimes, these systems have been around so long, even the Operations staff can overlook them. The crucial matter is that the data is related to video, from a content and timing perspective, meaning that these two elements are transmitted together to maintain that relationship.

Yet data with video, where the timing relationship is less critical, and where the viewer is aware of the data, also boasts some significant pedigree. The "teletext" services supported by WST (world system teletext) have thrived in a community craving information, but unwilling to attend night school to learn how to work the technology. WST, delivered in the VBI lines, is arguably the closest thing to a free lunch in the broadcast world. The VBI must be present in analog TV transmissions, and so WST offers more service in no more spectrum. Just what the regulators like to hear! Popular applications include news headlines and short stories, sports results, stock prices, foreign currency exchange rates, as well as a primitive TV guide. The localized capability of UHF and VHF transmission mean that local services - such as checking regional airplane departure and arrival timeliness, or artistic performance schedules and even advertising -

also find wide acceptance. Best of all, the service is essentially free in many cases. Teletext decoders reside in TV sets. The system involves typing in "page numbers" from the remote control, guided by an index or index tree. At any particular page address (e.g. page 131) several screens of information are sequentially displayed (e.g. pages 1/4 through 4/4). Color-coded keys can shorten the navigation process and even simple multi-choice "games" can be supported. For all its age and simplicity, the system works, and meets the need inexpensively. It is an early data carousel.

The graphics quality is poor, but the information is readily accessible through that most familiar of devices, the TV set. We learn from this that the knowledge is valued by the viewer, not "the multimedia experience". The success of such services demonstrates also the desire for data, or more accurately stated, for knowledge, and this not necessarily related to the accompanying video. In developing digital video products, Scientific-Atlanta recognized the need for continuity in all the above services and explicitly designed in their support. Industry groups such as DVB also have designed standardized methods for transmitting some of this information and the ATSC (Advanced Television Standards Committee) is also turning its attention in this direction.

THE COMMERCIAL APPLICATION

In the commercial environment, the demand for information, and its fulfillment, may take different forms. File transfer is a very popular usage, applicable to many forms of data, whether spreadsheets, text or graphics, and can be used to update catalogs or other data retrieval systems, kiosks, or distribute newscopy to regional print centers. Fast LAN interconnect is another dynamic growth area. These applications are in use today. Data channels available in existing deployed digital video compression systems can and do support these applications today.

The benefits of these technologies when using the satellite delivery medium are undisputed. When used in a broadcast mode, satellite is the most economical means of delivering identical information to many sites. Networks can be deployed rapidly with virtually no disturbance to the existing infrastructure, and practically no limitations in terms of location. No excuses about not living on the ring, or waiting for permits to dig up the road! Inexpensive nighttime transponder rates tip the economic balance, and error preventing algorithms such as the Kencast system dispel any reliability concerns. These benefits are apparent in the digital and analog domains, but for flexibility and control digital wins hands down.

What is this demand for?

We have already answered the question of whether the public wants data. Existing media speak to that need. Corporations on the other hand are not always interested in providing their staff with sports scores, but they do want to educate their staff to increase productivity, to reduce downtime due to accidents or poor communication, to reduce product cycle times - particularly product launch schedules, and they want their staff to have access to up-to-date accurate information on pricing, and availability, wherever they are. In fact, these are the classical uses of business television networks. Distance Education, Product Launches and Corporate Messaging are the strongholds of Business TV (BTV). And this is how Scientific-Atlanta developed its interest in PC decoder cards, as a natural development from the IRD (integrated receiver decoder) to take our service provider customer base where they want to take their corporate customers. Most of the companies attending NAB would consider video or audio as their core business, and together we comprise a sizable industry, but the PC decoder card serves a much larger marketplace, that of companies with multiple sites that need to communicate. In fact, the growing trend of distributor networks in the information technology and financial industry means that even the smaller organizations may find themselves utilizing this technology to keep in touch with their suppliers. In whatever industry growth occurs, this technology will enable service providers to pursue that growth, and the accompanying profit.

Controlled Access to Information

Clearly then, there are financially reconcilable applications in the commercial arena which demand solutions well supported by the PC decoder card. Consider the nature of the information propagated and it is clearly of a sensitive nature. Product launch information, pricing updates and detailed product training, identifying the benefits and pitfalls of a product or service are not the kind of information one would hand to one's competitor, so these networks demand security. Simple network management techniques such as grouping users, and defining tiers of programming - precisely the same methods used in the cable industry - allow Business Managers or Franchise Owners access to information not required by other levels of staff, equally securely, or more so, than mailing the information or sending it by e-mail.

BUT WHY DATA *WITH* VIDEO?

Delivery of some of the information above is most natural in a written form, "data", and some of it in the very visual format of video. Are not two independent mechanisms of delivery acceptable? Cannot video be sent just as well as "data files" and then played back? What is the significance of data *with* video? Firstly, from a technical and economic point of view there are the benefits of one conduit. One set of cables to run, regardless of whether video and data will be running simultaneously, or independently. One tuner and demodulator at the receiver. One card, with one OS (operating system) and one API (application programming interface), and one development environment. One set of performance characteristics - link availability, throughput, BER. One connection, which once made, need not be broken to gain the other service - have we not all suffered from leaving the fax machine on the main line at home, and then wondered why nobody calls? The homogeneity makes deployment and operations simpler and more reliable. Yet this is not all. By supporting video decoding in hardware on the board, the main PC processor is left free for other tasks, such as word processing. The end user can work on their normal activities - which may include accessing data through the satellite delivery mechanism - with video running in a small window. When their relevant video section is presented - the expedited method of processing expense reports perhaps - the video can be enlarged to full screen, and the other data activity temporarily suspended. This increases productivity because the end-user has not spent 30 minutes in the BTV viewing room, sitting through material of little or no interest, waiting for their section. Also there is in the long run an overhead reduction, as no dedicated viewing room is required for these purposes now.

Similarly, in distance education, video and data simultaneous delivery is invaluable. To be able to teach on a subject, and have every student receive and respond to a multi-choice test effectively reduces the perceived distance significantly, all through a network installed and managed by a service provider. To be able to send diagrams, which could then be animated, during the teacher's presentation, greatly increases the effectiveness of technical training in the automotive industry amongst others. The real time nature reduces the reliance on an administrator keeping track of the latest mailing. Real-time is also the best method to handle product recalls and adjustments, critical in the eye of the consuming public.

What are the applications?

Many of the applications in the commercial environment involve very large file transfers. PDF file format is often used, with file sizes of 20MB to 200MB being typical. One example of this was during the 1996 Olympic Games in Atlanta, where foreign language newspapers were transmitted from Europe across to Atlanta directly into the Broadcast Center. There they were printed out in full size to be made available to the public in a timely and economic fashion.

Data Integrity. The question is always asked about the reliability of a satellite link, particularly in Ku band because of the rain fade characteristics. It is worth noting that MPEG-2[2] video decoders require a BER at the transport or video decoder input of $1x\ 10^{-9}$. This rate is far beyond what any telecommunications company would promise on their voice circuits, and any link involving microwave transmission can not guarantee better than 1×10^{-6}. The Reed Solomon code used in the DVB satellite channel coding and modulation standard[3] guarantees even at threshold no worse than one uncorrected error per hour, in a multiplex. These numbers are not merely theoretical. We have had customers who could not believe they were offering a data channel with a BER as clean as 1x10-9, and ran BER tests through a 2Mbps link back to back for two days without a single error, before giving up! Rain fade can not be controlled, and so the use of error correcting protocols, such as those offered by Kencast, with overheads as low as 3% to 5% assure that even with intermittent fades, data deliveries arrive first time.

Government Use. Government agencies are voracious consumers of distance communications. AFRTS uses data channels in a virtual private network mode. The US Navy even delivers data to ships via satellite, using standard PowerVu digital video compression equipment with data capabilities. They transmit PDF and TXT files. Sailors are able to retrieve news stories and keep in touch with what is going on back at home. Using Kencast file protection, they are able to avoid "tricasting", sending the files three times, as was found necessary previously. The data arrives with integrity the first time.

Training. MPEG-1[4] compressed files are often delivered for training purposes. Land Rover and Volkswagen have such networks in place. MPEG-1 playback cards are used to decode the files, which may show how to perform a particular maintenance or repair activity. Training technicians is a critical part of the product introduction process, and speeding this up means a product can reach the market sooner, with the investments in development and tooling paying off sooner. A German auto manufacturer had required 9-12 months to introduce the C-Class automobile, without a business TV and communications network. With the network, they were able to introduce the E-Class in one month. All 6000 technicians were educated to the same level in that time, whereas previously groups had to cycle through the training facility, with the associated travel and lodging costs, and increased time off the job. Savings like that drop straight to the bottom line, and bring the kind of competitive edge any company would enjoy. The automotive industry also uses these same tools to download manuals and new procedure documentation.

Interactive training is very popular in the retail industry. Tools such as MacroMind Director are used to author multimedia programs. This method can be more efficient than full video. 33MB can provide a full three minutes of playback. Graphics studios prepare the material in accordance with the corporation's script. The system defines "actors" which appear "on stage" at given times for given periods. As a technologist, the term "object" can be interchanged. These objects can be video, audio, graphics or text. Hot buttons can lead to supplementary information beyond the elementary presentation. Kiosks also employ this technology. They are found in malls and food courts, particularly providing information on the travel and entertainment industry.

Corporate Messaging. Corporate Messaging is a key application. A "talking head" scenario can be coded very efficiently. The Chairman of GTE regularly records a short early morning address. It is encoded and broadcast out as an MPEG-1 file to the field offices, and is already on his staffs PCs by 8am. They can play back the message at their own convenience.

For these applications to work, to be financially rewarding, they must be used by staff. In other words, they must be reliable, secure and efficient. Those technologies which provide these attributes in the non-standard environment can also be ported to a standardized universe with no loss in performance.

TRANSFERABLE APPLICATIONS

Our experience has concentrated on delivery via the satellite medium. Inherently, there is nothing to prevent these approaches being applied to other delivery media such as cable or terrestrial television. Indeed, the historical example of WST Teletext services is employed in terrestrial TV. In that example the teletext decoder and its user interface, the remote control, are embedded in the normal viewing equipment. The user is not left to integrate a decoder with setup menus requiring acknowledgment of incomprehensible licensing conditions, as the PC industry is wont to demand. This distinction is important in assessing user acceptance.

Data in the Cable Industry

Cable Modem. Cable modems are already deployed by some MSOs (multiple system operators) in the US, requiring the user to supply their PC. There were about 100,000 cable modems deployed by the end of 1997. Cable modems present a challenge to the cable industry, as they are obliged to send two installers to each set up, one "cable guy" and one "PC/modem guy". The service may reap rewards for the operator, but the payback period is lengthened. Fast Internet access is the service provided, and so competition with ISPs prevent the MSOs from charging significant up-front connection fees.

VBI Internet Delivery. An "old fashioned" creative approach has arisen to provide Internet over cable without requiring the installer double act, and without requiring the user to provide a PC, and although prehistoric in concept, its attractiveness to the more technology-allergic section of the community warrants its mention. The Worldgate approach uses dormant VBI lines to carry email and Internet information. Via a Worldgate enabled set-top (supported by more than one manufacturer), the Internet information is presented directly onto the TV screen. As with WST, the text quality may not measure up to HDTV, but it works. A wireless keyboard can be used to type messages or enter addresses. It provides limited computer-like functionality without the complexity of a computer. Advanced graphics chips can enhance the visual performance of delivery systems such as Worldgate.

Digital Interactive Settops. The Cable industry is certainly diversifying its technology portfolio, as the third means - digital interactive set-tops - also debuted recently. Scientific-Atlanta's offering in this arena, the Explorer set-top performs all the functions of a traditional settop - TV decoding, access control, program guide - and its also supports various applications, including Internet access, and the delivery of data relevant to video. The Weather Channel could transmit data embedded in the video multiplex allowing the end user to call up the weather forecast for their region, or through navigation, for a different city, or even a different continent. A Sports channel could transmit player statistics and the settop would allow the retrieval and display of that information, on the TV set, with enhanced anti-aliasing graphics, without a PC - which after all most living rooms do not contain. Data with video certainly has its attractions in the consumer market, and these are the topic of much debate. Clearly, network television could employ the same techniques. Soap opera fan club webs sites could be transmitted in the video stream so that access is not gated by a slow backbone. Electronic commerce could enable the viewer to purchase their favorite team jacket before, during, or after the big game - whatever team sport. Purchase tickets for the finals of your favorite sporting event, or order tickets for the stage event which accompanies the movie, or buy the boardgame. The possibilities are endless, once secure outbound and back channels are established.

Data in the Broadcast Industry

The trend of connecting TV with web content as evidenced in the CNN-SI cooperation will surely continue, and the digital technology will allow creative ideas to run free, presenting the viewer with a smoother flow of information. To be able to conveniently access broadcast webpages would enhance the viewers feeling of participation in the programming. PBS is making a concerted effort to bring this to reality. This would lend itself well to the talk show environment, or the gameshow situation. Clearly, when true interactivity with a backchannel is required, this can be difficult in a domestic environment. But the viewer can have the "feel" of interactivity when they select what element of the broadcast information they receive. In a game show, multichoice questions could be offered, and simply through interaction with the remote control, the viewer could select an answer and have revealed whether this was right or not. Running scores could be maintained.

Sports scores and player and team statistics are obvious candidates for this technology. Finding out more about favorite actresses in soap operas and situation comedies also seems attractive to the end user. The potential for connecting advertisers with shoppers is perhaps the most appealing commercially. In a broadcast mode, recipes using particular food products could be transmitted, competitions run or advertised, and viewers could find telephone numbers to call to receive coupons for particular products, or where a backchannel, such as cable, is available, request coupons or samples or answer customer surveys directly. This kind of system would be far more effective in determining true audience attention than current Neilsen ratings.

The most concerning issue in North America is that action on data standards has significantly lagged traditional TV based standards. Work is now on-going in the ATSC and we look forward to seeing their results.

When such a large scale deployment is considered in an open network such as broadcast, standards are essential. No hardware will be built, no software written, until it is clearly demanded and supported by the marketplace. As broadcasters face the challenge of making money in the new world the appropriate developments will be forced to occur. Technology agreement will allow equipment onto the market, but content will drive the success of services. CNN Headline News has experimented with carrying viewer comments to bring interactivity simply. Web content however is often not suited for display on TV sets unless anti-aliasing techniques are applied. Classified ads are not glamorous but may be one of the earlier broadcast applications to be successful.

DEFINING A MARKET PROFILE

The market addressed here is for Corporate networks, for multi-site businesses, those wishing to maintain or improve communications. This broadens the purview of the video industry from video-centric companies to all medium and large businesses. Corporate networks are seldom operated by the end user corporation itself. They concentrate on their own business of retailing, automotive, financial dealings etc. A service provider contracts to provide the service, usually including the installation as well as the operation. It is the service provider which has the obligation to make the service work. The equipment decision is often in the hands of the service provider, although end users may voice their preferences based on past experience, parent company ownership, or any of the other reasons that influence sales in any environment.

This is not the same environment as consumer, or "Direct-to-home" Internet access, where data-only services are provided. Twenty four hour phone support is not required, and the PCs or servers the cards integrate into can be controlled. Rapid response is required - there is no tolerance of holding 45 minutes to talk to a service representative, as is still customary in the PC industry. Just as cable TV is not the same as Direct-to-home, this is not an environment of do-it-yourself installations. These are two distinct markets.

Equipment Attributes. Service Providers look for certain characteristics in their equipment. Reliability is important, because service calls due to failures are paid for by the provider, not the end user. Decoder management features beyond basic encryption techniques can add to that reliability, providing the operator flexibility in allocating bandwidth. For serving Corporate networks, there is a recognized need to deliver video and data, simultaneously. Data must be available direct to the

PC or via a LAN. Service providers want to control development costs, and so one development environment, whether for a data only card, or a video with data card must be retained. The information transferred between sites is often of a commercially sensitive nature, and therefore requires security on the delivery. Whilst complete files lend themselves well to passwording, streaming information is better protected by a traditional video conditional access and encryption system.

Every business wants to grow its revenues, and to continue to grow. In this sense, the PC decoder card can be the thin end of the wedge for the service provider. Once the hardware is installed, additional software could be downloaded to support additional services.

STANDARDS - THE BASIS FOR INTEROPERABILITY AND INTEGRATION

Standards are popular. Manufacturers like standards because in theory they spawn a commodity market for ASICs, interface drivers and other components which in theory bring product costs down. End users like standards because they expect this leads to cheaper goods. We observe however, that when new standards are deployed, conformant products often sell for more than their non-conformant counterparts of equivalent performance. Standards provide vendors with a level of comfort that a market does exist, as the effort involved in writing them is too great to be supported without a clear return on the horizon. Standards provide comfort to end users, who believe that the great brains behind the standards will have understood the technology and pitfalls perfectly and designed constructs which will support any eventuality. Having been intimately involved in standards development, I attest that we did our best to achieve this, but acknowledge that we are only human. Most clearly standards provide a level of interoperation in the communications environment and a promise of independence from a single service provider. The latter is often destroyed when conditional access is involved.

Interoperability has been a much addressed topic in recent years. MPEG-2 as a standard is highly flexible. This was its strength - without this capability we would not have enjoyed the dramatic improvement in video quality which has evolved over the past two years. It was also the challenge for MPEG. If different coders apply different search algorithms, is it really possible to interoperate? A common semantic and syntactic language affords this, but the other aspect required is a common approach to timing and buffering. Early testers did not take these into account, and so a seemingly compliant syntax was not necessarily a compliant stream. As these kinks have been worked out, and we have understood which "required options" are not supported by which parties, the level of interoperability between different vendors' current equipment is now so good that it often goes unnoticed.

Now we start a new round of standards implementation. The DVB data standard has been available for nearly a year and implementations are starting to appear. Again, a clear protocol exists, and the right foundation is there. Only time will tell how easy it is to misinterpret the standard.

DVB Data Methods

The DVB data standard defines five methods of transporting data. These are summarized as:

Data piping: this provides a simple asynchronous channel for passing data. It will be the method of choice to replace 19.2kbaud or 38.4kbaud circuits. Data is dumped straight into transport packets.

Data streaming: this supports streaming asynchronous, synchronous or synchronized data. It uses PES (packetized elementary stream) packets which contain timestamps. This provides a means to implement e.g. a 2Mbps data channel, but it also holds promise for sending heritage audio services, such as NICAM, retaining lip sync, alternative types of subtitles or in a multimedia mindset, karaoke.

Multiprotocol Encapsulation: This technique supports high speed outbound Internet access. The DSM-CC[5] (Digital Storage Media - Command and Control) private section format is employed.

Data carousel: This supports the periodic transmission of data modules through DVB systems. It could be used to implement a modern equivalent of WST teletext services, or to support kiosk based information retrieval services. It employs DSM-CC datagrams. Content can be prepared off-line

Object Carousel: It allows a means of transmitting DSM-CC User-to-User (U-U) Objects. It allows related chunks of information to be packaged together. This reduces errors in system implementation.

Multiprotocol Encapsulation has attracted most publicity because it allows Internet access. When "DVB data" is referred to generically, this is often the mode intended. In a Corporate environment, Intranet access holds a great attraction. A conditionally accessed secure network allows more material to be made available to more sites than otherwise, without the expense of full-time leased lines. It supports the creation of a virtual private network in the air.

PROJECT DEVELOPMENT

The MPEG-2 and DVB standards were the basis for Scientific-Atlanta's project implementation. The first time the PC card was tested with the existing PowerVu MPEG-2 DVB compression uplink, perfect video was produced, over a range of symbol rates (bandwidths) and over a wide range of video bitrates. This testifies to the compliance of standard deployed equipment. The high quality of components used on the board, including line doubling elements, produced video quality acceptable in the broadcast arena.

The next stage was to test the data implementations. Clearly, a lot of "data" is software. Whilst hardware support was present from day one, not all software was available. At the uplink side, command lines were used to control the development terminal, whereas the end product is offered with a graphical user interface. Similarly on the card side, the Multiprotocol encapsulation mode was available to test first. Data piping, the simplest, and one of the most popular methods of data delivery in use today, was also tested in the first round. The simplicity of the protocol also allowed straightforward interoperability, and testing using simple BER measurement equipment.

The basic throughput of data streaming is also simply tested, but special material must be prepared to guarantee the synchronized relationship is preserved. The simplest method is to use a PCR (program clock reference) timestamp as the payload of the test packets. Test equipment to serve development work and in-service operations is scarce in this area, making it difficult for smaller developers to start. Some companies have expressed an interest in developing existing MPEG test equipment in this direction, and I hope we will hear concrete announcements at NAB.

PROMOTING SUCCESS IN THE MARKET

Service Providers have traditionally played the role of value-added resellers, and that trend will continue, and increase. Service Providers of Business TV networks install receive antennas and decoders, often also monitors, VCRs, cabling, line amplifiers, and whatever else it takes to make the network perform to the customers' satisfaction. They offer help desks, many not only during the broadcasts, but throughout the working day. They help a corporation prepare its content, providing studio and production facilities and advising on techniques which work and those which don't. These skills will remain essential, and require enhancement. Help desks are indispensable, but their costs can be kept within a bearable limit through controlling the environment into which PC cards are deployed. With the $900 266MHz PC expected before year end, many Service Providers will choose to provide a complete package: new PC, data delivery mechanism, customized user interface and applications software will be tailored to a particular industry. Packaging defeats the demon of version control which dogs the PC industry in a multi-user environment. The ability to integrate and test these elements at the Service Provider's facility should mean that installers will require little more skill and training than they now have. There is a significant saving in the cost and rate of deployment if the installer or installation team can handle several installs in one day. Further, the packaged approach minimizes the incompatibilities which have strained forays into this market such as DirecPC. It seems human nature to recoil from technology advances, and so as BTV users have found, the initial launch must be as trouble-free as possible if the end-users are to be won over.

Network Management systems can offer features to improve on-air time and network reliability. The possibility of a modem or LAN return path has its promises, but management of large networks through such a method can require extensive infrastructure. Decoder management platforms, such as those designed explicitly for commercial networks prove their worth in this environment. Features such as force-tuning and search-and-find can support an ad hoc network without any user intervention, ensuring the receiver is tuned to the correct RF channel and further to the particular broadcast in that channel - watching the technical training rather than the sales presentation if the equipment is defined as a member of the technical group for example. Consumer-centric access control technologies do not always support these operator-friendly features, relying instead on the user to determine what they want to see and when. Of course, these system serve their purpose well, but it is not the same purpose.

Flexibility in terms of supported applications is also key. In spite of the favorable economics, money spent is still money missing from the bottom line, and any technology is carefully considered. The growth opportunities afforded by applications suites such as those available under DVB will serve the service providers and their customers well. Even if fast Intranet access is the Trojan horse application to bring the technology into the office or store, the data carousel ability to deliver news targeted to a particular segment of the user work-force will soon be appreciated and realized - privatized push technology being used to ensure notices are read. In the financial industry particularly, the commentary on interest rates and analysis of stock performance can never be too timely. The support of audio and video within the system - either onboard the PC decoder card, or via a supplementary card in the same chassis, means that in-store audio or video-to-the-checkout could be implemented economically, without significant additional expenditure. We will likely see the revival of some uses of the technology such as the latter, which were tried, and found effective but costly, as the services become more integrated and more affordable.

SUMMARY

Data services delivered with video have a long pedigree in the analog world, and will continue in the digital domain. DVB data is a standardized method of supporting these services, with the range of service formats allowing convenient support of many applications, some replacing existing services, and others birthed through the new technology. Distance learning, fast file transfer, fast Internet access and shared database access - such as form retrieval in the insurance industry - , including via in-store kiosks will likely be the leading applications in the commercial environment.

The simultaneous reception of video with data enables some applications which simply would not work without this capability, and enhances the effectiveness of others. Security of the channel enables the creation of a virtual private network, allowing commercially sensitive information to be passed from site to site, which would be jeopardized if transmitted over the public network.

The basic techniques can also be applied to a broadcast TV or cable environment. Experience gained in serving the commercial world offers the potential to secure a strong launch of data based or interactive services complementing broadcast television. When offering a new technology to the viewing public, the service must be reliable, useable and visually attractive if it is to succeed.

Scientific Atlanta looks forward to the successful deployment of DVB data based networks in North America, Europe and also in Latin America and the Asia Pacific Region, where the terrestrial infrastructure status and geography strongly promote the use of satellite technology, both in terms of economics and rate of deployment. The combined cost of new lower priced but highly functional PCs over the next twelve months when equipped with a suitable PC decoder card will allow all the functionality of current commercial or professional grade IRDs with the flexible capabilities of a computer. This makes the market ripe for rapid expansion beyond the traditional video bounds.

ACKNOWLEDGMENTS

The author would like to thank the DVB Project Office for their patience and efforts in seeing through the development and standardization not only of the data standard referred to, but also the many other standards which have enabled the digital video compression industry to get off to a solid start. Much of the work is invisible, but it is equally invaluable.

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A NEW MODEL FOR DIGITAL BROADCASTING – THE INTERACTIVE CONTAINER

Mark Krivocheev, Chairman ITU-R Study Group 11
David Wood, Chairman ITU-R SG 11/Working Party11A
Switzerland

SUMMARY

Until recently, the broadcasting world saw digital television as the means to multiply the number of channels available, and to provide HDTV when needed. This was a mistake. Digital broadcasting is more than this. Although it will not happen overnight, digital broadcasting will give the broadcaster the chance to create and deliver a flexible mix of programming and interactive multimedia. The digital channel becomes a container that can be filled by a range of complementary services. Arguably, the digital package should be provided by a single service provider, who can manage the total container in the most effective way. Standards for vision and sound coding and compression have been defined, but there are, as yet, no common systems for the software/multimedia interfaces for the digital receiver. These need to be found.

INTRODUCTION

It has always been clear that digital broadcasting will provide major benefits for broadcasters and viewers. But the complete nature of the benefits, and the shape that digital broadcasting will take is, arguably, only now beginning to emerge. The changes ahead are more than just technology or picture quality. They are likely to include a broadening of the range of potential services available from a broadcast channel, changes in the viewer's relationship to the services, and changes in program preparation and production. Among other measures, the broadcasting world needs to act quickly to specify a multimedia/software interface for digital television.

No one suggests that a switch will be thrown and television broadcasting as we know today it will disappear. Far from it. In any event, we need a finite period for digital delivery to be taken up by the public. Rules of thumb usually suggest that 50% penetration in ten years is a good batting average for new consumer media. But equally, it is always a mistake to think there is nothing over the horizon just because we cannot see beyond it. Eventually all broadcasting will be digital, and only digital. In time, television program channels will migrate to become flexible interactive multimedia channels which encompass linear programming, but have the option of more.

The ITU-R is a worldwide organisation that recommends specifications for broadcasting systems. It needs to stay relevant and valuable to the broadcasting community by responding to the evolving situation, and by staying ahead of the implementation cycles of digital broadcasting. The ITU-R needs to understand the evolution of digital broadcasting before it happens, so that it can be ready with specifications and guidance for users. The authors are the Chairmen of the ITU-R Study Group on Television Broadcasting, and its Working Party 11A, on Television Systems and Data Broadcasting.

Twenty years ago it was clear that the use of digital modulation would provide a dramatically easier environment for spectrum management - provided that the data rates needed for broadcast quality video and audio could be brought within bounds. This has been realised.

The development of motion compensated hybrid DCT, thanks to pioneering work by Professor Rao at the University of Texas, and others, led to the practical realisation of digital video compression systems for consumer equipment. The MPEG1 and 2 systems, and the similar proprietary systems that preceded them, were the result, and proved the key to the practical development of digital television broadcasting systems in the early 1990s.

The propagation, bandwidth, and power constraints of the bands used, define the modulation systems needed for digital broadcasting. For the satellite bands, where large bandwidths are available but broadcast power is limited, constant amplitude systems are needed. For current transponders for direct-to-home services which range from 27 MHz to 36 MHz in channel bandwidth, the deliverable data rate with consumer-acceptable dish sizes is about 30 – 40 MBit/s. QPSK is currently used for satellite broadcasting services, though some organisations are investigating the use of 8PSK, which if successful will double the deliverable data rate.

The various QPSK systems used for satellite broadcasting services are similar, and achieve much the same performance for the user, but there are some differences in error correction and multiplexing. The ITU-R has developed a Recommendation for a multi-system receiver for the QPSK systems. Manufacturers may choose to migrate to this receiver, although many digital satellite services use proprietary conditional access systems. The universal use of a common physical and logical system interface, however, would allow multi-system receivers to be used with different proprietary conditional access systems.

For cable systems, the channel bandwidths available are considerably smaller than for satellite, at least if current channel rasters of 6, 7 or 8 MHz are maintained. But, on the other hand, the signal-to-noise environment for a cable network is under control, and thus a modulation system with many amplitude/phase states, and a small decoding margin, can be used. The deliverable data rates for cable channels can be similar to those available for satellite channels, 30 – 40 Mbit/s.

For terrestrial systems, the channel bandwidths available are smaller than for satellite, but the available power is less constrained. There are several modulation systems currently specified in the ITU-R; OFDM and 8-VSB, and a further variant, band-segmented OFDM, may be added in 1998. In all cases, the deliverable data rates for a current terrestrial bandwidth channel will be 18 – 24 Mbit/s in practical circumstances to viewers with outdoor aerials.

The digital television systems thus provide not just channels for television; they provide what may be seen as two generic 'containers' for digital data. The terrestrial channel capacity provides about 20 Mbit/s, which is about half the capacity of the satellite channel, about 40 Mbit/s.

One of the strengths of the ITU-R is its ability to view delivery systems globally, across different broadcast platforms. This makes it easier to see trends, and to help arrange commonality across media.

THE CONTAINER

The container is a new concept for broadcasters to come to terms with. For fifty years, television broadcasting has been about providing a 525-line or 625-line linear programme chain in a given channel. This option will always be with us, and will always meet a public need. It will always be the mainstay of broadcasting. But the digital age offers us new options and new freedoms to experiment with broadcasting. The freedoms include providing a range of picture and sound quality options, and the means to put together any multimedia package we like, as a broadcast service.

THE OPTIONS FOR TELEVISION PICTURE QUALITY

The MPEG2 system is a variable bit rate coding system that encompasses a range of picture qualities from CIF (roughly equivalent to VHS quality) to HDTV.

The ITU-R has defined four quality 'bands' which cover the quality spectrum. The distinction is somewhat arbitrary, but it has served as a useful classification system. The ITU-R quality bands are Limited Definition television (LDTV), Conventional Definition Television (CDTV), Enhanced Definition

Television (EDTV), and High Definition Television (HDTV). Quality windows define these bands that come from the results of subjective evaluations, made at progressively closer viewing distances. The distinctions between the bands are essentially the viewing distances at which critical pictures look subjectively excellent.

MPEG2 coding is able to deliver all four quality bands, and allows wide or conventional aspect ratio. Progressive or interlace scanning is available, up to a ceiling on the number of luminance samples (though this is below that needed for 1080 active lines progressive, but includes 625 and 525 progressive).

The ITU-R has agreed production formats for 625 and 525 progressive, and for a High Definition Common Image Format (HD-CIF) 1080-line system, in both interlace and progressive formats. The production environment is able to cope with whatever the delivery environment needs.

In principle, we do not have to decide on a unique delivery quality. These can be chosen to suit the material being broadcast, and the intended-viewing environment.

The MPEG2 system also allows the transmission of stereoscopic pair signals, using a coded difference signal. The price for broadcasting a stereoscopic pair in the digital environment is thus not enormous. This will allow broadcasters to give serious consideration to stereoscopic television. Many problems remain, of course, in making practical domestic stereoscopic displays.

PROGRESSIVE VERSUS INTERLACE SCANNING

The choice of interlace or progressive scanning for the source prior to compression is a difficult question. It will always make good sense to make optimum use of the available delivery megabits. Therefore one of the factors affecting the choice of interlace or progressive scanning is 'bit rate efficiency'. For a given number of Mbit/s in the compressed delivery form, do we finally achieve a higher picture quality with a progressively scanned or an interlaced scanned source?

Many, though not all, engineers have a suspicion that the progressively scanned source will give the best bit rate efficiency, but this is not yet proven. An advantage of the progressive source is that the compression system has available the maximum vertical–temporal resolution, and therefore is able to make the most accurate motion compensation possible. In short, having the source in progressive form allows the compression to do the best job it can - and thus the whole signal chain is 'content adaptive'. In the case of the interlace scanned source, effectively one stage of non-adaptive compression has already been applied – the interlacing.

Seen from the overall chain, there should be 'coding gain' associated with the progressive scanning source compared to the interlaced scanning source. However, there are other factors to consider, and one is the noise bandwidth of the interlace and progressive signals. This may also affect the overall performance. The relative advantage of progressive or interlace scanning is probably difficult to predict without substantial practical tests and subjective evaluations of real picture material.

STATISTICAL MULTIPLEXING

MPEG2 compression is content dependent, and the largest instantaneous bit rates result from scene content that has the largest amount of detail and movement. Given a representative range of picture content, the bit rates needed show a variation over time of as much as 4:1. This variation can be exploited in several ways. If we allocate a fixed bit rate to a television program channel there will be substantial periods when that bit rate is not used efficiently.

The use of statistical multiplexing improves the efficiency of the delivery system, by transmitting more program channels than would be possible with a fixed bit rate for each. In a statistical multiplex system, a number of television programmes are delivered in the same channel, each with variable bit rate, creating a whole which is less than the sum of the peak bit rates needed. Since the statistical chances of all the channels needing the maximum bit rate all at the same time is low, statistical multiplexing can be successful. It is too risky to apply a 4:1 factor in increasing channel carrying capacity, but increases of 30-50% are possible. The use of statistical multiplexing is one advantage of a single service provider being responsible for the container.

In addition, the unused data capacity available when the entropy of the picture falls can be used for carrying data other than picture or sound data. Indeed, the use of the digital channel to carry types of data other than normal television programming is a major part of the new digital-broadcasting environment. These can be either as allocations in their own right, or the opportunistic data available at times with video coding.

MPEG4 AND 7

The ISO/IEC JTC 1 has now moved on from its MPEG2 coding to two important new areas. The ITU-R is viewing this work with interest. The MPEG4 system is, like MPEG2, a tool-kit of techniques that can be grouped to create a compression system. There are some new coding tricks that make the MPEG4 system slightly more efficient than MPEG2, but there is also the major new option of using object-based coding. If this can be implemented for broadcasting with acceptable results it will bring significantly greater compression than available with MPEG2.

The object based coding system works by separating a picture up into specific objects (segmentation), which can each be coded in different ways. For example, a picture could be split up into areas of human faces, which could be provided with greatest detail, and background areas, which could be provided with less detail. In this way, precious coding megabit/s could be used only for those items in the picture which really benefit from it. This will greatly reduce the net bit rate needed, but it remains to be seen if the segmentation can be effectively done with complex scenes.

MPEG4 is a step further down the road of making the compression system adapt itself to the picture content. For those organisations who do not need to broadcast some of the higher quality levels immediately, or can introduce services in a few years time when the technology is mature, MPEG4 could provide a future compression system with a lower bit rate.

The MPEG7 system will provide content labelling which will allow accurate searches to be made, and this may be useful for the future television equivalent of browsing and searching.

DATA AND MULTIMEDIA BROADCASTING

Many parts of the world – though not North America - have enjoyed relatively large 'teletext' services for up to twenty years. Teletext provides a TV-displayed electronic magazine or newspaper, broadcast as digital data in the vertical interval of an analogue television channel. This can be used for programme-related services, or a range of independent information services. This mode of operation is called a 'carousel', because the data is repeated regularly in a transmission cycle, The receiver grabs whatever part of it, or pages, the viewer calls for, as they come by.

Space in the analogue television vertical interval is limited, and the range of display possibilities with currently broadcast teletext is limited too. In the digital environment, data broadcasting with a wide scope is possible, and with formats that go well beyond today's teletext. Effectively we can move beyond the age of text and simple graphics services, to the era of multimedia and software services.

The options include using the more developed (higher level) versions of the current teletext system, or a system compatible with World Wide Web format. In the digital environment, the simple text based information services of teletext can become interactive multimedia services, with a wide range of applications. They can enhance, or interact with, the broadcast programme, or they can provide stand-alone services.

Data broadcasting services, or interactive multimedia delivery, may represent a major new growth area for digital broadcasting. The multimedia package can be seen as software Application that can be downloaded to the receiver, and runs on the receiver. Multimedia Applications can provide locally interactive services, or they can be connected via a return path to provide services that interact with the program service provider.

The television container can provide high demand web pages rapidly (such as via the standard-developer group DAVIC's Enhanced Digital Broadcast Protocol), or it can be used in conjunction with a telephone connection or other interaction channel media, to provide interaction with the program service provider (the DAVIC Interactive Digital Broadcast Protocol).

APPLICATIONS

A wide range of multimedia entertainment and information services can be envisaged, but one of the most useful broadcast multimedia Applications is likely to be an 'Electronic Program Guide' (EPG). This will provide background information on programmes and programme schedules. It seems entirely reasonable for the broadcaster to spend considerable energy on the guide, which may be the only comprehensible route through the multitude of content available. The EPG could include such features as Parental Control measures, and the means to create 'virtual channels' by selecting a personalised menu of programmes, that could be automatically found once the pre-selection was made. A household could have a 'mom's channel', and 'dad's channel' etc..

Other Applications could include interactive commercials, to hold the audience during a commercial break. A range of other interactive applications is also possible, including home responses to game shows, polling, etc..

Although linear or conventional programming will always be key services for the viewer, the addition of interactive multimedia to program services will be an important means to enhance the value of broadcasting.

THE AUDIENCE FOR INTERACTIVE MULTIMEDIA

The audiences for television programmes are assumed to fall into interest groups, largely associated with genres or programme types, or socio-economic groups. Programme chains or

programmes themselves can be focussed on particular segments of society. These distinctions will still apply in the digital world, but there may also be other distinguishable behaviour groups in society, related to the viewers' willingness to use an interactive system. These may vary from country to country and generation to generation. Work remains to be done to fully identify them. It is likely that some – or even much - programming will be needed which works well with or without the interactive multimedia elements.

THE SOFTWARE INTERFACE FOR MULTIMEDIA APPLICATIONS

The software interface for multimedia applications (the Application Program Interface) is one of the most critical issues for broadcast standards today. It is, arguably, as important to have an open and known software interface as it is to have open and known interfaces for vision and sound coding – which we do already have. Broadcasters need to be able to address multimedia software applications to receivers, and they need to know the constraints on processing capacity in the receiver.

If the receiving community uses different proprietary software interfaces, broadcasters will be faced with broadcasting multiple versions of software applications.

There are several proprietary APIs in use for digital broadcasting, but the specification-developer body 'DAVIC' is hoping to develop an open API based on the use of a system developed in the ISO/IEC JTC1, called MHEG5, in conjunction with elements of the JAVA system, which is used in multimedia applications for the world wide web and elsewhere. This seems to offer the best hope of a common worldwide API, and the ITU-R views this with great interest.

THE GLOBAL NEED FOR APIs

The need for a known API is not restricted to digital television broadcasting. It applies to all digital delivery systems. These include digital telephones, higher bit rate mobile systems, digital audio broadcasting, etc. The Applications and type of multimedia that each will manage will be different, depending on the deliverable bit rate, and the local processing capacity in the receiver. A global approach to all digital delivery systems would be in the public interest.

Worldwide leadership will be needed to achieve this, and this will go beyond the bounds of the ITU-R.

CONCLUSIONS

Digital television offers a new environment. The digital channel is a container that can be used for a flexible range of services. We are beginning to realise that it is not a means of providing more of the same, it is a step in the modernisation of television broadcasting. This is a message that must be carried to the regulators. A digital broadcaster can use his container for a complimentary set of services - services with qualities that match the needs of the content, and of the viewer.

The public will always need, and demand, conventional 'linear' programmes. There is no question about this. Television broadcasting cannot change overnight. But an element of the complete package of future digital broadcasting will be broadcast interactive multimedia. Broadcasters face a great challenge and a great opportunity with the new options that digital broadcasting allows.

In a sense, the baton is passing from the engineers providing the technical means, to the program maker - to select creatively from the potential options. The real job now is deciding what we want the technology to do, rather than working out how it can be done.

One of the key tasks of the broadcasting industry today is to specify a software/multimedia interface for digital broadcasting, and this is also true of other digital delivery media. Specifying a software interface will give value to the many years of hard work spent in developing the other elements of the digital broadcast systems.

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ADVANCED USE OF TV SET TOP BOXES

Raphael Nave
NDS Technologies, Israel
Jerusalem, Israel

ABSTRACT

The Set-Top Box (STB) or Integrated Receiver-Decoder (IRD), while evolving from analog to digital, is also making the transformation from an entertainment appliance to an information appliance.

Digital STBs use various kinds of delivery technologies to provide the consumer with many more channels, higher quality audio and video and a myriad of interactive capabilities. This opens a whole range of new business and application opportunities for broadcasters, subscribers and manufacturers.

TV SET-TOP BOXES GO DIGITAL

The last decade saw the proliferation of Direct-To-Home (DTH) TV via satellite in many countries. These services were characterised by more channels than Terrestrial or Cable services and higher quality programs and transmission. The majority of these services used analog transmission technology. In the 90's the services started expanding by providing subscriptions, Pay-Per-View and other conditional access services that provide marketing options and protect the broadcasters' revenues.

In the recent years digital broadcasting technology matured to a level which allowed broad commercial deployment. Digital broadcasting brought to the market very attractive traits such as better utilization of bandwidth, effective error correction and encryption techniques, and the ability to mix a variety of data and media types.

The application of compression techniques, such as the renowned MPEG II standard, yielded the ability to transmit many more entertainment channels than before. On one transponder one may transmit up to 16 movies or half a dozen Sports games. This development brings better efficiency and lower cost.

Beyond Compression

The real impact of digital broadcasting goes beyond the advantages of compression. When one comes to think of it - the compressed video data is actually a digital TV transmission bit-stream. It is almost indistinguishable from any other bit-stream and as with any digital data - it can be operated on using software techniques.

This opens a host of opportunities to process the information. Effective encryption techniques facilitate conditional access so that only the parties who are entitled to receive and enjoy the information being transmitted may do so. Furthermore, one may also add additional data such as subtitles, other languages, alerts of interesting viewing opportunities and so on.

Also, since a digital bit stream can be generated from a broad range of information sources, the transmission can contain a variety of data types. On top of the basic TV video and audio (movies, sports games, news...) one may add animated graphics, text, music and any information incarnation that the subscriber may desire to receive.

Thus the digital revolution opens a whole spectrum of opportunities to improve the DTH broadcast service quantity and quality, thereby making it a very attractive proposition to the average household. It is expected that this substantial additional richness of service will attract many subscribers, thereby generating extremely fast growth of this market. This may also increase the attractiveness to advertisers and thus generate additional revenues.

RICHER TV VIEWING

The primary application of the TV is, naturally, viewing of video programs, be it regular channel broadcasts or special offerings such as movies or Sports events. With the increase in number of channels, to several hundreds, the selection increases and with it the ability to provide special viewing opportunities or packages.

Program Packaging

Good packaging examples are Pay Per View (PPV) and Near Video On Demand (NVOD). These packaging options allow viewers to purchase the right to view, at their own convenience, premium TV events of their choice or on-circuit or classic movies from their armchair. The movies' quality is better than if viewed by the VCR, one does not need to go to the video store and the selection is rich.

Another example would be the various modes of subscription to Sports events. Such subscription may be in the form of a season ticket to a specific home team or to a specific league. It can also be a one-time purchase of viewing rights to special events such as the Olympic Games or a boxing match.

Advanced Program Guide

The Set-Top-Box (STB) is equipped with a user-friendly graphical interface, called the Electronic Program Guide (EPG), which allows subscribers to navigate the whole spectrum of programs in a convenient and friendly manner. The most common methods are using the mosaic, familiar from the cable world, or the newer and more powerful Grid-type. This is normally a two-dimensional graphical representation of the various channels' programs based on time of broadcast. Alternately, one may confine selection to a specific genre of programs (like Sports or Science or Children's programs) in order to decide what to view next.

The EPG may allow for advanced 'booking' of movies or events. It also enables creation of personalised program lists or favourite channels for the individual viewer. It also facilitates the application of parental control guidelines, spending limits etc.

The increased number of channels in the digital age of DTH TV makes the EPG a vital vehicle to facilitate navigation amongst the many entertainment opportunities. It provides an effective means of controlling the use of the STB and the services it offers.

The diverse and powerful features of the EPG entail a difficulty, since different broadcasters or manufacturers may opt to enrich the selection in unique manners. Thus the Service Information (S.I.) is being transmitted to the STB in different formats or structures thereby creating a 'Tower of Babel' for those who strive for standardization.

The STB features must be supported by complementary capabilities at the head-end. The head-end equipment needs to control the information flow and embed in the transmitted data the necessary attributes so the STB will properly handle the received bit-stream.

INTERACTIVE AND DATA BROADCASTING SERVICES

The digital age opens additional opportunities for the use of the TV Screen - for home applications that go beyond the passive viewing of TV programs and movies.

These additional applications are Interactive Services - since they entail interaction of the viewer with the service that is active on the screen.

In order to understand the role of these services, it may be best to digress a little and observe another display appliance at the home that is interactive by nature: The Personal Computer (PC).

In the last decade there has been a hot debate whether the PC or the TV should be THE universal multimedia appliance for the home. The PC has the advantage of processing power, large storage capability and natural interactivity using the Keyboard or Mouse. The PC has lately also been able to display motion pictures.

In the last several years - due to the Internet revolution - a large variety of services and networked interactive applications found their way to the home PC. Among such applications are home banking, home shopping, weather and data interrogations, distance learning, games etc.

PC Applications on TV

However, the spectacular growth in these areas has still been limited to the chosen PC few. Firstly, there is the frustration of slow response times due to the narrow bandwidth of the Internet - a phenomenon quite foreign to the TV viewer. Then there is the need for 'specialist' knowledge (not everyone knows Windows!) This has limited the take-off of interactive applications to the *cognicienti*. Not all homes have a PC or access to the Internet, and there are concerns about the security of monetary transactions over the 'open' Internet. So, while the PC is the natural interactive appliance the above limitations make a case for considering an interactive TV as well.

Direct satellite transmission to the home TV provides an excellent opportunity to overcome some of the deficiencies mentioned previously. It provides the ability to transmit data at about 40 Mbits/sec per transponder to the TV Screen. The viewer paradigm with control by the STB remote can be maintained, removing the 'black art' image of the PC. The whole market of TV viewers is opened up, even the ones not yet computer-literate or not owning a PC.

Therefore it becomes attractive to port many of those applications to the TV and thereby enrich the subscriber's experience and open the door for business opportunities by the service providers. The high data rate enables transmission of richer material, using dynamic video graphics and audio - describing the product or service being proposed. The viewer may request further details and data and get it instantaneously thru the satellite. Thus 'home shopping' and similar applications may become real, effective and entertaining.

A special example where Internet applications may benefit from the wide bandwidth to the home is the class of 'push' technologies. These are applications where a large amount of multimedia data is delivered to users such as a daily newspaper they subscribe to, budding games' subscriptions and services such as PointCast.

When selective information services are required, user profiles are established. For instance the user may outline the news that he would like to watch, the companies whose stock to monitor or whose news releases should be available, the cities where the weather is of interest or the Sports teams or leagues whose news should be selected. Each STB will then filter the relevant information off the air and display it upon user request.

Interactive Advertising

One advantage of the modern digital set-top boxes is that they provide the capability to overlay messages, or even a whole dialogue, between the broadcaster and the subscriber using the OSD messages and the Remote Controller. This capability is enhanced by the use of local secure storage, in the Smart Card, thereby deferring the need for on-line communication.

A great opportunity for such an application is Interactive Ads. The basic idea is that while a Video clip of an advertisement is playing, a message asks the subscriber, if (s)he is interested, to press a certain button which triggers follow-on activities. For instance, obtaining additional information on the product, winning a prize, participating in a game etc.

Such capabilities significantly enhance the effectiveness of advertising by better targeting the audience, counting eye-balls and thereby collecting feedback on who is watching which ads, and soliciting the subscriber's attention by lining up rewards.

A specific feature of such interactive ads is the immediate linkage to home shopping application, whereby the advertised product may be purchased 'on the spot'. There may also be book-marking, so that the subscriber continues watching the program uninterrupted, and goes back to the book-marked ad at his/her own leisure.

Asymmetrical Communications

The future STBs will support such services by enabling activation of multiple applications, and using the phone up-stream channel to the headend for the customer's queries and inputs. It should be noted that the up-stream communication bandwidth is limited to the speed of regular Modems on phone lines thereby limiting to those applications that need the high data rate only in one direction.

As the number of applications will substantially increase, they will not all reside in the STB. Rather, when the user expresses interest in an application it will be downloaded 'off the air' from the broadcast stream, and invoked momentarily to provide the desired service. This opens a whole new dimension to home business.

In the case of distance learning, a lecture with video, graphic and audio components be transmitted to many homes. Users may participate in the lesson by 'voting' on answers or by presenting questions to the lecturer, etc. Similarly, corporate training or dissemination of media-rich information for customer service may be distributed using the satellite channel and the STB.

Last but not least - the interactive STB may facilitate participation in interactive games. For instance - there may be a Jeopardy game going on in a certain channel on the TV monitor and the home subscriber may compete using 'electronic voting' on the right answers. There may be championships and competitions. Other interactive games such as Hugo or even Doom can be provided!

Paying the Piper

The STB will most likely also facilitate a second Smart Card slot - where users can insert their credit card or Mondex-like card and use it to perform transactions such as home banking, home shopping, lottery betting etc.

Another promising feature that will become prevalent in digital interactive set top boxes is 'pay per use'. In that model the subscriber does not pay for a full subscription or product but rather 'rents its use' and pays according to the number of times or the duration of use.

For instance: Attractive games may be charged on a 'pay per play' basis or on a 'pay per 10 minutes' basis (up to cumulative payment that equals the game's full price - after which the game is owned by the subscriber). This will allow users to 'taste' a game and toy with it before deciding if they really want to purchase it.

A whole spectrum of information and entertainment services may be provided on a similar basis. Lawyers may pay a central 'precedents' library for use of relevant case studies. Students or professors may 'borrow' research works, fans may watch Sports clips etc., etc.

Since the terrestrial broadcasts are heavily dependent on advertising for effective business, digital terrestrial boxes that facilitate such added functionality will provide attractive value added to the broadcasters and advertisers, thereby getting them higher return on the advertising budget.

We may take this approach a step further by proposing that the STB become the hub for entertainment at the home. On top of connecting to the TV - it may serve as the trunk for video transfer to other appliances. Thus the STB may connect to video recorders such as the new DVDs or Digital VCRs. It may feed modern video jukeboxes in the living room and it may link the satellite receptions with the home PC.

TOWARDS A WORLD WIDE BOX

The recent 'information age' evolutions have transformed the world into a 'Global Village'. The Internet is a clear demonstration of transcending thousands of miles and countries' borders in seconds and accessing information all over the world with no physical bounds. It is expected that TV broadcasts will assume similar characteristics, especially as the large number of channels available allow broadcast programs from all over the world to be carried.

But - there are other aspects of Globalization that are worth noting. Once a specific product or service is successful in one country or continent - there is no reason why it should not be marketed in other geographical areas, thereby making its potential market substantially bigger. Thus the idea is to deploy the future digital STB into as many countries as economically feasible.

It should be noted that such a proliferation will not be automatic since different countries and markets have built in different requirements. The most obvious one is the language: be it the texts engraved on the box, or the manuals or the remote control as well as the EPG and help texts displayed on the screen. The TV standards are different (PAL, NTSC, SECAM and their sub-schemes), electricity voltage and frequency, phone switches and dialling schemes etc.

Thus, while the same base design may be proliferated it will need customisation to the target markets.

It should be noted that other commercial considerations get in the way: In order to succeed in specific countries there is a need for local manufacturing. This reduces customs & taxes as well as shipping costs, and local distribution channels to facilitate delivery to the target audiences, as well as brand recognition.

Thus a World-Wide-Box [WWB] is an attractive prospect, but one that should be weighed with caution and the difficulties need to be considered before it is being touted as a panacea.

IN CONCLUSION

The emerging Digital revolution, coupled with right standardization trade-offs and the world-wide economy of scale makes the Direct To Home TV technology a fast evolving and extremely popular information and entertainment capability.

The combination of these developments, the investment in user friendly tools and environments and the addition of interactive services - will make the home TV a much more powerful and enjoyable appliance and will transform the TV viewing into a real experience and true home entertainment across the world!

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Internet Technologies for Radio

Tuesday, April 7, 1998
9:00 am to 10:30 am

Chairperson:
Barry Thomas
KCMG, Los Angeles, CA

Applications of Internet Service in Tokyo FM
Tsuyoshi Kawabuchi
Tokyo FM Broadcasting Co, Ltd.
Tokyo, Japan

***News from the Audio Webcasting Frontier**
Steve Church
Telos Systems
Cleveland, OH

***Internet Connectivity**
Robert Meuser
WHTZ-FM
Secaucus, NJ

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*Papers not available at the time of publication

Applications of Internet Service in Tokyo FM

Tsuyoshi Kawabuchi, Manager, Engineering Development
TOKYO FM Broadcasting CO.,LTD.
TOKYO JAPAN

ABSTRACT

Tokyo FM is an FM broadcasting station located in Tokyo with about 37.7 million listeners and is the key station organizing an FM broadcasting network called JFN (Japan FM Network) containing a total of 35 Japanese FM stations ranging from Hokkaido in the north to Okinawa in the south. Tokyo FM presents mainly Japanese pop musics on the air keeping the No.1 rating among the Japanese radio broadcasting stations for young people of twenties and thirties. Tokyo FM has not only broadcast radio programs by the conventional audio media but also presented teletext broadcasting services such as "Visual Information Radio" and "PAPARA G-COM" by using a DARC system, positively engaged in broadcasting services linked with other media such as the Internet, and realized various events, etc., so as to meet varieties of demands to be set out in the digital multimedia broadcasting age to come. Radio broadcasting is a medium, has an extremely high degree of fusion with other media and is expected to have increased joint event plans in the future. This paper describes the Tokyo FM's strategy for dealing with Internet, the configuration of the Tokyo FM station facilities and equipment, and the contents of service presented by Tokyo FM including the current Internet situation in Japan.

INTRODUCTION

In Japan, the Internet has spread gradually since three years ago and since 1996 when NTT introduced for the first time a discounted flat-rate charging system for late-night use of telephone lines, the number of Internet users has rapidly increased to be about 10 million in the end of 1997. The number of Internet users has reached about 10% of the total population of Japan and thus the Internet has become one of the new major media. Internet broadcasting stations also have commenced service. More than half of the Japanese broadcasting stations including TV, FM and AM broadcasting stations have already had Web servers and are positively using this new medium by making efforts in providing distinguished contents in linkage with the Internet while mainly aiming at promotion of their broadcasting services. Tokyo FM, commencing service with a Web server provided in its station building in November 1995, has presented such contents unavailable by other broadcasters than Tokyo FM by using live studio view cameras, etc., under the title of "TFM on the Web (http://www.tfm.co.jp)." As a result, the number of accesses has exceedingly increased with the increased number of Internet users (see Figs. 1 and 2) and as of 1997 broadcasting business has become achievable by using the Internet medium alone.

SYSTEM CONFIGURATION

The configuration of the Tokyo FM system is shown in Fig. 3. Contents are produced mainly by means of an INDY of SGI and personal computers while using SUN server for the Web server. For live view at the satellite studio, a VDOLive broadcast server and a CuSEEMe personal computer are used. For the server for live studio view purposes, a server located outside Tokyo FM is utilized since the circuit load of such a server is rather high. An ISDN line is used for transmission of still and dynamic picture information between the satellite studio and Tokyo FM.

ACCESS TRAFFIC ANALYSIS

The monthly trend of the number of page views of "TFM on the Web" is as shown in Fig. 1. Before March 1997 less change was observed in the number of page views but since April 1997 the number of page views has increased.

Fig-1 monthly pageview on TokyoFM website

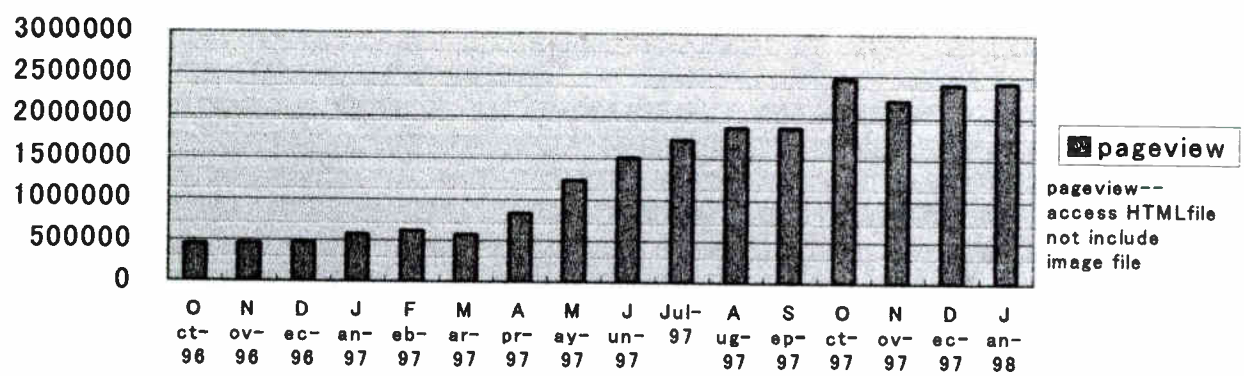


Fig-2 monthly hosts visited TokyoFM

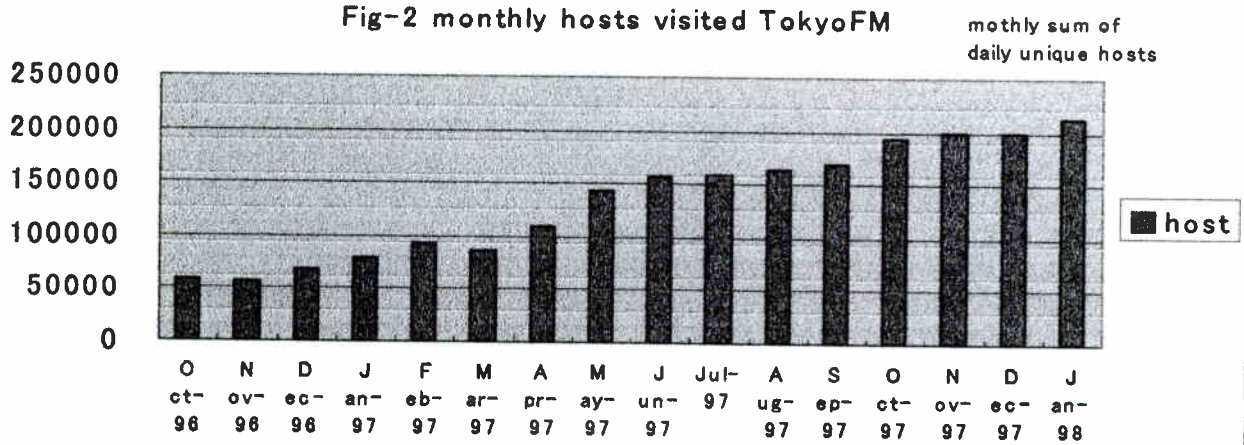
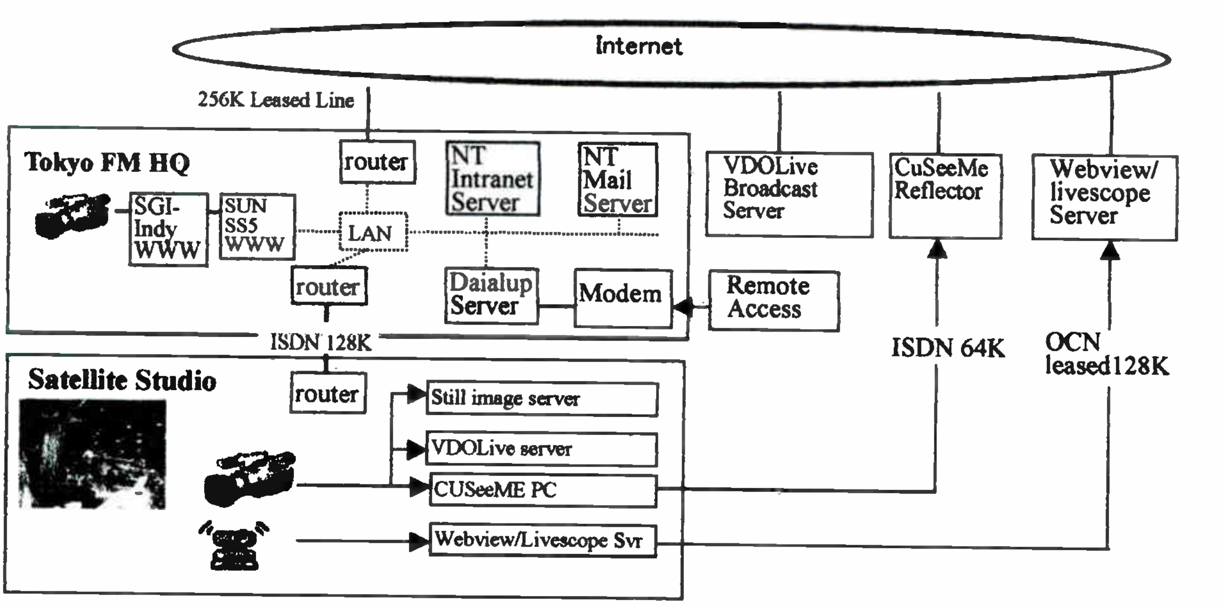


Fig-3 System Configuration



This is because the number of accesses has increased thanks to the response improvement achieved by raising the transmission rate of the leased lines from 128kbps to 256kbps in April but since October the utilization of the leased lines has partially saturated (100%) and increase in the number of accesses has stopped. In addition, the CPU load ratio of the Web server may sometimes exceed 100%. Because of the limits in both performances, increase in the number of accesses has been limited.

The weekly trend of the number of accesses is shown in Fig. 4. On Mondays more accesses are made than on other weekdays for check of updating of the contents by the company. On Saturdays the number of accesses somewhat increases because of the high popularity of the "Live Studio View" program described later and on Sundays the number of accesses from home somewhat increases also because of program linked pages prepared by popular artists. The number of accesses per hour is as shown in Fig. 5. The peak in the range from 10 a.m. to 20 at night corresponds to those accesses made by using Internet leased lines of companies or schools of the users and the peak in the range of 22 to 6 a.m. corresponds to accesses from home. This is because exchange lines are used mostly at home, and accesses are concentrated at the time zone of constant charging (from 23 to 8 o'clock) available by a special contract with NTT since otherwise subscribers are required to pay in the ordinary meter rate charging system. That is, it is important to present attractive Internet programs in that time zone of constant charging for further increasing the number of accesses in Japan.

Fig-4 daily pageview TokyoFM Website

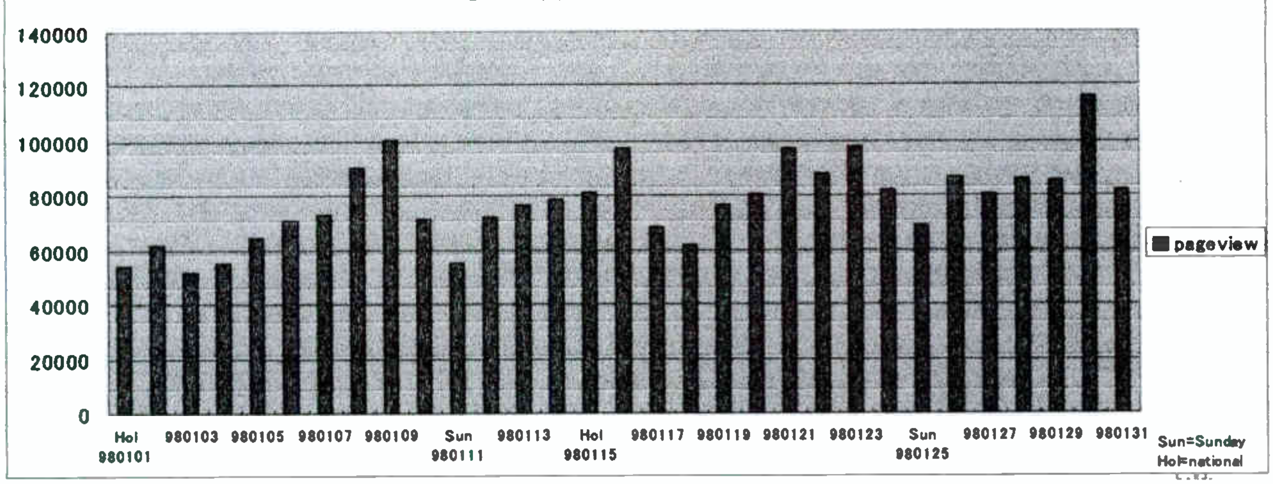
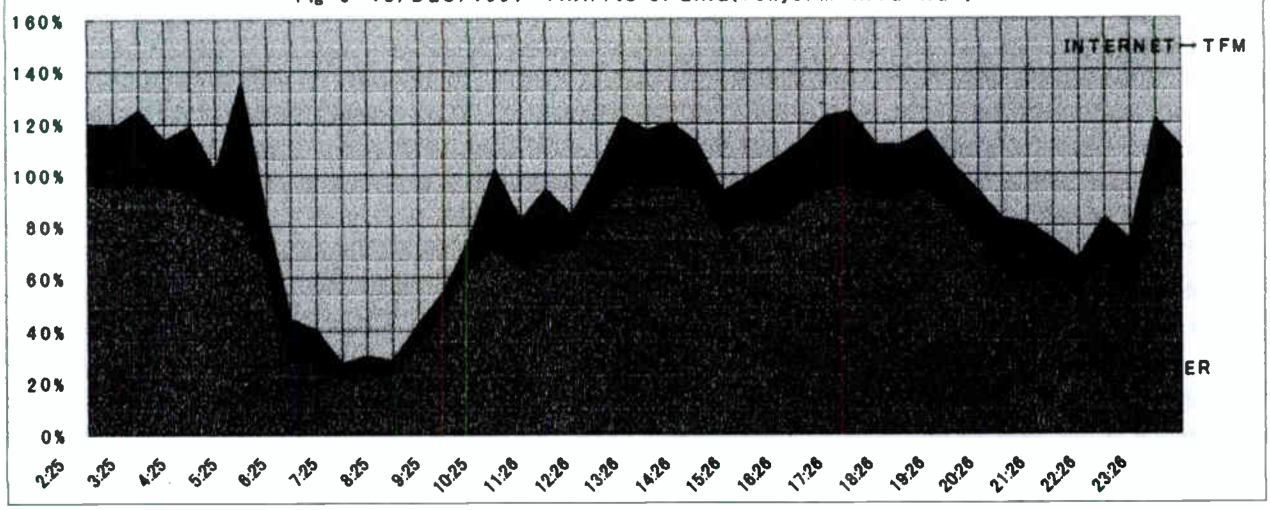


Fig-5 18/DEC/1997 TRAFFIC of LINE(TokyoFM-INTERNET)



TREND OF JAPANESE USERS

A survey on the trend of Japanese Internet users was made for about 7,000 people in October 1995. As a result of this survey, it has been found that (1) a total of 46.8% of people in a range of 21 to 30 years old use the Internet and this percentage becomes 70.2% in the case of people in a range of 21 to 35y ears old. The major listener bracket of Tokyo FM (that is, those who purchase by information from commercial messages) is twenties and meets the major user bracket of the Internet. Also, (2) the percentage of mail users of the Internet is as large as 91.4%, which meets the outstandingly large percentage of mail listeners of Tokyo FM. (3) The locations of accesses are office or place of work (44.2%), home (34.5%) and school (20.9%). Daytime accesses at user's office or school amount to 65.1%. In consideration of the environment of daytime connection with the provider, which is mostly made by leased lines at offices and by extension lines at home (more than 60% accesses are at transmission rates of less than 28.8kbps), it can be estimated that applications handling a large amount of data (such as still and dynamic pictures and audio) are accessed daytime and text pages and the like are accessed at night. However, the percentage of users of NTT's INS64 service (128kbps in bulk) is 15.2% and is increasing because of cost reduction in both leased line and ISDN equipment charges. The percentage of application users who handle large amounts of data at night also is expected to increase in the future.

"VISUAL INFORMATION RADIO" ON THE INTERNET

Tokyo FM has presented a teletext broadcasting service using the DARC (DAta Radio Channel) system called "Visual Information Radio" since April 1996. This broadcasting service presents information of 15 double byte characters × 2 lines per page and can transmit a maximum of about 60 pages per channel. The programs of the "Visual Information Radio" linked with audio broadcasting and independent programs not related with audio broadcasting, such as news reporting, weather forecast, traffic information, racing information, etc., are broadcast. Programs with unique contents are produced at each station of JFN as in the case of music broadcasting, and reuse of contents in other media can be expected from the standpoint of the single-source, multi-use policy of program contents in the future for an integrated group of Tokyo FM and JFN. The information and contents prepared initially for the "Visible Information Radio" can be subjected to secondary use in the Internet, or vice versa, which is expected to be realized in the very near future.

CONTENTS

Some popular contents of "TFM on the Web" are described below.

①Linkage with programs

• Real-time linkage with programs

While a program is broadcast on the air, opinions or comments are written in, questionnaires are answered, voting is made, radio drama selection is made by tuning into a particular program and the contents of these activities are adopted real time in live programs. By this, two-way communication programs with listeners' participation can be produced. By allowing other listeners to make comments or give opinions in a visible way, a close linkage among listeners can be established. By providing an E-mail address for each program, Tokyo FM has come to have more positive listeners' participation in programs, have more convenient means of expressing listeners' opinions and comments and be able to have listeners of wider brackets participated in its programs.

• Use of Internet as a complementary means for programs

FM broadcasting has been considered as an audio radio service to be listened but now use of the Internet in combination allows program contents and materials used in the past broadcast programs to be displayed to listeners. Introducing program hosts, personalities, guests, staff members, etc., by audio and video reviews allows the distance between the program and each listener to be much shortened. In shopping by audio broadcasting, goods or products to be purchased cannot be usually seen but can be looked at and confirmed on the Internet if utilized in combination, contributing to raising the consumer purchasing power.

②Video Information Distribution Service

After 5 p.m. on every Monday through Friday and after 0 p.m. on every Saturday Tokyo FM has a relay broadcast from Tokyo FM Satellite Studio located at Shibuya in Tokyo (shown in Fig. 6). This satellite studio, located on the first floor of a department store on a busy street, forms an open studio and is crowded every day with sightseeing and shopping people. Thousands of people make a long line waiting in front of the satellite studio when a popular

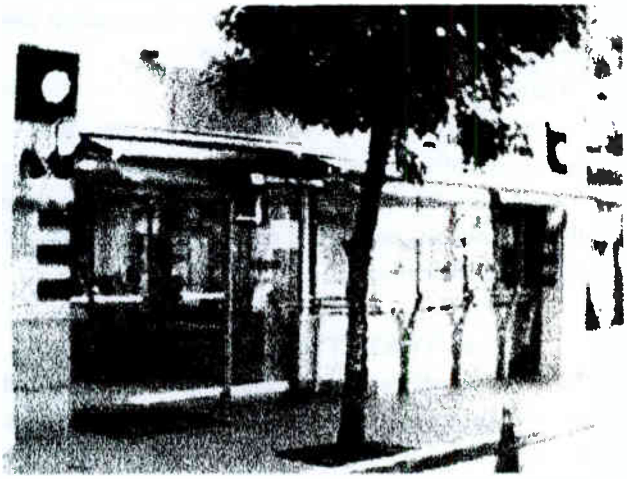


Fig-6
SPAIN-ZAKA studio
'OPEN' LIVE ON AIR STUDIO
at Shibuya,TOKYO.

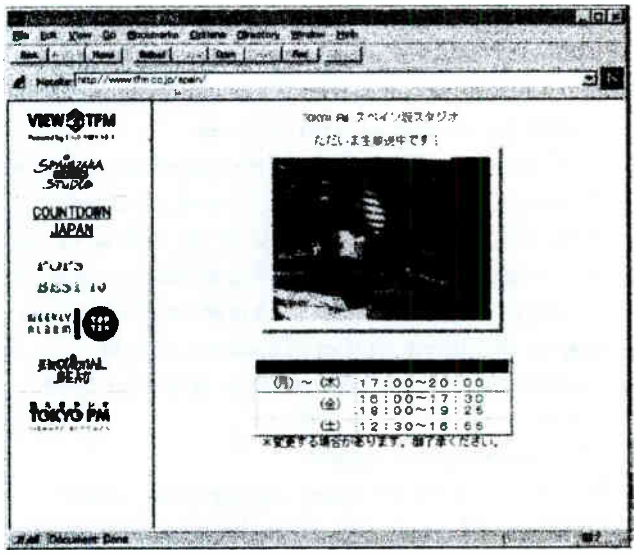


Fig-7
LIVE ON-AIR booth at SPAIN-ZAKA studio.
still JPEG image reloading per a minute

Fig-8
LIVE ON-AIR booth at SPAIN-ZAKA studio.
Motion picture,client user can move camera.
(CANON Webview/Livescope system)

star or personality is expected to perform as a guest. Indeed, the satellite studio is one of the noted places in Shibuya for people's big gathering around it. The scene in the satellite studio can be seen on "TFM on the Web." The satellite studio and the Web server are interconnected through an ISDN line and allows JPEG still pictures (Fig. 7) to be updated every minute, "CuSEEMe," "VDOLive," and dynamic picture video utilizing the "WEBVIEW/LIVESCOPE" shown in Fig. 8 (which is a dynamic picture distribution system developed by Canon and with which the camera angle can be remotely controlled) to be watched.

By introducing these contents, listeners usually incapable of visiting the satellite studio can now watch the studio. Indeed, these contents have a good reputation. Apart from the studio, the whole station building of Tokyo FM is located next to the Imperial Palace nearly at the center of Tokyo, so that the camera set on the roof of the station building is continuously shooting nature of the Imperial Palace in all the four seasons of the year. The video output from the camera is passed to a live camera connected to the INDY and can be watched on the Web as the latest picture captured and updated every minute. This content is also popular among overseas Internet users also.

③Audio Program Distribution Service

Some music programs broadcast in the past can be listened to although depending on the program. At present, CD sound sources cannot be broadcast on the Internet because of the current copyright restrictions in Japan, and desired free talking portions without music or unbroadcasted portions in the past programs on some particular days or in some particular program corners can be listened to form of real audio streaming. Broadcast programs which a listener failed to listen to in the past can be listened to at any place in the world where access to the Internet is achievable. If the copyright problem is solved in the future, not only past programs but also live programs can be listened to as they are on the Web by connection to the sound source database server in the digitized studio currently designed to commence service in the near future.

④Event Information and Tokyo FM Publication Information

Information on the events, concerts and movies (such as the Gallery of Takeshi Kitano who received the Grand Prix in the International Movie Festival in Venice) sponsored or supported by Tokyo FM, information on photograph collections and books such as essays sold by Tokyo FM Publication and information on related businesses appear. Some books published by Tokyo FM Publication can be purchased through "TFM on the Web."

⑤Listener registration

Listener registration is achievable on the Web, and registered listeners can enjoy special services such as reception of Christmas cards. By this, the listener attributes can be grasped and data meeting the needs of the sponsor can be collected.

FUTURE "TFM ON THE WEB"

The current problems will be improved and services will be added so as to further develop "TFM on the Web" as a medium.

• System redundancy

The current system is not furnished with redundancy, so that system down may be encountered in the event of failure. Therefore, it is desired to arrange the redundancy of the system including the storage medium and CPU.

• Development in system performance

Applications mainly in video are expected to increase in the future, which will require to have an increased processing capability of the CPU for the Web server use. The number of accesses is aimed at increasing up to 10 million page views/month which is about four times as large as the current number of accesses, and the performance of the whole system must be developed further.

• Development in line speed

In order to remove the current saturation in the line utilization efficiency and realize the target in the number of accesses about four times as large as the current number of accesses, it is necessary to have a line speed of 256kbps, or about five times as high as the current line speed, and, if possible, a line speed exceeding 2Mbps.

• Reinforcement in production system

The major causes for users' dissatisfaction with home pages provided by companies are: (1) less information than expected : 55.6%, (2) less updating frequency of information: 50.3%, and (3) difficult to have desired information: 49.4%, and even if the infrastructure for improving these insufficiencies is arranged, it is still doubtful and important to consider whether the contents provided are ultimately accepted or not. There are rather few successes in business achievements from the Internet for the time being, so that broadcasters tend to run their

Internet businesses at low costs. However, in order to make successes as media, it may be necessary to positively expand and reinforce the contents to be provided in the future.

• Electronic settlement of accounts

One of the major applications for positively utilizing the on-demand function which is a special feature of the Internet is electronic settlement of accounts. For the time being, electronic settlement in Japan is being under experiment and verification but we like to be positively engaged in developing suitable electronic settlement systems which will surely be very important for future Internet business.

• Development as media handler

"TFM on the Web" may be developed into as important a service as the existing broadcasting services. While the existing radio broadcasting services feature broadcasting type and temporary passing type in sending information, the Internet is capable of presenting individual services such as the on-demand service, storage capability, and multimedia services such as sending/receiving of video, audio, characters, etc. In the current Internet, though having these remarkable possibilities, the low line speeds between users and Internet providers actually cause a bottleneck in the development of applications which require a large volume of data for audio/video distribution, etc., as the prerequisite. If the up-link is used mainly for broadcasting type service including PUSH type service and the up-link mainly for on-demand type service using lines, both for distribution of a large amount of data, by future spread of satellite Internet, digitalization of terrestrial communication/broadcasting services, ISDN lines, etc., then the possibility of development of "TFM on the Web" as a medium surpassing the existing radio broadcasting services is extremely high. This infrastructure can also be used for distributing program data to network stations or selling audio program data, etc., for general purposes.

Virtual Production for Television

Tuesday, April 7, 1998
1:00 pm - 5:30 pm

Chairperson:
Robert Seidel
CBS Inc., New York, NY

Adding Value to a Virtual Set with Animatable 3D Graphics
Christopher Young
Discreet Logic, Inc.
Montreal, Canada

***How It's Done In Hollywood**
Bill Panepinto
Polhemus
Colchester, VT

Open Digital Video Systems for the Broadcast Market
Beth Rogozinski and Tim Myers
MACROMEDIA, Inc.
Redwood City, CA

***Implementing Virtual Set Technology: A User's Perspective**
William Chapman
Turner Entertainment Networks
Atlanta, GA

Understanding the Future of Video Editing and Compositing Tools in an Digital Era
Jared Vishney
Silicon Graphics, Inc.
Mountain View, CA

***Sliding Your Studio on to the Virtual Edge**
Bruce Erickson
Evans & Sutherland
Salt Lake City, UT

***Three Dimensional Content Outsourcing and Realistic CGI Car Creation in the Broadcast Industry**
Walter Noot
Viewpoint DataLabs Inc.
Orem, UT

An Innovative New Solution for Shadow Browsing for Preview and Broadcast Automation
Alan Chaney and Dr. Tony King
Telemedia Systems Ltd
Cambridge, England

***New Graphics Systems for the Digital Production Environment**

Christopher Young
Discreet Logic
Montreal, Canada

*Papers not available at the time of publication

ADDING VALUE TO A VIRTUAL STUDIO WITH ANIMATABLE 3D GRAPHICS

Christopher Young, Product Manager
Discreet Logic Inc.
Montreal, Quebec

ABSTRACT

With new advances in real-time rendering 3D technology, the production value achieved with virtual studio applications is becoming more and more attractive to broadcast program producers. However, one of the major drawbacks of virtual studios has been the inability to present timely information in the form of animated 3D graphics. New developments in virtual studio software are providing new ways of presenting newsworthy information. Producers are accepting this new technology that not only allows them to create a unique virtual world but also continue to provide services that a broadcaster must provide: presenting timely and accurate information.

INTRODUCTION

This paper will examine the current state of the technology behind real-time 3D graphics systems as they are used in broadcast video applications as important components in a virtual studio system. In addition, a detailed discussion of the hardware, software, and supporting technology for the implementation of this kind of real-time graphics creation is presented. Finally, we will present numerous examples of how to use this technology to add value to broadcast productions.

For illustration, we will be using examples from Discreet Logic's VAPOUR™ virtual set technology which benefits from the external control interface and internal real-time power of out FROST™ 3D Graphics systems. Both systems utilize non-compressed video and a modular functional structure, encouraging future integration with editing and effects products from Discreet Logic and other manufacturers.

Virtual studio systems use different technologies that, when combined, create the illusion of actors immersed in 3D virtual reality environments. In these environments, all movements of the studio camera are precisely replicated by a "virtual" camera that resides inside a high-performance graphics computer. By combining live action and virtual elements, broadcast professionals can create a dynamic on-air environments that is an attractive alternative or complement to physical sets. Depending on the platform used, graphics spanning a wide spectrum of complexity can be created, with various numbers of video I/O and scaleable degrees and anti-aliasing, all of which affect the level of realism and image quality that can be achieved.

Previously, the challenge has been that the 3D virtual reality environments were limited by the requirement of rendering pre-defined scenes to disk. Although the people investing in virtual studios utilizing blue screen technologies have primarily been broadcasters, the requirements of pre-building the intended 3D graphics has meant that the tools which have been provided don't meet the needs of a broadcast user. Therefore, "live" inserts of real time data visualization during on-air productions such as sports shows, news programming, and special events such as election returns have not been practical until now, which is one reason virtual set technology has not found as broad an acceptance in the United States as it is abroad. This paper will examine how virtual reality and real-time data visualization technology can be combined to provide a truly interactive 3D environment which can allow TV program producers and presenters to convey more information with greater clarity while maintaining a consistently smooth feel to their graphical presentations.

DESIGNING FOR REAL-TIME

By now, 3D modeling and animation techniques are well-known to the entertainment industry. Simply put, 3D modeling and animation enable graphic elements to be drawn and animated in three-dimensional space. Once a basic geometric shape, or primitive, is drawn, properties, such as colour, material, and texture, can be applied to it. The addition of light sources to the scene can enhance perspective and realism, causing objects to cast realistic shadows or be shadowed by another object. Moreover, objects can be animated, allowing size, position, and shape to be dynamically modified over time. But none of this is particularly new.

What is new is the ability to draw, model, and animate objects in "real time". In the context of motion video, real-time rendering means that a new image must be generated at a rate of 30 frames per second for 525 line systems (720 x 486 image size) and 25 frames per second for 625 line systems (720 x 576 image size). Utilizing today's more robust computing power images no longer have to be pre-rendered to disk storage, thereby enabling real-time interactivity. Each image is calculated either according to the point-of-view of the virtual camera provided by the user, or by software coupling to studio cameras or other motion control devices. This way, the combination of high-performance graphics computing engines and rendering algorithms enable real-time 3D solutions for broadcast video users.

Modeler. At the heart of this real-time virtual set technology is the modeler interface as shown in Figure 1. The modeler's main purpose is to:

- Manage 3D projects and libraries,
- Create and edit 3D model files,
- Create and edit 3D animations,
- Pre-visualize graphic models and animations, and
- Register virtual sets.

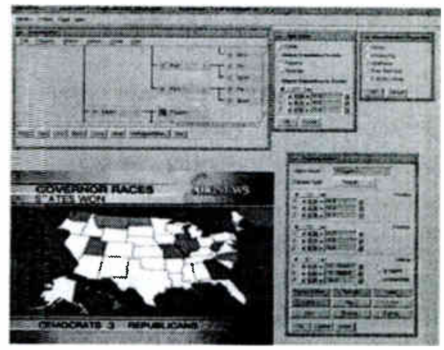


Figure 1: VAPOUR user interface

Ideally, a simple 3D modeler offers both artists and producers enhanced creative freedom when performing 3D authoring for video production. Graphic primitives and basic editing tools should allow artists to import models and attributes from such traditional post-production 3D authoring tools as SoftImage 3D and Alias PowerAnimator, letting the virtual set system optimize them for real-time applications . 3D editing tools should enable producers to make quick scene changes during rehearsals, with minimal production delay.

To optimize a scene for real-time performance, it is crucial to know the hardware well enough to trade off performance-limiting factors. Only a system that provides total external control over managing both the total polygon count and texture management will provide performance at maximum efficiency. Let's consider these factors individually.

Polygon Count. The number of "polygons" — individual triangles or rectangles comprising a set of 3D objects — in a scene will substantially effect the amount of time and effort required by the computer and the graphics engine to render the scene. If the number of polygons in a scene grows too high, even the most powerful system will be unable to draw the scene in the time available between video frames. Invariably, the update rate will be unacceptable, requiring the designer to reduce the number of polygons. However, designers usually want to use as many polygons as possible since the number of polygons in an object helps determine the realism of the object's appearance. It is therefore very important to possess tools and an environment that will effectively analyze polygon counts, determine hot spots, and enable intelligent reduction of polygon counts utilized to create an object.

Texture Management. Texturing is used to decrease polygon count, while adding detail and realism to a scene. For this reason, texture management is a primary component for real-time systems. Artists have access to many texture creation tools, such as Adobe Photoshop in a small graphics suite, or Discreet Logic's FLINT™ or FLAME™ in a larger post-production facility.

To ensure that the systems are used as efficiently as possible in terms of using texture storage (in the raster manager) and reducing drawing time (a function of the pixel fill rate of the graphics subsystem), graphics systems usually have strict rules on how textures must be set up and used. The software application should make nearly all of this transparent to users, without preventing designers from providing hints or controls when appropriate.

Even on the highest-end systems, texture memory is limited. This requires designers to closely monitor how texture storage is used. For example, large amounts of texture memory can be saved if they are not used for texturing objects in the distance. A good system should provide tools to analyze texture wastage, as well as easy ways to trim and shrink textures, further improving texture memory usage.

Animations and External Control. To facilitate real-time virtual set production, External Control provides access to all object parameters and animations and computer subsystem operations through an ASCII control protocol very similar to traditional broadcast automation protocols.

This means that all object properties, including animations defined in the animation channel editor shown in figure 2, within a 3D graphic will be accessible to external systems. The graphics system can therefore be controlled by automation systems or new asset management protocols, providing the connection to other hosts in the same studio (for playback control and/or data acquisition) or in another city (remote control and/or data acquisition).

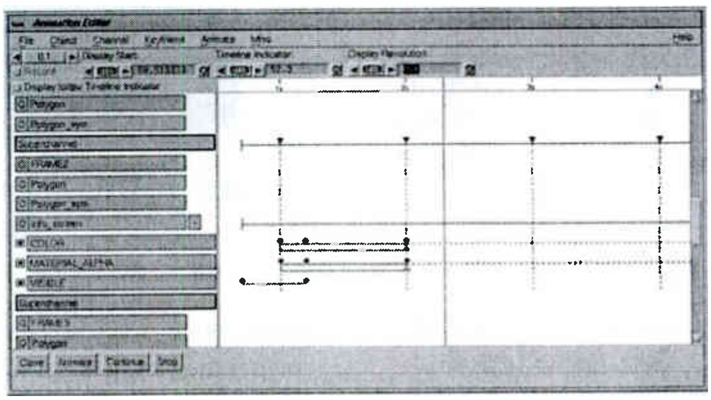


Figure 2: VAPOUR Animation Editor

Some typical external real-time control requirements involve the remote triggering of graphic sequences and an unlimited number of mix effects and animations at the object level, or adding intelligence to the handling of the graphics and animations. For example, if the Dallas Cowboys take the lead in a football game, an external control application might reverse the order of the teams, change the colour of the leading team, and reveal the words JUST IN without changing the graphic template.

Another example of external control applications would involve connecting to another host that receives news wires, financial data, sports scores, election results and weather information and changing graphics, text and attributes. In this way, data can be retrieved and graphic attributes can be modified accordingly. Still another instance would be connecting the 3D graphics system to other databases—such as Oracle, Informix, Sybase, MS Access or any other SQL or ODBC compliant database—to extract sports statistics, team rosters, or template attributes for a particular item such as all colours, textures, spelling, and fonts associated with the Dallas Cowboys during the 1996 season. This could be a very powerful tool for communicating football, hockey, basketball and baseball statistics.

TOWARD A REAL-TIME VIRTUAL STUDIO SYSTEM

What is needed in today's real time production environment is a 3D Virtual Studio Solution designed for seamlessly integrating actors into 3D environments that can then be animated interactively via external control at the touch of a button.

Using an Application Programmer's Interface (API), this Virtual Studio Solution should render 3D imagery based on cues received from outside sources: touch-screens or data sources, such as voting and polling data services, stock quote services, news databases, or interactive telephone systems for game shows or children's programming.

This 3D imagery should combine all the important features required for titling, DVE moves and effects, ideally generated in real-time without the need to pre-render. This drastically reduces the time required to generate graphics, since rendering time is eliminated. By producing its visuals on-the-fly and in real-time such a 3D graphics system also reduces the need for mass storage and external still stores. Text crawls, reveals, clocks and stock tickers can all be built as templates for daily reuse with other information supplied from external sources.

Real-time DVE moves can be used to fly live video through a 3D scene, then mixed to full screen to emulate traditional 3D DVE moves typical in news and sports programming. "Virtual flight", which is a simulated camera movement that has no correlation with any real camera movement, can be used in the introduction to a show, prior to the actor's appearance on set, giving the effect that the camera is traveling from a great distance—a much greater distance than is possible in the studio. A similar virtual flight was used to fly down the ABC News "space tunnel", passing billboards displaying live video and live election results as shown in figure 3.

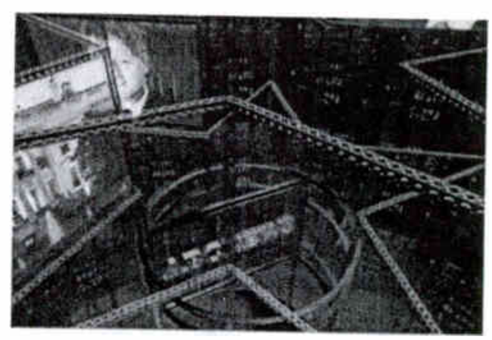


Figure 3: ABC News space tunnel virtual flight

This Virtual Studio Solution must also incorporate a set registration module required to match the physical set to the computer generated virtual set. It should also include a sophisticated real-time rendering algorithm and the ability for on-the-fly camera switching.

Virtual Studio Architecture.
There are five major components to a virtual studio facility (figure 4):

- Renderer
- Camera tracking system
- Real-time compositing
- Camera switching
- The physical blue-screen studio

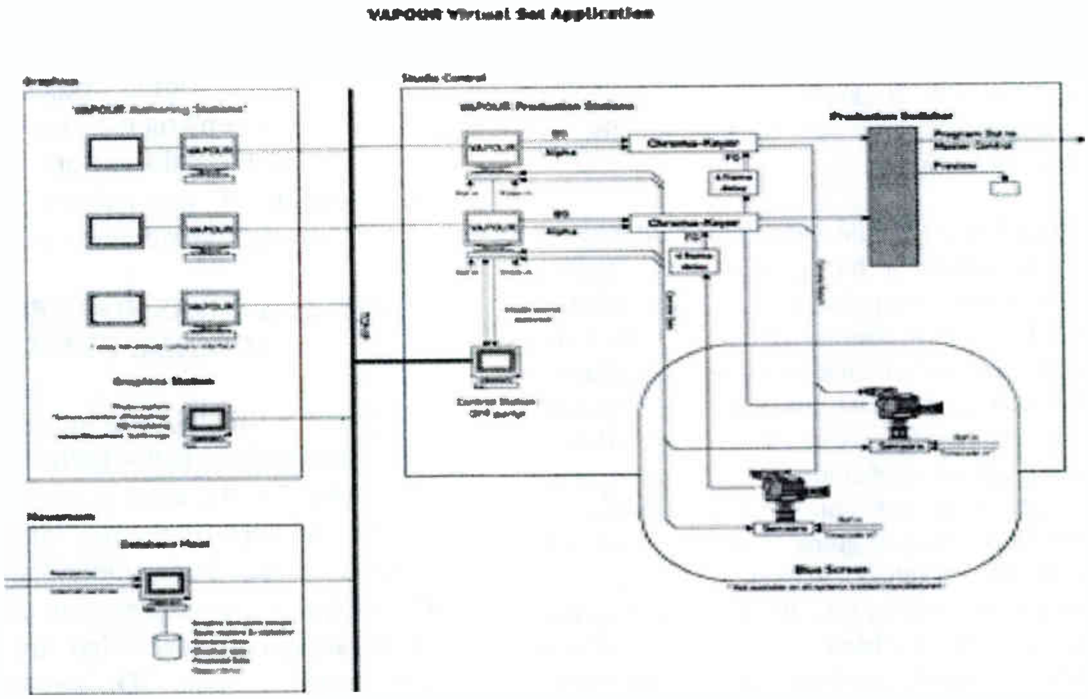


Figure 4: Virtual studio technical block diagram

The Renderer. The renderer generates the background scene in real-time. The rendered scene is then combined with foreground video provided by the real camera to create the final composited scene. The renderer needs to produce the program channel synchronized to field rate for the selected video output format and perform stress management at the rendering channel level to assure real-time output production through level of detail (LOD), dynamic resolution, and channel property controls. The key features of a renderer for virtual set applications include:

- Double buffering,
- Z-buffering,
- RGB colouring,
- Anti-aliasing,
- User defined background colour and matte Infinite, local spotlight and simulated spotlight lighting,
- Texturing transparency and blending,
- Matte information,
- Layering,
- Live video,
- Shadows,
- Fog,
- Depth of field.

Camera Switching. Camera switching is an important component in any virtual studio facility. Virtual studio camera switching provides two main advantages:

- The ability to switch camera foregrounds and virtual backgrounds using a single, high-end graphics workstation
- The ability to preview a camera foreground and virtual background using a lower-performance graphics workstation for the next-to-air camera.

The ideal Virtual Studio Solution should have the ability to render a background image from a single camera viewpoint at any given time. It should be able to frame-accurately switch this viewpoint to another camera on the same graphics workstation to generate the virtual background, thereby eliminating any need for two real-time, on-line graphics workstation. A preview channel also ought to be available, so that camera shots can be framed before going to air using a lower-performance graphics workstation. This will permit the operator to take the previewed camera to air, and leave the ideal Virtual Studio Solution to make the appropriate switches between the on-line and preview graphics workstation.

Set Registration. Set registration matches the real-world coordinate system—the cameras and the blue screen—with the virtual system, or scene database. This transformation must be specified by the user, and is unique to every combination of virtual scene and blue screen. A given scene, in other words, will require a new registration process whenever it runs on a new bluescreen setup. At any time, a single registration is in effect for the entire production, regardless of the number of cameras used.

Each camera tracking system has its own method of calibration, each with its own level of precision. The most precise calibration procedure requires the use of a Theodolite, a standard surveyor's distance measuring tool. The first time setup of a physical bluescreen studio requires Theodolite measurement of physical points in and around the cyclorama. These become reference points, enabling the cameramen to zoom into them when locating camera positions. All movements of the cameras will be sensed relative to those initial locations. The initial process when the bluescreen is constructed can take up to two hours. Once the cameras are positioned, the triangulation of the camera locations take approximately five minutes each.

APPLICATIONS FOR SPORTS AND SPECIAL EVENTS

A system with its own modeling tools and real-time renderer saves both time and money by eliminating the need to pre-render graphics for the various outcomes of Sporting and Special Events. The same system that creates the virtual environment can also create the informational graphics that are presented to the viewer. The 3D, keying and audio triggering capabilities this system would also provide sports broadcasters with the ability to animate plays and lineups in innovative ways.

When incorporated into Special Events telecasts, this kind of a real time graphics system used in a live-to-air context would let its 3D images drive the production in a way far more visually interesting than the pre-rendered graphics used today. Producers could then call, preview, and take graphics as much as they would with cameras. During special event coverage, the producer would be guaranteed that the data and graphic content is up to date and reflects the current outcome of any race. In fact, external devices such as traditional CG keyboards and touch-screen panels can be used to automatically send cues to the 3D graphics system.

Implementing real-time broadcast-quality 3D graphics is an investment. Some of the tangible and intangible ways in which real-time 3D systems add value to a users' productions are presented below:

ADDING VALUE TO A PRODUCTION WITH 3D GRAPHICS

One of the most important characteristics of a news presenter is the ability to convey the right information quickly and accurately. Sometimes, it is simply too time-consuming to convey vast amounts of information with traditional 2D type graphics. Viewers understand relative, not absolute, information. Being able to present numbers and objects in 3D with relative proportions and perspectives conveys much more information, more quickly than any 2D image. By placing the control of what data and graphic is being displayed with the presenter in real-time, it becomes even easier to effectively communicate with the viewer.

The next step is, of course, extending the 3D imagery to a full virtual environment that immerses and tracks the presenter. Most broadcast professionals and production facilities are familiar with bluescreen productions for weather and background replacement. In traditional systems, however, animations are usually pre-rendered and the camera is locked down to maintain realism. With full 3D virtual studios, presenters have the freedom to interact with the information that they are trying to convey, adding a higher level of clarity to the presentation.

Increase Studio Operating Efficiencies. A real-time virtual set system can reduce studio downtime and reduce operating budgets, since it allows your artists and production crew to quickly add or change scenery at any stage during the production process, including during rehearsals, without delaying the crew or schedule. Sets can be created in an off-line authoring environment, keeping the main production workstations and studio operational and generating revenue at all times.

With a real-time system 3D graphics and animations do not need to be pre-rendered, an expensive process in broadcast production cycles where massive amounts of storage are required to accommodate all possible graphic combinations.

Invest in and Protect a Program's "Look and Feel". Creating that special look for a program is a task for the art director and the graphics department. It is a difficult process, requiring a significant amount of time. The approved look must remain consistent from day-to-day and from person-to-person. To protect this investment in time, energy and money, the ability to template styles, fix fonts and lock down designs is important in any graphics system.

A real-time virtual set graphics system would have the ability to create and store graphic templates without rendering the image to disk. This means that the data, textures, fonts and imagery in the graphic can be changed at anytime, including as it is being rendered live-to-air. The change can come directly from the operator or remotely from an external database. The system would then immediately update its scene database for the template and render the change to the program output when the graphic is cued, while keeping all the other template information fixed for consistency.

Increase the Productivity of the Graphics Department. Since these real-time systems work with graphic templates, the job of supplying the content can be turned over to those who are accountable for the content, as TV2 Norway has done. Journalists can select the imagery and text they would like to see to enhance the impact of the presentation of their story. This way, the graphics department can continue to create new program graphic styles, or build libraries of new objects and textures, that can be used by the journalists.

Author: Christopher Young holds the position of Product Manager for Broadcast Production products with Discreet Logic, an award-winning provider of effects, editing and graphics solutions for the visual production industry located in Montreal, Canada. Mr. Young has been with Discreet Logic since 1995, and been involved in defining Discreet Logic virtual studio and broadcast graphics product strategies. Mr. Young has experience designing production switchers and satellite transmission systems and holds a Bachelor of Applied Science degree in Electrical Engineering from the University of Waterloo and a Master of Business Administration from McGill University.

OPEN DIGITAL VIDEO SYSTEMS FOR THE BROADCAST MARKET

Beth Rogozinski and Tim Myers
Macromedia, Inc
101 Redwood Shores Parkway
Redwood City, CA 94065

ABSTRACT

Open digital video systems are based on standard media layers such as QuickTime or DirectShow. They allow customers to run relatively inexpensive software and hardware—mixing and matching according to creative and productive needs. New technologies make open systems reliable, faster, and more affordable than ever, causing the broadcast industry to expand its use of open, digital video systems in production environments. Proprietary digital video systems cost up to $500,000. Depending on your production needs, open systems range from $20,000 - 50,000. Open systems have additional advantages over other production tools as well, including: 1)promulgation of greater competition resulting in faster innovation and competitive prices; 2) facilitation of data exchange between all the various and available products via software and hardware interchange; 3) access to new technologies including streaming video for the Internet and Intranets;
4) new delivery options including repurposing media for the web, multimedia and other applications. Open systems media tools facilitate the creative processes and simultaneously grow and improve whole new methods of communication and technology.

This paper reviews the state of the art of digital media production tools. It discusses open media technology solutions using Windows and Macintosh systems, digital video boards, editing and compositing software, hard drives, alternative storage, and more. A point by point review of the benefits of using open media systems explicates their current and future advantage over other production tools and solutions. On the software side, the necessary features of professional productivity systems are also analyzed. Finally, the return on investment for open media system solutions is reviewed.

OPEN SYSTEMS TRENDS

As technology tools become prevalent in the realms of art and communications, a new trend towards open systems is quickly emerging. Where once there were separate tools designed to do disparate jobs, there are now interconnectable tools that do many jobs– and do them faster and more efficiently. Open systems, based on these interconnected tools, are reliant upon effective media layers to bridge the communication between the OS, the hardware, the media and the production software. Robust software packages that power the needed solutions are also mandated for effective production. While open systems tools have existed for quite sometime, a number of barriers have recently been overcome that make this technology pathway the most desirable. A primary facilitator of open systems is the maturation of digital media layers such as QuickTime. Among the many other advances that add to the attractiveness of open systems solutions are the increasing speed of processors, increasing throughput of data, lowering costs of storage and the acceptance of delivery and exchange protocol standards for easy networking, collaborative work environments and distribution. The advent of open systems technology will hasten the evolution of broadcast and media creation facilities into efficient digital production facilities. The direct and immediate results will be: a more efficient workforce; faster and cheaper implementation of new technologies due to increased competition in the market; increasingly productive and creative system solutions; and faster transitions to alternative delivery mechanisms and alternative revenue streams.

Digital Media Technology

Digital media technology came into use in the 1980's with the burgeoning of desktop publishing and the beginning of the multimedia phenomenon. These nascent media technologies were expensive, difficult to use and unreliable. In addition, the first digital media solutions to appear in production environments were little more than replacement technologies that could do the titling and effects that were, until that time, done manually. While these early systems did not dramatically augment the creative process, after sufficient time and under the auspices of creative talent, the usefulness of digital solutions began to evolve as professionals pushed them to create the objects of their imagination. Within the past decade these systems have moved from the replacement business and are now serving as the creative, revolutionary tools that all experts regard as essential.

Black Box/Proprietary Systems

The early media technology systems, and most of the systems in place today, are considered proprietary or black box systems. These systems, most notably from Avid and Quantel, offer creative production people great advantages over their existing analog equipment. Among the many advantages provided by new digital systems are that they are faster and more productive than their analog cousins and thereby provide more creative freedom. In a digital environment, there is no image deterioration between versions and edits, eliminating the destructive nature of the original creative process and retaining the highest value of the media assets. Digital systems are also able to efficiently database assets and recall from both on-line and off-line, a feat that both adds value to existing assets and increases productivity as assets are easier to find and use. Increased productivity and decreased costs are also extended by the fact that digital systems can proffer editing, effects and compositing in one environment eliminating the need for multiple work areas staffed by multiple artists and technicians.

While these black box systems have done much to advance the productivity of many studios, they are fraught with impediments. The most notable obstacles inherent in black box solutions are that they are simultaneously very expensive and are often not at all flexible or scalable. This means that users who purchase these systems spend a lot of money on a solution that does not easily work with other systems they have in place or may want to add. Additionally, any types of upgrade paths to these systems must come from the original manufacturer and/or their authorized partners. (See the example in **Return on Investment** section of this paper.) While this situation has, to date, been acceptable, it does not take advantage of the wide range of industry advances that continue to occur at an ever-increasing speed. In addition, proprietary systems require training and support that are, again, likely only available from the original vendor or their authorized agents. Because of the closed nature of these systems, they lock the customer into a price point, support system and upgrade path that may not reflect their needs or the general market trends.

Open Systems

Open systems technologies for media production provide all the advantages of the established black box solutions with the added value of being open. The term "open systems" indicates a system that uses standard API's and is open to all hardware configurations and software applications. The ideal of the open systems architecture is that all existing and subsequent technologies that embrace the standard will be connectable, integratable and to some degree interchangeable. The facilitation of technology interchange and connectivity will lead to dramatic decreases in cost and increases in feature sets of media technology solutions. While technology is only beginning to approach the ideal of open systems, some significant advances have been made that will facilitate achieving this goal.

OPEN MEDIA LAYERS

Open media layers, system software extensions that make media, most notably video, usable on a computer, are the key to a truly open digital video production system. Media layers are an API (Application Programming Interface) that allow software developers to write applications that deal with media files. They contain built in codecs (compression/decompression algorithms), that translate video, audio, and image files into a form that the processor understands. They are extendible as well, allowing additional codecs to be plugged in for customization and expansion of options. They also facilitate communication between the software applications and a variety of hardware products, including I/O boards and video digitizers. Open system media API's run on a variety of personal computers and are adept on the standard systems used in business and production: Macintosh, Windows and UNIX. As the media layer is common across platforms, these disparate systems are connectable making the entire open system media production environment very adaptable and very scalable.

Robust and standard media layer API's allow software vendors to focus their development efforts on user interface, work flow and productivity features instead of writing driver code for a variety of devices. Utilizing open media layers, applications operate with software codecs or very elaborate hardware capable of supporting uncompressed ITVR 601, as well as other standard production codecs. For example, QuickTime, an open media layer, supports JPEG, MPEG, DV, Cinepak, Sorrenson, Duck, Uncompressed*, amongst other codecs and formats. For hardware manufacturers, conforming to standard media API's means that their boards and peripherals will interface with all compatible systems without hardware conformations, software drivers or patches. Like software developers, hardware manufacturers benefit by concentrating their development cycles on innovative features rather than conformation issues. For all technology concerns, open media layers can result in greater penetration of their product in the marketplace, hence greater revenues. End-users reap the greatest rewards of the open API, as hardware and software products are evaluated in similar environs and systems. In similar environments, innovative features are rapidly demarcated, design flaws are easier to recognize and pricing becomes extremely competitive. While these development architectures are created by OS companies interested in making set-up and maintenance easy – thereby making their products ubiquitous - it is clear that open systems also help hardware and software developers and this, in turn, benefits media producers.

* Uncompressed format, while not a codec, is a crucial production option.

Open Media layers have recently matured to new levels of professional functionality. They are currently capable of running multiple media streams at sustained data rates of 40 MB/s. They support a growing number of formats and file types and are easily extended. They support real-time effects such as two channel dissolves with alpha keying and can play long spans of synchronized audio and video. Some open media layers allow time-code and other metadata to synchronize and track with the video data.

Common Media Layers

Currently, there are two common open media layers, Microsoft's DirectShow and Apple's QuickTime. DirectShow is a fairly recent product that Microsoft developed to replace Video for Windows. While Video for Windows was designed as a playout media layer (a facilitator for playing video and media on the computer screen) and not as a creation environment, Microsoft regards DirectShow as a true API (though it too has its roots as a playout media layer.) DirectShow is currently limited to the Windows OS. It has no common file format for interchange and no standard API for the integration of high-end video cards. Microsoft continues to develop DirectShow and coming versions of this media layer should support all the needs of open systems for broadcast production. QuickTime, now 6 years old and inversion 3.0, already meets all of these needs. In its new incarnation, QuickTime is truly cross platform. It has a standardized API on Macintosh and Windows systems and is easily extended via plug-in components. These components allow QuickTime to have additional codecs, work with real-time hardware, track timecode and other logging information, and manage and extend the types of media files that can be used. With QuickTime, extensions don't alter the standard API, they simply add functionality, such as new codec and video capture options. The standard QuickTime file is interchangeable across many applications and platforms. QuickTime files are very flexible and can support all types of media - video, audio, animation, graphics and 3D images. The files also support augmentary file information such as thumbnails, pointers, media metadata, and media versioning. This data is especially critical when video assets are repurposed for alternative distribution such as the web and interactive television. QuickTime 3.0 also supports JPEG A and B standards, allowing data transfer between most boards without the need for transcoding.

Chart 1: QuickTime 3.0Features Chart

Allows use of custom clock & timers for sync
Designed to seamlessly play back multiple video files without processing
Animation APIs
File format with metadata
Multi-file live audio mixing support
MPEG playback (OM1-Compliant) and editing; Multilingual Track, Text Video CD 2.0; Non standard MPEG (VBR, Audio only, Layer1, etc.)
DV support for encode and decode for most camcorders and DV device control
Reference movies to allow several versions without using additional diskspace
High-end video formats such ITVR 601,YUV422, Open DML, OMF
Timecode support (Including SMPTE)
Support for major hardware codecs (C-cube, LSI, Zoran), dynamic codec exchange, and easy to add proprietary codecs
Vendor interpretable real time effects and effects acceleration
Cut list support
Edit by reference
Bookmarking
Internet features, i.e. URL Chasing and auto-go to
File streaming
Alternate data rate delivery
Poster frame embedding
Live broadcast streaming
Transcoding
Multi-platform file format (SGI, Win3.1, NT, Win 95, Mac)
Backward compatible to previous OS & System files and compatibility support(Win 3.1/16bit)(Mac system 7, 8) 486/66, Pentium, Pentium Pro, PII, Mac020, 030, 040, PPC

Source: Mitchell Weinstock, Senior QuickTime Product Manager. 1/98

TECHNOLOGY ADVANCES FACILITATING OPEN SYSTEMS

Computer component systems have advanced rapidly in the past several years both decreasing in costs and increasing in speed, capacity and reliability. Many components, including fixed disk, removable media, RAID storage (redundant arrays of inexpensive disks) and RAM are now considered commodities having become somewhat standardized and available according to free market systems such as market demand and international pricing structures. All of these commodity items are essential components of a video production tool set and all benefit open systems solutions by driving the total costs (system purchase plus cost of ownership - the costs associated with training, support and upgrades) down. Lower prices of these components will persuade more video and communications professionals into the digital realm. The trend of decreased cost and improved reliability and speed promises to continue.

Storage and Memory

A basic media production system requires tremendous amounts of both storage and RAM, previously an enormous expense in digital video systems. These prices have come down dramatically. Storage costs alone have dropped by up to 40%-50% in 1996. (Pipar Jaffray, 11/96)

One gigabyte of storage holds only about 2-3 minutes of high quality video and now costs about $130.00. (300KB/frames x 30fps x 60 = 540 MB per minute.) Not too long ago, the cost of the RAM in media technology systems represented more than the total costs of all other components. The mean average price of RAM has dropped for all computer systems, even high end and proprietary systems, though due to the proliferation of consumer and business systems and the large numbers of units being sold, the overall drop in the price of RAM for workstations and personal computers, as represented by Macintosh and Windows OS systems, has been extraordinary. Five years ago, a two-8MB SIMM RAM upgrade cost over $600.00. Today the same upgrade costs about $60.00. (The ChipMerchant) The minimum amount of RAM for a video production system is generally considered to be 128MB. Today, for Macintosh or Windows systems, this costs about $380.00. (The ChipMerchant)

In addition to decreases in costs, there have been great improvements in these standard technologies. RAM has been segregated into task specific chip sets, such as VRAM and WRAM, (video random access memory and window random access memory) which both are optimized and dedicated to facilitate video playback. Storage systems have also been optimized to meet the needs of media producers with the advent of A/V drives, fast disk arrays and more reliable removable storage systems. An A/V drive is a hard drive that has audio/video optimization, such as suspended thermal recalibration and specialized frame buffering to prevent frame loss. Disk arrays are combinations of hard disks that the system regards as a single disk. These can be arranged as RAID arrays that have built in redundancy from levels 0-5 for asset protection and system reliability. Removable storage devices, such as Jazz drives or magneto-optical disks, now have access times (the amount of time required to access a piece of information from the storage medium) that are almost equal to principal drives. Evolving alternative means of storage and networking, such as Ultra2SCSI, (80MB/s and a potential cable length up to 12 meters), SSA – Serial Storage Architecture (80MB/s) and Fibre Channel (100MB/s) are creating more efficient networked storage options that will further drive down the overall costs of storing and managing media data.

Processors and Throughput

In addition to access of inexpensive storage and memory, media production environments require fast processor speed, and fast data throughput. Processor speed (measured by clock speed of chips as well as chip architecture) has increased exponentially and has a theoretical improvement rate of about double every 18months. Known as Moore's law, processor improvements have consistently improved at this rate.

System buses have also improved dramatically in the past several years. Internal system bus speed is the measure of the speed of data flow from one board set to another - each board set containing hundreds if not thousands of chips that process data. Internal throughput of bus systems have gone through a process of advancement similar to the evolution of chip architecture, processor speed, and I/O capacity. This trend is accelerating as media production environments demand faster access to data. The measurement of throughput for internal buses is different for each system, but many hardware developers are creating systems with improved internal bandwidth specifically for large graphic and media files. The new 64 bit PCI bus standard, which is designed to be four times faster than the current 32 bit PCI bus, is a recent example of such advances.

Chart 2: Bus Bandwidth Evolution

Year	Bus	Width (bits)	Bus Speed (MHz)	Bus Bandwidth (MB/sec)
1979	8-bit ISA	8	8.33	8.3
1984 (still a mainstay)	16-bit ISA	16	8.33	16.6
1987	EISA	32	8.33	33.3
1992	VLB	32	33	133.3
1993	PCI	32	33	133.3
1997	3.3 V 64-bit PCI 2.1	64	66	533.3

Source: PC Webopeadia, 1998

Video I/O Boards

In addition to the rapid improvements and price reductions in computer components and systems, getting video and media into your system has never been easier, or less expensive. There are currently a number of video I/O cards and boards that are as scalable as the systems they work with. Beginning at a rock bottom price point of $499 for motion JPEG, the new video boards are reliable, easy to install and use, and offer professional results. Without detailed exploration of each board's facilities and benefits, suffice to say that where once creating a digital video system was left to the capable hands of a few experts, (who mostly worked for the proprietary systems manufacturers), now most technically savvy people can pull together their own systems.

Chart 3: Open System Video Boards

I/O Board	Manufacturer	Platform	I/O Capacity	Price
TARGA 2000 RTX	Truevision	Mac/PC	18 MB/s	$7995
TARGA 2000 DTX	Truevision	Mac/PC	8.0/8.4	$5495
Media 100qx	Media 100	Mac	8.1	$1900
VideoVision PCI	Radius	Mac	4.5	$2489
DPS Perception Video recorder	DPS	PC	6.0+	$1590
AV Master	FAST	PC	4.5	$649
BRAVADO	Truevision	PC	3.1	$599
miroVIDEO DC30	miro	Mac/PC	4.5	$845
miroVIDEO DC20	miro	PC	2.6	$499
miroMOTION DC20	miro	MAC	2.2	$499

Source: DV Magazine, 1/97 and Macromedia 1998

BENEFITS OF USING OPEN SYSTEMS

It is clear that open systems are an economically attractive alternative to proprietary digital production solutions. In addition to the less expensive initial price tag, open systems are less expensive to maintain, less expensive to train on and less expensive to upgrade. These cost factors, considered the costs of ownership, are estimated to be approximately 20% of total system cost. Computing the costs of ownership according to this standard (Pipar Jaffray) reveals a significant savings over the lifetime of open versus proprietary systems. For modular open systems, the total costs of ownership can be even less than 20% total system cost as components can be upgraded individually. As technology markets trends continue along their current path, this expense will likely decrease even further.

Accelerated Creativity

While cost effectiveness is a central concern for all production facilities today, media professionals measure the value of their tools by what they bring to the creative and productive processes. If cost were the only issue, most would continue to opt for the tried and true analog methods. But just as proprietary systems brought increased productivity and accelerated creativity to video artists, open systems offer even greater potential for productivity and creativity. As open solutions are more flexible, comprehensive, and easy to learn, a video editor can add effects and graphics directly into their production. A sound designer can quickly place visual images with multiple audio tracks to test which scenario works best. A reporter or anchor, with little or no training, can quickly mock up an EDL or rough of the feature they just recorded - and they can do this right at the scene of the action. Open media technology systems make the production environment more yielding and more efficient and this means that more time can be spent with the content.

Flexible, Scalable and Integratable

Costs of solution tool sets seem the most significant measure of the benefit of open media systems over proprietary systems, but it is only the most obvious. Open systems provide videographers and professional communicators features and properties that black box solutions could never provide. A primary benefit to using open systems is that they are flexible, scalable and integratable. Regardless of how inexpensive systems have become, it is still costly to replace equipment and upgrading an entire facility to digital systems would be unheard of for most production houses. Open systems offer a stair step approach to modernizing and upgrading a facility. Systems can be added as need and budget warrant, all without fear of rapid obsolescence or incompatibility. If remote production is the primary concern, most open system solutions can be used on laptop computers for a fraction of the costs of proprietary remote production equipment. If an asset management server is the most pressing need, there are open systems solutions that offer all the required features. Additionally, the remote production systems will work with the in-house digital asset server, and indeed with all other systems that are in place from yesterday and the ones that will inevitably come tomorrow. Open systems allow the producer to add according to their needs while taking advantage of new trends in the industry. If more storage is required, off the shelf drives can be added. If faster I/O is required, new board sets can be added as they become available.

Open architecture solutions, using open systems media layers are adept at integrating with all existing, open hardware and software solutions. Open media solutions also work with all standard analog systems and automation and control devices. Open systems are particularly adept at integration because they share open media layer and networking formats. This integration of systems means that you don't replace what doesn't need replacing. The most prevalent open media layer, QuickTime from Apple, is designed to support continuity with other file formats and standards. In this architecture, all data formats are extant, so you can continue to use your existing content as you switch over to or start using a QuickTime-based, open system solution. QuickTime even provides transparent access to multiple file formats – not a translation that doubles the amount of disk space required and can loose data. Legacy data and content can not be abandoned and data conversions can be time-consuming and expensive. Open systems effectively solve this issue with their inherent flexible, scalable and integratable nature.

Additionally, most new innovations are forged on open systems, and open systems can easily integrate nearly all new advances immediately. For example, the Radius VideoVision Studio was the first video codec card to provide record and play of 60-field video in JPEG compression. AVID offered only 30-field video for many years. Adobe Premiere has offered editing, compositing, and effects in one complete package since 1991. Avid is just now offering fully comprehensive packages in their upcoming releases. DV is another example. DV is available on open systems today whereas AVID has no DV solutions at this time. With proprietary systems, consumers must await the integration by the system vendor of new tool sets or features. In today's atmosphere of increased competition, having access to the latest tool can give a company great advantage over the competition, making this level of integratability an inestimable benefit.

Multi-Purpose and Comprehensive

Open systems are multi-purpose systems that can be transitioned to various duties. With proprietary systems, you buy a single tool: editing, content creation, effects, or sometimes an aggregate of two or more of these solutions. When there is no need for that particular service, that system is off-line and not available for any other functions. Open systems solutions are specified by the software application that is running. With open systems, if the opportunity arises, an available solution system could be transformed with a mouse click into any number of media production, media management or even business systems. While black box solutions have become more comprehensive in their recent revisions, many are still single-solution based. While many see this as an advantage of these systems as they are able to focus all computing power on the task at hand, they also are severely limited in the features that they can offer. Video editing is not only about cutting and pasting clips to generate EDL's, roughs and finals. The full production of a finished piece often includes effects, graphics and compositing, bottom third titles, main titles, logo impositions and even 3D animation. With open systems, more comprehensive applications are available that offer all in one production environments. Additionally, most software solutions utilizing open media layers allow for the quick import of files from other predominant applications. This comprehensive and multi-purpose environment facilitates the creative process and increases productivity.

Open System Applications

The most critical applications in the video production environment are the editing application, followed closely by compositing applications. While there are currently editing applications available for open systems, most notably Adobe Premiere, these lack the productivity and quality features necessary for professional production. The most popular open system compositing application is Adobe AfterEffects. Designed specifically for compositing, it can be used to create stunning 2D animations of composited stills and graphics. While, by definition, different open systems applications do work together, a new trend toward integrated editing, effects and compositing solutions is emerging. Working with an application that performs all the regularly necessary functions for production, facilities can save steps, time, and money. With a single interface, there is no need to save files and interchange them. With a single application, training, support and upgrades are simplified. Final Cut, a powerful new software solution from Macromedia, embodies all the professional producers' daily requirements.

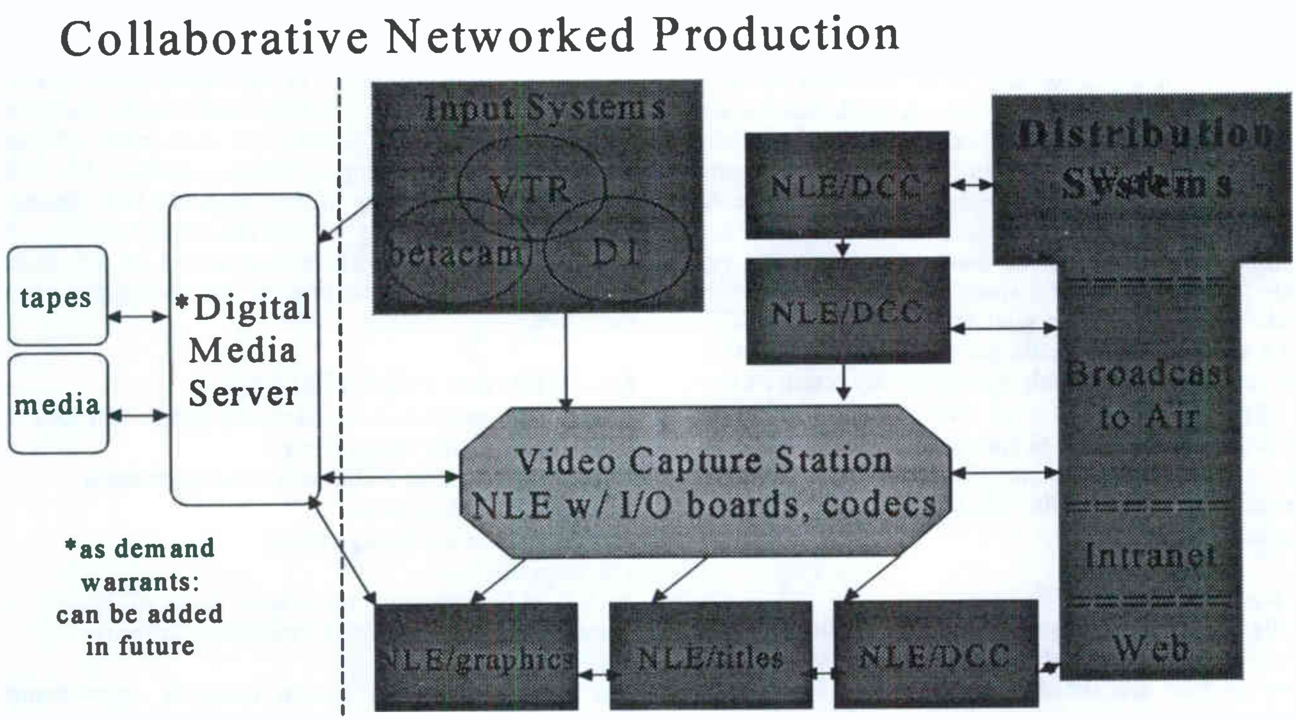
Final Cut is designed as a professional production environment that provides single step productivity features such as three point editing with insert, overlay, and compositing. Allowing edits to be modified quickly with minimal steps is de rigeur in fast-paced production environments and Final Cut edits can be adjusted in a single operation that can be performed in a variety of ways. Final Cut simplifies the process of adding transitions, allowing transitions to be inserted inline as opposed to applied in a special track. Transitions can be selected in the timeline, trimming, or effects viewers. All modifications of duration, intensity, etceteras, can be made either graphically via drag and drop or with the keyboard. Match frame, replace, and trimming features are available for selecting an applied edit, adjusting and returning the modified edit to the program. Professional trimming functions that work across multiple edits are also available.

Macromedia Final Cut also embodies many of the most popular AfterEffects features. Manipulation of elements is done in similar style to popular illustration applications. Items are selected and manipulated as if they are on a canvas. Motion paths are created by drawing straight or curved paths. The paths can be manipulated using Bezier control points. Acceleration can be graphically or numerically controlled at keyframes that are set anywhere on the path. Macromedia Final Cut also supports all third party AfterEffects plug-ins.

Collaborative Work Environments

Open system media solutions, unlike some black box solutions, facilitate collaborative work environments by allowing networked production across platforms. In such an environment, one system might be set up as a capture station, while other stations serve as editing, compositing and/or effects stations. Potentially all might be linked via a digital asset management server. The networked production environment is fiscally efficient, enhances productivity and fosters creativity. This environment increases productivity and cuts costs from the bottom line by more efficiently deploying the creative workforce. In a networked environment, producers can have simultaneous access to tools and media assets to perform all required modifications. Workstations can be alternatively used as effects, graphics, audio, or editing systems allowing artists with disparate talents to collaborate on productions and simultaneously contribute their talents to building a winning piece.

Diagram 1: Collaborative Workflow



Support, Installation and Training

Installation, training, and support are often not the first measurements of systems as they are being evaluated. However, in the bustling broadcast and production environments, these factors are crucial to productivity and are essential for accurately computing the total costs of ownership for production solutions. If a black box solution is down for some reason, the production facility is reliant upon the original vendor for service. And purchased service and support programs from black box vendors can be very expensive. While support services for open systems can also be expensive, there are many more avenues to pursue to quickly get a system performing as needed. Additionally, in the case of open systems, other functioning systems can be substituted for the system with problems.

For installation, training and support, proprietary systems offer users who are artists, producers or business people easy set-up and installation, one stop support avenues, and specialized training. However, most production facilities these days have access to technology professionals, either on staff or used as consultants. Additionally, many VAD and VAR outfits will work with independent production facilities to set up, install train on, and support customized open system solutions. Black box solutions can often only be serviced by the vendor or their authorized agents. The same is often true for training. With black box solutions, the end-user has a continued, locked-in need to utilize the limited support and training personnel that the vendor makes available. While open media systems will also require new technical knowledge that many facilities might not have in house, there is a large and growing pool of talent from which to choose. Training in house personnel in the use of these new systems has also become a valid option. (This is an option that many broadcast facilities will deem necessary as unions will demand that tenured employees not be out placed by new technology.) Where once training and support for any technology was scarce and expensive, there now exists in trade schools, universities, consulting agencies, and even users and industry groups, a number of training programs that can quickly get people up to speed with new technology systems. Added to this is a plethora of books, videos, CD-ROM's and web sites dedicated to technology training. These concerns and materials simultaneously prepare a new pool of talented people to work with all aspects of media technology. Additionally, as open systems are based on the same IT technology that most businesses use regularly, technicians for these tools are far more abundant and reasonably priced.

New Revenue Streams

As digital technologies advance even further into production and broadcast facilities, new benefits and income streams are already emerging. As open media layers offer formats for multiple markets – film, broadcasting, CD-ROM, Internet, games, etceteras, delivering media over multiple modes becomes more efficient and attractive. The installed base and business models for many of these new revenue streams are still being developed, but in the near future, these will be avenues that every production facility will pursue. Internet distribution is already causing many broadcast facilities and concerns to establish web sites and expand their advertising offerings. These technology delivery platforms are very enticing to some advertisers, as their messages are delivered to a quantified, segmented population that has a standard of education and income that is very attractive for many product advertising concerns. While currently only about 35% of all households have computers capable of receiving alternative media via CD-ROM and/or Internet (approximately 98% of all households have televisions - a greater percentage than have telephones - CES Survey, 1997), this number will continue to grow as the prices of personal computers continue to plummet. With open media formats, the same content that was broadcast as NTSC, PAL or SECAM can be streamed across the Internet. These new delivery mechanisms will eventually require new means of delivering their media messages to differentiate them from the traditional linearity that television and films provide. This interactive content creation will then, in turn, influence how information is prepared and produced by increasing the need for versioning and localization, among the many other changes that may come. Eventually, the Internet and web, business intranets, interactive television, family learning centers, museums, and schools will likely take advantage of the prospects that open systems are now availing professional media communicators.

Without awaiting the future of communications systems, open media technology solutions make more traditional methods of distributing information even more efficient. Moreover, quickly emerging business models for video presentations and training in both corporate and education sectors are also supplying possible additional revenues. All of this means a potentially greater return on investment while simultaneously preparing for and paving the way to the future.

RETURN ON INVESTMENT

A fully configured open digital media system will have the following or similar components:
300MHz Workstation - Windows NT or Macintosh
18 Gigabyte Storage Drive
Jazz Drive External Storage Drive
128 Megs of RAM
Real-Time Compressed, Dual Stream Video I/O Board
Macromedia FinalCut: Video Production Software

The total cost of this system today is approximately $20,000 including dealer set up, limited service and support. A $20,000 system, with a cost of ownership of 20%, will cost a facility $4,000 a year to own and will have a principal life span of 3.5-5 years (Pipar Jaffray). (The principal life span is the amount of time that the system is used as a state of the art production device. After it's principal life, open system workstations are often used for tertiary purposes for an additional 5-7 years.) The total primary costs of this system are $34,000. Calculating the return that comes from such an investment requires considering the many ways that digital video systems increase productivity and retain and even increase the value of media assets. Increases in productivity involve decreases in the amount of time that the producer spends fixing mistakes, toggling and rewinding through footage, creating versions, adding effects, and distributing the final product. In the single example of fixing mistakes, a digital production environment can increase productivity by 20% or more. In an analog video suite, if there is a flaw in the production, the producer is required to go back to the point of the mistake, correct it and re-do all of the production work past that point. Depending on where this flaw occurs in the production, this can costs hours or even days of labor. In a digital environment, the error can be corrected in minutes without effecting any other portion of the project. Toggling and rewinding through footage, a process that can represent 15-35% of the time required for production, is essentially eliminated in the digital domain. The processes of versioning, adding effects, finalizing, and distributing the project are also dramatically hastened in the digital realm. With these increases in productivity, each producer spends less time in mundane endeavors and more time being creative. This also means that products are completed more quickly, increasing the potential revenues that can be generated. Furthermore, as open media systems are less costly to buy and own, facilities can equip every producer with his or her own tool set, utilizing the talents and efficacy of the labor pool to its fullest.

Conservatively, these increases in productivity can amount to over 1,000 production hours, per production employee, per year – over $50,000 worth of labor - quickly covering all of the costs associated with digital equipment, including training, support, upgrades and even additional personnel. Beyond the upsurge in productivity, the value of assets also increases as they are not destroyed in the production process, they are readily re-used and re-purposed for evolving distribution schema, and they are more easily stored, sorted and found in digital asset management systems. Additionally, an indefinable potential increase in value also results from the added creativity, experimentation and inventiveness that digital systems facilitate.

Proprietary digital systems can provide a similar return on investment, but these systems are more expensive to own, more expensive to support and much more expensive to upgrade. The difference in upgrade costs between open and closed systems is dramatic. The cost of upgrading some Avid systems from a NuBus version to a PCI version cost approximately $30,000. The same upgrade in an open system would cost the price of a new board - from $500-$8,000. Trained production professionals are also more readily available for open systems, facilitating the finding, hiring and retaining of talent. Once the talent is employed, open systems, due to their relatively inexpensive costs and ease of use, also promote individual productivity as each contributor is afforded their own workstation and tool set. With proprietary systems, as costs are prohibitive, often many producers have to share a system, potentially debilitating the overall effectiveness of every user. While broadcasters devise their own means to measure the return on investment for information and media technology, all methods will lead to the same conclusion: Open digital media systems afford a salient and gainful endeavor.

CONCLUSION

Taking the temperature of the business environment has never been an easy course of action for any business executive or market strategist, yet the opportunities that open systems present are obvious. The professional requirements for all production tools in the studio environment are that they can perform under mission critical circumstances, that they offer high productivity and that they afford an overall return on investment. Open media system production tools promote cost effective replacement strategies, increased creative possibilities and potential new revenue streams. Additionally, during these times of transition to digital television open media production environments provide an adaptable solution to meet the many coming challenges. In the US, the FCC has mandated a rapid transition to digital television distribution. The ramp up phase for this technology is likely to be long and confusing. A great deal of this confusion will surround the issue of which digital standards will be adopted by a majority of the nation's broadcasters. (There are 18 standards, all based in MPEG-2, that the FCC accepted from the ATSC's recommendations.) It is likely that many stations will, of necessity, adopt a bouquet of these standards to use during their fluctuating daily schedules. The DVB standards of the European Broadcasting Union also provide various ways to distribute digital television, and the Internet and telephony will add even more opportunities for dispersion of information to consumers. The production systems used to create this multi-formatted, multi-purpose media information need to be flexible enough to adapt to all the future needs of stations as well as be appropriate for today's needs. Open systems offer that level of adaptability and flexibility in a cost-effective package. Today open systems can support DV, DVC Pro, MPEG-2, MPEG-1, M-JPEG, web compression H323 and RealVideo, amongst many other formats. As the future inevitably brings more means to communicate and more developments to facilitate that communication, open media technology will adapt, grow, expedite and advance the processes.

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UNDERSTANDING THE FUTURE OF VIDEO EDITING AND COMPOSITING TOOLS IN A DIGITAL ERA

Jared Vishney
Silicon Graphics, Inc.
Mountain View, California

This paper explores how computer technology impacts video editing and compositing tools. It provides a new framework for evaluating video editing and compositing tools in the digital era, and explains how new tools are changing the industry's business model.

INTRODUCTION

Television studios, broadcasters, post-production facilities and other entertainment professionals agree that digital video editing and compositing tools save time and expand production capabilities beyond traditional analog systems. They also allow a move away from the proprietary solutions previously required to handle real-time, frame accurate, uncompressed video solutions. The question becomes how to choose the right digital system? A broad selection of editing and compositing tools are now available based on generic computing platforms. Widely varying feature sets and prices make it difficult to compare and select a system while ensuring that the fundamental video capabilities will be supported.

The first step is to evaluate how the system will be used. The demands placed on the editing or compositing system will determine the required feature set of the platform. Video professionals should consider the need for high-bandwidth input and output channels and internal buses, high-performance CPUs, accelerated graphics, and a predictable operating system. At the foundation of any video editing and compositing solution, the performance of each computer subsystem and their interdependence directly determine the system's suitability as a video computing platform.

After considering the suitability of a particular computer platform, then evaluate the overall solution. A set of metrics helps to rank the effectiveness of the digital technologies embedded in video editing and compositing tools. These metrics include: productivity, scalability, longevity, reliability, interoperability with existing and future tools, standards support, asset management capabilities, and distribution protocols. Although each metric addresses a critical aspect of the final solution, this paper also discusses how the relative importance of each metric will vary with specific applications.

Finally, this paper suggests a number of larger issues to consider when assessing the overall business value of a solution. In addition to delivering a quick return on investment, a good solution enhances both short- and long-term business opportunities and facilitates the incorporation of future advances as they become available.

THE IDEAL DIGITAL PLATFORM

The proliferation of affordable computers—costing $5,000 or less—allows businesses to choose from a wide selection of potential platforms for video editing and compositing tools. PC and Macintosh® computers, while providing inexpensive computing power, offer very limited video processing technology. PC and Macintosh

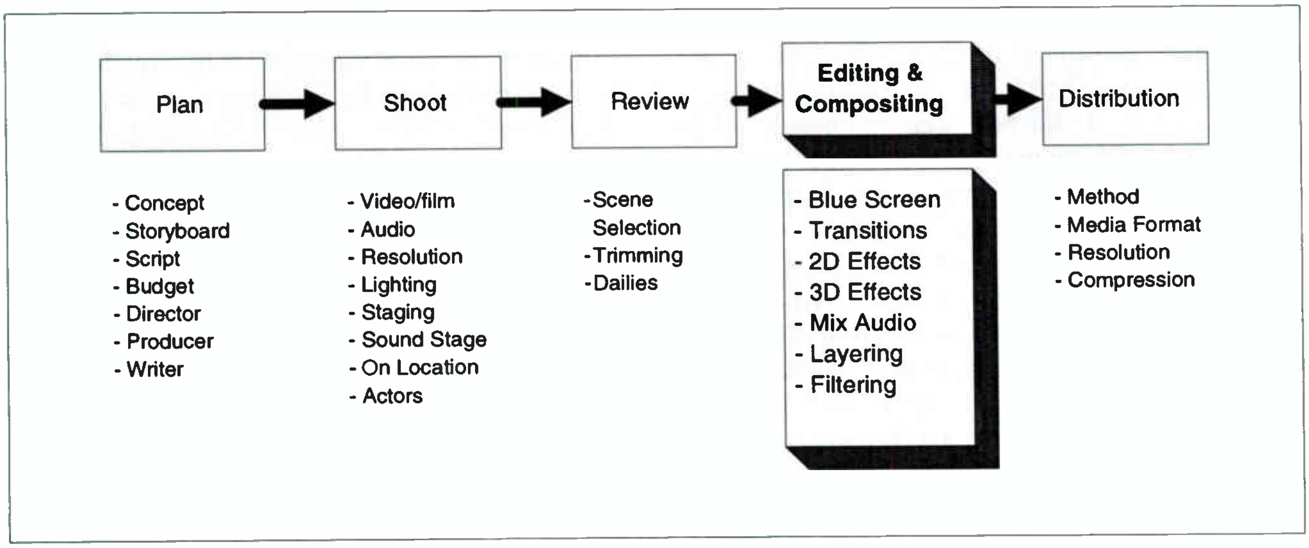


Figure 1. Production Workflow

computers rely on third-party add-on hardware components—e.g., video capture and compression boards, video-enabled graphics boards, and special effects rendering boards—to increase processing power, but this is essentially retaining a proprietary approach. Video and media content never makes it into the computer; data is not accessible by other applications unless they are specifically created to make use of thespecialized hardware. Other general-purpose solutions, such as graphics workstations based on the UNIX® system, support video hardware by integrating video processing features directly into the baseline system. Although UNIX is not as widely used, all the components usually come

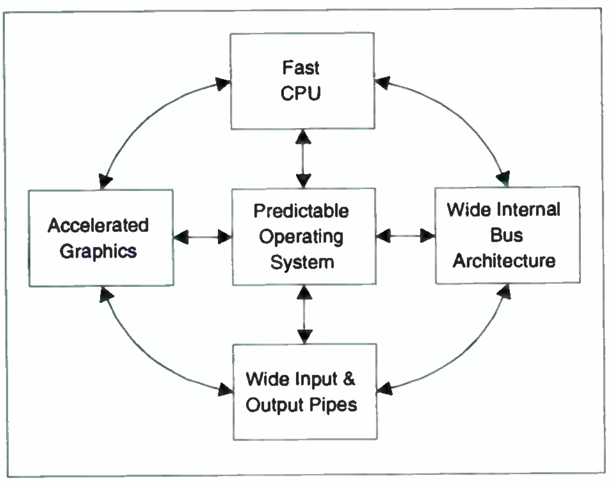


Figure 2. Balanced Performance for Digital Content Creation

from a single source that provides an opportunity for a fully integrated, powerful environment.

How can these platforms best be evaluated? While each system must be considered in light of a specific need, the ideal computing platform for any application offers balanced performance as illustrated in Figure 2. The ability of each of the computer's major subsystems to manipulate large amounts of data directly impacts the platform's suitability as a video editing and compositing tool.

Relying only on a high-speed CPU will not help if the system cannot move video data in and out quickly, and efficiently move video data within the system. Likewise, a large intemal bandwidth does not necessarily mean the system has enough power to process video data in a reasonable amount of time. Like a plumbing system where the smallest pipe determines the flow rate, the slowest system component directly affects the maximum throughput achievable.

Wide Input and Output Channels

When editing using a traditional on-line linear editing suite, the source video material already resides on the working storage media and loading video material onto another media is not necessary. On the other hand, the first step when working with any non-linear digital editing suite is to load the source video material from a video tape recorder (VTR) or other video device to the working storage media for the editing system. For typical production, video (ITU-R BT.601

uncompressed 16-bit YUV 4:2:2 format) has an average data rate of 20.74 megabytes per second (MB/s) and 41.44MB/s for increased color depth and key (ITU-R BT.601 uncompressed 24-bit YUVA 4:4:4:4 format). To accept uncompressed video from a VTR in real time, the system's input channel must accept at least 21MB/s sustained throughput from end to end. An uncompressed HDTV video data stream requires anywhere from 120–180MB/s, depending upon color depth and sampling. As broadcasters move towards DTV, the video I/O problem is compounded. Even the least-stringent proposed production standard requires I/O performance in excess of 165MB/s.

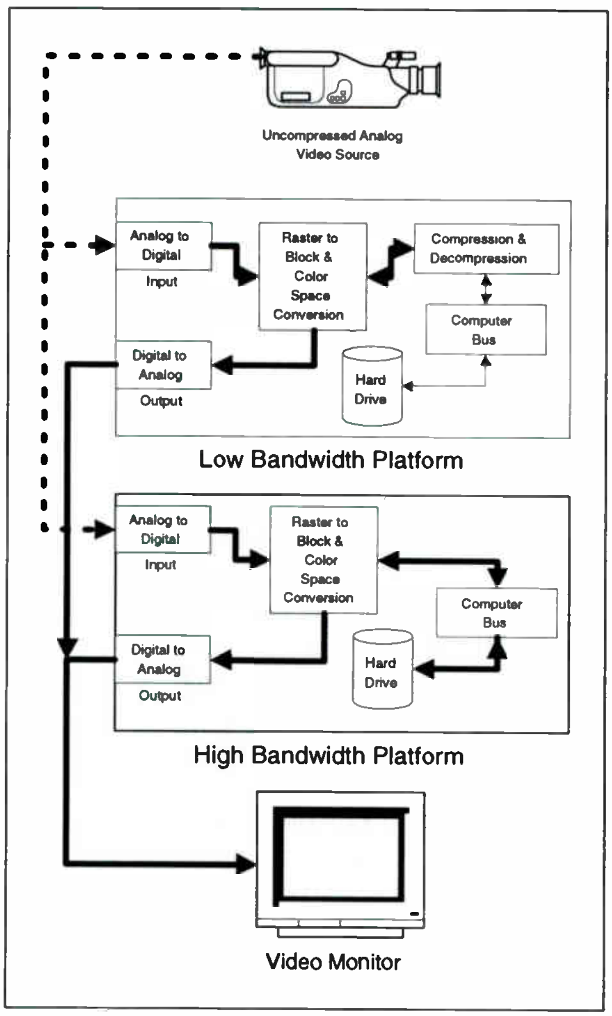


Figure 3. Video Data Flow

Actual video input and output rates vary greatly from low-end PCs and Macintosh systems to higher-performance PCs and UNIX workstations. PCs and most Macintosh computers perform video I/O functions by relying on third-party boards that plug into the PCI bus. Higher-end systems' video I/O boards may also connect to standard buses, such as PCI, or use a proprietary bus to deliver higher throughput. For analog video, most video capture boards store the incoming signals in a frame buffer (memory) on the I/O board until the field, frame, or predetermined portion has arrived. The video material must then be converted from a raster to a block format, possibly color space converted, then sent to disk in either a compressed or uncompressed format, depending upon the speed of the system. Most PC designs also require thevideo material to pass through memory, which can seriously slow the process.

Incoming video moves to disk storage across the I/O bus. Most system technical specifications describe theoretical bandwidths across the bus (or backplane), but consideration should be more importantly given to actual throughput. A system's actual I/O rates will determine whether or not it is suitable for a particular task. Working with a single stream of uncompressed video requires guaranteed bandwidth of at least 20MB/s. For most PCs or Macintosh systems, this means the video must be compressed during input and decompressed during output. Even then, the video streaming process consumes virtually all of the system resources and limits the number and complexity of concurrent processes. Depending upon the final image quality required, or the type of project, compressed video may provide acceptable quality. If not, the alternative is to use a higher-bandwidth workstation as the platform for the editing tool, compositing tool, or proprietary solution (e.g., Quantel Henry™).

Wide Internal Buses

Compared with the video I/O, a platform's internal bus architecture has an even more significant impact on the overall effectiveness of a system's performance. Once the video material is in the system, the internal buses must provide enough throughput to supply data to the CPU and other subsystems as fast as possible. Any advantage of an extremely fast CPU or other subsystem is lost if the system cannot move data within the machine as fast as the CPU or subsystem can process it.

The PC's linear-bus architecture, pictured in Figure 4, severely limits both input and internal data movement. Even as CPU speeds increase, bus bottlenecks constrain system responsiveness. For example, the PCI32 bus running at 33MHz has a theoretical bandwidth of 133MB/s, but that number is the peak bandwidth. Adding operating system constraints to the equation can easily limit the usable (sustainable) bandwidth to less than 10MB/s for a PC. Older bus architectures have even lower sustainable data rates. These I/O bandwidths are not sufficient for uncompressed video I/O.

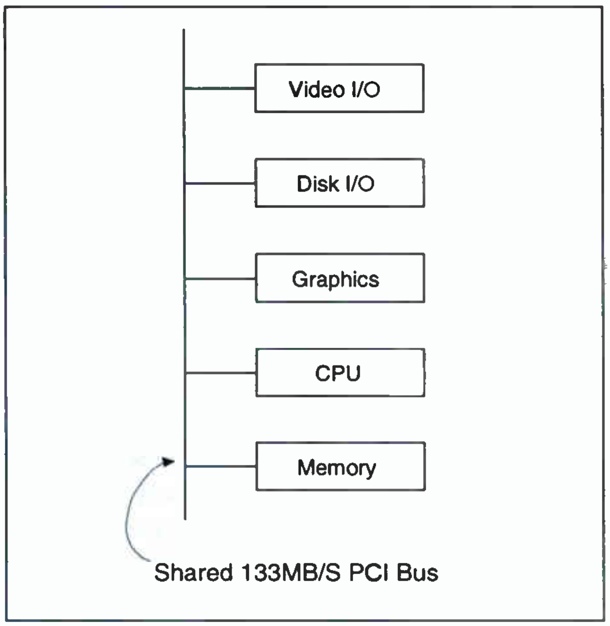


Figure 4. Linear Internal Bus Architecture

The newer 33MHz and 66MHz PCI64 bus (specified at 266MB/s and 512MB/s respectively), combined with operating system improvements, offers more promise. A Unified Memory Architecture is one way to avoid linear bus limitations (see Figure 5). This design places high-bandwidth memory at the core of the system and provides better overall performance.

In contrast to a PC, high-end workstations or dedicated "black box" hardware have much higher bandwidths. For instance, the Silicon Graphics® OCTANE™ system's XIO bus architecture has a theoretical bandwidth of 1.6GB/s (800MB/s in each direction) and can realize bandwidth of 1.2GB/s. The Silicon Graphics XIO architecture uses a high-performance switch to link any two computer subsystems, giving them a high-speed path without interfering or competing with other system activity. This allows the system to internally manage multiple streams of uncompressed D1 and one uncompressed HD video stream in real-time.

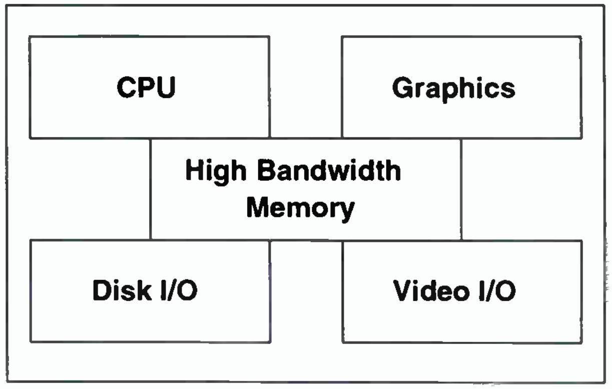


Figure 5. Unified Memory Architecture

The internal bus structure directly effects an editing or compositing tool's efficiency. It directly determines if and how video streams of compressed or uncompressed video can be processed in real-time. For example, a simple dissolve between two video streams requires that the machine manage three video streams internally, as well as make the necessary calculations for the effect. With Stream A and Stream B active simultaneously, the tool must calculate the effect and generate a resulting third stream, Stream C. High-end systems accomplish such a rendering effect in real-time, but non-professional systems (i.e., low-end systems) require additional hardware and can only operate on compressed video off-line (without real-time capabilities). Systems using third-party plug-in boards that could calculate the effect in real time cannot take advantage of this because the data cannot be delivered in real time. Other factors, such as the quality of the bus architecture implementation, other subsystems on the bus, and operating system and application bus management efficiencies, also effect overall performance.

Fast CPUs

After the compressed or uncompressed source video material is loaded on the system, and if the bus architecture has sufficient bandwidth to manage the video, only then does a generic CPU become a major factor for the editing or compositing tool. If it is not possible to get the video into the system or effectively manage it once it is there, having the fastest CPU available does not contribute to optimal tool performance.

The ability to get the most out of the CPU depends on the efficiency of the application software and the speed at which data can be fed into and from the CPU. An editing or compositing application may or may not make extensive use of a CPU.

Due to the cost consideration, the CPU-dominant approach is put to good use by most PC and Macintosh editing and compositing applications. The CPU is used for all aspects of managing the application, managing the system hardware and performing the necessary calculations (rendering) to manipulate the video. This is a suitable approach for off-line projects where compressed image quality is acceptable.

In a typical digital editing or compositing application, the most CPU-demanding tasks include calculating transitions, effects, or filters applied to the video. Some tasks involve floating-point calculations and others only integer calculations; the same CPU may perform very differently in each of these two cases. For instance, pushing one stream of video to cover another might involve integer-intensive math if the movement of the video is controlled in finite incremental steps. For true 3D effects, floating-point calculations are required because the proper representation of a video image in a 3D space occurs in increasingly small (infinite) increments. Likewise, representing a complicated filter effect requires floating-point math to calculate the single-frame interpixel results and the interframe results over time.

Another processing option is to off-load the video computations to specially designed processors. This is accomplished in the PC and Macintosh world by adding a plug-in board with a dedicated video effect processor to render the desired effect. While these add-ons may significantly expand capabilities, they also add expense to the system. They can bring the total cost close to the price of more capable workstations or proprietary solutions, but without compensating for bottlenecks in bandwidth surrounding the CPU and other parts of the system.

Accelerated Graphics System

If properly implemented, an integrated high-performance graphics system opens new possibilities for video editing or compositing systems. 2D graphics performance, for instance, will improve the rendering of any video effect in a single plane, such as frame painting (air brushing graphics onto video) or creating titles. 3D graphics performance improves transition effects that move the video or streams of video in x-, y- and z-planes, such as a page turn or barn door effect.

As previously discussed, many software packages rely only on the CPU for rendering video effects. This method is adequate for many non-professional tools. Another option, depending upon the system's overall design, is to use the graphics subsystem for video processing. Since full motion video is only 25 or 30 still images per second, a graphics subsystem can contribute to video-related performance.

Using a 3D graphics subsystem for video processing also offers some less apparent benefits. Some systems depend upon the use of 2.5D effects—basically creating a 3D effect in a 2D plane—to compensate for the lack of full 3D functionality. Using 2.5D restricts the flexibility for video editing. While a 2.5D image can appear to be a high-quality 3D view, the rotated image will reveal only a flat surface. A true 3D effect will have three dimensions even as it is rotated in space, allowing more choices and creativity when editing and compositing, and ultimately allows the delivery of a more professional image.

This is one arena in which a system like the Silicon Graphics workstation differentiates itself most from PC and Macintosh solutions. Transitions and special effects are rendered using the OpenGL® application programming interface. OpenGL is an open, flexible graphics standard available on a variety of platforms including PCs. Silicon Graphics uses a high-speed graphics engine, that performs extremely fast floating-point calculations, to achieve the best performance. With high-performance graphics engines tied directly into a wide, high-performance internal bus, the Silicon Graphics workstations are capable of manipulating multiple streams of uncompressed video in real-time. While not all applications require this level of performance, it demonstrates the full benefits of a system (workstation or "black box") with true video capabilities.

Operating System

All of the care taken in the design of balanced subsystems is wasted without effective communications between the subsystems. The operating system ties together the various hardware components within a system and needs to be manageable, predictable and stable. Most video editing and compositing tools shield the artist from any operating system, but it is important to understand how an operating system impacts the overall solution.

The ideal editing and compositing platform gives complete control of system resources to the tool designer. If, for example, video is streaming in or out between the hard drive and the VTR, unforeseen system interruptions should not suspend the video stream. Likewise, it is useful to be able to schedule events or dedicate resources as needed. It may be desirable to stop, suspend, or reschedule certain processes, including housekeeping procedures initiated by the operating system, during specific phases of operation. For example, if the editing or compositing tool needs to play back single or multiple video streams from storage, it is desirable to suspend a process that automatically performs periodic checks for new e-mail.

The Windows® and Windows NT® operating systems are well suited for off-line editing and compositing applications. While the operating systems offer limited real-time video performance (partially due to system design limitations), they are good platform solutions when cost is the deciding factor. These systems do not, however, accommodate any deterministic or time-sensitive scheduling; working with any specific time references requires complex software work-arounds. This includes the ability of the operating system to be locked to a time-sensitive reference, such as a house reference signal, and perform tasks based on the signal. Likewise, the Macintosh operating system works fine for off-line applications that do not require deterministic scheduling. In contrast, the UNIX system has many features that make it more suitable for real-time, on-line applications. The UNIX system gives developers more control over the computer system and squeezes every last drop of performance out of the system. Without being religious, the operating system choice should be driven by the needs of the system. If cost is the driving force, then Windows is a good starting point. If performance is the driving force, UNIX platform-based tools will better serve the artist's needs.

SOLUTION EVALUATION METRICS

In addition to choosing a platform with balanced performance, it is also imperative to evaluate the potential benefits from the combined platform and software tools. Ideal editing and compositing solutions should be characterized with features that benefit every phase of a project and optimally suit overall company operations. After evaluating and selecting the most appropriate platform, video professionals should consider the following additional factors that will determine the overall usefulness of a particular solution.

Productivity

Raw performance obviously affects productivity. If, for example, one system takes ten minutes to display the results of a process on the screen, and another can immediately display the same results, the second system is clearly a more productive environment. But other less apparent factors influence productivity as well. To effectively rate the productivity of an editing or compositing system, consider the following questions.

In general, consider:

- How intuitive is the user interface?
- How much training is required to learn to use the tool?
- Is there a large pool of available operators trained on the tool?

For real-time capabilities, consider:

- Does the platform readily suit the type of work done most frequently?
- Is uncompressed or compressed video needed?
- How many layers of uncompressed video can be composited in real-time?
- How many layers of video are available for editing at a single point in time?

These questions will expand the evaluation of productivity beyond performance specifications and benchmark results—an important step since numbers alone do not tell the whole story. The answers to the questions shed light on the other factors affecting productivity, including application functionality, ease of use, real-time responsiveness of the interface, and the level of intuitiveness offered by the tool.

Longevity

When will the system encounter a critical performance limitation? Is there sufficient headroom to accommodate emerging trends in editing and compositing and the evolving high-definition industry standards that will quickly consume more system bandwidth? Consider the limits of the system both in terms of today's work, as well as in terms of these fast-approaching business and industry directions. Perhaps a business centers on VHS, S-VHS or DV formats and equipment today. But as the industry moves toward higher-definition, higher-speed formats, will the system have to be replaced, or can it be upgraded or expanded? When expansion is required, will existing work move seamlessly to the new or upgraded system?

Consider the point at which the system will reach performance saturation. Can application tools be added as needed, or does the cost of the system justify permanently dedicating it to a particular task or project type? These considerations and questions help in the preparation for inevitable business changes, and also point out significant differences between PCs and workstations.

When considering the longevity of a video editing and compositing platform, PCs and workstations should be sorted into three basic categories: compressed, uncompressed, and HD ready. Consideration must be given to how today's system investment can or cannot be leveraged for meeting future needs.

For example, entry-level PC systems offer limited headroom and must be replaced to take advantage of new technology. This makes entry-level PC systems more suitable to compressed applications where technology is not changing rapidly. If a new compression technology does emerge, it will probably be available as a plug-in board that is easily added to the system.

For uncompressed video solutions, workstations and higher-end PCs are more appropriate. For example, workstations and PCs are suitable as cuts-only editing systems (with the workstation solution able to grow beyond this basic application). However, the cost of PC systems quickly increases when additional hardware, software packages, and special video options are included in order to match the built-in functionality of workstations.

In the case of HDTV, the situation changes dramatically. Some of today's workstations provide a foundation that will ultimately translate to HDTV, whereas current PCs and "black box" systems do not offer enough performance. For instance, a Silicon Graphics OCTANE system currently handles multiple streams of uncompressed video in real time with the same hardware that can also support a single stream of uncompressed HD video in real time. The OCTANE system's inherent HD ability make it a good choice as a long-term editing or compositing system.

Reliability

Digital video applications push the performance edge of most computers. If the job is done off-line, system glitches may not be as significant as they would in a real-time environment. If interruptions to editing or compositing work are unacceptable, then choose the most stable platform possible. Some people evaluate reliability by gathering individual reliability/availability/serviceability (RAS) features and statistics for the various components of the solution. Experience shows that platforms with parts from the least number of vendors tend to be the most dependable.

In the PC world, the final system tends to incorporate components from a variety of vendors. While this expands user choices, it also poses support and service challenges that must be taken into account. Turn-key solutions assembled by a solution provider help minimize reliability issues, but reliability and performance are only as good as the sum of the parts in the system. Fully or tightly integrated hardware solutions have the advantage of being designed to work together from the beginning, and typically avoid many of the issues common to multivendor solutions.

Interoperability and Standards

Most video editing and compositing environments include a broad mix of electronic and video equipment. Will the platform handle incoming data feeds from existing equipment at full speed and full resolution? Will new equipment interface easily? If not, can it be upgraded with add-on boards, or will the platform be isolated from the new gear? Make sure to consider the size of the data—including resolution and frequency—that will be communicated among devices, and check that the platform has the bandwidth to support it. To support the move to true end-to-end digital, consider each stage of the solution and avoid those that currently require compression in order to meet performance requirements.

Do all components of the solution adhere to published and accepted standards? This will simplify connection and communication, and also increase the likelihood of compatibility with future technology and formats. Furthermore, does the solution allow flexibility in the standards supported. For example, how would an editing or compositing tool installed today for HDTV cope with a change in the HDTV specification in three years?

Asset Management Capabilities

How and where will video assets be stored? Simply archiving to tape can be reasonably done from PCs, Macintosh systems and workstations alike. As asset volume grows, a data warehousing solution may be more appropriate. A data warehouse, or central repository, of video material offers very fast (almost instant) access to material stored in a digital file. The challenge then becomes the management and control of the material. Various vendors offer products, including the StudioCentral™ asset management system from Silicon Graphics and Digital Library™ from IBM, aimed at meeting these challenges.

What a video library (asset) management system needs to do for editing and compositing is straight forward. First, the system must provide the infrastructure to manage the library of source material in the system. For example, can the system simultaneously provide the correct material to an Avid® MediaComposer® in the edit suite and to a Discreet Logic® Flame™ in the composite suite? Other requirements that the video library must meet include access to data from all editing and compositing tools (does the system offer the connectivity required by these tools?), avoiding data incompatibilities among the tools, and the capability to adapt to future industry standards. If fast access to a library of work will contribute to productivity and competitiveness, look for a data warehouse solution that integrates well with editing and compositing tools.

Distribution Protocols

How will the final product be delivered to the customer? The system should make it easy to export the final product using the customer's requested delivery vehicle—CD-ROM, film, tape, or via the Internet. Check to see that the computer system supports the network or media distribution protocols that meet customer requirements.

VALUE = MORE THAN ROI

Once the performance characteristics and issues described have been evaluated, the final phase in the process is to step back from the technical specifications and appraise the platform in light of the highest-level business goals. It is not unreasonable to expect the system to deliver quantifiable payback—for example, can it double the number of projects through the studio, reduce average project time by 25 percent, or cut production costs by 10 percent?

In reality, time savings may not translate into a reduction in editing hours, but rather into the creation of a more sophisticated product with better special effects, a more refined image, or a more complicated scene. Increased productivity also allows clients to explore more "what-if" scenarios. The better the tool, the better the final product. While enhanced product quality and innovation may be difficult to quantify, the contribution to the bottom line in terms of customer attraction and retention are real and significant.

Consider the cost of the platform in relation to potential business benefits. For producing corporate videos, the system need not support uncompressed video capabilities, real-time motion video, or true 3D digital video effects. More and more, videographers are turning to 3D effects to provide a fresh, sophisticated look for corporate clients. Developing a 30-second commercial for a major soft drink company calls for multilayering and other sophisticated special effects that require high-performance capabilities to remain competitive and to deliver the quality the customer demands and expects.

When looking at cost, consider also the peripheral requirements. For example, does the solution use standard hard disk drives, or does it require high-cost drives, or even drives that are only available from the software solution provider? These "hidden" costs need to be taken into consideration when evaluating a system's total cost.

Finally, in addition to assessing the ROI, take into account how the new tool can improve the way video editors and artists work. Will it help re-purpose assets—i.e., does the system get more mileage out of specific graphics or special effects? Digital tools make it easier to re-purpose material. Perhaps a realistic jungle background was developed for a commercial. That background might be able to be re-used in a video game or promotional spot. Or plug existing background plates or logos into new projects for regular customers. Re-purposing will save time by using existing content, rather than creating new pieces from scratch. Digitally searching and browsing for the material is faster than traditional analog video searching methods. By estimating how much these capabilities are worth, the longer-term value of the platform investment can be determined.

SUMMARY

The landscape at the end of this century will include an even broader range of choices for editing and compositing tools. While the selection may seem overwhelming, it really represents a wealth of opportunities to advance editing and compositing capabilities in all application areas. Careful assessment today of unique requirements and the available solutions will allow a smooth transition to a successful business in the year 2000.

In review, at the start of an evaluation process, first consider the job to be done. At a minimum, the system must help do the editing or compositing work faster or better, and hopefully at a lower cost compared to current processes. You must also be able to recoup your investment before the tool wears out or becomes obsolete.

PC- and Macintosh-based solutions are good for consumers and semi-professionals who are not constrained by time. These platforms also work well for semi-professionals and professionals who are working in a compressed, off-line environment. If a business requires true on-line, uncompressed tools, customers must consider either a custom hardware design (e.g., Quantel) or more powerful workstation-based solutions.

The following checklist summarizes the recommendations in this paper:

- Look for balanced platform performance.
- Evaluate individual platform components—CPU, internal buses, input/output channels, operating system, and graphics system—in relation to balance, and against specific application requirements.
- Rate the system's distinguishing characteristics—productivity, reliability, interopera-

bility, support for industry standards, asset management capabilities, and distribution protocols—in terms of applicability to the production environment.

- Consider quantitative ROI measurements, as well as more esoteric values, such as longevity, asset re-use, and enhancing "sex appeal" for better client retention.
- Relate each purchase to short- and long-term business goals.

An Innovative New Solution for Shadow Browsing for Preview and Broadcast Automation.

Alan Chaney, C.Eng MIEE
Dr. Tony King.

Telemedia Systems Limited,
Cambridge, England

1. What is Shadow Browsing?

The spectacular proliferation of broadcast and film material over the last few years, together with the need to satisfy the demands of an increasing number of channels for ever more sophisticated material, has posed new and difficult problems for the broadcast industry. In the face of tight deadlines and tighter budgets the need to make effective commercial use of programme material is growing ever more pressing, and broadcast companies all over the world are faced with the need to provide more effective management and control of their most valuable assets.

In this paper we describe a technique known as "shadow browsing", together with a commercial implementation of the technique which we claim will solve some of these problems. The idea is not new, but the system we propose is the first complete and effective implementation made available to meet a wide range of broadcasting requirements.

This *Distributed Digital Shadow Browsing* (DDSB) system distributes preview-resolution copies of broadcast material from a special video server (the "Shadow Video Server") across a local area network and into standard PC workstations for display. For such a technique to be of any use, however, there must be an absolute guarantee that there is a one-to-one correspondence between the frames that comprise the original material and those that comprise the shadow version. Where SMPTE time-codes are present on the source these also must be reproduced faithfully.

How can we implement such a system? Typically, where a broadcast application employs shadow browsing there will be the following components:

- A source of broadcast material, such as a tape library, an off-air feed, or a BVS.
- An encoder, to create the shadow material from the broadcast source material.
- Mass storage for the shadow material.
- A set of networked PC workstations to which the shadow store delivers video, and on which run the applications that control the acquisition and distribution of the shadow material.

Some applications may also require a database to relate the shadow material to the source material; in other situations, this may not be necessary. A database is not included as a part of our system but is easily integrated into it.

Finally, it must be emphasised that a shadow browsing system such as is described here is not a solution to a particular problem in the broadcast environment, but an underlying technology that can be used by a third party to build a wide range of such solutions. This paper explores the reasons why shadow browsing is needed, discusses the main principles and engineering issues behind such a system, and describes in detail some aspects of a commercial product called SpectreView which embodies all the principles described here.

2. The Need for Shadow Browsing.

The film, entertainment and broadcast industry spends million of dollars every year on producing audio-visual material of incredible value. Ironically, in the broadcast environment, this material is often difficult to access, as it is held on videotape or on a broadcast video server. There are many situations where staff would like to be able to view the material and make decisions about the content, but frequently this requires the location of a tape, the occupation of an edit suite and a considerable amount of time copying a segment from one format to another.

Staff in broadcast companies can easily be overwhelmed by the vast quantity of videotapes used for preview purposes. This can quickly become an administrative nightmare, and cause expensive and unacceptable delays to broadcasters. What is needed is a cheap, easy and flexible way of using available technology to locate and manage material. This is where shadow browsing becomes so useful:

- Helping improve access to valuable media assets.
- Helping skilled and creative staff use their time more effectively.
- Reducing or eliminating the overheads associated with copying, storing and retrieving media for preview.

3. Main Principles.

The broadcast industry is a demanding customer as regards equipment intended studio or on-air use. The designers of a new technique or technology must have a clear idea of what they are setting out to achieve before they start in order to achieve high standard of product. We propose here five main principals that may be used to guide an effective implementation of a DDSB system. These are *accuracy*, *usability*, *flexibility*, *economy*, and adherence to *standards*.

3.1 Accuracy.

The system must guarantee the accuracy of the shadow copy with respect to the original material, both in terms of the <u>location</u> of frames within a video segment, and the <u>fidelity</u> of image content within a frame.

The shadow copy must track the original source segment which will be composed of a sequence of frames, each having a SMPTE time-code associated with it. The shadow browsing system must guarantee that time-codes on the source material are preserved on the shadow material. There are two common formats for time-code, VITC and LTC, and the system must always cope correctly with the situation where a segment has missing or inconsistent time-codes.

It is essential that each and every frame of the source material be shadowed. The shadow copy must also render all the detail demanded by the application, and be of sufficient quality to enable comfortable prolonged viewing by the operator of the browsing system.

3.2 Usability.

The responsiveness of the system should be good enough to conceal the system processes from the operator. Also, the system should provide a sufficiently complete set of primitives that any reasonable operation can be efficiently implemented.

There must be provision in the system to handle the basic actions necessary to locate, identify and view each frame, and the response of the system to commands issued by the operator should not distract them from the task in hand. The tools should support seeking by both time-code and frame number, and allow user interfaces which match the expectations and experience of broadcast operatives – for example, a digital implementation of a jog/shuttle mechanism should have characteristics not too dissimilar to familiar solutions.

The system should be structured in such a way that the controlling application is not directly involved in the flow of data. This allows processes located in two parts of the system to be coordinated by a controlling application located at a third point.

An example of this is the situation where time-codes in the original source material control acquisition and storage of material. The controlling application specifies that the acquisition at the encoder takes place in parallel with recording material onto the server with a connection between the two, but stands back from the actual transfer of data. This is particularly important in applications such as automation, where a database can be used to track media as it enters the system, and moves between storage servers within the system.

This way of organising a number of cooperating processes is called a *peer-to-peer* architecture, as opposed to the more commonly seen *client-server* arrangement where client applications issue requests for services which are carried out on a server machine on the network. Such an arrangement could not efficiently support the coordination of activities as described.

3.3 Flexibility.

All the performance and functionality of the system should be exposed through the interfaces, and all the complexity hidden.

The system should be flexible, both as regards configuration, and in the way in which it can be adapted to a wide range of applications. One application, for example, may be a sub-titling operation where "jog and shuttle" and the ability to view and cue the sub-titles by time-code is essential. A different application may demand the functionality of a non-linear editor, and the performance will depend critically on the speed and accuracy with which multiple clips of video can be accessed and browsed.

Different applications will have widely different demands for the number of simultaneous encoding streams, the amount of shadow material to be stored and the number of workstations at which the material can be viewed.

The best approach to obtaining the required flexibility is that of the *toolkit*. As its name suggests, a toolkit presents its user with a complete set of tools for building a custom application. The exact function of each tool - where, how and why it is used - has to be extremely carefully documented. The tool must require no special knowledge, by the user, of the system. A toolkit can conveniently be structured in terms of a set of Application Programming Interfaces (APIs) as described in section 5.

3.4 Economy.

The system must justify itself in terms of increased efficiency and reduced costs.

Shadow browsing can cut out various slow and inefficient operations associated with managing tape-based media and can save money, time and effort. The type of saving depends on the particular application; for example, in a studio or production facility it can make significant impact upon capital equipment costs. A typical UK subtitling operation often has four or five people working on one item of material concurrently. Each one of those needs an expensive VCR with time code support. Every time they wish to view material, it must be laboriously copied and manually distributed.

When calculating the relative costs of shadow browsing versus a traditional tape-based method several other issues should be considered:

- The lifetime of the tape heads of professional tape machines.
- The time spent in managing the storage of tapes, and the cost of the floor-space dedicated to storage.
- The time spent by staff waiting upon the availability of an edit suite.
- The speed and efficiency with which staff can perform their tasks.

3.5 Adherence to Standards.

A shadow browsing implementation should benefit both from the favourable price/performance trend of industry-standard computer components, and from the adoption of standard computer interfaces and formats.

The technology upon which a DDSB system is based should use industry standard components and make use of formal interface standards where applicable. The benefits of doing this include:

- Competitive pricing for a component due to a wide choice of vendors.
- The ability to import and export material using common formats and interfaces.
- The use of a rapidly improving and mass-market computer technology (i.e. multimedia-enabled PCs).
- The ability to evolve the product as the enabling technology improves.

4. Engineering a DDSB System.

4.1 *Meeting the Requirements.*

A Distributed Digital Shadow Browsing system design must embody the principals of accuracy, flexibility, usability, economy and standards compliance. The decision that has the greatest impact on the ability of an implementation to do this is the choice of video encoding standard, closely followed by the types of network to support and the protocols to run over those networks. Of the encoding options available (including raw video, motion JPEG, H.263) the decision to use MPEG-1 clearly is justified as regards economy (there are many competitive, commercial MPEG-1 products), and standards compliance [4].

MPEG-1 specifies a syntax for compressed digital video that originally was intended to enable delivery from a standard audio compact disc. Its use in a system such as is described here was not foreseen. In fact, given careful, and most importantly, end-to-end system design, MPEG-1 video can support the 'hands-on' approach to video manipulation that is implicit in our requirements for accuracy and usability.

The flexibility of a DDSB system will reflect in the effectiveness with which applications using DDSB components can be built. The Application Programming Interface (API) is the key to providing this flexibility and the SpectreView API design is described in some detail in section 5. The present section describes how a DDSB system is engineered, with emphasis on video transport and networking, and outlines the approach taken by SpectreView to satisfy the requirements that we have identified.

4.2 Streaming MPEG Over Networks.

The first issue to consider in designing a network for continuous media such as video and audio is the obvious question of how to get data from one place to another. Fortunately a satisfactory solution exists in the form of the Internet Protocol (IP). IP is a network protocol that provides a connectionless datagram data delivery service to a Transport Layer process.

A connectionless protocol treats every item of data (a datagram) individually, attempting to deliver it to the proper recipient, but discarding it if this fails for some reason. There is no logical connection between the sending and receiving processes so neither can be notified of the failure. For this reason a connectionless protocol is referred to as being *unreliable* because there is no way of recovering the lost data. Even so, it may be the most appropriate protocol to use in many cases for reasons given below. One example of this type of protocol is the *User Datagram* Protocol (UDP).

A connection-oriented protocol, on the other hand, provides a *reliable* and *sequential* service, in that sender is informed of failures, and the data are delivered in the same order as they are sent. To do this a connection between the communicating parties is set up prior to any messages being sent, in much the same way that a connection is made in a telephone system. Setting up such a connection involves overhead, and if problems in the network cause the connection frequently to be lost then the overhead of repeatedly setting these up can lead to a worse service than that provided by a connectionless protocol where data is occasionally lost.

IP is an example of a connectionless protocol and may be visualised as a 'black box' into which the sender places data, and which then takes care of all the routing issues involved in delivering the data to its destination. However, IP only provides 'best effort' delivery services: it does not guarantee that data arrives in the correct order, or even that it arrives at all. If reliable transport is required then the connection-oriented Transport Control Protocol (TCP) can be used in conjunction with IP (the common way to denote this is 'TCP/IP'). TCP requires acknowledgement of the receipt of each data packet and if none comes, or if there is a checksum error, the packet is retransmitted.

In SpectreView TCP/IP is used throughout for control purposes. Whether this is appropriate for handling video as well, however, depends very much on the characteristics of particular networks.

In some situations, as described below in connection with PC playback, the act of discarding and retransmitting entire blocks can itself lead to error conditions. On the other hand a missed MPEG header could also cause catastrophic failure in some decoders. MPEG was designed for use in a lossless environment such as that found in a PC where files are read by a decoder directly from a local disc. Synchronisation points exist in the stream to aid error recovery but PC decoders are not built to be resilient to the kinds of errors, such as block loss, that can be expected in a networking environment.

The only real approach to this problem is to be pragmatic: to design all the components in the system from the ground up for reliability, to qualify encoders, switches and decoders in terms of their error behaviour, and to support both reliable and unreliable network protocols. For SpectreView both UDP and the reliable TCP protocols are supported for streaming MPEG over the network.

As far as the physical network is concerned the obvious choices are Ethernet and ATM - Ethernet because of its huge installed base and familiarity, and ATM because of its extremely good performance and flexibility as a multi-service network. Both have long been proven to be capable of supporting real-time traffic [1], [2], [3].

4.3 Frame-Accuracy.

Fundamental to a Distributed Digital Shadow Browsing system is the ability to access individual frames in the video stream. MPEG-1 delineates frames by inserting a header at the start of each new frame, so in principle these frames can be indexed and the indices used for random access. The problem comes when these frames need to be decoded, bearing in mind the inter-frame nature of MPEG compression. Complete and self-contained frames (I-frames) appear only once every N frames while the remainder (B and P-frames) are reconstituted using motion compensation applied to rectangular blocks of image from adjacent I or P frames. This collection of N frames is called a GOP (Group Of Pictures) and in general, in order to decode a given frame, two entire GOPs are required to be buffered (since the last frame of a GOP may depend both on the current and the next GOP).

There are two situations where the GOP structure is important. The first is where a trigger point has been set and the system is monitoring incoming video for a match. If two GOPs worth of video are always buffered at this point before being discarded, then, on the occurrence of a match, all the information needed to reconstitute the required frame is available to be recorded. The second situation is where it is required to play back a clip of video starting at a particular frame. In this case the server commences playback at the start of the GOP. It is then the responsibility of the playback client to decode the GOP into a sequence of images, discard any which are superfluous, and then send the required frames to the display device. Since clips can be collected into a list to be played back-to-back this needs to be a fast and efficient implementation

4.4 Searching and Previewing.

The SpectreView API supports near instantaneous seek operations using frame or timecode indexing, and variable-rate video playback, both forwards and in reverse. Processing an MPEG file in real time into variable-rate video is impractical so SpectreView takes an alternative approach - transparently to process the captured MPEG, as it is recorded onto the Server, into a second, much smaller, and much more conveniently structured video file. This file suffers a loss of spatial resolution, but can be played easily at any speed forwards or backwards by the Client, and is of good enough quality to be used for rapid browsing of the material.

4.5 PC Client Playback.

Software and hardware PC codecs are usually designed to replay from hard disc or CD-ROM. Playback from a real-time network stream is different in two important respects. The first concerns the way the video reconstruction clock is derived from the data stream. MPEG specifies that a clock reference - a snapshot of a counter driven by the encoder sampling clock - regularly be transmitted alongside the data so that the decoder can refer to this to play at the same rate as the encoder samples. There is, however, no way of enforcing this mechanism on all players. Some players use a fixed local clock, some derive a clock from the arrival rate of audio data. These techniques work fine when a local disc is the source - more or less data is read as is needed to satisfy the demand of the device - but can cause serious problems of buffer starvation or overflow when using a constant bit-rate source on the network.

The second thing to appreciate is that software codecs are reasonably CPU-hungry. Again, this presents no problems when reading from an autonomous and error-free disc subsystem. A network device driver, however, often has to cope with error conditions due to data loss. When this happens it is possible for the recovery mechanism to take enough of the CPU to cause the codec to lag behind the incoming data, causing yet more data to be lost and possibly leading to catastrophic failure.

These and other problems, while not insurmountable, are quite subtle, and require careful system-level design to ensure correct and reliable operation.

5. SpectreView Components.

5.1 The SpectreView API.

5.1.1 Hiding System Complexity.

SpectreView is a media capture storage and distribution system built from standard computer networks and platforms in conjunction with recently developed high-performance hardware and software multimedia technology. In such a system there are inevitably a large number of co-operating components - some very complex - which need to be controlled carefully in order to give good performance.

A fundamental design challenge, then, is to supply an interface that provides access to all of the facilities and performance that the system offers, while not requiring any special, in-depth knowledge by the implementers of the system. To achieve this the SpectreView Application Programming Interface (API) uses two main abstractions. A *Record Object* provides control over the acquisition of shadow video to a video server and a *Playback Object* provides control over the playback of shadow video from one or more shadow servers. The Record and Playback Objects are examples of a *Stream Object* class that contains features common to both Objects.

The SpectreView API allows OEMs to take complete control of the components of SpectreView and assemble them into their own, quite individual system-level solutions. In general these solutions will include one *Acquisition Manager* and some number of *Browse Clients*. In the following we will outline the SpectreView objects, and try to give an idea of how their APIs can be used to build an Acquisition Manager and Browse Clients.

5.1.2 Stream Object.

The Stream Object describes video flowing across the network either between a shadow video encoder and a shadow video server, or between the shadow video server and the client PC. An instance of a Stream Object always has a current *State* that describes the way video is buffered in the network. At any time an application program can read the current state of an object and can cause transitions of that object to a new state by invoking the appropriate method. By manipulating the objects in this way the application program can control the flow of video between various points on the network. Figure 1 shows the Stream States for the Record and Playback Objects.

A Stream Object represents an ordered sequence of video frames stored in various parts of the system, possibly in different formats - in the encoder, on the network and within the video server. Indeed, the video described by the Stream Object may not yet be acquired, but undergoing monitoring for a trigger condition to be met which then causes acquisition to commence.

Where a frame of video needs to be specified for such a purpose - as a trigger point, or as the target of a seek operation during browsing - an index into the stream is used. The Stream Object allows an index to be of four kinds: an absolute frame number from the stream start, an absolute SMPTE timecode, a relative frame number or a relative SMPTE timecode.

5.1.3 Record and Playback Objects.

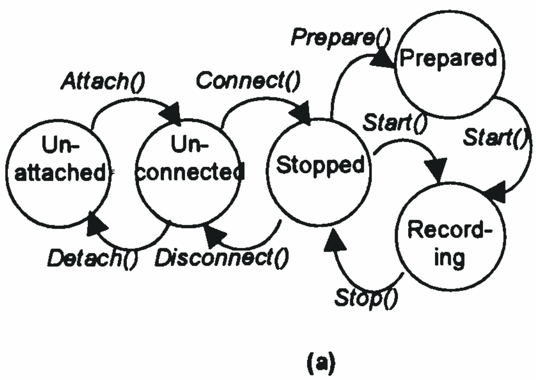
The Record Object provides the basic mechanisms for recording video onto a shadow video server. Where a stream is to be recorded the stream state diagram is as shown in Figure 1(a). Invoking the appropriate method upon the object causes transitions between states of that object. For example, in the 'Stopped' state an invocation of the *start()* method will cause a file of video to be recorded. A detailed description of the exact meanings of the states is beyond the scope of this paper, but in general, resources are committed as the states make transitions from 'Unattached' towards 'Recording' and the readiness of the system to start recording increases.

One stream state merits special attention and that is the 'Prepared' state. One of the most important functions which SpectreView performs is that of *Acquisition* where video is loaded into the SpectreView system. In the 'Prepared' state the system is monitoring incoming video for an index which matches a pre-set start point, at which time the stream object moves to the 'Recording' state. Alternatively the *Start()* method can be invoked at any time which circumvents the trigger mechanism and causes recording to start immediately.

The Methods shown in the figure are not the only ones; there are other mechanisms, for example for specifying the video source and destination, for setting start and end trigger points, and for obtaining information about various aspects of the system.

The Playback Object provides the basic mechanisms for playing back video from a shadow video server. In this case the stream state diagram is as shown in Figure 1(b). As can be seen, the process of preparing the stream in readiness for playback is similar to that for recording, and indeed, many of the underlying operations are the same. The Playback Object, however, does differ fundamentally from the Record Object in that it supports the concept of a *Clip List* - an ordered collection of video segments which can be played back as a single stream.

The main functionality of the Playback Object lies in its support for managing clip lists and in the mechanisms provided for seeking to specific points in a clip (specified by frame number or timecode), and playing these clips at variable speeds, both forwards and in reverse.



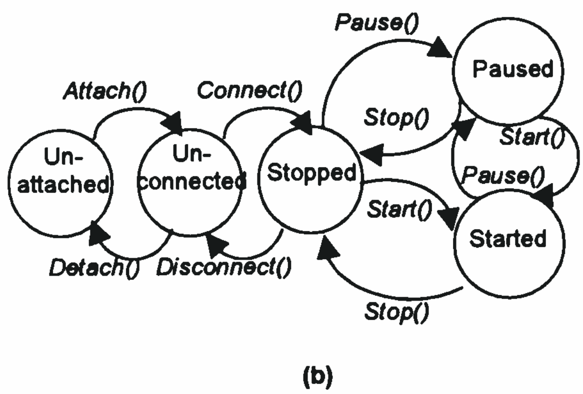


Figure 1. Stream State Transitions for a Record (a) and a Playback (b) Object.

5.1.1 Other Objects.

Other Objects are defined within the system, for example, a Management Object presents APIs for remotely managing resources in the system, for MPEG file naming, copying, deletion and access control. APIs are also provided to allow OEMs to install Servers, nSpectre units or Clients automatically.

5.2 The Capture Station - nSpectre.

The nSpectre is a networked media capture and encoding hardware unit designed specifically for use in the SpectreView system. It can capture and digitise two independent video and audio channels and process them into two independent MPEG-1 video and MPEG layer 2 audio system streams at 1.5 Mbits/sec each. It also captures two channels of SMPTE LTC and VITC timecodes and performs frame-accurate annotation of these onto the MPEG system streams.

An important aspect of nSpectre is that it is *remotely administered.* Although it is a sophisticated computing device in its own right, there is no conventional operating system running on it, nor is there a user interface in the normal sense. The nSpectre is designed to operate as a stand-alone unit in a rack of equipment and as such is configured and controlled through the network using a proprietary protocol (called OMDCI) which provides the SpectreView API with a mechanism for controlling all the functions that the hardware presents.

Physically, the nSpectre is housed in a standard 1U case suitable for rack mounting. Video input standards supported are analogue component (RGB and YUV), composite and S-Video, with SMPTE 259M digital video to be supported in the near future.

5.3 The Video Server - SpectreView Server.

The SpectreView Server implements a streaming video service designed to avoid the problems associated with file based access mechanisms such as video glitches, dropout and corruption. These undesirable effects quickly become evident once more than a small number of users attempt to use a file-based server. The SpectreView Server runs as an ordinary service on a standard Microsoft Windows NT platform using an NTFS file system.

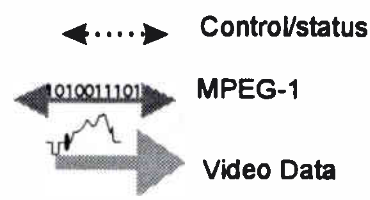
A reasonably high-performance (200 MHz Pentium) machine with an efficient network interface card is recommended for good performance, but no non-standard hardware or software is required. The Server supports up to sixteen concurrent recordings from the nSpectre Capture Stations and thirty or more concurrent playback streams. It also supports play-while-record streams, a facility that is particularly useful for applications requiring 'live' captioning and play-out of broadcast news or sports material.

Apart from supporting record and playback streams, the server presents functionality to the management APIs for 'housekeeping' operations. For example, it is possible to catalogue material on a Server, to reserve space for a future recording, or to control the archiving or restoration of material to and from backup media.

6. Applications

The DDSB solution can be applied to a number of real-world applications. Here we illustrate two of these, and also discuss other possible opportunities for deployment.

In the figures there are two types of network communication, compressed video data, and the controls and status messages necessary to manage the interaction of the components of the system. The video data can be either analogue or digital. In either case, the associated time-codes and audio will also be input to the system. This is indicated by different types of arrow, thus:



6.1 An Automation Application

In Figure 2 below is an illustration of a typical automation system.

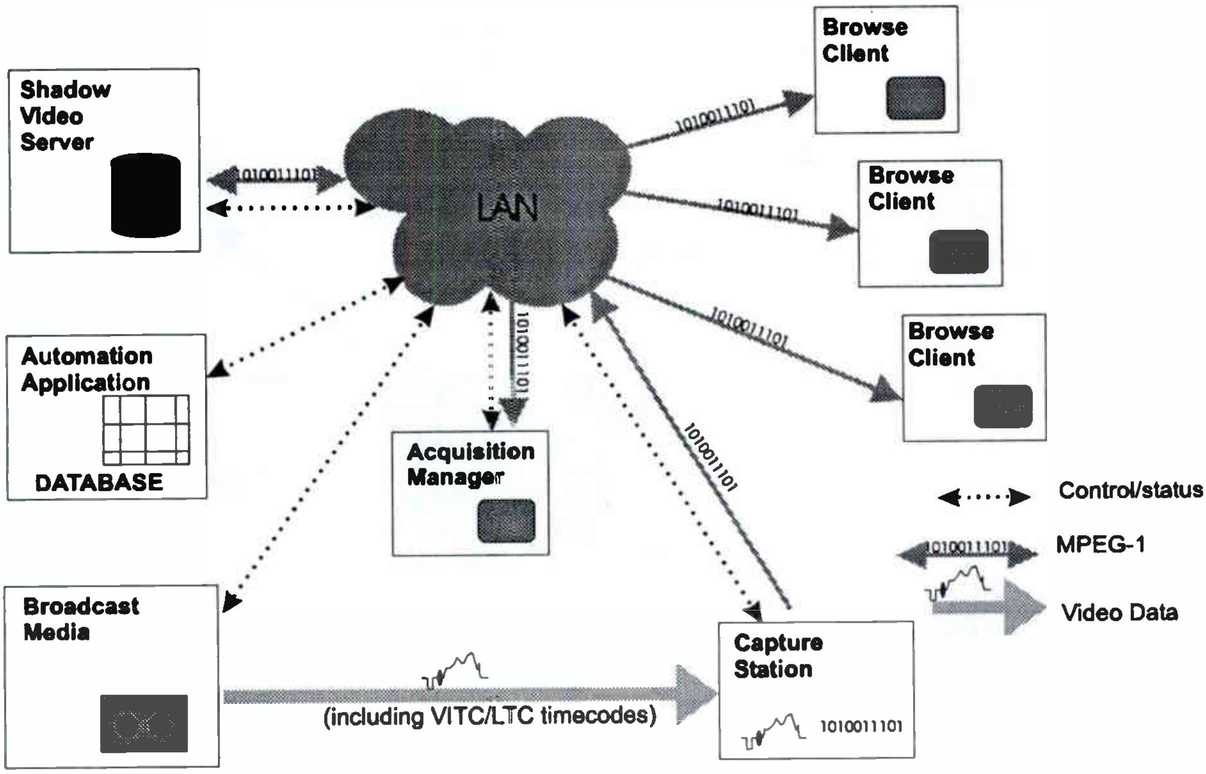


Figure 2 - An automation example

The broadcast media source is controlled by an *Automation Application*, which maintains a database of all the material loaded into the system, and controls the play list. This application would also manage a video routing and switching system, which is not shown for clarity. When new material is available, the Automation Application will communicate with the *Acquisition Manager* via the local area network to instruct it to record the segment to the *Shadow Video Server* at a rate of 1.5Mbits/sec. The time-code interface ensures that each and every time-code corresponding to the original source material is stored with the shadow copy. The database will also be updated with the relation between the shadow copy and the original source material.

After this any *Browse Client* can preview material from the shadow server. If any scheduling or playout changes are required, these will be signalled back to the automation application, and the play list will be updated. These changes can be examined, and their effects confirmed to be as expected.

It should be noted that the use of a shadow copy means that editing operations are not performed directly on the source material. This is, of course, similar to the techniques employed in non-linear editing. However, the major differences are the combination of accessing preview quality images in real time by a local area network, and the integration into the automation application to provide a means whereby users of the system may view and manipulate data remotely without needing to interfere with the normal operation of the broadcast source in its play out function.

Different Browse Clients can be provided with different functionalities, dependent upon their role in the system. For example, a Continuity Announcer may be provided with the ability to preview programme material to rehearse cues, and a Transmission Manager can be provided with tools allowing for the previewing and rescheduling of interstitials to allow for last minute running order changes.

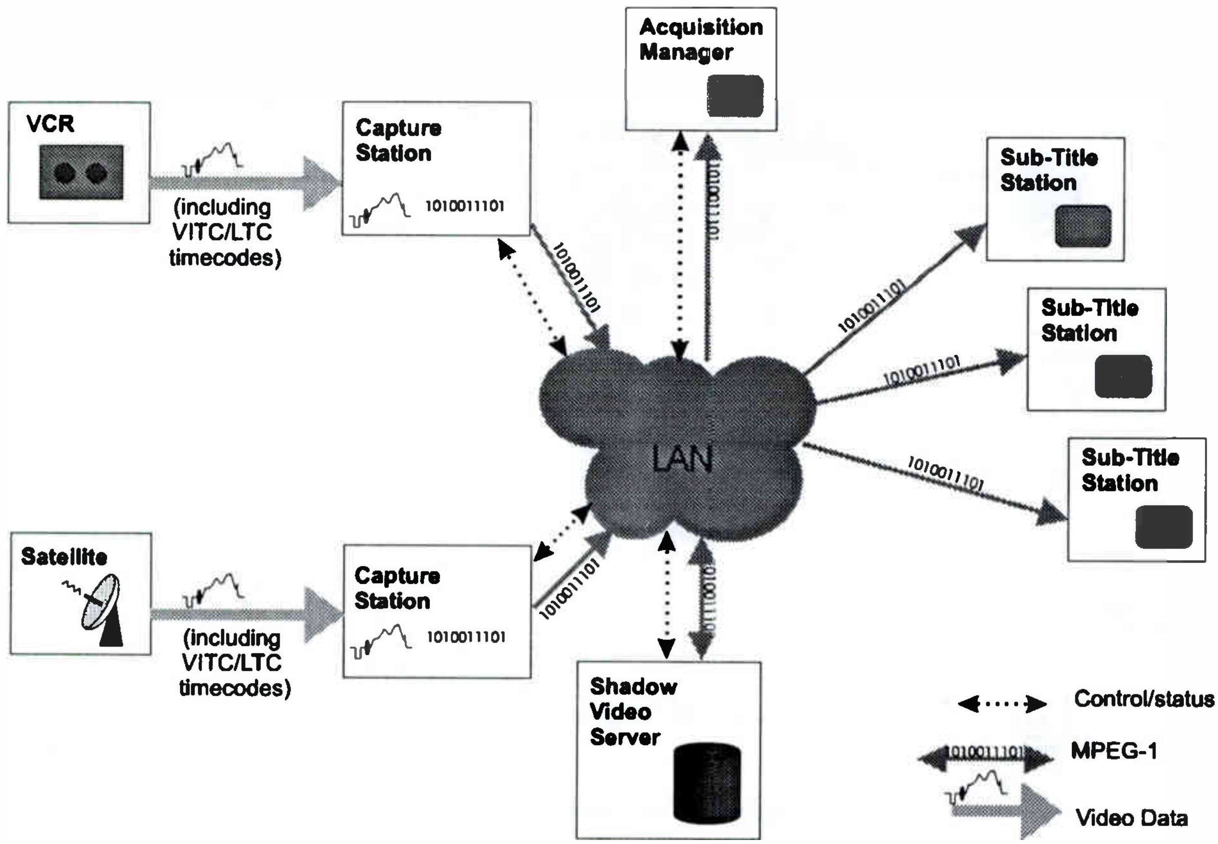


Figure 3 - A sub-titling example

6.2 Sub-Titling (Closed Caption)

A sub-titling, or closed caption, system will be similar, but in this case the shadow material will be input to the server either from tapes, or from a high-bandwidth long distance link – perhaps a satellite feed. The system in Figure 3 will allow groups of sub-titlers to work together on one source programme. The output from the subtitling system will be a set of sub-titles, which may be passed to a transmission suite for inclusion in the broadcast programme.

The advantages of this system are considerable:

- Each sub-titler can be working on an identical copy of the material.
- Staff can be easily re-allocated to priority tasks.
- No time is lost in the process of duplicating the input material.
- Changes may be quickly updated if edits are required in the source material.
- The tools are easier, cheaper and more effective than the conventional alternative of video tape and scrubbing.

6.3 Other applications.

The same techniques can be applied to systems that are being used for such purposes as newsrooms, compliance monitoring, pre-production editorial work and general asset management.

The full extent of the application of this technology will only be realised in the fullness of time, but any broadcast management operation which does not require the manipulation of full bit-rate source material can be made easier and cheaper by the adoption of a solution incorporating shadow browsing.

7. Conclusions.

In this paper we have described in detail the key features of the new technology of DDSB, and how it relates to the broadcast industry.

Shadow browsing will improve the ability of broadcasters to manage their material, increase productivity and reduce the time delays at all levels in the production process, from initial editorial content to final transmission. An effective DDSB system meets most of the management needs throughout a facility, at low cost and with incredible flexibility, and in combination with the appropriate automation and media management tools will greatly benefit programme producers.

The increasing availability of higher bandwidth wide-area communications will also allow organisations to use shadow browsing as a way of viewing, manipulating and managing material over a significant geographical distance. In many cases, the activities of broadcast organisations are hampered by the necessity to copy material to tape, and then physically transfer this tape from one place to another.

8. References

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Building the Digital Radio Station

Tuesday, April 7, 1998
1:00 pm-5:30 pm

Chairperson:
Tom McGinley
WPGC-FM/WPGC-AM, Greenbelt, MD

***Avoiding The Pitfalls of Upgrading an FM Station to Digital**
Daryl Buechting
Harris Corporation
Quincy, IL

New Electronic Music Production Studio, Computer Aided Music Studio
Shoji Akita
NHK Broadcast Engineering Department
Tokyo, Japan

***From Analog to Digital: Proper Planning and Implementation Through Consolidation**
Ben Brinitzer
SFX
Raleigh, NC

***Moving Radio People, Programs, and Pieces from A to D**
Al Korn
Georgia Public Broadcasting
Atlanta, GA

***Rules of the Road for the Digital Audio Superhighway: Equipment Selection and Design Criteria for Digital Air and Production Studios**
Russ Mundschenk
WBEB-FM
Bala Cynwyd, PA

ISDN for Studio Call-in Talk Systems
Steve Church
Telos Systems
Cleveland, OH

***Design Considerations of a Digital On-Air Radio**
Ted Staros
Pacific Research & Engineering Corp.
Carlsbad, CA

Eliminating Delayed Audio Feedback in Live Broadcasts
Eric B. Lane
Avocet Instruments, Inc.
Beaverton, OR

***Total Cost of Ownership (TCO) for Windows-Based PCs**

Skip Pizzi
Intertec Publications
Redmond, WA

*Papers not available at the time of publication

NEW ELECTRONIC MUSIC PRODUCTION STUDIO, COMPUTER AIDED MUSIC STUDIO

Shoji Akita, Kimio Hamasaki
(Sound Engineer)
Production Operations Center, Broadcast Engineering Dept., NHK
Masami Fujita
Engineering Development Center, Engineering Administration Dept.,NHK
Tokyo, Japan

ABSTRACT

The electronic music studio of NHK had been renewed with the full digital facilities. A general plan of renewal was as follows; tape less production by using the non-linear recorder, the introduction of an integrated digital mixing console combined with Digital Audio Workstation, multi-channel sound production, improvement and expansion of music production equipments. This paper describes an outline of the full digital facilities and the latest music production activities.

INTRODUCTION

In the electronic music studio called CC-500, from the 1950's, a lot of contemporary musical works were produced by many composers, Toru Takemitsu, Toshiro Mayuzumi and Maki Ishii, and so on. In the 1950's and the 1960's, composers were hard on studying new way of composing and creating new sound by the various electronic components such as oscillators and filters. In the 1970's, the synthesizer appeared on the market and it had been indispensable to the field of popular music in the 1980's. Since we can inexpensively get various synthesizers recently, this studio had lost the originality in creating sound.

Before the renewal of this studio, it took enormous time to create original sound and to edit sound materials, because every production were done by using all kinds of tape recorders. There were also many analogue equipments to create various sound materials, however those were too old and old-fashioned to comply with a composer's wishes. Therefore, last year, this studio had been renewed with the full digital facilities.

DESIGNING STUDIO CC-500

This studio is expected to serve for composers to create various kind of music, and contribute to growth of the musical culture of Japan. The basic conceptions for renewing this studio are as follows;

1. The facilities of this studio can meet the needs of all sorts of composers.
2. In this studio we can produce various kind of music, not only electronic music but also other kind of music such as music for the theme of television and radio programs.
3. This studio has to be constructed by small cost and in the limited space.

For realizing these basic conceptions, we designed this studio according to the following plans.

TAPE LESS PRODUCTION

Over fifty Digital Audio Workstations (DAWs) have been used in NHK. DAW is non-linear recorder which utilizes a hard disk and a magneto-optical disk as the recording media.

At first DAW was introduced for audio post-production studio of TV programs. As DAW has powerful editing functions, it is quite useful for editing of narration, dialogue, sound effects and music. On the other hand, DAW has been also used for editing music in musical productions, because of its effectiveness in the productions of music. Whilst it is very rare to apply DAW for multi-track music recording, this is the first time that we have introduced DAW in a studio of recording music.

During the production of contemporary music, sound materials are copied and mixed over and over. Therefore it is very difficult to keep the sound quality and quick processing by making use of conventional magnetic tape recorders. In order to realize high quality production and streamlining production. The recording equipments of this studio are consist of the DAWs without any magnetic tape recorders.

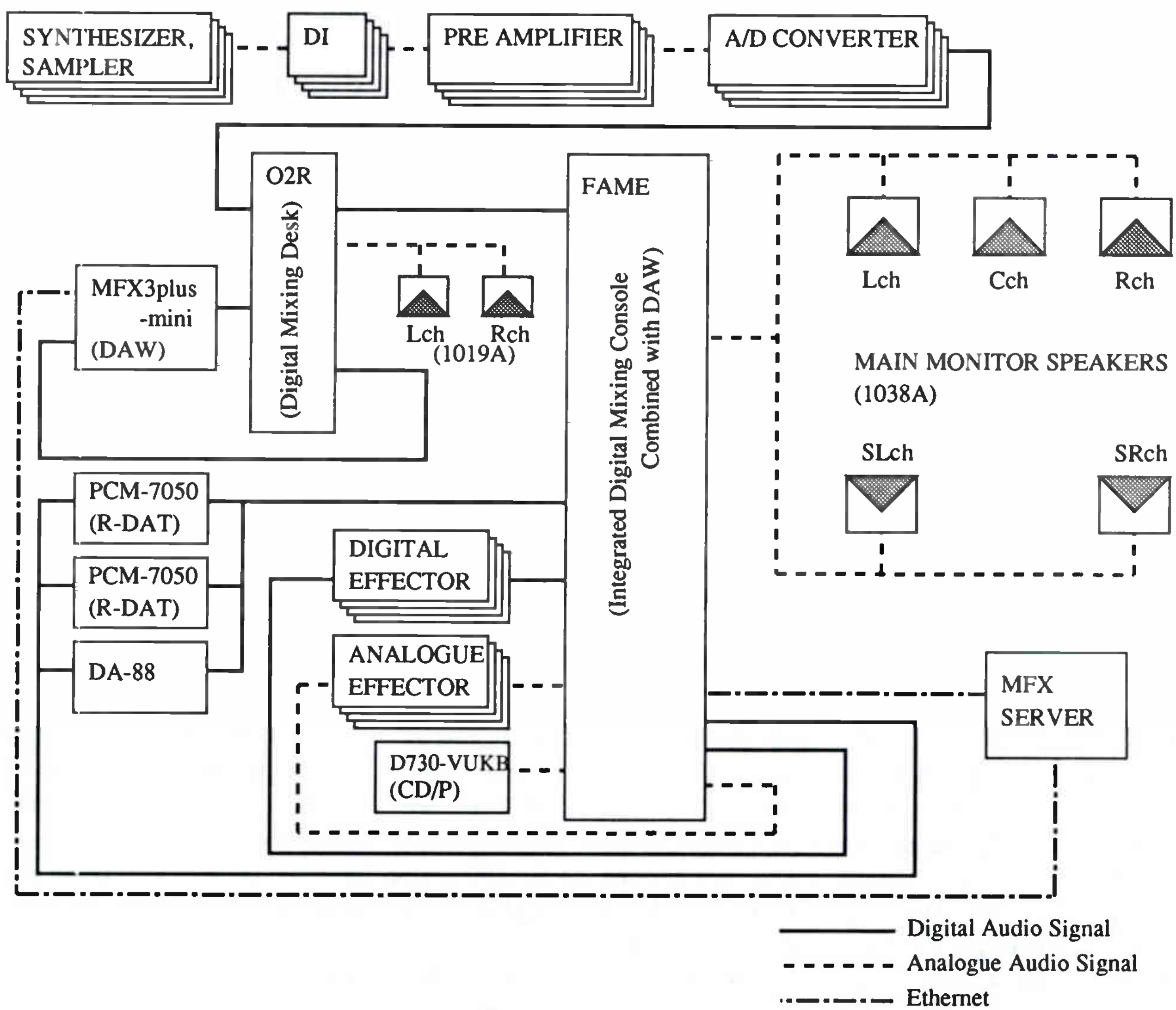


Figure 1. Schematic diagram of the sound system of studio CC-500

AN INTEGRATED DIGITAL MIXING CONSOLE COMBINED WITH DAW

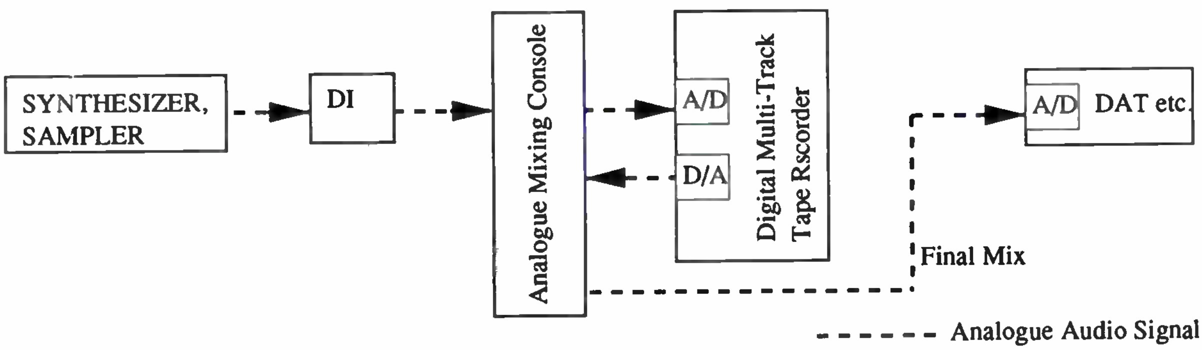
An integrated digital mixing console combined with DAW was introduced into this studio. The reasons why we had chosen the integrated digital mixing console combined with DAW were as follows;

1. On this type of console, data of mixing control are related to data of DAW. Therefore it is much more efficient in sound production than conventional mixing desk.
2. An integrated system is less expensive than a separate system.
3. An integrated console is more compact than the other console and it is not necessary to keep wide space for installation.

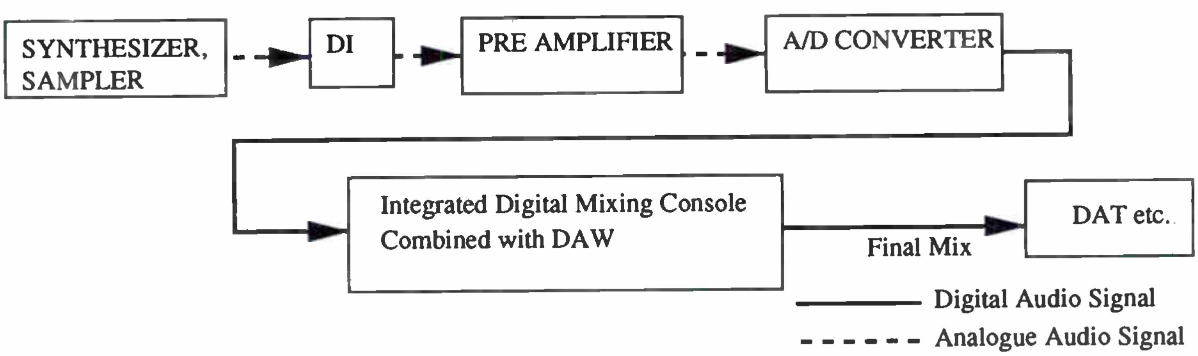
MULTI-CHANNEL SOUND PRODUCTION

In NHK many television and radio programs have been broadcasted in stereo, and some HDTV programs are in 3-1 quadraphonic surround sound. For example, we have been regularly broadcasting a show of Japanese popular songs, "Kayo Concert", once a week,in 3-1 quadraphonic surround sound.

Recently the multi-channel sound production is discussed very often, because of the development of digital broadcasting and DVD. The multi-channel sound production is very important for the contemporary music production, too. Composer usually design their sound production of music not only in stereo image but also in spatial image. Therefore in this studio, 3-2 surround sound production can be done fundamentally and also up to 8 channel sound production can be produced by free arrangement of loudspeakers.



(a) the conventional analogue mixing console with a digital multi-track tape recorder



(b) the full digital facilities of studio CC-500

Figure 2. Schematic diagram of the sound system of compares the sound processing

THE FACILITIES OF AUDIO SIGNAL PROCESSING

Schematic diagram of the sound system of studio CC-500 is shown in Figure 1. It is very important to maintain the quality of sound source of the synthesizers and the samplers. Analogue audio signal of sound sources is fed directly to ADC (analogue to digital converter) via DI (direct injection) and pre amplifier. ADC is processed by quantization of twenty bits and the sampling frequency is 48kHz. After converting to digital signal every processing is done in the digital domain.

Schematic diagram, figure 2, of the sound system compares the sound processing between the conventional analogue mixing console with a digital multi-track tape recorder and the full digital facilities of studio CC-500. Whilst processing of sound is repeated over and over in the electronic music production, ADC and DAC should be repeated in the conventional analogue mixing system and it looses the quality of sound. On the other hand, it is not necessary to convert between analogue and digital any more after the first digitizing from analogue feed of sound sources by the full digital facilities.

THE FACILITIES OF MUSICAL PRODUCTION

The most important function of this studio is to realize any demands of composers in the creation of new sound. Following is the MIDI system and musical sound sources of studio CC-500.

MIDI SYSTEM

Schematic diagram of the MIDI system of studio CC-500 is shown in Figure 3. MIDI (Musical Instrument Digital Interface) signal is distributed to each equipment via two MIDI Time Piece which are matrix for MIDI signal. The master keyboard is used to play each sound source and to feed data of musical notes onto a personal computer. Generally musical sequence software for PC which are Mac, Power Mac, and Windows, is utilized in order to control musical sound source equipments according to the musical score.

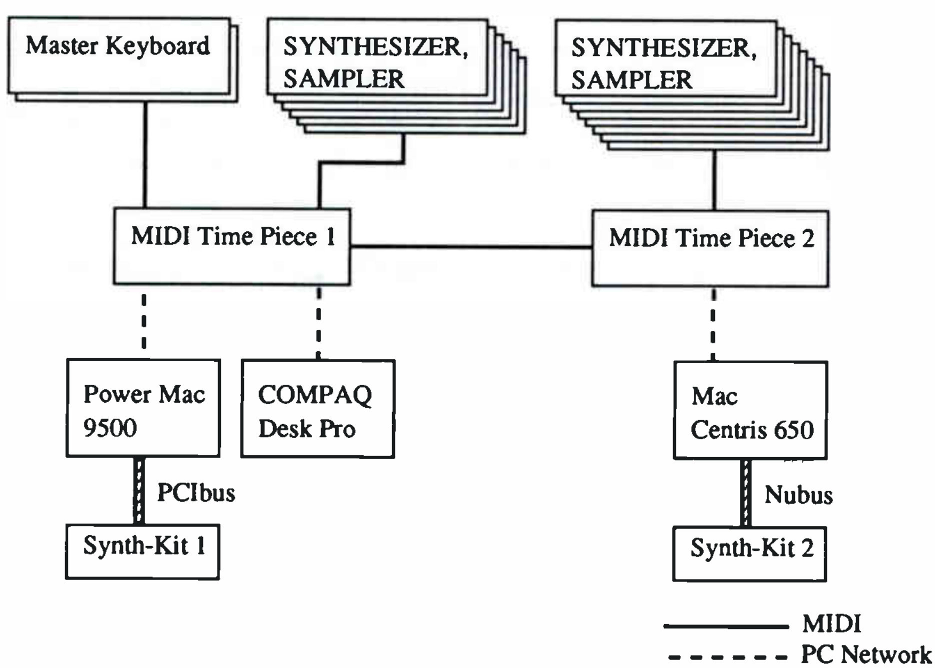


Figure 3. Schematic diagram of the MIDI system of studio CC-500

MUSICAL SOUND SOURCES

Musical sound sources are categorized as analog synthesizers, samplers, and digital synthesizers. Analogue synthesizer gives an old-fashioned impression, but it is still attractive to generate various sounds.

Samplers are most popular equipments. Sound of real musical instruments and voices, etc. are sampled by a sampler and those are used as musical sound sources. It is possible to create some new sound sources by this procedure.

Digital synthesizers generate various sounds by means of DSP (Digital Signal Processor) algorithm. These algorithms are designed according to the musical instrument acoustic theory and they have many possibility to create high quality original sound. Of course it is not so easy to develop any algorithms. But we are trying hard to develop new original algorithm by using a developing tool.

VIDEO DISK RECORDER

A video disk recorder was introduced in this studio. This video disk recorder shortens the time for searching any certain video and transport control. On the other side, this video recorder can be used to prescribe a tempo of music. By the musical sequence software it is not possible to keep flexible tempo. Therefore picture of a conductor who gives certain tempo is recorded on this video disk recorder and it is enable us to vary the tempo of music during the performance.

SERVER SYSTEM

The hard disks and magneto-optical disks are used for the recording media of DAW. The DAW equipped for this studio has two hard disks, each disk has a capacity of 4 gigabytes and a magneto-optical disk has a capacity of 1.3 gigabytes.
But it is not enough to keep several projects at the same time by using only these attached recording devices. The electronic music production needs a large amount of disk capacity.

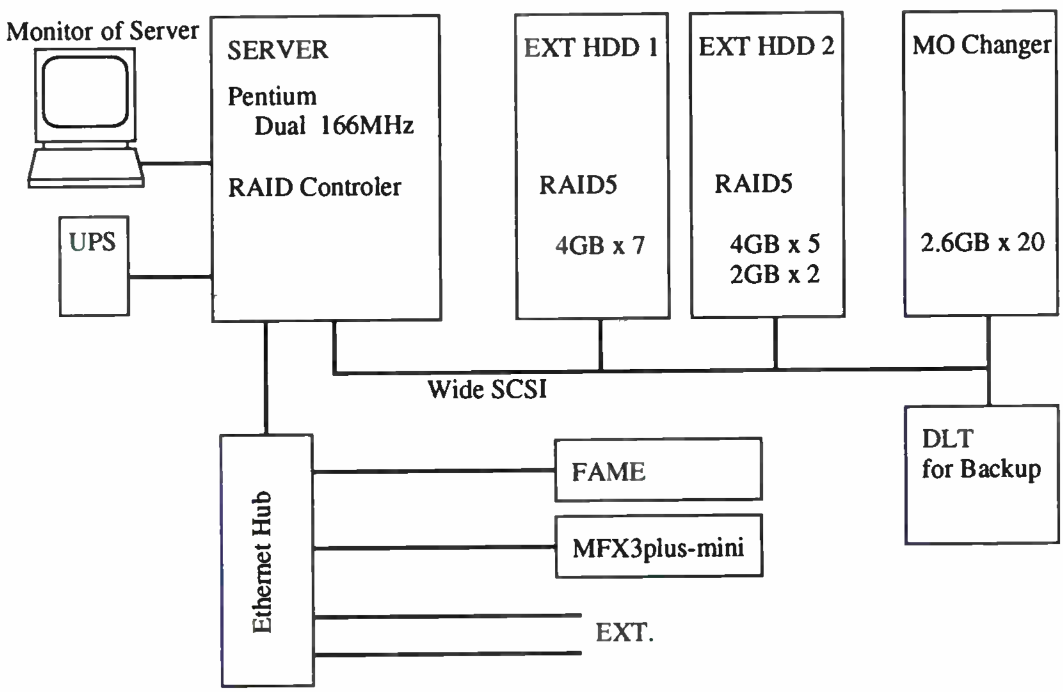


Figure 4. Schematic diagram of the server system of studio CC-500

For such reason the server system which has a huge capacity has been introduced. Schematic diagram of the server system of studio CC-500 is shown in Figure 3. Windows 95/NT server is selected and the server has dual CPU of Pentium of 166 MHz. There are twelve hard disks of 4 gigabytes and two hard disks of 2 gigabytes. The all capacity of these hard disks is 52 gigabytes. There is also an automatic changer of twenty magneto-optical disks. The hard disks were connected by Wide SCSI and the server is connected to DAW by Ethernet of 100 Mbps via a switcher hub. Some DAWs can access to the server independently and they can share sound clips in the server without copying.

CONCLUSION

The first recording session after renewal was music production for radio drama. We used only the DAWs for this recording without any tape recorders. The non-linear recorder is very convenient, because the time for locating is almost zero. It makes total time for recording and mixing much shorter than before. The non-linear recorder has another powerful function which is non-destructive recording and editing. We can retake any part of the music without erasing a previous take and chose best take from every session. In the future every DAWs which are installed in each studios shall be connected to the server via DAW network. It will be possible to access any sound files from any studios easily. Tape less recording and network of the DAW should be key technology of sound production in 21th Century.

ACKNOWLEDGMENTS

The authors would like to thank all partners in the studio CC-500 project. And he would like to express his sincere thanks to Fairlight Corp. and Fairlight Japan Corp. they designed and produced the equipments of studio CC-500.

ISDN FOR STUDIO CALL-IN TALK SYSTEMS

Steve Church
Telos Systems
Cleveland, Ohio

ISDN has become widely used for remote broadcasting. But does it offer anything to stations who want to improve the quality of call-in programming? The answer is yes, most definitely. When the telephone network is extended to studios in pure digital form, quality is improved in a number of dimensions. Studio telephone interfaces using ISDN are being introduced now. These are especially suitable for installations which intend to keep audio signals as much as possible in the digital domain, and for the combined studio facilities which are becoming common under consolidation.

ISDN for Call-in Systems?

Broadcasters have discovered that ISDN service offers valuable digital access to the dial-up telephone network. High-fidelity ISDN MPEG codecs are widely used for remote and inter-studio feeds, and most major-market stations have ISDN lines installed for this purpose. The wide use of ISDN is one expression of the natural progression of analog to digital connectivity and electronics for moving and processing audio signals.

ISDN can also be used to enhance the quality of call-in shows. While not apparent to the many users of ISDN for MPEG codecs, the telephone network is able to cross-connect Plain Old Telephone Service (POTS) analog lines and ISDN. (Telos has had a POTS calling feature in our Zephyr codec since its introduction, but, according to our support engineers, most users are surprised to discover that it can be used to connect to analog phones.)

The cost of ISDN service is not a barrier. In most parts of the USA, the benefits of ISDN can be had for about the same cost as analog. The usual ISDN service has two channels and costs about twice as much as a POTS line. (Pricing varies around the country, at from a 20% discount to a 30% premium. The average is probably around a 10% premium.)

The Telco network is almost entirely digital–except for the "last-mile" copper connections from the central office to the customer site. Telephone switches are digital, long-distance calls travel over digital fiber strands, and many local phone paths use digital T1 lines. Despite this, the vast majority of users interface to the network via an analog technology that is little different from that employed in Alexander Bell's days. To support the immense installed base of analog telephones, central office equipment includes a stage which converts the digital signals to analog for connection to POTS subscriber lines.

Within the Telco network, calls are routed over 64 kbps channels. A sampling rate of 8kHz is used, with a word length of 8 bits. The 8kHz sampling rate supports a Nyquist (audio cut-off) frequency of 4kHz. In practice, telephone systems are designed to have audio frequency response extending to 3.4kHz in order to allow relatively simple roll-off filters to be used. The word length is what determines dynamic range–and 8 bits would only permit 48 dB were it used in standard PCM linear fashion. A primitive kind of compression is used to stretch the dynamic range: µLaw in North America and much of Asia, and A-law in Europe. This is a scheme that equalizes the step-size in dB terms across the dynamic range–a smaller step-size on low-level signals reduces quantization noise and improves effective dynamic range to the equivalent of about 13 bits.

Benefits of ISDN

So much for the technology; what we really want to know is "What's it going to do for me?" The answer is multifold:

- ISDN lines are inherently 4-wire, maintaining separation between send and receive audio signals and improving hybrid performance
- Better digital-analog conversion quality
- Lower noise
- Better, faster call set-up and supervision
- Higher gain and reduced feedback during conferencing
- Caller ID
- Line monitoring capability

ISDN Lines are Inherently "4-wire"

Analog lines use a single pair of wires (hence are a "2-wire circuit") for both signal directions, mixing the announcer and caller audio. This causes the famous "leakage" problem–where the announcer's audio is present on the interface output, where we desire that there only be caller audio. The announcer audio, having been affected by the phone network is filtered and phase-shifted. When it combines with the original caller audio in the on-air mix, the result can be a significant distortion of the announcer's voice.

A "4-wire" circuit has two wire pairs, and therefore two independent audio paths. Some telephone networks are actually implemented that way–the military's AUTOVON being the major example in the USA. Digital circuits inherently offer independent and separated signal paths because it is not possible to have bits moving in both directions without separating them somehow. Though a digital circuit may today not use wires at all, but rather fiber, or microwave radio, or satellite, telephone engineers, bowing to tradition, continue to refer to all separated speech paths as being "4-wire."

While the application of DSP to the problem of separating the signals–used in digital hybrid interfaces–has made a dramatic improvement over analog systems, ISDN enables a yet further improved performance. That is because it offers a fully independent path for each speech direction. In the case where both ends of a connection are digital, there is no mixing whatsoever. In the call-in application, the far-end from the studio will still be 2-wire, so the audio paths will not be fully independent and we will still need a digital hybrid function to cancel residual leakage. Moving the studio side connection away from mixed analog can help tremendously because it is a much better starting point.

Better Digital-Analog Conversion Quality

The codec (Coder/Decoder) conversion chips used in telephone central offices are not very good compared to the converters commonly used in audio equipment. Fidelity is not an important consideration when designers choose parts for this function. And much of the problem is due to telephone standards: these codecs have 8-bit digital inputs and effectively only 13-bit analog outputs.

In a professional interface for studio application, we are able to afford to design-in much better converters than available in the phone company's equipment. Noise-shaping functions permit a larger word-length converter to provide significantly better distortion and signal-to-noise performance.

In all-digital installations, the phone interface can maintain a digital path all the way. AES/EBU can be provided on the interface to accomplish the connection to the studio gear.

Lower Noise

Being digital circuits, ISDN lines are not susceptible to induced noise. Analog lines are exposed to a wide variety of noise and impulse trouble-causers as they move across town on poles and through your building. Hum is the main one, given the lines proximity to transformers and power lines, but there are also sources of impulse noise from motors, switches, and other sources. Digital lines convey the bits precisely and accurately from the network to your studio equipment without any perturbation–so the audio remains clean.

Call Setup and Supervision are Better

Analog lines use a strange mix of signaling to convey call status. Loop current drop and returned dialtone signal that a far-end caller has disconnected; blasts of 100 volts at 20 Hz mean someone wants you to answer. Why should we be using a mechanism designed to bang a gong against a metal bell to transmit network status information in the 90s? ISDN uses a modern digital approach to controlling calls and conveying status information about them. The sophisticated transactions on the D channel are able to keep both ends of a call accurately informed about what is happening.

For starters, ISDN call set-up times are often a few 10's of milliseconds, enhancing production of a fast-paced show.

Perhaps more importantly, when a caller disconnects while waiting on hold, the ISDN channel communicates this status change instantly. This contrasts with the usual 11-second delay on most analog lines. One of the most common complaints of talk hosts is that they go to a line where they expect a caller to be waiting, only to be met with a blaring, annoying dialtone. The chance of this happening with an ISDN line is reduced to near zero.

Another common error is the condition where a talent goes to punch-up a line that looks free, but which actually is just about to begin ringing and connects to a surprised caller. This condition results from the delay in the ring signaling which comes from the nature of the analog line's ringing cadence. This is much less likely with ISDN because the ambiguous status period is eliminated.

Higher Gain and Reduced Feedback During Multi-line Conferencing

When conferencing is required on 2-wire circuits, very good hybrids are needed to separate the two audio paths in order to add gain in each direction. When the gain around the loop exceeds unity, there is the possibility of feedback "singing." Since the conference path usually includes four AGC functions, the hybrid must be sufficiently good to cover the additional gain that may be dynamically inserted.

Because of the 4-wire nature of ISDN, the hybrid function is more effective–and more reliably so across a variety of calls. That means more gain can be inserted between calls before feedback becomes a problem.

Line Monitoring

Since there is a full-time connection between the central office and the terminal on the D channel, it is possible to detect when a line is a not working. On an analog line, one discovers a problem only from a failed attempt to use the line.

ISDN 101

ISDN lines come in two varieties: Basic Rate Interface (BRI) and Primary Rate Interface (PRI). BRI lines are the kind we normally see in broadcast stations, as these are what is used with MPEG codecs. BRIs have a capability of one or two 64 kbps channels. The line from the central office is a single copper pair identical to a POTS line. When it arrives at the subscriber, this is called the "U" interface. The U interface converts to an "S/T" interface with a small box called an "NT-1." In the USA, NT-1 functionality is usually included in the terminal equipment. In Europe, the telephone company provides the NT-1. Only one NT-1 may be connected to a U interface. As many as eight terminals may be paralleled onto an S bus.

Professional equipment should usually provide access to the S interface, so that it is possible to parallel multiple terminals. You can use either an external NT-1, or the equipment may have an internal NT-1 with both U and S/T connectors.

PRI lines can have as many as 23 channels (31 in Europe). These travel over T1 (E1 in Europe) lines, which require two wire pairs. Normally, this service is intended for bulk-line connection to PBXs.

The B "Bearer" Channels are the 64 kbps paths which carry the voice audio. The D "Data" channel is the path between the central office and terminal equipment that is used for call set-up and status communication. It operates at 16 kbps.

Broadcast interfaces may use either type of line. A simple interface for the newsroom could use a single BRI. Even sophisticated multi-line systems could use BRIs, with enough of them to achieve the desired number of lines. While PRIs would seem to be a more technically appropriate solution for a multi-line system, BRIs may be more cost-effective, more readily available, and would serve to provide a measure of redundancy.

Data and Voice

ISDN lines may be used for voice signals encoded in standard fashion to allow interworking with analog phones, as proposed here, or may be used to transmit digital data streams. The latter mode is used for such applications as high-speed Internet access. It is also the mode used with MPEG codecs. In that case, the ISDN line may be carrying voice signals, but is doing so in a format which is not compatible with the POTS network.

The distinction is made in the "Setup" message which begins each call. Of course the ISDN call-in interface should always use the "voice" setup configuration.

(Incidentally, the voice mode may well be able to convey digital data–and some Telcos charge more money for data service. That is why some ISDN "modems" have a special data-over-voice option which fools the central office into billing for voice call, even when the payload is data.)

The ISDN Broadcast Interface: What will it do?

Send/Receive Separation

This is the traditional hybrid function provided by broadcast telephone interfaces. Despite the fact that ISDN lines naturally have two independent send and receive paths, there is still the need to provide additional functions to further reduce "leakage." The reason is that almost all calls will originate with telephone sets connected via two-wire analog lines, and so there will still be a mixing of both speech directions.

Acoustic Coupling Reduction

There is often an acoustic path between the received caller audio and the send audio signal. This results from having a loudspeaker in the studio that produces sound which couples into the microphones. When the talent use headphones for monitoring callers, this is not a problem. But sometimes it is not practical to convince guests to wear headphones, and television stations generally don't want talk show talent to wear earplugs. In these cases, it is desirable to have a mechanism to reduce the coupling electronically. This can be accomplished with a combination of adaptive cancellation and dynamic gain reduction.

High-grade Digital-to-Analog Conversion

When an analog connection to studio equipment is required, pro-grade converters can be used to provide much better quality than the usual Telco conversion. At minimum, 16-bit parts should be used, but 18-20 bit parts may not be overkill given their current reasonable cost.

Sampling-rate Conversion

When the studio connection is via a digital AES/EBU channel, no analog-digital conversion is required, but it will be necessary to adapt the sampling rate of the telephone network to the studio rate. Telco sampling rate is 8 kHz and studio equipment will usually operate at 32, 44.1, or 48 kHz. A process is required to perform the required up-and-down sampling, while suppressing aliasing and reconstruction audio components.

Automatic Gain Control

This function should be provided on both the send and receive audio paths. On the send side, it is necessary to smooth the wide level variations which arise from usual studio practices. Unlike a telephone handset microphone which is placed fairly consistently from the mouth and can be relied upon to produce reasonably consistent levels toward the telephone network, studio microphones and mixing consoles produce a wide range of levels. Talent are used to having on-air processing take care of level variations and are generally not very careful at riding gain.

On the receive side, AGC is essential to deal with the very different levels that can result from the many types of phone sets and Telco analog network components. Our experience is that audio volume can vary as much as 30 dB from call-to-call on a given studio line. This degree of difference requires a careful approach to audio leveling. An AGC that maintains a constant compression ratio regardless of average gain reduction produces more consistency. Freeze gating is also important, so that gain does not increase during caller speech pauses.

Dynamic Equalization

The Telos Delta POTS hybrid interface has included for some time a feature which balances the frequency spectrum of caller audio. With phone sets having a very wide variety of microphone characteristics, this function helps callers to have a reasonable consistency, and has been proven valuable in the "real world." We have found a three-band dynamic equalization processor to have the right trade-off of enough power and not too much undesired audible shifting of frequency characteristics.

Caller "Ducking"

This is an "aesthetic" requirement of many talk hosts. The function reduces the level of the caller when the host talks, allowing her an automatic control over a caller who wants to "carry on." This is a matter of taste: some talents and programmers prefer no ducking so that hosts and callers can conduct heated exchanges without impediment, while others want to exercise control. Equipment therefore needs to have a control which can adjust the effect to the desired level.

The time constants of this operation should be carefully crafted so as to mask the audibility of the gain change with the talent's voice.

Caller ID

ISDN naturally conveys caller ID information. This is transmitted instantly in the setup message, and is much faster than the 1200 baud modem method used in analog caller ID.

Conference Linking

With two B channels available on one BRI line, broadcast interfaces will be dual units, making possible high quality conferencing between the two potential callers. Some systems will probably support larger numbers of conferenced callers.

CONCLUSION

ISDN is now widely available, cost-effective, and offers many advantages for call-in talk systems. It is yet another example of digital technology enhancing broadcast operations.

ELIMINATING DELAYED AUDIO FEEDBACK IN LIVE BROADCASTS

Eric B. Lane
Avocet Instruments, Inc.
Beaverton, OR

ABSTRACT

A unique technique has been developed to overcome delayed audio feedback in live broadcast situations. Delayed audio feedback is the psychoacoustic phenomenon that occurs when a speaker hears his own voice delayed by a specific amount of time. Monitoring this delayed version of his own speech causes the speaker to stammer, change his speech cadence and make mistakes.

This new delay cancelling technique is a digital process in which both the originating and monitored signals are digitized, compared in a DSP, and the echo is removed.

INTRODUCTION

Avocet Instruments has developed a technique to eliminate the problems caused by delayed audio feedback in live broadcasts. The technique can identify a voice match between microphone and off-air return signals, and effectively removes the delayed voice from the return. With the technology provided by Avocet Instruments, the echo of a speakers voice can be suppressed by up to 40 dB and with up to 800 milliseconds of delay.

The Nature Of Delay

Two general types of echo cancelling exist: Acoustic echo cancelling, and line echo cancelling

Acoustic echo cancelling controls audio feedback in an acoustically confined area. Acoustic echo cancellers are used in speaker-phones where the microphone and speaker assemblies are located near each other. The Acoustic echo canceller removes the local talker's speech before the audio goes to the speaker to avoid feedback

Line echo cancelling is used when a signal on a transmission line either electronically or acoustically leaks into the return channel going in the opposite direction.

While both acoustic and line echo cancellers can use the same technologies, their applications and environments are often very different. Some semiconductor components and DSP algorithms are available for echo-cancelling, but the parameters we find in broadcasting are different from those in telephone transmission and speakerphone applications. In telephone applications, the returning signal is always at a lower signal level than the transmitted signal (typically about 15 dB of attenuation). Also, the timing of the echo is always less than 250 milliseconds.

Broadcast echo problems do not follow these criteria. Because the original signal will be mixed with other programming, and the return path is independent from the original path, the signal level of the echo is often greater than the signal level of the original. Also, since several factors cause the delay timing (not only telephone circuit response times), the return echo in a broadcast application can easily be more than 250 milliseconds.

Where Is The Problem?

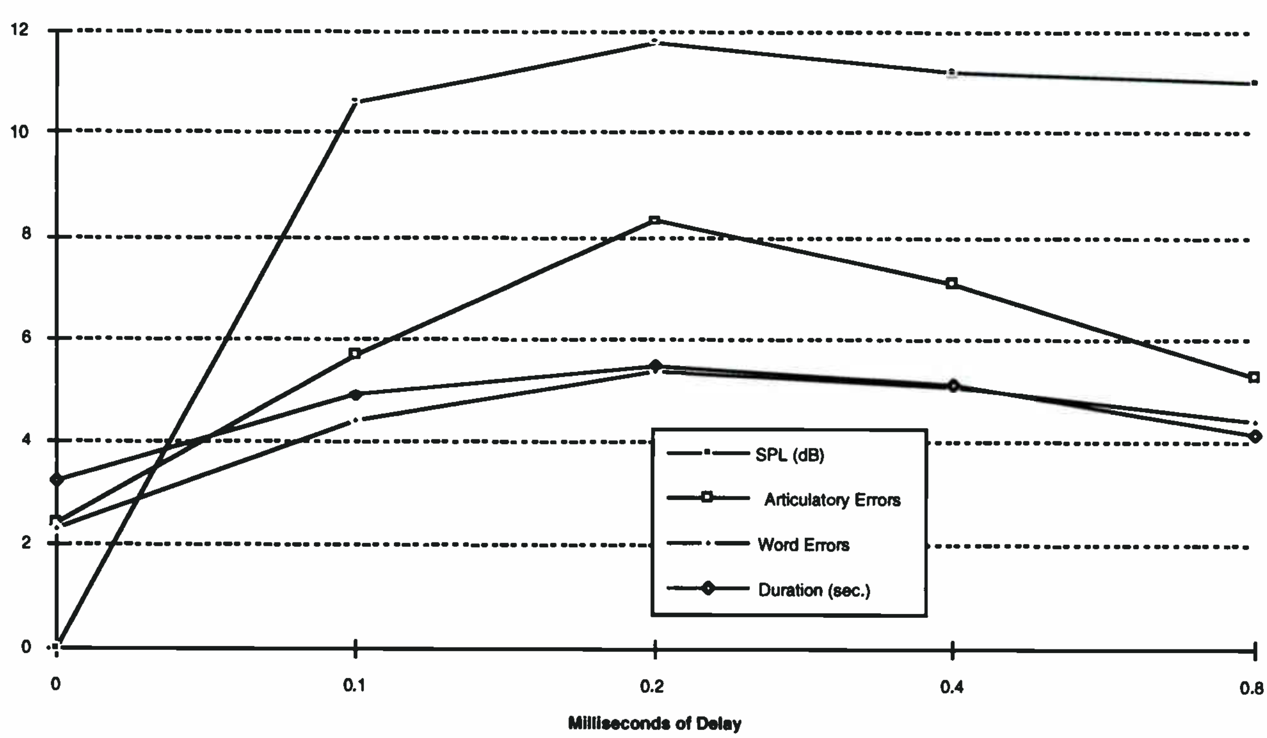
Delays are introduced into a signal path by a variety of factors. Television stations use frame synchronizers that introduce about 30 milliseconds of delay. Radio stations using digital compression can experience up to 50 milliseconds of delay in their audio. A single satellite hop (2-way) adds 250 milliseconds of delay, and ISDN and PCS lines add 50 to 80 milliseconds A Digicipher video encoder adds 650 milliseconds. When a speaker receives a delayed version of his own voice for monitoring, it is very difficult, and sometimes impossible, for him to talk. This is true for field news reporters, sports announcers, radio jocks and news anchors.

Effects Of Delay

Speech and hearing professionals have studied delayed audio feedback since the early fifties. The initial motivation in this research was to determine false responses to hearing tests by the military. Since that time, delayed audio feedback has shown relevance in the treatment of stuttering, autism, and Parkinson's disease.

The early research revealed specific effects that occur when a speaker is exposed to delayed audio feedback. The effects on normal speech include increases in the number of word and sentence structure errors,

Figure 1: Measured effects of delayed audio feedback (Fairbanks)



slowing of the cadence of the speaker, and changes in the loudness of the speaker.

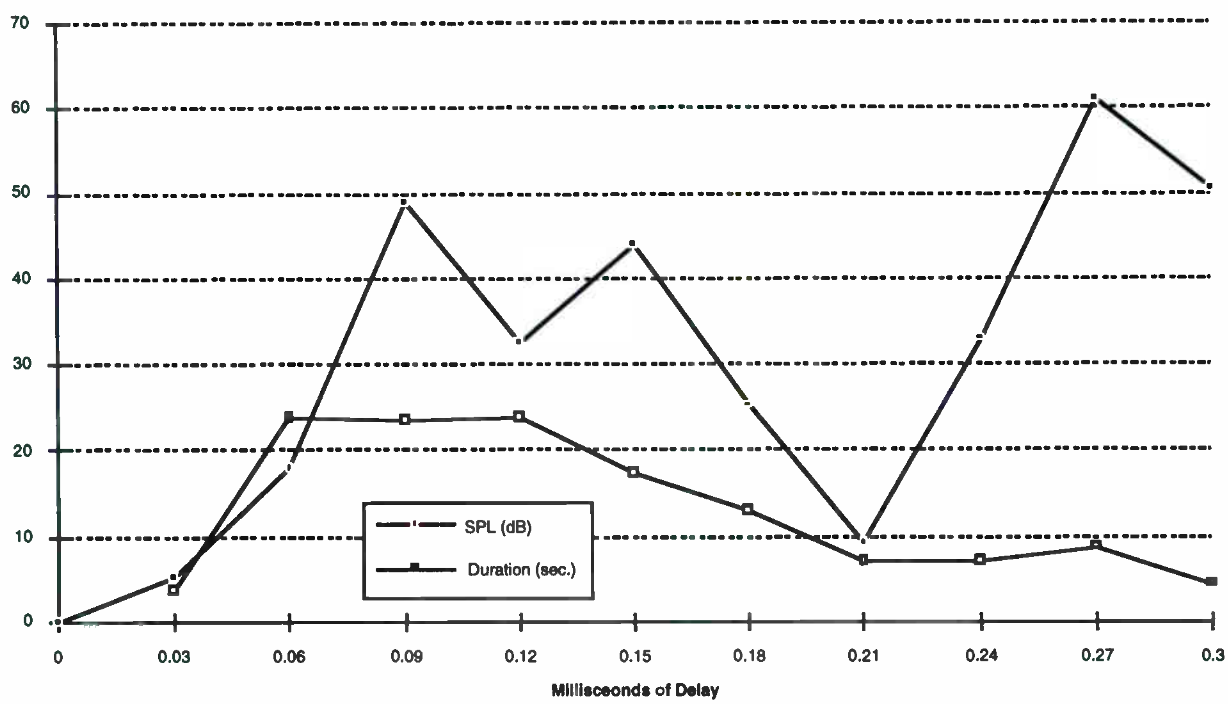
Figure 1 shows the results of tests published by Dr. Grant Fairbanks of the University of Illinois.[1] Using a prepared script and several speakers, articulation errors were counted under delayed audio feedback. They were most numerous at a delay of 200 milliseconds, increasing from 2 errors to 8.3 errors. Word errors increased from 2.3 to 5.4; time duration from 3.24 minutes to 5.52 minutes; loudness (SPL) increased by 11.8 dB.

Test results performed by Dr. C. J. Atkinson of the University of Iowa showed similar results (Figure 2).

These tests confirm what we know from experience: exposure of a speaker to delayed audio feedback causes slurred speech, word errors, and changes in the pace and loudness of speech.

Tests show that speakers exposed to delayed audio feedback experiences a 10% increase in the periodic heart rate, a 9%/18% increase in blood pressure, 68% and 49% rises in plasma concentrations of norepinephrine and epinephrine respectively, and a 25% increased activity of dopamine-beta-hydroxylase in plasma. The conclusion of these results is that delayed audio feedback consistently induces mental stress in normal speakers.[2]

Figure 2: Measured effects of delayed audio feedback (Atkinson)



What The Users Say

Several technologies being use in radio and television broadcasting can interfere with a performer's ability to monitor off-air. These technologies introduce delay into the broadcast signal. A small amount of delay makes the signal sound as if it is going through a tunnel. Larger amounts of delay cause a severe impact on the brain's ability to control clear, concise speech.

Different users report different abilities to react to delayed audio feedback. The following chart is a summary of the responses to recent inquiries about air personalities' ability to listen to delay.

Delay (msec.)	Problem	Reaction
0 - 30	No problem. Easy to monitor.	May hear some tunnel effect
10-40	Problems begin. May be able to train yourself to listen.	Some tunnel effect, some reverb effect, may slow dow speaker's speech
20 - 50	Can't listen.	Brain can't correlate listening and talking; wants to finish hearing before speaking. Speech very choppy.

Replacing A Mix-Minus

In the past, a mix-minus often solved echo problems in live remote broadcasts. Since a voice signal was coming from a specific microphone feed, a separate return feed could be made for one of the speakers by mixing the inverse of his voice with the signal ready to go on-air. The inverse of his voice cancels his voice, so he could then hear the entire program content less his own voice.

The limitations of a mix-minus are that it can only be done back at the studio, not at the remote site, and that it requires a separate feed for each talent. The new delay cancelling technique presented here is the first method that offers broadcasters the ability to perform this cancelling-type function at the remote site. Because delay cancelling is done at each remote site, an indefinite number of users can use the technique to remove their individual voices.

How It Works

In this patent-pending process for cancelling delay, both the local microphone input and the off-air return input are digitized. A DSP search looks for similar patterns shifted in time and level (Figure 3). Upon finding a match, the DSP performs a correlation and the removes the unwanted signal from the off-air return audio. The local microphone audio can then either be mixed into the off-air return audio or left out.

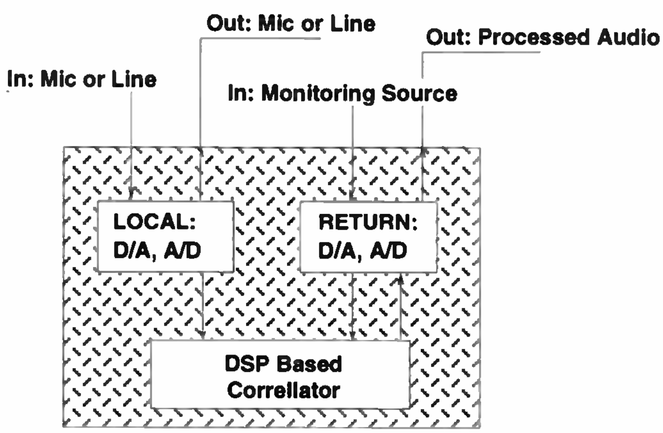


Figure 3: By digitizing the local and return audio signals, they can be compared for echo in a DSP.

VOX Mode

We must prevent the echo from reaching the talent before the cancellation algorithm can cancel it. There is a short interval in which the speaker is on-air, but the cancellation has not yet taken effect. This can be extremely distracting or disconcerting, and we must make sure that the speaker's original delayed voice does not reach him. To do this, we start with a "smart" voice activated switch (VOX). This VOX switches from the off-air return signal to the speaker own voice whenever the speaker activates his microphone. Once the speaker stops talking, the VOX keeps his voice active for an additional interval equal to the estimated delay interval before it switches back. Thus, if the speaker should suddenly be live, he will never hear his own voice echoed. The time delay is short enough to be transparent to anyone but the speaker.

Correlation Times

If we know the approximate delay time, correlating to a delayed signal takes 2-to-3 seconds plus the amount of the delay. An auto-correlation can also be performed which first searches for the delay time. This process adds about three seconds to the total correlation time. Since we know the correlation time in most cases, it is a simple task for the user to enter the anticipated delay time and thus reduce the total time to correlate. Nonvolatile memory should retain this setting whenever power is recycled.

Avocet Instruments has developed a Delay Canceller™ which can correlate the timing of the local and return signals to perform echo suppression. A correlation takes 2-to 3-seconds if the search takes place within a 100 millisecond window. If the search window is not known, the Delay Canceller™ uses an Auto Mode to auto-correlate and find the window first. This takes an additional 2-to-3 seconds. In Auto Mode, a search is made at a reduced sampling bandwidth until a presumed match is made. When enough confidence is achieved that the match is a good one, the 100 millisecond timing window is defined and the normal correlation procedure is followed. (Figure 4).

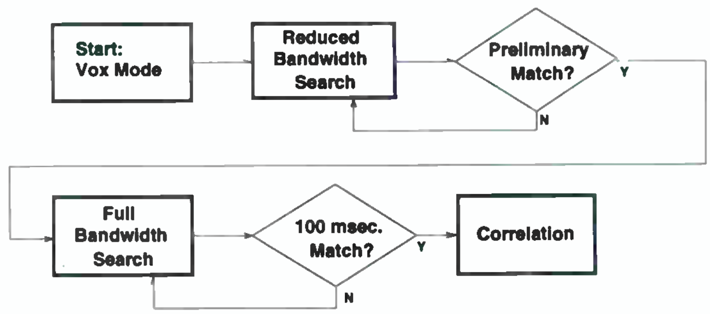


Figure 4: A correlation search can home-in on the exact delay.

Once we determine a correlation, we display the correct correlation setting for future reference. While a specified 100 millisecond timing window can speed up a correlation, choosing an incorrect timing window will result in no correlation; when a timing window is specified, only that window is searched for a correlation. In most practical instances, the user knows the delay timing, and the timing is the same whenever they do a remote.

Voice Processing

In live broadcasts, the audio source can be either at microphone or line level. We must accommodate both in a delay cancelling device.

The DSP algorithm for delay cancelling works very well with linear shifts in time or amplitude, but not with non-linear shifts. Down stream compression and limiting pose a problem since they are both non-linear. It is useful to place a compressor/limiter before the local side of a Delay Canceller™ to minimize voice peaking, and thus minimize processing that might occur later in the equipment chain.

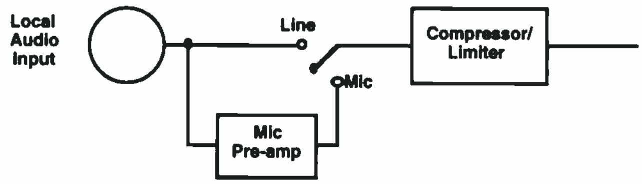


Figure 5: A mic pre-amp and audio processor have useful purposes in delay cancelling

We optimize performance of a delay cancelling device when we scale the local and return audio signals to each other. As long as the time and level differences are linear, we can achieve a good correlation and echo suppression. When the differences are non-linear, we can experience anomalies to the processed audio coming out of the return output. The biggest potential for non-linear variations is compression or limiting occurring after the local loop-through (typically done back at the studio). The Delay Canceller™ provides a gentle compressor/limiter circuit "before" the local input signal digitizer to minimize any compression or limiting that may be required later in the audio chain.

Listen Mode

Some speakers are used to hearing their own voices off-air, and some are not. The latter have gotten used to never listening to their own voices live because their feed has been a mix-minus. A good delay cancelling solution offers a choice of either listening or not.

Masking Tests

Avocet Instruments developed a delay generator and used it to test several factors related to delayed audio feedback. We ran a series of tests to assess the effects of mixing music or another voice signal with the delay-reduced audio. We found that music mixed with the performer's voice provided masking of the partially attenuated echo signal. Another voice, as from a 2-way conversation, also helped to mask the echo.

We believed that assessing the states of the audio path in both directions gives us a key to doing successful delay cancellation. By looking at the content of both the local and off-air return signals, we can use a sequence of cancellation techniques to improved the cases where echo was objectionable, and make it (and all the other cases) listenable.

Avocet Instruments has proven the feasibility of this technology with the introduction of their AV-2000 Delay Canceller™. While the technology is not simple, the state of the art available today, makes it possible,and the capability is now available to broadcast users.

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Computer Networking and Media Management

Wednesday, April 8, 1998
9:00 am -12:00 pm

Chairperson:
Barry Thomas
KCMG, Los Angeles, CA

Information Technology: Supporting Broadcast Media
Sky Kruse
Innerlinx Technologies
Seattle, WA

Acquisition and Management of Digital Assets for the Transitioning Broadcast Facility
Beth Rogozinski
Systrum Media
San Francisco, CA

Technology Options for Information Management in Radio Group Operations
Jeffrey Kimmel
CBSI/Custom Business Systems
Reedsport, OR

***Computer Systems for Radio**
Robert Meuser
WHTZ-FM
Secaucus, NJ

***Automated Control of the DTV Station**
Brad Gilmer
Gilmer & Associates
Atlanta, GA

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*Papers not available at the time of publication

Information Technology: Supporting Broadcast Media

Sky Kruse
Vice-President, Innerlinx Technologies
Seattle, WA

ABSTRACT

This paper considers the impact of information technology on the broadcast industry, with an emphasis on radio markets. Issues related to successful planning and implementation of an Information Technology solution is addressed, covering three primary categories: open systems, immediacy of information and automation.

INTRODUCTION

Efficiency. The mantra of automation. The promise and allure of information technology is clearly defined in the time savings and efficiency it delivers. Whether the purpose is to reduce time needed to go from concept to on-air, improve the success rate of the sales force, or allow sharing of resources across a broadcast group spread amongst several cities, such solutions have become essential to industry success. Information technology has evolved as a "labor-saving device" for the corporate world, a tool to allow greater productivity across the enterprise. Turning these claims and aspirations into reality has proved of varying efficacy over the years, but certain elements are consistently successful. Core success within the broadcast field today seems to draw from three major IT components: open systems, immediacy of information, and automation.

OPEN SYSTEMS

Open system architecture is a relatively recent achievement. Mainframes and minicomputers dominated traditional information resources and were well out of the range of most broadcast groups. Although major organizations could and did use "big iron", it was not an information resource available to the average employee. The last twenty years has seen considerable change in this, as the advent of the personal computer brought information processing to individual users. More recently, widespread deployment of Local Area Networks (LANs) made information transfer a speedy process and standardization of software formats made it useful. And currently, another paradigm shift is occuring as these disparate networks are tied together within broadcast groups in Wide Area Networks (WANs) and to the Internet as well.

Standardization of technology is not a problem exclusive to broadcast, nor any other field, but the need to support a wide variety of audio and video formats is only exacerbated by conflicting standards within the computer industry. Platforms have generally stabilized to the array of Windows platforms (Windows 3.1, Windows 95, and Windows NT) and Macintosh, often with a Unix architecture behind the scenes hidden from the average worker. Standardizing platforms and connecting them mandated that they be able to transfer information effectively, a quandary that evolved office suites, email and groupware. The growth of the Internet put browser technology on the map as a means to present information to anyone regardless of what system they were using, and Java continues to promise software that functions on any machine as well.

There is still considerable diversity of software, and though there is no shortage of different methods of handling information, there is generally an open interchange between different packages. The first examples of this came from word processors, which developed the ability to read file formats other than their own in order to allow greater compatibility with other systems -- and to ensure that users of a rival system had a migration path to their own. Spreadsheets and other common packages followed, soon becoming components of software bundles, and eventually these developed into today's office suites: related products that work together well, if not always seamlessly. Groupware such as Lotus Notes and information-sharing mechanisms such as corporate intranets have defined a software path for collaboration with coworkers. Expandability and interoperability have become necessary components of intranets, hence, recent generations of software have moved to take advantage of such utility.

Wide-scale usage of databases has been another crucial component in bringing information technology from esoteric to utilitarian. Whether the impact is as simple as centralizing information so that multiple people can access the same data at the same time, or as advanced as correlating on-air content for multiple stations with each station's demographic research in order to produce automated direct-to-broadcast multimedia feeds, the time savings from a proper centralization of data can enable startling productivity enhancements. Also worth noting, with a sufficiently advanced WAN and information distribution system, this can be done from a distributed environment as well.

Databases and data manipulation tools are evolving past the traditional limitations of textual and numeric manipulation. Though traditionally databases have been used to store numeric, textual, and financial data, more powerful computer systems have made it feasible to manipulate and retrieve information of other sorts. Informix has perhaps provided the best example of this with its DataBlade technology, enabling targeted informational retrieval for audio and video. As these technologies develop further, information-processing techniques have begun to enable search and delivery of multimedia data. Though not yet affordable for wide-scale use, this technology will become an important tool in the management of broadcast information in the near future. The capability to find a particular sound clip within a hour-long interview by merely requesting the text thereof represents a considerable time savings.

IMMEDIACY OF INFORMATION

The cardinal goal of information technology is simple: deliver all (and only) relevant information to the people who need it as soon as they need it. Given the inherent immediacy of the broadcast industry, bringing news, live events, call-in interactivity, or even simply the latest music, a role clearly exists for a well-tuned information delivery vehicle to aid in this process.

At simplest, digital media storage has made it convenient to deliver canned information with very little preparatory time. The ability to queue a radio show by merely racking several CDs is a substantial convenience, and the ability

No commercial station goes without the feedback provided by ratings services. But other feedback is garnered by time-intensive methods: listener feedback, advertiser feedback, local market feedback (clubs, promotions, and so forth), and market research focus groups. If information can be gathered, indexed appropriately and delivered to the proper people without human intervention, this represents considerable savings of time. For this reason, stations are using web technology to communicate more internally and to the community: they provide means to gather feedback directly into a central database, as well as polling listeners, expressing the station's culture within the community, providing additional promotional opportunities and revenue, and keeping station prominence high in the minds of listeners. Community presence through a web site can keep listeners more in tune with station events and promotions, increases listener retention, and assists information gathering of audience demographics. As web sites are increasingly part of the cost of doing business, some consideration should be given to how they can gather data.

Content distribution can be made more convenient by automating it as well. Many radio stations have content components delivered in broadcast stereo format, such as Digital Courier's commercial delivery system. This is still relatively slow and proves infeasible for delivery of larger or instant content, but such technology is being developed. Even simple content distribution can be a considerable assistance: as an example, incoming faxes can be received by computer and routed either to the email box of the recipients or to the corporate intranet. Likewise, placing audio clips of sponsored events or advertisements on the Internet, so that the advertisers can listen to them without requiring audio tapes to be cut, can reduce production and distribution expenses and time without much effort.

At an extreme end of automation, such products as StarTrak from ADC Labs convert the entire radio station or series of repeater stations to a digital playback format. Any or all components are assembled ahead of time, and audio broadcast is done directly from computer system. Similar technology has been used to deliver sound or video clips from a database in real time using the new "universal servers" from Informix, Oracle, and IBM. Perspectives vary on this, of course: some broadcasters are very tentative about using a substantial digital broadcast base and others are jumping wholeheartedly into this new technology to downsize and gain efficiency.

CONCLUSION

Information does not exist in a void. Information technology cannot, either. Any attempt to revolutionize performance by scrapping an existing process and replacing it will cause great stress, which means that it should be done as seldom as possible. Proper consideration of the desired impact and the means of achieving it should always be the first step in any information technology process. Nearly always, immediate access to information and automation of tasks that do not require human intervention will be at the top of any wish list. Open systems are becoming more powerful and more accepted, and as time passes, with standardization of audio, video, and other media content, open systems architecture will reduce migration costs and speed user training. These technologies in synthesis will yield incredible efficiency gains through better resource management, access to distributed talent, and faster access to information needed.

to ready that same radio show for distribution by burning CDs straight from audio archive and mixing board has decreased the overhead complexity and production time of doing so. Digital information can be much more conveniently transferred from one machine to another: files can be immediately sent from mixing board to on-air studio, without the necessity of creating an intervening tape.

Advances in technology have also made distance much less of a barrier to delivery of this information. High-bandwidth ISDN phone lines allow usable-quality vocals to be transmitted, enabling sound production distributed across studios in different cities. ATM switching or satellite links, though considerably more expensive, enable similar video engineering. Time savings quickly compound when travel times can be reduced from cross-town or cross-country trips to merely a walk down to the production studio. This technology has been used to produce content for multiple stations in different cities from a single studio using on-air talent present only by ISDN connection. Likewise, the ability to collaborate to produce broadcast quality audio or video has made it feasible to decentralize media archives and share them as necessary, although – as has been noted – this requires substantial WAN architecture.

Of course, more widespread means of information transfer have proved to be valuable to the broadcast market as well. Real-time stock quotes, via tickertape or data line, news feeds, and traffic reports are all easily available and represent an important component of the offerings many stations bring to their listeners or viewers. Where originally, teletype machines provided breaking news every couple of hours, the ability to review news in real-time via computer has allowed more timely and often more useful reports, particularly with software selecting certain stories of potential interest.

The ability to transfer information immediately within a station or a broadcast group should also not be overlooked. E-mail has rapidly become a de facto standard for transferring information and small files from one person to another. Groupware and workflow software enforce process flow of vital documents, routing them along the chain of command, and should certainly be considered if information is largely managed by computer. Though tools for distributed media creation and engineering are still nascent, the more-standard suites of productivity tools allow multiple users to collaborate on documents, presentations, and the like. The traditional process of routing memos to multiple individuals for comment and elaboration, then evaluating and rekeying changes, can be streamlined to allow comments from readers or collaboration among employees in far less time than before. There is also significant value in sales automation, with software managing contacts, client history, pricing, and important sales data in order to make the salesperson's job easier and reduce the time necessary to become familiar with a new account.

AUTOMATION

Although this goal is often elusive in practice, the implementation of a comprehensive information technology solution should have the result of simplifying processes. Increased automation is often used to deliver this result, relegating tasks that do not require human intervention to electronic agents. For those looking to downsize or enable an existing staff to do a larger job, the investment in these resources may represent a path to success.

ACQUISITION AND MANAGEMENT OF DIGITAL ASSETS FOR THE TRANSITIONING BROADCAST FACILITY

Beth Rogozinski, Systrum Media
Technical Contributions and Review – Paul Stevens, Silicon Graphics
Systrum Media
1515 Pine Street, #3
San Francisco, CA 94109

ABSTRACT

Television stations and all entertainment industries are becoming more reliant upon digital tools for the acquisition, creation and management of media assets. This trend will continue and rapidly expand within the broadcasting realm as broadcasters around the world begin to transition their facilities to all digital workplaces in preparation for all digital transmission. While the future of "all digital" is still quite some time away for most broadcasters, it is inevitable also, that most, if not all, are in need of digital solutions to replace aging analog equipment. In addition, all broadcasters are concerned with and interested in beginning their transition to digital facilities in order to maintain an edge in this increasingly competitive industry. As broadcasters face the task of upgrading their facilities, they are all concerned with similar questions: How will digital equipment work with existing equipment, both analog and digital, that doesn't have to be replaced yet? How might new digital solutions increase productivity in broadcast stations, thereby cutting costs? How might digital systems help to increase and retain the value of media assets? How will the digital solutions purchased today prepare broadcast stations for the future? This paper addresses the questions facing broadcasters wishing to upgrade their facilities with digital equipment, both from an engineering and logistical perspective, as well as from a financial and business perspective.

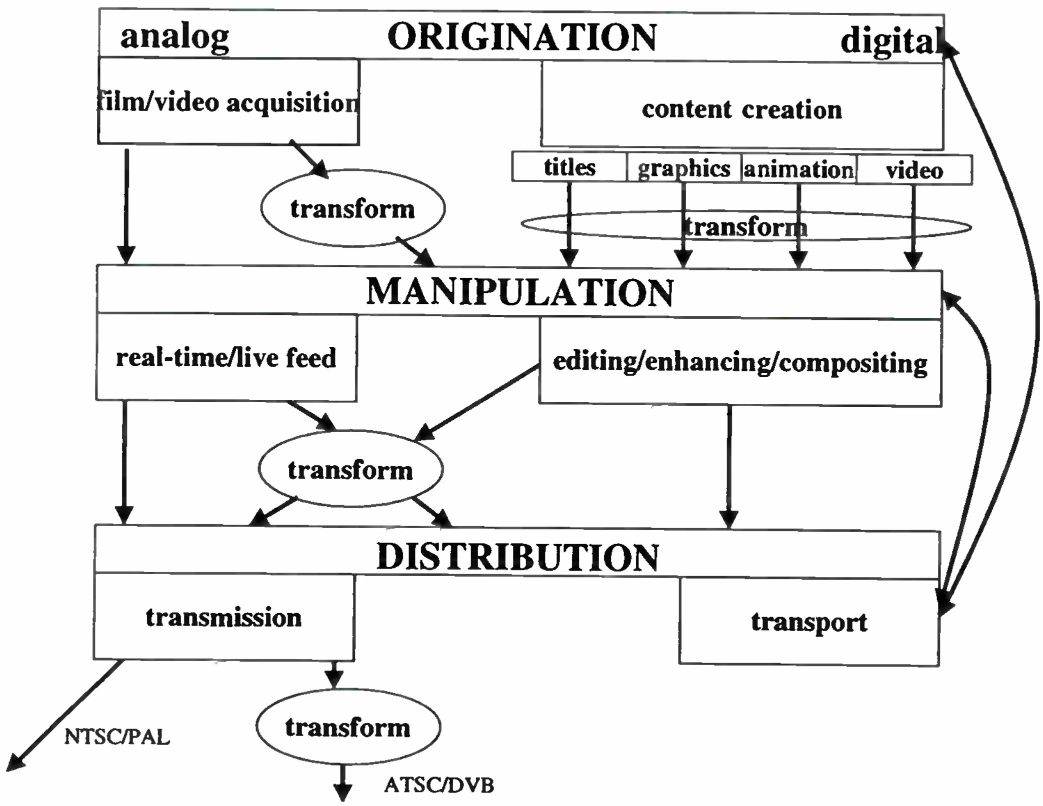
THE BUSINESS OF TELEVISION

Most of the world's television stations are for-profit businesses which base their business models on the advertising fee system. While upgrading a television station to all digital transmission might, in the long run, generate new income streams, immediate income from advertising fees will not likely increase dramatically. (The historical example of the introduction of color attests to this hypothesis. Advertisers did not dramatically increase the amount of on-air fees they paid to television stations for the privilege of color.) The lack of an immediate return on this significant infrastructure investment will lead broadcasters to upgrade their facilities in various stages to amortize the associated costs over many quarters or even years. This situation is made additionally troublesome as digital equipment simultaneously improves and decreases in cost on an average six month cycle. Broadcasters are therefore forced to consider and plan the entire upgrade of their stations now, envisioning what the final result will be in three to five years. Plans for immediate purchases must be based on traditional IT and business considerations, such as cost and features, but extensive care must also be taken in consideration of how well today's component purchases will serve as cornerstone and/or integrated systems into future scenarios. Flexible, scalable, open, maintainable - these are the descriptors of the digital systems that are the provident, economical and sound business purchases for broadcast stations as they transition into the digital future.

Television Station Workflow Dynamic

Television stations operate as dynamic workflow environments in which content is brought in via many means and methods; manipulated, edited and enhanced by many people, sometimes simultaneously; then finally amassed, organized and assembled, for final transmission. Digital solutions will not change this overall workflow edifice, but when properly structured, new digital solutions facilitate these existing processes, increase productivity, and increase the residual value of media assets. The return on investment that broadcasters get immediately for installing digital solutions are these increases in productivity and media asset value. Over the long term, open and scalable digital equipment offers broadcasters continued access to new digital tools which further facilitate creation and distribution processes, and a base environment that can grow and change with the facilities changing needs.

Diagram 1: Broadcast Station Workflow



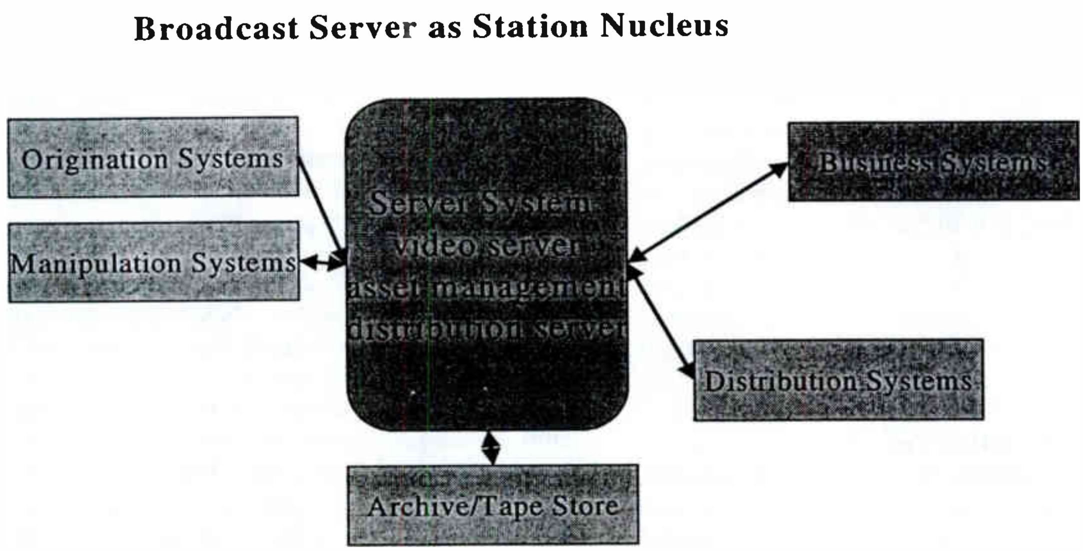
Current Broadcast Station Systems

The content origination element of broadcast station workflow involves creating and collecting recorded video footage, voice over narratives, music and other audio enhancements, and static and motive graphical elements. In most broadcast stations today, all elements of content origination are analog, save for the creation of graphical elements, which is typically done digitally. The manipulation element of broadcast station workflow involves editing, enhancing, and compositing the four types of content assets to a complete ensemble piece. Manipulation of media assets needs to occur in real time for immediate playout to air as well as in the more standard off-line, post-production mode. While some affiliate facilities and LPTV stations still do all manipulation on analog systems, most stations use some digital solutions in their manipulation processes. In the distribution and transmission element of workflow, composited media content is either distributed to another internal workflow environment, transported to another facility, or transmitted as playout to air. Transportation and distribution of processed content internally and externally is still mostly accomplished manually and transmission is via analog methods.

DIGITAL BROADCAST STATIONS

The cornerstone system in a transitioning and even a fully digital broadcast environment, will be the server system. In broadcast facilities, the digital server system will interface with all aspects of workflow, from content creation, to manipulation, to transport and eventually - direct to air transmission. The server systems should also link production systems with business systems such as internal communications, billing and revenue systems. While video servers will introduce a new level of complexity to the television station environment, they also promise significant operational improvements and increased productivity. In addition, digital servers can increase, extend and maintain the value of created media assets by making them easier to find, access, work with and by retaining original material integrity after even multiple uses. While video servers must have computational might, cost effective servers for broadcast stations must focus as well on bus speed, I/O bandwidth, scalability, flexibility and interoperability.

Diagram 2: Broadcast Server as Station Nucleus



An adaptable broadcast server provides broadcasters with a powerful, flexible and scalable aggregation of tools that can be customized and configured to meet the needs of every growing broadcast station. The nucleus of the facility, the video server must be an open, highly-scalable environment that supports a wide and growing range of software applications, HDTV-capable graphics, high-speed networking, RAID storage, and additional video peripherals. In addition, the system must be able to combine video peripherals for digital asset management and media distribution. ROI insurance for the future of this type of system would be its ability to handle multiple channels of uncompressed ITR-U 601 video I/O, dozens of channels of compressed video and audio I/O, all relevant disk I/O standards including SCSI, HiPPI, Serial HiPPI, and FibreChannel, and LAN and WAN networking standard protocol systems. With a scalable, flexible and integratable digital video server system in place, the traditional workflow processes of broadcast stations can be supported while component systems are individually transitioned to digital.

From a close examination of today's average broadcast station, it is likely that the manipulation phase systems will be the first systems to be all digital, while origination phase systems will likely be mixed for sometime. Distribution systems will also be mixed for quite sometime as digital distribution will begin immediately internally, yet external distribution and transmission will be both analog and digital until the vast majority of consumers have digital receiving devices.

DIGITAL CONTENT ACQUISITION

Due to the transformative nature of the shift to digital broadcasting, economics require broadcasters to phase in digital technology that will be compatible with and befitting analog means of gathering content. While, as stated, creation of most graphic elements is currently a digital process, the most prevalent media asset gathered is still analog video and most stored assets are either analog video or film. While some stations will quickly ramp up to the sole use of digital cameras and begin digitizing all media assets stores, most stations will require systems on site that can quickly and easily digitize media assets. This conversion will be required for mixing analog media with digital elements, for digital editing and enhancing, to facilitate collaborative working environments and digital distribution and transmission. Analog to digital conversion technology is already available and can, to date, support uncompressed digital media at film resolution. Digital cameras are also currently available, affording broadcasters manifold transition paths to the digital morrow.

Conversion and Workflow Concerns

Among the main concerns of broadcast engineers and production managers when transitioning to digital systems is the perceived amount of time that is required to digitize or "load" analog footage into the digital systems. While this is a viable concern, if we examine the workflow time of traditional means of creating an EDL or rough edit and compare it to working with digital video, we can clearly see that the time required is approximately the same and that digital video offers more reliability, flexibility and potential quality. In addition, automated video cataloging systems now available, such as the Virage Video Cataloger, not only dramatically cut down on the amount of time required for logging incoming footage, but they effectively database the assets, thereby increasing their potential value for versioning and archiving for future use.

Analog EDL/Rough Edit Process in a News Gathering Scenario:

1. Tape comes back from field and gets delivered to editor. Reporter goes off to finish writing VO.
2. Editor fast rewinds the tape making mental notes regarding which scenes work and which scenes clearly do not work. Editor stops and plays at normal speed all interview and reporter talking head footage to check for usability.
3. Using only the "by memory" list of shots, the editor jogs the tape back and forth to create an EDL and rough of the final.
4. Reporter returns with final VO ready, record VO and final coalition of media assets is made in the edit bay.

Digital EDL/Rough Edit Process in a News Gathering Scenario, using the Virage Video Cataloger:

1. Tape comes back from the field and is installed at digitizing station. Reporter goes off to finish writing VO.
2. Footage is digitized and simultaneously logged. A scene change template database is created with thumbnail images and timecode.
3. EDL can be made using the visual and timecode reference data and EDL for various versions can be made consecutively.
4. Reporter records versions of the VO directly into digital format and final coalition is made immediately with all versions ready to go out to air.

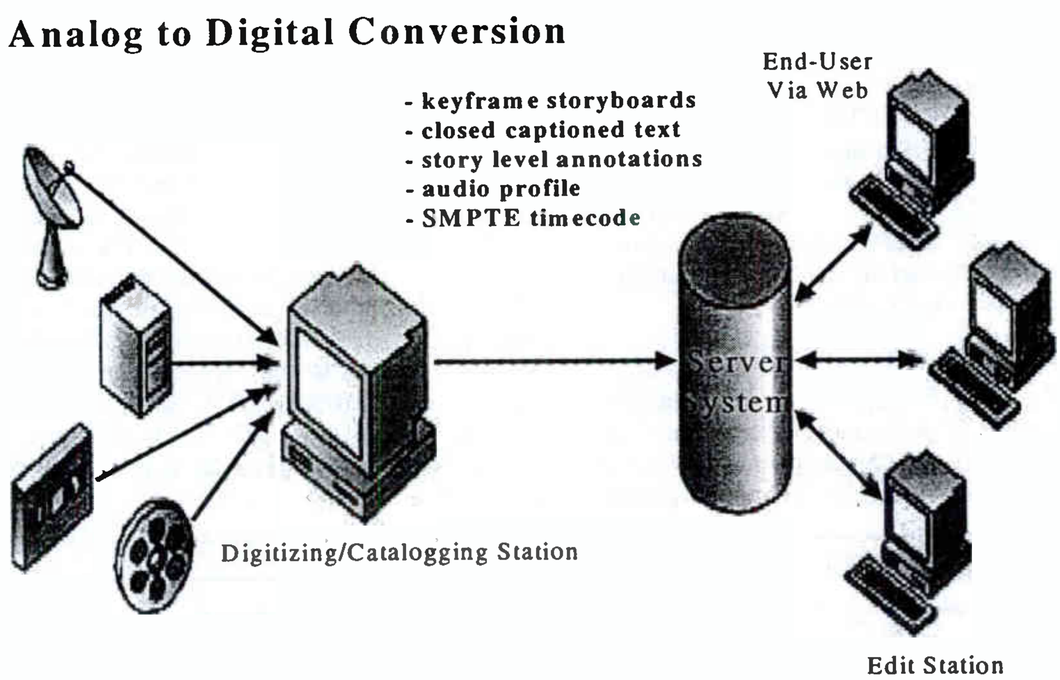
The two processes described here can take approximately the same time, though with the digital system, there is far less valuable staff time required. In addition to cutting the amount of labor required to prepare the EDL, the digital system offers a definitive record of scene changes and timecode, instead of relying on the editor's memory. The catalog of scene changes can also quickly and easily be made in to a number of EDL's for different versions to be used throughout the day and/or for localization. The digitized content, ready to be edited and composited with audio, titles and other graphics, can also simultaneously be distributed via the web to internal end-users for further use and/or it can be published to a web site for sale or distribution outside of the station.

Analog to Digital Conversion Devices

The most common analog to digital conversion systems are telecine, datacine and film scanning. As most broadcast facilities will require only video transfer on a regular basis, it is likely that most will opt to purchase a telecine device and forego the more expensive datacine and film scanning devices. (This type of digitization will likely be outsourced to post-house facilities and new facilities that are forming to meet this increasing need.) Regardless of the resolution of the system that broadcasters opt for, key evaluation concerns for analog to digital conversion systems are centrally involved in the computer system specification. These include: bandwidth capabilities of the system I/O boards and

cards; internal throughput and processing power of the system; the availability of appropriate and powerful software solutions; and the ability of the system to interface with all other production systems.
Telecine systems, such as those from Kodak and Philips, which interface with Silicon Graphics desktop systems O2 and/or OCTANE, as well as dedicated, internal video capture and compression boards, afford broadcast stations the ability to quickly and cleanly digitize media footage, create a log of the incoming footage, quickly and easily network the digital video to content creation seats for amalgamation and DVE systems for editing and enhancement. Installing this type of system is also future proof. If and when a broadcast station upgrades their facilities to all digital cameras, the CPU and networking components of the digitizing system can be converted to either additional DVE or digital content creation seats.

Diagram 3: Analog to Digital Conversion Workflow



Direct to Digital Acquisition
As most video footage generated by affiliate stations is remotely captured news and weather footage, it seems likely that the first direct to digital acquisition systems will be remote capture/field cameras. These digital cameras must be light, durable and able to hold large stores of content. Additionally, these systems must also easily interface with in-house studio cameras, digital content creation workstations, and media asset servers.

Networking digital camera systems to video servers requires the computer systems have real-time capabilities and that the computer system have as many

There are a growing number of digital cameras available for broadcasters to evaluate based on size and weight of camera equipment, size and resolution of the CCD, and price. Digital cameras interfacing with computer manipulation and management systems is a fairly straight forward proposition that concerns the broadcaster with two primary issues: networking connections/throughput and compression compliance between systems.

I/O pathways as possible. Current trends are to input/output digital video through the PCI bus (the Peripheral Component Interface). While the latest advancement in PCI technology, the PCI 64 bus, has a theoretical bandwidth of 266MBps, it has typically been limited to three to four slots in the computer. Most broadcasters will require systems that can simultaneously handle I/O of 20+ streams of digital data and while it is possible for PCI cards to be stacked to this level, the result would likely be bus congestion into and out of the central processing or storage systems-even with a dedicated bus manager. By using multiple systems, broadcasters can skirt the bus congestion issues, but this type of system will quickly fall short for systems beyond 20 streams. For moving more streams, there currently exists a number of developing proprietary systems that are effective for increasing the possible number of I/O ports. These "switched-fabric" systems are not scaleable however, nor can they handle data rates sufficient to move large-scale files such as those associated with HDTV. Additionally, I/O bandwidth requirements of media needs to be addressed by all components of the computer system. High I/O capacity of a single interface is insufficient if the internal I/O bandwidth or CPU processing of the system can not support the required sustained and/or concurrent requirements of the overall system.

While many groups are currently working on various solution sets for I/O of large video and data files, broadcasters need to ascertain the scalability and flexibility of any and all solutions purchased. Large scale solutions tailored to meet the current and growing needs of broadcasters, while initially appearing more costly, can actually result in significant savings as they can be adapted over the years as opposed to having to be completely replaced. A recent development in the I/O arena is the XIO bus which has a theoretical throughput of 1.6GBps and, in the high end server systems such as the Origin 2000, can scale up to 24 simultaneous streams in and out. This level of throughput means that broadcasters can push 24 channels of 601 video, or 2 channels of HDTV, across the systems at the same time. For even larger systems, beyond 20 streams of 601 or beyond a few streams of HDTV, the capabilities of even the most advanced I/O buses available today will be reached. In this range it is important to use a video server architecture that does not rely on a single backplane for memory access nor a single I/O bus. The Origin 2000 line is one example of a system which employs a Non-Uniform Memory Architecture (NUMA) and CPU and I/O node boards which can be expanded in an almost unlimited way. NUMA architectures can scale to 100's of processors while maintaining a single address space. These types of system architectures allow for the configuration of very large servers which can scale up for handling large numbers of media streams of very high media rates - such as 720I and 1080I H/DTV. A NUMA architecture coupled with independent CPU and I/O nodes eliminates any single point as a bottleneck for I/O bandwidth.

Chart 1: I/O Bus Comparisons

Card/Board	PCI - 32	PCI - 64	XIO
Theoretical Throughput	133MB/s	266MB/s	1.6GB/s

ASSET MANAGEMENT SYSTEMS

For broadcasters, asset management is an essential business requirement. The bulk of expenses for broadcasters revolve around the creation, storage of and access to media assets. In the analog world, the "storage of and access to" parts of this equation are largely inadequate and leave broadcasters unable to fully take advantage of the assets they have accumulated. Transitioning to digital asset management creates a new systemization that requires internal training and set up time, but the benefits of digital asset management are countless and the return on investment in terms of work hours saved and the increase and retention of asset value will be seen almost immediately.

Current Systems of Asset Management

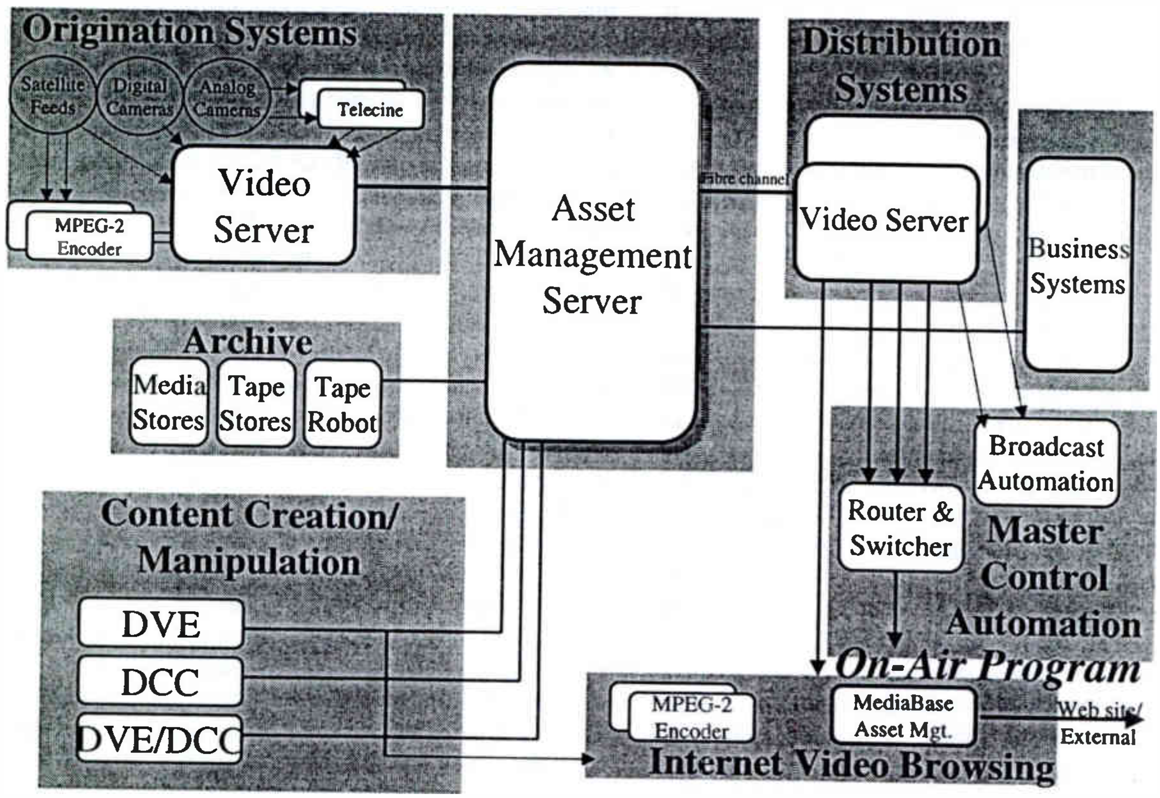
Asset management in most broadcast stations today consists of a number of ways and places where different media assets and files are kept. Digitally created graphics and animations will be kept on individual workstations and/or backed up onto some form of removable storage media, such as magneto-optical disks or DAT tape (both of which have definite drawbacks and problems inherent in their designs). Video footage, audio clips and voice overs, text documents and print images will be kept in one or more asset libraries, often placing certain assets quite a distance from the production work areas. All of these assets are stored in their original - read non-compatible and time degenerating - formats, that, when and if found, have to be converted and cleaned before they can be used.

Digital Asset Management

A digital asset management system allows broadcasters to combine all of the station's assets onto a central storage facility with various levels of access via multiple access paths. The effective digital asset management system handles the input of all types of digital assets, accommodates various formats of digital media, (i.e. GIF, JPEG, TIFF, EPS, MPEG, Quicktime) and various resolutions for input and output, (i.e., film resolution, HDTV, NTSC, PAL, thumbnails, et cetera). In addition to accommodating all formats and resolutions, to be truly effective, the system must support format conversions, file type recognition, creation of assets with multiple resolutions, as well as the capacity to group together all formats and resolutions of the same asset. An asset management server system with these capabilities increases productivity by decreasing the amount of time required to find a given asset, decreasing the time to convert an asset to appropriate format and resolution, and facilitating versioning of assets for playout to air. This type of system will also help to retain and even increase the value of stored media assets by making them more accessible, easier to find and use. A ready-to-use video clip that can be found in 30 seconds by the person who needs it is far more valuable than one that requires research by the producer or editor, searching shelves and stores by support staff and preparation by production staff. Digital asset management also adds value by making assets simultaneously available to many users and making them trackable. The continuing problem of "lost" or between shelves assets is completely removed with a digital system. Additionally, there is no danger of generation loss or compromising the original asset in a digital system as the true original is only ever virtually accessed and manipulated. Saved versions of each manipulation and use are always associated with the original asset set, thereby adding further value to the media store.

A digital asset management system requires large amounts of digital storage capacity at on-line or near-line status, with secure back up and redundancy built in. Depending on the size of the facility being served, the asset management system can also serve as a playout to air server, but in most larger facilities, these two server systems will more likely be separate, with separate RAID stores, but connected via high speed networking lines. The asset management system should be future-proof with faculties to create complex object hierarchies, have seamless publishing and subscribing procedures, track the usage of assets and send out bulletins to end-users. The two other main systems requirements for effective asset management systems are the interoperability of the system with other facility systems and the creation of consistent metadata fields. As the asset management system must be able to work with all other information systems in the broadcast facility, including business systems, the environment must run on all existing, installed platforms and integrate with other software infrastructures such as different types of file systems, other business systems, content creation and manipulation systems, etc.

Digital Asset Management Workflow



Content and Metadata

The main principle underlying the management of assets is the separation of metadata and content. An asset has contents and metadata - data about the asset. A video clip is an example of contents. The information that the image is DVCPro25 file, V1, 60 seconds, is an example of metadata. Metadata is stored typically in a database system (RDBMS - relational database management system, ORDBMS, object relational database management system, or even an OODBMS, object-oriented database management system). Content is typically stored directly in a file system, but these file systems may be spread out across an installation's network. Furthermore, the file systems may be tuned for the type of content that they store. In some cases, content may need to be stored in application specific formats to enable real-time streaming, recording or automated playback. The asset management system must facilitate the use of a wide variety of content stores as well as maintain the association of metadata and content. While most file servers and object-oriented relational databases currently perform search and retrieve functions on metadata, a digital asset management file system must also use metadata to retrieve assets, open appropriate applications by icon clicking, create a summary statement of media that is in the bank, and be "intelligent" enough to automatically link and code new versions of an asset with appropriate information. As the exchange of digital assets and their metatdata becomes more widespread the need for common metadata definitions and wrapper formats is becoming critical. There are several groups and organizations working to establish standards in this area. The Metadata Coalition (http://207.33.3.206/welcome.html) has been pursuing the standardization of metadata for enterprise, and government science and defense systems. The European Broadcasting Union (EBU) and SMPTE have formed a task force for the "Harmonization of Standards for the Exchange of Program Material as Bit Streams" (http://www.ebu.ch/pmc_es_tf.html and http://www.smpte.org/engr/ebumeet1.html) to provide recommendations for metadata definitions and wrapper formats pertinent to digital assets in the broadcast environment. In addition, the SMPTE/EBU task force is studying file transfer, asset streaming, compression and systems issues as well. To date the metadata standards committees have received full cooperation from a wide range of concerned parties (broadcasters, content creators, software and hardware vendors) to ensure that standards are set and adhered to within the realm of digital media asset management.

Asset Management Environments

As asset management will continue to grow as a central business concern for broadcasters, the asset management system that they choose must be open, customizable and scaleable to meet current and growing needs. In pursuit of such an environment, four years ago, Silicon Graphics began the development of StudioCentral, an asset management framework, that can be likened to a "software bus". Connected to this bus are industrial strength databases such as Oracle and Informix, and content storage servers such as RAID and HSM systems. The open architecture of the system provides "plug-in" access to many content creation applications (such as Avid, Adobe, etc.) and asset and workflow management applications (such as Bulldog). The advantage of the "software bus" approach is that components can be added to the bus as the system scales. Providing an open framework that supports content creation, management, and delivery of digital assets, this environment allows content creation tools, asset management applications, databases, and storage devices to work together through a common interface. The StudioCentral environment consists of three main layers, the Applications Layer, the Foundation Layer and the Asset Bank Layer. Developed in close association with entertainment customers needs and desires, this layered environment provides an industry perspective on the digital media asset management requirements for the entertainment industries.

The Application Layer contains tools and applications developed by Silicon Graphics' software partners as well as 3rd party VAR's. Tools for cataloging, archiving, browsing as well as plug-ins that enable content
creation applications (such as editing, cel animations and compositing applications) make up this layer. Applications (and plug-ins) interface with StudioCentral using various means such as native C++ libraries, Java, Perl or HTML templates.

The Foundation Layer provides a warehouse for metadata, content and workflow management. In the Foundation Layer, developers can create objects (such as "voice over", "video", "script") that contain metadata for the assets. With customized objects in place, developers then perform operations on the warehouse repository to store, retrieve and query the metadata and the content on the newly standardized metadata object attributes. Creation of these objects are easily accomplished using the set of "data-modeling" capabilities built into the package. In addition, the Foundation Layer also provides for content management and tracking, event notification for enabling work-flow routing and project management, plug-in support for connecting with legacy databases and other applications, and access control and Java libraries for accessing this repository from any platform.

The Asset Bank Layer consists of a set of customizable and adaptable metadata and content stores for caching a digital asset's metadata and content. Metadata stores are implemented using RDBMS (such as Oracle and Informix OnLineServer) or ORDBS (such as Informix Universal Server), that support indexing and query features. Content stores are implemented using file system volumes in various content servers, possibly in different machines on the network.

INTEROPERABILITY OF DIGITAL AND ANALOG ASSET MANAGEMENT AND STORAGE

While digital asset management is now a viable and affordable alternative to tape storage, most stations will retain analog tape and off-line systems for storage and archiving. The prime considerations for integrated server/tape systems is that the asset management systems can record and track analog and/or off-line digital assets and that automation systems can interface with the server. A key component of digital asset management environments, is that they track all types of assets and can be programmed to read bar code data for checking off-line assets in and out so that assets don't get "lost" when they are pulled out for production uses. The systems also must have capabilities to communicate with hardware control devices, such as the Avalon Archive Manager and the Emass hardware control peripheral made by Amass.

All broadcast facilities today are beginning their evaluations of digital solutions. Integral to their success in the pursuit of a smooth transition to a digital broadcast facility is the interoperability of digital and analog equipment, new and legacy databases, and traditional and modern methods for acquiring, manipulating and distributing media. The business of television is one in which content is produced to inform, to educate, to entertain the masses. The industry of television is a substantial and successful one that has refined the process of information dissemination. As television transitions to its use of new digital solutions, it is critical that the analog systems that work, that will continue to work, are regarded as integral components of the immediate future. All digital solutions must conform to the sound idea of interoperability with analog systems. While it is de rigeur for digital solutions to be scalable and upgradable, they also must be optimized for today by being able to take advantage of the capacity of existing analog systems. As the amount and level of much digital equipment available today is insufficient to replace analog systems, analog systems must be integrated or the show will not go on. As the cost of much digital equipment is exceptional, analog systems must be integrated or the business will not go on.

CONCLUSION

While transitioning to become a digital studio for most broadcast stations will be a multi-year process, planning for the all digital station should begin before new digital equipment is installed to ensure that each component system will work within the final, ultimate digital otential.

facility. Broadcast stations should anticipate being in digital transition phase for at least 3-5 years and systems purchased today must be able to withstand the future needs five years out, including the anticipated need for HDTV. All new digital systems that are put in place must be able to seamlessly and easily interface with the remaining analog systems as well as being capable of scaling to meet future needs. The key to a successful transition to a digital facility in both business and technical terms is to transition first the aging analog equipment that

needs replacing in any event and to create a strong foundation on which the digital facility can grow and flourish. The foundation of all transitioning and ultimate digital broadcast facilities will be the server system running a comprehensive asset management application. The asset management server system will act as I-beam support for the digital facility and will promote communication between all business and production systems, and provide for the free transfer of information and assets in a seamless,

expeditious manner. The central server systems and all component digital solutions must increase work flow productivity, media asset value, and revenue p

Technology Options for Information Management in Radio Group Operations

Jeffrey C. Kimmel
CBSI/Custom Business Systems, Inc.
Reedsport, OR

Abstract

The challenge that faces today's radio operator is how to handle the incredible increase in information that has resulted from the rapid growth of broadcast groups. More effective ways of processing as well as analyzing this data must be found in order to make these large business entities manageable on a day-to-day basis.

This paper outlines a blueprint for radio stations to effectively manage the information flow from the point at which an order is generated by the salesperson, through the traffic phase, through billing, and finally to managerial analysis of both historical and future data.

Today's Information Flow

An order for broadcast advertising typically begins with the salesperson's preparation of a proposal for a prospective client. This might be a manual process whereby the salesperson's handwritten notes are prepared for a more formal presentation by a sales assistant, or the salesperson may directly prepare the proposal on a PC. After the client reviews and accepts the proposal, a contract is then created through similar means for the client's signature and the station sales manager's approval. The contract then typically goes to the traffic department, where the data is re-keyed a third time, into the station's computerized scheduling system. Scheduled spots appear on the program log, which must subsequently be reconciled with the programming as-aired for verification that the spot actually ran as scheduled; in most stations this is still a manual procedure. Invoices are then prepared reflecting the air times and spots; this takes the data from the traffic to the accounting/billing department. Affidavits or other paperwork directly related to the spots scheduled may also be required a this stage. And, finally, management will want to review the data generated by this contract as part of their sales or fiscal analysis. If the accounting or traffic

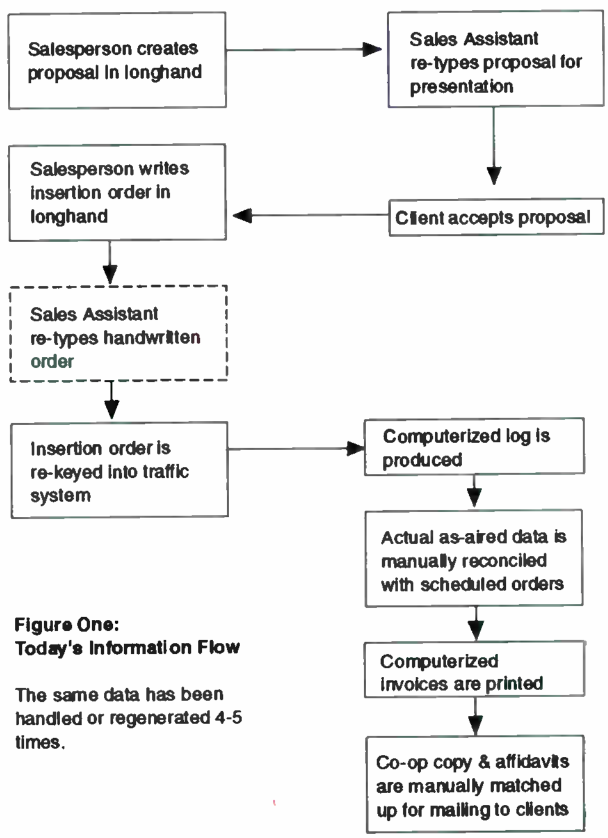


Figure One: Today's Information Flow

The same data has been handled or regenerated 4-5 times.

systems cannot generate these top-level reports, this may well involve further manipulation of the data as figures are entered in a spreadsheet or database.

As we can see (Figure 1), this process currently involves many individuals repeatedly handling and re-keying the same data.

There are some clear inefficiencies and perils in this kind of repetitive data handling. Each time the data passes manually from one individual or department to another, the opportunity for error is introduced – the station is, essentially, "playing telephone" with its revenue stream. The man-hours involved in repeatedly handling the same data also multiply significantly when re-keying or working frequently between manual notes and a PC-based system plays a role in the process.

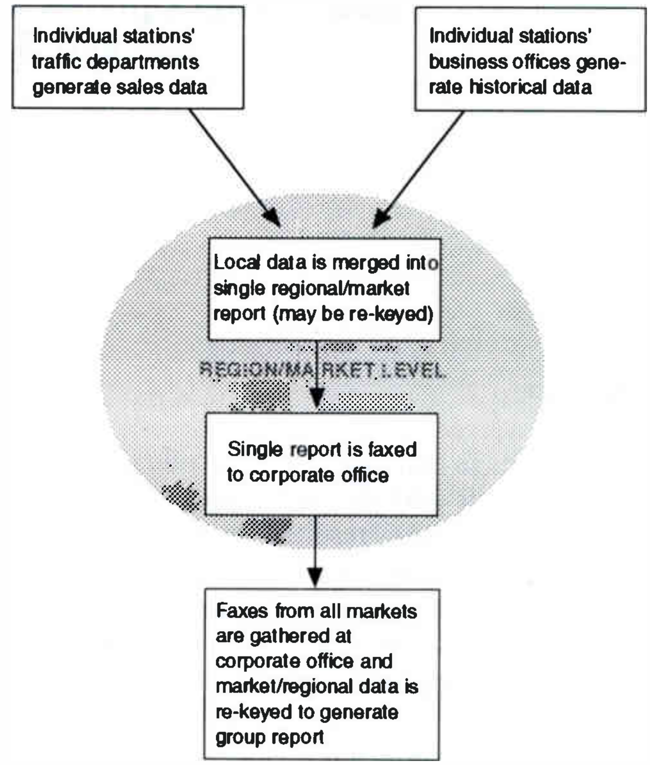


Figure Two:
Group Information Flow

Manually adding group reporting to the station's existing workflow means several additional layers of data handling & regeneration.

The Plot Thickens

The emerging mega-groups in radio broadcasting add a further layer of complexity to this already challenging information flow. Each group will have its own management strategy, but whether that strategy involves stations reporting to market or regional centers, or directly to a headquarters location, group management will inevitably want to see some sort of financial and sales data from each station on a regular basis. This can add at least one, and probably more, layers of data handling – again a mixture of manual and computerized – as the data from the source station makes its way to the regional and corporate levels (Figure 2). In addition, the formats for presenting the information may vary between stations, so that at the regional or group level the data is not being simply re-keyed, but also manipulated to fit a different template. As with the information flow within the station, each transition to a new department or individual introduces the potential for errors and adds significant time to the process.

Demonstrating the Solution

There are many environments in which the computer has enabled us to both speed our communications and make them more accurate – electronic mail, computerized financial processing, and so forth, work to automate what were formerly manual systems involving significant time and repetitive handling of the same data. When a station applies that computing power to the process of proposing, entering, verifying, and then invoicing the sale of a commercial spot, as well as evaluating the financial and sales results afterward, the improvement in the time and accuracy of the workflow can be dramatic. The potential gains can be illustrated with a detailed look at each of these areas:

- Proposal creation and presentation
- Sales management approval
- Electronic transfer
- Traffic and Billing
- Subsequent management reporting

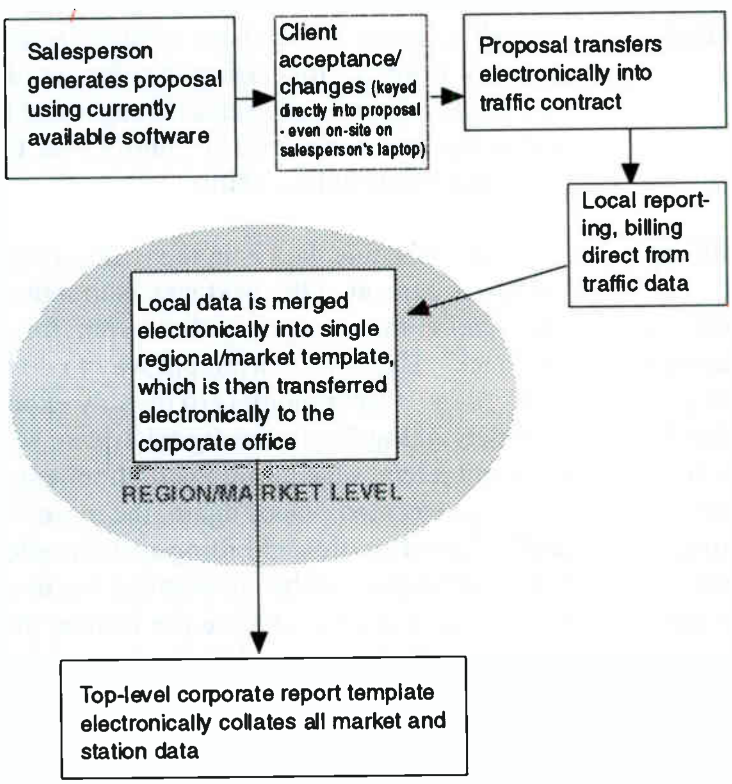


Figure Three:
A New Model for Information Flow

This scenario conveys information electronically and seamlessly from the salesperson's laptop through traffic, billing and station/market/corporate reporting. Manual data handling occurs only at proposal/acceptance.

Proposal Creation and Presentation

Those of us who have worked in or worked closely with a radio sales operation know that this has been one of the last areas of the station to be computerized. And, indeed, personal contract with clients and telephone follow-up continue to be two key components of this demanding position. At the same time, however, computerizing a radio station's sales force – particularly if it is one which is spread out geographically – gains significant benefits in accuracy and communication with the office staff, while time spent on proposal generation is greatly reduced in favor of actual sales calls. Let's illustrate one possible scenario for how currently available software, running in a Windows environment; can add efficiency and speed to this area.

Sales Management Approval

Now that our sales force is PC-literate and capable of networked communication with sales management, whether within the office facilities or remotely via modem, we need to address the key step of sales management approval of the contract the field sales rep has negotiated with the client. Here, again, currently available software provides a means whereby electronic transfer of information allows the Sales Manager to efficiently and quickly review and approve contracts as well as communicate them to traffic.

This transfer involves the creation of a "software bridge" between existing proposal writing software and existing traffic and billing software. This bridge technique minimizes the need for new staff training: both the salesperson and the traffic operator are working in familiar media, only using the link program to transfer data. Electronic transfer ensures that there is no discrepancy between what was presented to the client and the actual data entered in traffic. Further labor savings and improvements in accuracy are possible by creating a single, multi-station proposal for a given client and then authomatically converting it to individual insertion orders for each station as the data is transferred. Of course, a "safety net" between the sales and business departments is essential, lest an unacceptable proposal be confirmed and entered without sales management approval. Accordingly, the link program holds orders that are exported from the proposal writing software in a pending file until ultimately approved by the sales manager, and in a further pending file before being verified by traffic. This ensures that there are no spots being scheduled without final approval from both the sales manager and the traffic director.

Traffic and Billing

Many software alternatives for traffic and billing exist today, and most provide the common denominator of placing data which is entered onto a schedule and generating logs and invoices from that data. Where real efficiencies come into play, however, are with capabilities that permit the data to travel electronically through **all** stages of the traffic and billing system, from contract entry through invoices and statements, with minimal intervention by a human operator. There are several key points in the traffic workflow where this is possible: at the time of contract entry; during the log reconciliation; and the merging of invoice and co-op/affidavit information.

We've already seen how electronic transfer of contract data from the sales force to the business office speeds the workflow. Just as critically, however, it also ensures that the data is accurate and up-to-date…eliminating the often comic scene of the traffic director frantically searching for a salesperson to interpret the handwriting on an insertion order or to explain why the number of spots and the total dollars do not agree. When we allow the computer to "double-check" our work as it is created we can be assured that the number of errors is reduced and the time spent correcting them can now be spent on more productive pursuits. Once the mundane task of data entry has been relegated to the computers themselves the traffic director can now concentrate on the manipulation of scheduling priorities and conflict codes. Skillful assignment of scheduling priorities that allows the computer to accurately distinguish a station's high priority business from the lower priority business also increases the computer's ability to create the maximum revenue producing program log. Most traffic systems have from seven to ten priority levels. There are more sophisticated systems available, however, that have as many as sixteen levels of priority. Increasing the number of priority levels gives the sales manager and the traffic director more specific control over the computer's scheduling ability.

Once the schedule data is in the traffic system and the log is generated the next step is to move that information to the digital audio system for playback. The most convenient way to do this using today's technology is to directly export the commercial log file to the digital system so that the commercial scheduling instructions can be properly executed. Once again, the more sophisticated traffic and billing systems allow for this direct export of the commercial log in the digital audio system's native file format, thus ensuring that there is no data loss.

Finally, the task of log reconciliation may also be handled electronically. Most digital audio systems have the ability to generate an "as played" report that the traffic director refers to while manually reconciling the log. Optimally, the traffic system should have the capability – as at least one currently available system does – to accept the "as played" data in file form and electronically compare it with the original schedule data, making the necessary adjustments automatically. The only spots that are not automatically reconciled are those that require human intervention, such as spots that did not play at all or spots that were inadvertently moved into the wrong daypart by the announcer. Here, as with the priority assignments described above, operator control of the criteria for when and how to automatically reconcile a spot guarantees that the adjustments are not made capriciously: the traffic director sets the parameters and the computer executes them efficiently and quickly.

With the spots having been reconciled, affidavits may now need to be generated for the advertiser along with billing. Most available traffic programs incorporate this feature: the more advanced ones allow several formatting options. Particularly as stations which may have varying needs for this kind of documentation continue to come together under common banners, this kind of flexibility will be important in any station management system – because it will allow each station within a group to retain its own identity and familiar appearance to clients, while still enabling the group to realize the efficiencies of standardizing on a single system.

Management Reporting

The spots have been ordered, scheduled, broadcast, reconciled, billed – and, hopefully, paid for. Now, at the end of the week, month, quarter or year, the station (or regional, or group) manager needs to review the financial and sales data. His/her questions of that data are myriad and they are the lifeblood of that station. Are we getting the best possible rate for every commercial avail? Are spots bumping excessively and resulting in lost revenue? Are my salespeople all performing to expectation? Are they selling our clients the schedules, which will bring them the most successful exposure while also bringing the station profit? Answers to these questions are critical to the success of any radio operation and they depend on accurate data delivered in a timely fashion to managers qualified to make the necessary decisions.

Two elements are critical for management reports to be generated which will meet these and similar information needs. First, techniques must be employed which will bring down to a manageable and interpretable level the vast storehouse of data a station or group possesses on the sales, placement and pricing of spots. In this age of information overload, selective reporting is key. Second – and, though it should go without saying, this is an element too often ignored or poorly implemented – ease of use must be incorporated. The most informative report will be useless if generating it is so cumbersome and complex a process that it never becomes a regular part of a station's routine.

In this area we can take our cue from the computer database industry, where sophisticated tools permit a "data warehouse" to be queried on a selective basis for useful information. This is particularly important in a troubleshooting situation. Perhaps you have only a handful of potential problems, among hundreds of accounts. Selective querying lets you omit that data so you are working from a cleaner picture at the beginning. Instead of reviewing every advertising contract in detail to glean the information needed to assemble the top-level picture, you are in effect *reporting by exception* – I want only to see data on spots which fall below my target average yield, or salespeople whose performance is outside such-and-such a parameter. Especially where a general or sales manager may now have responsibilities for dozens of stations rather than one or two, the potential time savings and gain in efficiency from his looking only at pertinent data is significant.

Ease of use is perhaps the more difficult consideration in that the human factor plays a large role. For all that we live in a technology-rich environment, many individuals at the executive level in all industries, broadcasting included, may not have the comfort level with databases and PC technology that would allow them to generate these queries and reports themselves – even if they had the time to do so, which they likely do not. It remains for their own technical staff or the original software providers to step into the breach and add the user-friendliness to these tools that makes them tools which will actually be used. The latter probably have a more immediate understanding of the software's capabilities and how best to package a report; and there has been a clear trend for manufacturers to move from standard reports, to custom reports, to user-defined reports, and finally to reports which present user-requested data in a form as intuitive and fast as a well-planned web page or "help" system. Here, as in so many other areas, more will be done on screen than on paper as the available software continues to evolve – and, in general, done faster, "smarter" and with less extraneous or irrelevant data being presented.

Conclusion

We have now clearly illustrated how a computerized approach to managing the information workflow, within a station as well as across a group of stations, yields benefits such as reduced time spent on repetitive manual tasks, improved accuracy, and greater timeliness of information. All of these benefits, however, are ultimately reflected in the station or group's bottom line. By utilizing current technology, and by making business decisions based upon more complete and timely information, a station can close the cracks through which significant revenues, in the form of data, can otherwise disappear.

DTV AUDIO WORKSHOP: AC-3 TECHNOLOGY AND DEMONSTRATION

Wednesday, April 8, 1998
9:00 am - 12:00 pm

Chairperson:
Dave Burns,
Harris Corporation, Richmond, IN

***Welcome and Overview of Digital Audio Technology**
Harris Broadcast

***Where Are We Headed in the Digital Arena - On Air Processing Advantages and Opportunities for the** Broadcaster -George Schlatter, Director/ Producer/Writer, Los Angeles, CA

***AC-3 AUDIO: Audio Services, Applications- Routing AC-3 through The Studio - How Does Dolby See the Direction of the Industry**

***Studio Options**
Chip Schneider
Harris Broadcast
Florence, KY

***Audio Monitoring,**
Will Wohler
Wohler Technology
San Francisco, CA

***Panel**
All Presenters

*Papers not available at the time of publication

Hot Topics: Regulatory Issues in the Real World: Part I

Wednesday, April 8, 1998
9:00 am -11:30 pm

Chairperson:
Dane Ericksen, P.E.
Hammett & Edison, Inc., San Francisco, CA

***Unattended Operations**
Harold Hallikainen
Hallikainen & Friends
San Luis Obispo, CA

***Relaxed LPTV/TV Translator Rules for DTV Refugees**
Keith Larson
Federal Communications Commission
Washington, DC

Your DTV Assignment - Does It Provide Optimum Coverage?
Robert D. Weller, P.E.
Hammett & Edison, Inc.
San Francisco, CA

***UHF DTV Power Disparities**
Nat Ostroff
Sinclair Broadcast Group Inc.
Baltimore, MD

***OET 69 and Interference Studies for Non-Sixth R&O Facilities: What the FCC Expects**
Keith Larson
Federal Communications Commission
Washington, DC
Richard Smith
Federal Communications Commission
Washington, DC

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*Papers not available at the time of publication

Your DTV Assignment – Does it Provide Optimum Coverage?

Robert D. Weller, P.E.
Hammett & Edison, Inc.
Consulting Engineers
San Francisco

ABSTRACT

A method for evaluating allocation conditions for DTV stations has been specified by the FCC in its Rules and in OET Bulletin No. 69. This method has been used to make initial DTV allocations for TV stations in the U.S. While stations are free to propose alternative allocations, rote application of this method may lead to unacceptable interference and to design of a facility having less than optimum coverage. By judicious application of the allocation criteria, opportunities may exist for improved DTV facilities.

The FCC allocation method for DTV stations is reviewed, providing detailed explanations of concepts such as replication directional antenna pattern, replication power level, assumed receive antenna, and DTV coverage. Some of the various assumptions made by the FCC in its allocation software are identified, as well as the inherent limitations of the specified method. Supplemental analysis methods are identified that can be used to design stations having optimum coverage, while still meeting the allocation criteria.

INTRODUCTION

Last year, the FCC culminated its ten-year investigation into advanced television and its impact upon the existing (NTSC) broadcast service with the adoption of a Table of Allotments for digital television (DTV) and rules for modifying that table.[1] Following release of the 5th and 6th Report and Order (R&O) documents, some 231 petitions were filed requesting reconsideration of various aspects of the Commission's decisions, as well as a number of oppositions to those petitions. Significantly, based upon the Commission's adoption of a spurious emission "mask,"[2] the Advanced Television Test Center (ATTC) revised its recommended DTV protection requirements for adjacent channel DTV operations.[3] Despite the presence of the pending petitions, the FCC issued a number of construction permits for DTV stations. With the February 1998 issuance of a Memorandum Opinion and Order, which changed some 71 of the entries in the original DTV Table, it appears that a few of the early applicants will have to apply to amend their DTV construction permits to conform to the revised table.

The actual coverage, both noise- and interference-limited, of a DTV station, and indeed the actual field performance of the Grand Alliance DTV system itself, is largely unproved. However, by focusing on coverage optimization at the allocation and design stages, one can maximize the probability of satisfactory DTV coverage. Examination of the methods used to produce the Table of DTV Allotments is a useful starting point for determining whether optimum coverage will be achieved.

DEFINITIONS

Use of the complex method described below for DTV coverage and interference analysis leads to several new terms-of-art that will appear in this paper and which one may encounter in general discussions about DTV coverage and allocation. These include:

DTV Threshold Coverage. An equivalent field strength that provides a "just usable" DTV signal most of the time. For UHF Channel 39, that value is 40.8 decibels above 1 microvolt per meter at the best 50% of locations 90% of the time, *i.e.*, an F(50,90) value of 40.8 dBu. The threshold field strength varies by channel. While analog systems, such as NTSC television, are evaluated by fidelity criteria, such as signal-to-noise ratio and TASO grade, digital systems are evaluated by probability of error, which is commonly expressed in terms of bit error rate (BER). A BER threshold value of 3×10^{-6} determines whether the DTV signal is usable or not. This BER threshold has been determined experimentally to exist at an equivalent carrier-to-noise ratio of 15.2 dB, which can be used to infer an equivalent field strength value, occurring statistically at a certain percentage of receive sites and a certain percentage of the time, with certain assumptions about the receiving system.

DTV Replication Pattern. The transmitting antenna azimuth pattern, calculated by the FCC on a radial basis, that causes the F(50,90) DTV coverage contour to match the F(50,50) Grade B coverage contour. Because of the non-linear relationships between these contours (particularly when the DTV channel allocation is in a different frequency band from the associated NTSC channel), the replication pattern may have a different shape from the associated NTSC antenna azimuth pattern.

Replication Power Level. The maximum effective radiated power (ERP) level assigned by the FCC to a DTV allotment. Because of the DTV replication pattern, described above, most DTV assignments assume the use of a directional antenna. The DTV replication power level is the maximum permitted in the direction of maximum radiation.

Assumed Receive Antenna. For purposes of coverage and interference calculation, use was assumed of a directional receiving antenna having specified gain and radiation pattern characteristics. Different gains and front-to-back (F/B) ratios were assumed for the NTSC and DTV antennas.

FCC ANALYSIS METHOD

A new Section of the FCC Rules, §73.623(c), references both Appendix B of the 6th R&O and Office of Engineering and Technology (OET) Bulletin No. 69 as sources for the procedure required to evaluate proposed modifications to allotted DTV facilities. However, neither source provides adequately thorough guidance for conducting interference evaluations involving the newly-allotted DTV channels, with regard to potential interference to and from existing authorized NTSC facilities, and other allotted DTV facilities. In fact, many details are contained only in the actual computer code and input files used by the FCC and could therefore be subject to substantial change with just a few keystrokes.

The Commission's program utilized a complex set of analysis tools to generate the Table, that may be briefly described as follows:

For any given NTSC or DTV station to be studied, the FCC analysis model first determines the location of the conventional F(50,50) Grade B contour of the NTSC station, or of the NTSC station associated with an assigned DTV station, using antenna azimuth pattern information contained in the FCC engineering database and an assumed antenna elevation pattern. The model treats that contour as an envelope, outside of which no protection from interference is implied or afforded. The location of the Grade B contour is also used to determine the assigned power for the DTV station, once again using the F(50,50) curves to produce single-valued (but sometimes discontinuous) contours and then working backward to determine the power necessary on a radial basis to generate the associated F(50,90) DTV coverage contour for the assigned DTV channel (40.8 dBu for UHF adjusted by a dipole factor, 35.8 dBu for high-VHF Channels 7–13, and 27.8 dBu for low-VHF Channels 2–6). The maximum power determined using this method was assigned as the DTV operating power, provided it was calculated to be above established minimum power levels; otherwise, a minimum power level (50 kilowatts for UHF channels, 3.2 kW for high-VHF channels, and 1.0 kW for low-VHF channels) was assigned. Note that the use of this method usually creates a directional antenna pattern, *even for DTV assignments to NTSC TV stations that are presently omnidirectional.* The FCC requires that a DTV facility employ an antenna design that "fits" within the calculated pattern, or that a nondirectional antenna be employed that does not exceed the Grade B envelope in any direction, unless it can be demonstrated that no new interference is created.

In addition to the use of the Grade B envelope and an assumed directional transmitting antenna for all DTV facilities, the model assumes the use of directional receiving antennas at each studied location, or "cell." The characteristics of the receiving antennas are different not only for the low VHF, high VHF, and UHF frequency bands, but also for NTSC and DTV reception, where, based on the FCC model, more directive antennas are assumed to be used to receive DTV signals.

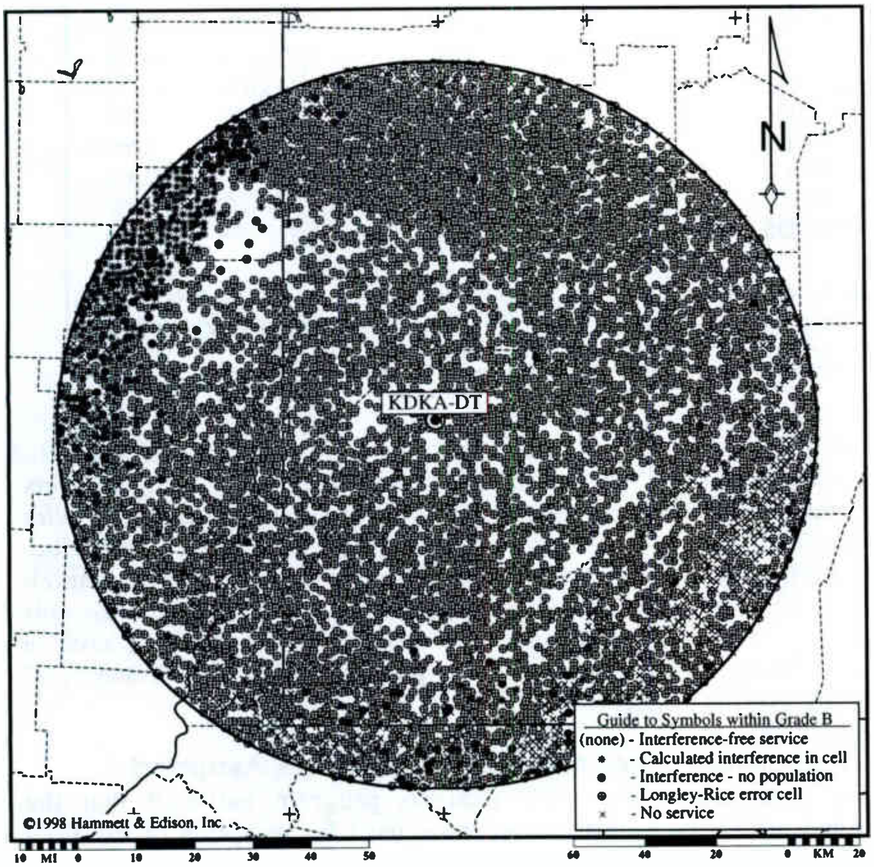

Figure 1. As discussed in the text, the Longley-Rice propagation loss prediction algorithm cannot always calculate (within certain confidence limits) a field strength level at each cell. As an example, the DTV coverage area of KDKA-DT contains many cells where the program reported results that were "dubious or unusable." For purposes of its replication calculations, the FCC always assumed interference-free coverage in such situations.

The FCC analysis technique employs terrain-sensitive calculation methods based on Version 1.2.2 of the ITS Irregular Terrain Model, also known as the Longley-Rice model. For each NTSC or DTV station to be studied, a grid of cells, each two kilometers on a side, fills the associated Grade B contour. The program first determines which of the cells is predicted to receive service from the associated station, using Longley-Rice with F(50,50) statistical weighting for NTSC stations and F(50,90) statistical weighting for DTV stations. Cells determined to have no service are not studied for interference from other stations. Once cells having service are determined, the software analyzes potential interference from other NTSC or DTV stations, again using the Longley-Rice propagation algorithm and F(50,10) statistical weighting for all potential interfering signals. Each cell is evaluated using the desired-to-undesired ratios presented in FCC Rules §73.623 for each channel relationship, and cells determined to have interference are flagged and summed with the study results of other cells, resulting in the generation of total interference area figures and tabulations of total population contained within the summed cells.

LIMITATIONS OF THE FCC METHOD

Experience gained by using the FCC analysis program has led to the identification of several factors that are, at least, unusual, and do raise significant concerns about the validity of some of the assumptions made.

"Interference-Free" Areas

The interference analysis technique employed by the FCC and specified for study of proposed DTV facility changes employs terrain-sensitive calculation methods based on the Longley-Rice model. The model is used to analyze paths between the transmitter and assumed receiver locations that are contained within a grid of square cells that fill the entire protected service area. However, the Longley-Rice model is not always capable of determining, within certain confidence limits, whether a particular cell has service.[4] Specifically, in cases where the actual horizon from a given receive cell or transmitter location is less than 0.1 times or greater than 3 times the distance to the smooth earth horizon, the Longley-Rice algorithm will return an "Error Marker 3." According to the program documentation, this means that internal program

Market		Example Station	Longley-Rice Errors (as % of Grade B) Area, sq. km		Population (1990)	
#2	Los Angeles, CA	KABC(N07/D53)	9,002	*26.8%*	768,975	*5.3%*
#5	San Francisco, CA	KDTV(N14/D51)	6,134	*32.5%*	990,688	*15.8%*
#6	Boston, MA	WGBH(N02/D19)	6,697	*27.1%*	1,373,952	*19.7%*
#12	Seattle, WA	KTZZ(N22/D25)	7,360	*37.3%*	385,679	*13.0%*
#17	Phoenix, AZ	KPHO(N05/D17)	7,862	*16.5%*	40,830	*1.8%*
#18	Denver, CO	KCNC(N04/D35)	9,211	*22.0%*	204,553	*7.8%*
#19	Pittsburgh, PA	KDKA(N02/D25)	17,074	*50.5%*	2,041,954	*52.3%*
#24	Portland, OR	KOPB(N10/D27)	13,739	*35.8%*	121,680	*5.8%*
#29	Raleigh, NC	WLFL(N22/D57)	3,964	*12.6%*	239,358	*11.3%*

Table 1. There are many markets where terrain causes significant Longley-Rice errors. This table lists markets where the assumption of interference-free service contributed greatly to the apparent close replication of NTSC service area and population listed in the Table of Allotments.

calculations show parameters out of range, and any reported results are dubious or unusable. Incredibly, the procedure used by the FCC when such a Longley-Rice error occurs, whether during determination of potential service or potential interference, is to assume that cell is enjoying "interference-free service."

While this assumption appears not to introduce significant overall errors in areas of relatively flat terrain, it has been found that the error code is returned much more often for studies involving mountainous or even hilly terrain. For example, Figure 1 shows predicted interference to the coverage area of the DTV Channel 25 allocation assigned to Station KDKA-TV, Pittsburgh, from various other NTSC stations, and DTV assignments. While the FCC DTV analysis returned a result of interference to about 172,000 persons, it simply ignored possible interference to nearly 2.6 million persons within the KDKA-DT coverage area, considering all of that area to be "interference-free" coverage due to Longley-Rice errors.

This problem is not restricted to a few markets. Table 1 summarizes for a number of top markets the tremendous arbitrary designation of service occurring due to Longley-Rice errors. It is obvious that there are situations within the top 30 markets where Longley-Rice errors are the only justification for classifying over half of the area or over half of the population with "interference-free" service.

Receive Antenna Models Assumed

The DTV analysis program assumed that the consumer antennas used for reception of the DTV signals would be better-performing than the ones used for reception of the NTSC signals, despite the fact that both operations are on the same general frequencies and that no one, in fact, had suggested that consumers would be at all inclined to replace their existing antennas or to install second, larger antennas just for DTV. The off-axis rejections (front-to-back ratios) of the DTV antennas are also higher, as shown in Table 2.

Receive Antenna Characteristic	Low VHF	High VHF	UHF (all)
NTSC Gain	0 dBd	0 dBd	0 dBd
DTV Gain	4	6	10
DTV-NTSC Difference	4	6	10
NTSC Front-to-back ratio	6 dB	6 dB	6 dB
DTV Front-to-back ratio	10	12	14
DTV-NTSC Difference	4	6	8

Table 2. In calculating coverage and interference, different receiving antennas were assumed for NTSC and DTV service.

However, such antennas would not necessarily be purchased and installed by consumers viewing the DTV stations off-air. In fact, a more reasonable assumption might be that they will *not*. Therefore, DTV service will be more interference-limited than assumed and replication of NTSC service will not, in practice, be achieved.

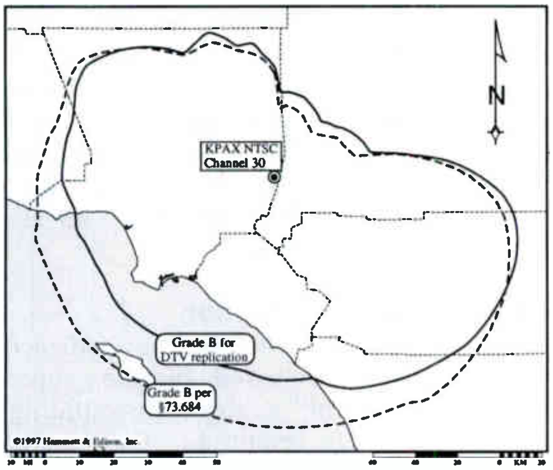

Figure 2. Because the Grade B contour was not calculated in accordance with FCC Rules in developing the Table of DTV Allotments, the allotted DTV power level and directional pattern often do not replicate the true Grade B coverage. This is demonstrated by the actual and replication Grade B contours calculated for a station in the Los Angeles market.

Establishment of Grade B Contour

Several simplifying calculations were made that are not consistent with the definition of the Grade B coverage contour contained in the FCC Rules, including:

Elevation Plane Pattern. The method by which the FCC projected NTSC Grade B contours for the DTV allotment project does not comply with Rule §73.684, which defines prediction of coverage for NTSC facilities. First, the requirements of §73.684(c)(2) were not followed, such that full radiation is to be assumed whenever the radiation toward the radio horizon is at least 90% of the maximum. This would tend to understate the distance to the Grade B contour. Of course, without the actual elevation plane pattern employed by the station, the determination of radiation toward the radio horizon would be expected to be in error. Even though a standard elevation plan radiation pattern was used, it may not be appropriate for high-gain antennas, for which it would overstate the radiation, and certainly would not be appropriate when electrical beam tilt and/or mechanical tilt is employed, for which the error could be in either direction.

Azimuth Plane Pattern. Second, the horizontal plane azimuth pattern, as taken from the FCC database, was employed with an assumed standard elevation radiation pattern, to generate a protected NTSC Grade B contours for an associated DTV facility. However, the database contains only the projection of the actual azimuth pattern onto the horizontal plane and so may significantly understate the radiation in particular directions if mechanical tilt is employed.

HAAT. Third, the FCC program did not truncate, as required by §73.684(d), radials used to determine height above average terrain (HAAT) when they extended over large bodies of water or over foreign territory.

For example, Figure 2 shows the projection of the Grade B contour for TV Station KPAX, NTSC Channel 30, San Bernardino, California, in accordance with the FCC Rules (dashed line) and as determined by the FCC's DTV replication program (solid line). The obvious errors represent the cumulative effect of two of the three problems identified above and have a significant affect on the station's DTV replication power and pattern.

Scaling of Replication Pattern

The FCC replication program used a procedure that derived the Grade B contour for an existing NTSC station, and then redefined that contour as the limit of protected service for the DTV facility. Using the appropriate F(50,90) curves, the DTV power necessary to reach the NTSC Grade B contour location was determined radially. When the maximum calculated power was found to be above the maximum power allowed for a given channel, the pattern was scaled to that maximum. However, the scaling process will necessarily reduce the directional replication pattern to power levels *below* that maximum for all other azimuths, even though the replication power at those azimuths may not have exceeded the maximum power. Therefore, by scaling the pattern instead of truncating it at the maximum power level, the DTV station is further limited from replicating its Grade B coverage.

Some Protected Areas Not Studied

The FCC replication program studied for interference protection only those stations within certain distances, which are specified in OET-69. Many of the distances specified are often not sufficient to allow the entire Grade B area of a station to be studied. This effect can result in truncation of a station's Grade B contour, and also in areas of further interference within the Grade B contour that have not even been checked, leading to further exaggeration of the degree of replication.

DTV FACILITY OPTIMIZATION

The 6th R&O encourages DTV stations to "maximize" their facilities,[5] with coverage up to that of the largest station within each market, such that no significant new interference is caused to other stations. Fundamental to the prediction (or optimization) of coverage is knowledge of the effective sensitivity of the receiving system. As previously mentioned, only scant data is available concerning the field performance of the DTV system, so it is prudent to make conservative choices when estimating coverage and interference performance.

Definition of Coverage

Whereas the coverage of NTSC television stations is often described in terms of three types of service (City Grade, Grade A, and Grade B), which are related to the quality of the assumed receiving installation and picture fidelity, the FCC has provided only one definition of coverage for DTV: the threshold. The well-known "cliff effect," whereby a small reduction in received signal strength causes a perfect DTV picture to become abruptly unusable, suggests that an adequate margin is necessary to ensure that a usable DTV picture will exist over the long term. Implicit in the FCC's definition of DTV coverage is that the specified field strength would be available at least 90% of the time. Obviously, households receiving no picture 10% of the time will not be dedicated viewers of your station.

Improvement of the long-term reliability of the signal can be ensured by adding an additional fade margin factor to the threshold coverage signal level. Additional fade margin factors can be derived by assuming that the fading ratios tend to follow log-normal distributions.[6] For example, a DTV station assigned to a UHF channel might apply an additional factor of 3–9 dB to increase the time variability factor from 90% to 99%.

Similarly, if one believes that some viewers may use back-of-set antennas for DTV reception, the assumed receive antenna gain might be reduced by 10 dB (at UHF). Additionally, building penetration losses might be expected in the range 6–12 dB.[7] Accounting for just these three factors, one might increase the required field strength by 19–31 dB above the threshold value. So, DTV coverage should be optimized not at the threshold value (around 41 dBu at UHF), but 20–40 dB above it (around 60–80 dBu at UHF).

Antenna Pattern Selection

In order to maximize the potential audience of a DTV station, knowledge of the population distribution in the region surrounding the transmitter site is required. One method of visualizing the population distribution is to utilize the U.S. Census data, calculating the azimuth and depression angles from the transmitting antenna to each Census Block. An example of this technique is shown in Figure 3. This type of analysis suggests an appropriate elevation pattern, including appropriate values of electrical and mechanical beam tilt.

The specification of optimum elevation-plane radiation characteristics (including beam tilts) is especially critical for stations allotted facilities of less than 1,000 kilowatts at UHF, but desiring to increase to the maximum value. The FCC recognizes the use of beam tilting techniques for improving coverage through increased main-beam ERP within a market, while limiting interference outside of that market.

Other Propagation Models

Because of the inherent limitations of Longley-Rice, as discussed above, it is prudent to explore other propagation prediction models when evaluating methods to optimize coverage. Proprietary, commercial, and public-domain analysis software has been developed that employs all of the analysis features described above, as well as several other more subtle elements employed in the FCC allotment program. Such programs can provide a graphical element that allows the identification of all interference cells on a map with an associated tabulation, and generates a DTV antenna pattern envelope that shows areas that can be maximized without creating interference in any cells that were not already receiving interference. Such programs can be used to test implementation scenarios that

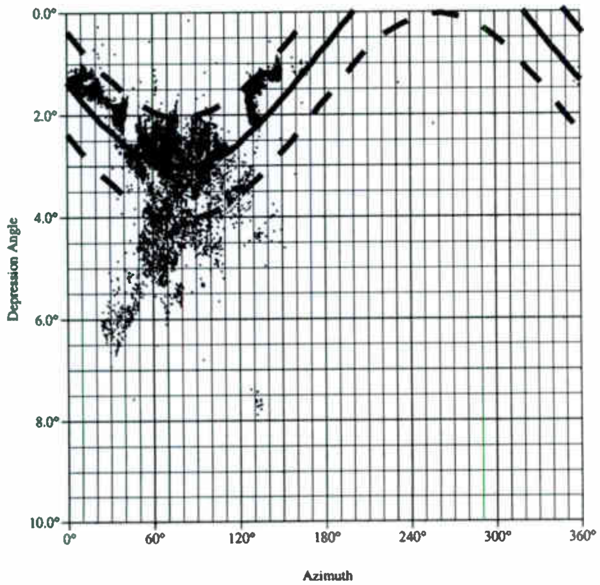

Figure 3. Each dot represents a U.S. Census Block (*i.e.*, one or more potential viewers). The depression angle from the transmitting antenna to each Census Block was calculated as a function of azimuth. The main beam elevation pattern of a hypothetical antenna is shown as the solid black line, with the half-power points shown as dashed lines. In this case (Farnsworth Peak, near Salt Lake City), use of both electrical beam tilt (1°) and mechanical tilt (2° toward 80°T) seems appropriate for optimum coverage. Population at depression angles exceeding 4° (*i.e.*, close to the transmitter site) would be treated with null fill.

involve changes to antenna height, antenna pattern, channel number, and transmitter location. Additionally, such programs can determine coverage areas of DTV and NTSC stations, with interference cells omitted (*i.e.*, noise-limited coverage) at almost arbitrarily fine resolution, rather than the relatively coarse 2-kilometer square cells used in an FCC-style analysis.

CONCLUSION

In developing its Table of Allotments, the FCC made a number of simplifying assumptions, which are entirely appropriate for allotment and allocation use. One should not, however, rely upon such methods for facility design or optimization, since inherent in many of those assumptions are factors that can reduce coverage, exacerbate interference, or both. Proper facility design requires an understanding both of the procedures used to establish the DTV allotments and of their limitations. Design of optimum DTV transmitting facilities should take advantage of opportunities existing within the regulatory framework, while realizing that the regulatory procedures are intended as spectrum management tools rather than station design procedures.

ACKNOWLEDGMENTS

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The author can be reached at Hammett & Edison, Inc., Box 280068, San Francisco 94128, telephone 707/996-5200 or e-mail rweller@h-e.com.

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1 FCC, Fifth and Sixth Reports and Orders, MM Docket 87-268, both adopted April 3, 1997, amended by the Memorandum Opinion and Order on Reconsideration of the Sixth Report and Order, adopted February 17, 1998.

2 Section 73.622(h) specifies attenuation in dB of $46 + (\Delta f^2)/1.44$ out to 6 MHz from the channel edge, with 71 dB being required at all frequencies beyond 6 MHz. Δf is the frequency difference in MHz from the edge of the channel.

3 ATTC Document #97-04, "An Evaluation of the FCC RF Mask for the Protection of DTV Signals for Adjacent Channel DTV Interference," July 16, 1997.

4 This is one of the reasons that H&E uses TIREM (Terrain-Integrated Rough Earth Model), a more sophisticated propagation loss algorithm of which the Longley-Rice routine is only a part. See "Coverage Prediction for Advanced Television," Proceedings of the 1994 NAB Engineering Conference for further information.

5 6th R&O, at ¶31.

6 Harry Fine, The Normal Distribution as Applied to VHF Broadcast Service Problems, FCC T.I.D. Report No. 4.2.2 (Washington, DC: FCC, 1949).

7 Macario, "How Building Penetration Loss Varies with Frequency," IEEE Vehicular Technology Society News, Vol. 40, No. 4 (1993), p. 26.

Hot Topics: Regulatory Issues in the Real World: Part II

Wednesday, April 8, 1998
2:00 pm - 5:30 pm

Chairperson:
Dane Ericksen, P.E.
Hammett & Edison, Inc., San Francisco, CA

***Fine-Tuning EAS**
Richard Rudman
KFWB-AM/KTWV-FM
Los Angeles, CA
Leonard Charles
SBE EAS Committee
Madison, WI
Frank Lucia
Federal Communications Commission
Washington, DC
Dave Wilson
NAB
Washington, DC

***Satisfying FCC Expectations For The New RFR Guidelines**
Keith Larson
Federal Communications Commission
Washington, DC
William Hammett
Hammett & Edison, Inc.
San Francisco, CA
Robert Cleveland
Federal Communications Commission
Washington, DC

***Broadcast Auxiliary Spectrum: Fighting for Frequencies and Using them Efficiently**
Moderator: Jack Goodman
NAB
Washington, DC
Presenters: Ben Fisher
Fisher Wayland, Cooper, Leader and Zaragoza L.L.P.,
Washington, DC
Kelly Williams
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Christopher Imlay
Booth, Freret, Imlay & Tepper, P.C.
Washington, DC
Victor Tawil
MSTV
Washington, DC

*Papers not available at the time of publication

DTV in the Real World: Digital Television Case Studies and Standards

Wednesday, April 8, 1998
2:00 pm - 5:30 pm

Chairperson:
Jerry Butler
PBS/Public Broadcasting Service, Alexandria, VA

***Transitioning to DTV: Network Contribution and Intra Studio Distribution Issues**
Tom Jordan
Leitch Incorporated
Chesapeake, VA

Naming, Numbering and Navigating the Channels in the Digital World
Bernie Lechner
Consultant
Princeton, NJ

ATV Implementation- An Engineering Perspective On a Real World Project
Garrison Cavell, Joseph Davis, P.E. and Thomas Mann
Cavell, Mertz & Perryman, Inc.
Fairfax, VA
Greg Johnson
KITV
Honolulu, HI

***Designing and Building a Real DTV Station - The KCPQ Story**
Stuart Loberg and Greg Doyle
Sparling Communications
Seattle, WA

***WRAL-HD DTV Field Testing**
Luther Ritchie
WRAL-TV
Raleigh, NC

***Field Results: Field Testing at the Model Station Project - WHD**
Dennis Wallace
Wallace & Associates
Washington, DC

***Field Testing Digital Television Panel**
Victor Tawil
MSTV
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Dennis Wallace
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Gary Sgrignoli
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Raleigh, NC

*Papers not available at the time of publication

NAMING, NUMBERING AND NAVIGATING THE CHANNELS IN THE DIGITAL WORLD

Bernard J. Lechner
Consultant
Princeton, New Jersey

ABSTRACT

The ATSC has adopted a Program and System Information Protocol Standard that defines a number of tables and descriptors that will be included in the transmitted signal of each digital television station. The information in these tables and descriptors provides a list of current and future programs being transmitted and defines a two-part channel number that is associated with each program. The first part of the number identifies the broadcaster and the second part identifies a specific program. The receiver collects this information and associates the channel number with the viewer's tuning commands to select the desired program. The receiver can also use the transmitted information to build an on-screen program guide. This paper describes these functions of the ATSC Standard as well as its other features including descriptors for content advisory and closed caption information.

INTRODUCTION

The introduction of digital television broadcasting has presented many challenges to the television industry. One of the more important of these is how to name and number the digital channels and how to make it possible for consumers to easily navigate the new channels. In the current NTSC world, broadcasters identify themselves with call letters and their assigned RF channel number, and consumers navigate the channels by entering the channel number and/or using up-down buttons. In the new world of digital television, each broadcaster will be transmitting his digital signal on a different RF channel than his NTSC signal. Furthermore, each broadcaster may be simultaneously transmitting several different digital television programs within that single RF channel. How are these programs to be identified? Will the channels and programs be labelled the same way on cable TV as they are over the air? What will happen when the consumer pushes the up button? Will he get the next program in the RF channel to which he is tuned, or will he move up to the next RF channel? If he moves up to the next RF channel, which of the several programs being transmitted in that RF channel will he get? How will consumers learn that the new digital television station on RF channel 39 belongs to the same broadcaster whose NTSC signal has been on RF channel 5 all these years? These are but a few of the questions that arise relating to the naming, numbering and navigation of the channels in the digital world.

Beginning in early 1997, the Advanced Television Systems Committee (ATSC) undertook an effort to develop a standard that would provide a unified solution to naming, numbering and navigating that would satisfy the requirements of terrestrial broadcasters, cable TV operators, consumer electronics manufacturers and consumers. The work was carried out within the ATSC Technology Group on Distribution (T3) by the Specialist Group on Service Multiplex and Transport Systems (T3/S8) and was completed in November, 1997. The resulting Standard, "Program and System Information

Protocol for Terrestrial Broadcast and Cable", ATSC Document A/65, was approved by ATSC on December 22, 1997. The Standard is generally referred to as either (A/65) or the PSIP Standard.

REQUIREMENTS

The first step in developing the PSIP Standard was to develop a list of requirements. Inputs were solicited from a wide variety of sources, including terrestrial broadcasters, cable TV operators and equipment suppliers, consumer electronics manufacturers and others. Some of the most important requirements were the following:

Preserve Channel Number Branding. Terrestrial broadcasters have invested heavily in developing brand names, logotypes, slogans, etc. that are based on their existing NTSC channel number and they expressed a strong desire to retain that number for their digital service. If you have been "Newscenter 5" for the last 25 years or more, "Newscenter 39" doesn't quite sound right. Likewise, "Live at 5 on 39" doesn't have the same appeal as "Live at 5 on 5" for the station with an NTSC assignment of channel 5.

Harmonization Between Terrestrial Broadcast and Cable TV. When carried on cable TV, it should be possible to preserve the names and numbers associated with terrestrial broadcast programs so that consumers can tune them identically in both media.

Direct Access to Any Channel. This means that the consumer must be able with a simple set of key-stroke entries to go directly to a desired program regardless of what RF channel it is being carried on and which program it is within that RF channel.

Grouping of Digital and Analog Services. There should be a simple relationship between the identification of a broadcaster's digital television service and his analog NTSC service so that they can be found so-to-speak "at the same place on the dial". Also, on cable TV systems it may be desirable to group two separate digital services, e.g., NBC and CNBC or ABC and Disney.

Compatibility with Both Printed and Electronic Program Guides. It must be possible for the consumer to tune a program by looking it up in a printed program guide and making a simple set of key-stroke entries that take him directly to the correct RF channel and to the desired program within that channel. On-screen electronic program guides must also be supported.

Support a Variety of User-Friendly Navigation Paradigms. Consumer electronics manufacturers take pride in their ability to provide a wide variety of products that satisfy the needs and desires of consumers. Consequentially they expressed a strong desire that the channel labeling system support a wide variety of navigation paradigms but not require any specific paradigm.

Extensibility to Data Broadcasting and Other Services. It is anticipated that digital broadcasting will include not only television programs, but also audio-only programs and data broadcasts. These programs must also be labelled so that they can be tuned by the consumer, or in the case of some data broadcasts, by the device, e.g., a computer, for which they are intended.

THE PSIP STANDARD

The PSIP Standard meets the above requirements. The Standard defines a small collection of tables and descriptors which each broadcaster must transmit continuously. These tables contain information about the identity of the broadcaster and provide a list of current and future programs being transmitted. The receiver collects and stores this information from all of the broadcast signals that are available to it. The receiver uses this stored information to implement a navigation paradigm that will allow the consumer to select a desired program. A key feature of the PSIP Standard is the use of a two-part channel number. The first part, the major channel number, identifies the broadcaster and the second part, the minor channel number, identifies a particular program.

Virtual Channel Table

In order to preserve channel-number branding and to provide for harmonization between terrestrial broadcast and cable TV, the PSIP Standard uses the concept of virtual channels. A virtual channel is the designation, the two-part channel number in the PSIP Standard, recognized by the user as the single entity that will provide access to a particular program. It is called "virtual" because its identification (name and number) may be defined independently from its physical location. Examples of virtual channels are: digital radio (audio only), a typical analog TV channel, a typical digital TV channel (composed of one audio and one video stream), multi-visual digital channels (composed of several video streams and one or more audio tracks), or a data broadcast channel (composed of one or more data streams). In the case of an analog TV channel, the virtual channel designation will link to a specific physical transmission channel. In the case of a digital TV channel, the virtual channel designation will link both to the physical transmission channel and to the particular video and audio streams within that physical transmission channel.

One of the tables defined by the PSIP Standard is a Virtual Channel Table (VCT). For each virtual channel (major number plus minor number) the VCT specifies the carrier frequency of the RF carrier that carries the stream that contains that channel's program, the MPEG Transport Stream Identifier (TSID) for the stream, the MPEG Program Number and the Service Type. Other entries in the VCT include the Short Name (up to 7 characters) for the channel and a list of associated descriptors. The use of a VCT allows the channel numbers to be assigned independently of which transport stream the program is carried on and independently of the carrier frequency on which the transport stream is carried.

For terrestrial broadcast, this will allow broadcasters to use their existing NTSC channel number as the major channel number to identify their new digital transmissions, irrespective of the actual RF channel assigned to the digital transmitter. Minor channel number "0" is reserved to designate the NTSC channel; minor channel numbers 1-99 can be assigned to programs by the individual broadcaster at will without the need to coordinate with other broadcasters in his local area. Receivers will collect the VCT's from all of the available broadcast signals and build a master table to properly interpret tuning commands.

Figure 1 gives a simplified example of a VCT. In this example the broadcaster (XYZ Television) operates an NTSC station on channel 5 and has been assigned RF channel 41 for his digital service. He uses the major channel number 5 to identify both his analog NTSC service and his digital broadcast services. Minor channel number "0" identifies the NTSC service and the minor channel numbers 1, 2, 3 and 10 identify four digital services. Note that the minor channel numbers do not need to be in order; they can be chosen by the broadcaster arbitrarily within the range of 1 to 99. Minor channel numbers from 100 to 999 are reserved for other services such as data broadcasting. In addition to its main digital program XYZD on channel 5-1, XYZ Television is broadcasting a news program XYZ-N on channel 5-2, a sports program XYZ-S on channel 5-3 and is providing an audio-only music service XYZ-M on channel 5-10. Although the two parts of the two-part channel number are shown in this paper as being separated by a hyphen, the PSIP Standard does not define a delimiter. A *de facto* standard will no doubt evolve for printed program guides, probably a hyphen in the author's opinion. Consumer electronics manufacturers, however, are likely to use a number of different schemes, just as they do today

Major Channel Number	Minor Channel Number	Short Name	Carrier Frequency (MHZ)	Channel TSID	MPEG Program Number	Service Type
5	0	XYZ	77.25	0x0AA0	0xFFFF	analog
5	1	XYZD	632.31	0x0AA1	0x00F1	digital
5	2	XYZ-N	632.31	0x0AA1	0x00F2	digital
5	3	XYZ-S	632.31	0x0AA1	0x00F5	digital
5	10	XYZ-M	632.31	0x0AA1	0x00F7	audio only

Figure 1. Simplified Example of a Virtual Channel Table

to distinguish between single-digit and two-digit channel numbers.

The Channel TSID is the MPEG Transport Stream Identifier for this broadcast signal, the digital signal being transmitted on RF channel 41. It is a 16-bit number which must be unique to each television station. It is expected that the FCC will assign these numbers and it has been recommended that only odd numbers be used (least significant bit always equal to "1" as shown in Figure 1). This will allow broadcasters to use the related even number (least significant bit equal to "0") to identify their analog NTSC signal if they choose to include a 16-bit Transmission Signal Identifier in an XDS packet in accordance with EIA Standard EIA-752.

The program number associates the virtual channel with the MPEG-2 Program Association and Program Map Tables. For the analog service this number has no meaning and is always forced to the value 0xFFFF. Each digital virtual channel listed in the VCT of a terrestrial broadcast signal must have an associated Service Location Descriptor that carries information about the PID's that apply to the program being carried on this virtual channel. This feature will facilitate faster acquisition of the program by the receiver. Optionally, each virtual channel listed in the VCT may have an Extended Channel Name Descriptor to allow the broadcaster to provide a name for this channel that is longer than the 7 characters allowed in the VCT for the short channel name.

Major Channel Numbers

Annex B of the PSIP Standard provides an algorithm for assigning major channel numbers to terrestrial broadcasters in the United States:

- Broadcasters with existing NTSC licenses will use their current NTSC RF channel number as their major channel number.
- New broadcasters without existing NTSC licenses will use the number of the RF channel assigned to them by the FCC for digital transmission as their major channel number.

These two provisions will assign major channel numbers between 2 and 69 to all digital television stations in a manner that guarantees no duplication within a market. It also allows existing broadcasters to preserve their channel number brand identity regardless of the RF channel number assigned by the FCC for their digital transmissions.

Since the PSIP Standard allows the use of any number from 1 to 99 as the major channel number for terrestrial broadcast, the numbers 1 and 70 through 99 are available for other use. For example, a broadcaster might lease a portion of his digital broadcast stream, say, one standard definition virtual channel, to an educational institution or a religious organization, that has no transmission facilities of its own but wishes to be identified by a major channel number. The PSIP Standard is flexible enough to permit this in that both the major and minor channel numbers can be independently specified for each virtual channel in the VCT. However, if this capability is to be used, coordination would be required within each market to avoid conflicts.

Another example of the use of more than one major channel number in a VCT is cable TV. Because of the higher guaranteed signal-to-noise ratio on cable TV, a single 6-MHZ RF channel can carry two digital terrestrial broadcast signals. The VCT for the double-rate digital cable TV stream would list the virtual channels for both of the broadcast signals using the exact same major and minor channel numbers assigned by the broadcasters. The carrier frequency would, of course, be different, corresponding to that of the RF channel being used on the cable TV system to carry this signal. Other entries in the VCT would also be different.

Event Information Table

Another important table defined in the PSIP Standard is the Event Information Table (EIT). The purpose of the EIT is to list all of the events or programs that occur on each virtual channel over a specified time period. The television receiver can cache this information and use it to build an on-screen program guide. As defined in the PSIP Standard, there is a separate EIT for each virtual channel and the time period covered by each EIT is 3 hours. The start times of each table are every 3 hours on the hour starting at midnight UTC (Coordinated Universal Time). Terrestrial broadcasters are required to send the EIT for the current 3-hour time period and the EIT's for the next three 3-hour periods. Thus, the receiver always has at least 9 hours of advance information available to it. Optionally a broadcaster may transmit up to 128 EIT's providing 16 days of advance program information.

Figure 2 gives a simplified example of an EIT. It is the EIT for XYZ-S from 18:00 UTC to 21:00 UTC. If XYZ-S operates on the East Coast of the United States, this will be from 1:00 PM to 4:00 PM EST. This example shows five events during this 3-hour period. Notice that although this EIT is for the time period from 18:00 UTC to 21:00 UTC, the first event listed has a start time of 17:30. This is because that event will not end until 18:15. It was also listed in the previous EIT. Likewise, the last event listed in this EIT which starts at 20:45 will also be listed in the next EIT since it does not end until 22:45. Each event listed in the

Event ID	Start Time	Length (seconds)	ETM location	Title	Descriptors
31	17:30	2700	01	Automobile Racing Report	Closed Captions Content Advisory
32	18:15	900	00	Pre-Game Warm-Up	Closed Captions Content Advisory
33	18:30	6300	01	Basketball Live	Content Advisory
34	20:15	1800	01	Sports Round-Up	Closed Captions Content Advisory
35	20:45	7200	00	Ice Skating Championships	Content Advisory

Figure 2. Simplified Example Event Information Table

EIT may have an associated Extended Text Message (ETM) that can provide a textual description, typically a few sentences long, of the event. The 2-bit field, ETM location, in the EIT signals whether there is an ETM associated with the event and if there is, where to find it. In the example, events 32 and 35 do not have ETM's; the other three events do have ETM's.

The PSIP Standard defines two descriptors that may be associated with each event listed in the EIT. The first is the Caption Service Descriptor which provides information about the closed captioning, if present, associated with the event. This information includes the number and type of closed captioning services present and the language code for each service. The second descriptor is the Content Advisory Descriptor which indicates the ratings that apply to this event. In the example of Figure 2, there is a Closed Caption Descriptor associated with events 31, 32 and 34. A Content Advisory Descriptor is associated with all of the events. In addition to these two descriptors, the AC-3 audio descriptor defined in ATSC Standard A/52 as constrained in Annex B of ATSC Standard A/53 may be included in the EIT.

Other PSIP Tables

In addition to the VCT and the EIT, the PSIP Standard defines three other required tables and one other optional table. A required table is the System Time Table (STT) which provides current date and time of day information. The time is specified as the number of GPS seconds since midnight, January 6, 1980. The current offset in whole seconds between GPS time and UTC time is also provided and a means to control the transitions into and out of Daylight Savings Time is provided. Another required Table is the Rating Region Table (RRT). The RRT carries the definition of the content advisory or rating system that applies to a specific geographic region. The PSIP Standard defines the United States (50 states plus possessions) as a rating region and provides the means to carry the definition of the United States rating system, but it does not define the rating system itself. Once the FCC finally approves a rating system for the United States, a separate document will be needed to define the exact contents of the RRT.

The third required table is the Master Guide Table (MGT). The MGT is an index to all of the other tables except the STT which is independent. Except for the STT and the MGT itself, there is an entry in the MGT for each table currently being transmitted. For each table listed the MGT provides the PID of the transport packet carrying the table, the version number of the table and the size of the table in bytes. By monitoring the MGT, a receiver can readily determine if a table has been added or dropped, or if a table has been replaced by a new version. Knowing the size of the tables helps the receiver allocate memory to store them.

The MGT also plays an important role in allowing for future extensions of the PSIP Standard. If additional tables are defined in the future, e.g., to provide information about data broadcasts or interactive services, these tables will be listed in the MGT, allowing a receiver that recognizes them to manage its storage and updating functions in a coordinated fashion.

The optional table is the Extended Text Table (ETT) and like the EIT's, there can be multiple ETT's. ETT's carry Extended Text Messages (ETM's). Thus, they can be associated with events listed in EIT's to provide an extended textual description of the event. ETT's can also be associated with the VCT to provide an extended textual description of a virtual channel.

CONCLUSION

The PSIP Standard provides a unified solution to naming, numbering and navigating the digital channels that satisfies the important requirements of broadcasters, cable TV operators, consumer electronics manufacturers and consumers. Broadcasters can retain their channel-number branding and they can group their digital and analog services under a common major channel number. Channel numbering and naming can be harmonized between terrestrial broadcast and cable TV. Consumers will be able to directly access programs by entering the two-part virtual channel number. Viewers can use printed program guides or on-screen electronic program guides. The presence of the PSIP information in the digital broadcast signal will provide consumer electronics manufacturers with a wide variety of options for implementing user-friendly navigation paradigms. Finally the PSIP Standard can be readily extended to encompass data broadcasting and interactive services.

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Note: The Advanced Television Systems Committee Standard - Program and System Information Protocol for Terrestrial Broadcast and Cable is included in the last section of this publication.

Advanced Digital Television Implementation

An Engineering Perspective on a Real-World Project

Garrison C. Cavell, Joseph Davis, P.E. and Thomas L. Mann
Cavell, Mertz and Perryman, Inc.
Consulting Engineers
Fairfax, VA – Los Angeles, CA – San Antonio, TX

Abstract

KITV, Honolulu, Hawaii was the first television station in the United States, which was designed and built from the ground-up as an Advanced Digital Television station. This is a practical account of the blood, sweat, toil and tears involved in it's conception, design and construction.

Present Analog Facility

When Argyle Television agreed to acquire KITV, Honolulu, Hawaii and it's satellite stations, KMAU, Wailuku (Maui) and KHVO, Hilo, in December 1995, it knew from it's due diligence inspection that the property lease for the studio headquarters was up in less than three years, and that there was little choice except to relocate the station's studio and office facility to a property with double the existing space, room to grow, and adequate parking. The real question, other than locating a suitable and affordable property, was what to do about advanced Television (ATV).

The existing studio building contained 15,000 square feet of awkwardly constructed space, grown haphazardly through the 1960's, 1970's and 1980's. There was there was simply no land to make any sensible add-on. The only realistic solution at the old site was to split the facility into two (or more) parts, and that made no operational sense to anyone.

The old landlord was one of the large landholders on Oahu, with large shopping centers to nurture, and little reason to offer their television station tenant any cost-effective alternative. In fact, the landlord was down right convinced that the TV station tenant was not paying it's fair share of the profits from their enormous landholdings.

Rationale for Near-Term Conversion to Digital

KITV reasoned that whether the FCC adopted ATV in 1997 or some subsequent year, it would ultimately be adopted, and therefore, all technical and production planning for the new studio and technical facility must take into account digital television. All the while, aspect ratios, scanning formats and numbers lines were occupying the technical world in great debate when this project was undertaken in 1995, but for the designers, only three thing really mattered: NTSC, HDTV and SDTV. This facility had to accommodate all three and it had to be able to originate more than one at a time.

For KITV, digital was an easy decision: almost all of the analog technical plant was approaching 20 years of age, and had been kept operating through the bankruptcy of the previous owner. Only a few pieces of equipment were even worth carrying to a new plant, whether that plant was analog or digital.

Analysis showed that the cost curves of digital and analog studio equipment were crossing, and that analog equipment was no longer cheaper. All of the existing equipment which would be brought to the new plant either had digital inputs and outputs or could be equipped with a digital I/O card easily and cheaply. A study was made of the analog world which had to interface to the plant: the ABC network, remotes, ENG pickups, satellite feeds, etc. It was determined that a certain amount of conversion equipment would residually be required to get in and out of the digital world.

The station quickly came to the conclusion that the basic station infrastructure would be digital, that is, in accordance with the CCIR-656 serial component standard (known to most as "601" digital), with analog islands where required, maintained at a bit rate capacity of 360 mB/sec., which would be then converted into and out of digital as required. So, the master control, router, studio control, cameras, graphics, commercial insertion and net delay would all be digital. The largest single consideration the station had to make was tape format, which was a hodge-podge at best on acquisition, having one-inch, Beta SP and ¾" U-Matic all in operation.

Planning Factors

A space budget was developed with the help of Frank Rees and his associates, which was essentially an update of an earlier study done by that firm for a prior owner. A variety of properties were investigated in Honolulu over a one year period, but few were suitable for the station's purposes. The market situation was that everything was either too small or two big for the station's needs, or else it was located too far to the west to be convenient for news coverage with Honolulu's horrible traffic jams.

Two properties continued to resurface in our study: a new high-rise condominium project then under development (which was a purchase) and a shopping center/office complex that was in the process of failing (which was a leasehold). After intense negotiations with each property, and concurrent engineering and architectural studies of the candidate properties, KITV decided that the purchase of the ground floor in the Myers Corporation's ***One Archer Lane*** condominium project offered the least cost over the life of the studio as well as a more flexible building.

Although there is little rhyme or reason for it, most television studio buildings are kept in operation long after their useful lives have been exceeded. KITV's old studio was built in 1953, and still operating for the station, even though people were almost stacked on top of each other.

The condo project's architects, Mel Choy and Bill James of Media 5 based in Honolulu had given the station some very intriguing tentative designs as part Myers' business proposals, and the station ended up using Media 5 to design the space layout. The total space is approximately 35,000 square feet on four levels, intersplayed much like a split level house due to the fact that the space is in the parking garage portion of the building.

The ground floor houses the lobby, public affairs, promotions and production departments as well as the showcase studio and the *Star Trek* like technical area. The intermediate level contains the news department, graphics, production control room and edit bays. The second floor contains the executive offices and main conference room, which overlook the studio, the sales and traffic departments and the business office. A fourth, lower level of 5,000 square feet is completely unused and available for future expansion.

It was decided at once that the facility should be designed for both multicasting and eventual high definition television. Multicasting is a natural in the Hawaiian Islands, where there are satellite transmitting stations dispersed over three of the eight islands, and where the economic communities on Maui (in particular) and the Big Island are rapidly developing independently of Honolulu.

There is a natural need to provide different news, weather and commercial content, as well as specially localized programming to the neighbor islands.

Studio Planning Issues

The studio technical complex at KITV is made up of four islands of activity which are (or can be) relatively independent of each other:

- Master Control – The core of the television station containing 80% of the technical equipment.
- Production Control/Studio – Studio/control room complex for originating local news programs as well as public affairs and other live or tape delayed broadcasts
- Newsgathering/editing – Field newsgathering, transmission and editing of news stories
- Production Editing – Independent edit suites based upon Avid MediaComposer[1].

These areas are divided along basic departmental responsibilities in the station and logical boundaries. The news department gathers stories and edits them independently of the engineering or production departments. The engineering department runs master control independently of production or news, and so forth.

The collaborative effort, of course, comes about daily when the three departments come together to produce and air the local news broadcasts.

"People factors" were very important to the design. The facility was designed to originate multiple program streams regardless of the format. Certainly all stations implementing ATV will have to originate both NTSC and ATV for some years during the transition. We believe that many stations will end up 'multicasting", doing their NTSC stream and multiple SDTV streams during parts of the day, and probably HDTV pass-thru from their networks during prime time. It becomes critical, considering these probabilities, for stations to install 'people friendly' and 'people efficient' systems, in order to keep personnel costs from multiplying when you add an additional output channel.

Station automation systems are highly desirable ways of controlling multiple output channels simultaneously, even if only NTSC and ATV, although some disk-based commercial origination systems may well take over that role as time progresses and their software matures.

NEWS:

Early in the game, ownership committed to digital news editing for KITV with a substantial investment in Avid NewsCutters, rather than fostering continued use of tape.

Network delay, a unique situation to affiliates in Hawaii, was already being

[1] The Avid MediaComposer products were already in service at KITV before the new plant was constructed

done with a Philips BTS MediaPool. It was decided to keep things as tapeless as possible, but we also recognized that tape would be a necessity at least into about the 2003 time frame, because the cost of storage for long-form product (i.e. program length components) in disk-based servers is just too high for the present.

A digital tape format would have to be selected. In the end, after great consideration, only two formats were economically acceptable: DVC-Pro and Digital SVHS.

Hawaii is a tropical environment, with consistently high humidity and temperature, plus the salt-air atmosphere of being surrounded by the Pacific Ocean. In the end, questions about how the narrower tape of DVC-Pro would behave in the oceanic environment and the fact that DVC-Pro50 (50 mB/sec data rate) was not going to be available until NAB 1998 (about 6 months past when the station needed delivery) persuaded the station to go with JVC's Digital S-VHS, which already had the 50 mB/sec data rate and ½" tape. The station also had prior excellent experience with JVC from a service point of view. The station was able to convert it's newsgathering format to Digital S-VHS at the same time.

So, for the preset, spot, news and program storage at KITV rely on a combination of Disk based storage and digital tape.

In the future, the station hopes to integrate a mass storage device such as Storagetek to the Philips MediaPools so that long-form materials can be shuffled in and out of the MediaPool for air playback and tape management as well.

Digital Infrastructure – the heart of it all

The secret to a successful integration is how the interface between analog and digital systems is handled.

In most cases, the routing switcher, master control switcher, machine control system and production switcher make up the video technical infrastructure. The vast majority of the video sources in a station come into the routing switcher. In KITV's case, all external feeds (ABC network, remote microwave receivers, fiber optics from GTE and other vendors) are converted to digital and brought into time with the station as they come into the building. Outgoing signals are kept digital to the transmitter, with the exception of the NTSC signal which is converted to analog just before the transmitter input.

Some sort of digital infrastructure is required even in stations with less aggressive digital plans than KITV's, though, and in those cases, where considerable analog equipment maybe retained well into the future, a routing system with smaller analog and digital grids may be economically employed, utilizing pathfinding between the two worlds. This technique allows a smaller number of Analog to Digital (and Digital to Analog) converters to be dynamically allocated by the switcher/machine control software on an as-needed basis rather than a hardwired basis. Although somewhat more expensive initially, where appropriate, pathfinding pays for itself in short order.

Properly planned, this approach allows smaller stations to have gradual evolutions to digital, at least as far as the digital infrastructure is concerned.

What Digital Infrastructure provides

Most network affiliate television stations made up of what can be thought of as technical islands:

- Master control, where the network and local programming are switched and interspersed with commercial content, essentially, the boiler room which gets the product on the air to the customer
- Production Control, which packages news and local production content into a live stream to be passed through the boiler room to air or recorded for later telecast.
- News gathering and editing, which shoots, takes the stories in and edits the packages to take to air, usually in live news broadcasts

In converting a station to digital, the Master Control island is the one which must be converted to digital first. Otherwise, we couldn't get an ATV signal on the air. In small stations, this may at first consist only of a smart digital bitstream splicer to insert local NTSC commercials and programs, up-converted to SDTV, into the digital bitstream provided by the network.

In KITV's case, the old master control island was so old that it needed replacement, so a fully integrated digital solution from Philips/BTS was specified. This simplified the design of the digital infrastructure considerably. Since the routing switcher was to be the central core of the station, most of the initial design consisted of allotting sources to the input side of the router. In the Philips system, the master control switcher(s)[2] is(are) virtual, in that the switcher's matrix is actually the entire matrix of the router, dynamically allocated to the Saturn operator control panel, therefore, all inputs to the router are available to go on-the-air.[3]

Similarly the Digital Diamond production control switcher *can* be configured in the same way, wherein all inputs are dynamically allocated from the router. In KITV's case, 10 of the thirty inputs to the DD-30 are dynamically allocated, the remaining twenty are hardwired from the sources (i.e. cameras, chyron, still store, etc.).

This methodology requires that the router be realistically sized for the present and have room for future growth. The KITV router is 128 x 128 and the chassis is sized to handle cards for 196 x 128. Many readers will undoubtedly gasp when they hear the size of this matrix, however, you must remember that crosspoints are cheap in the beginning, never wear out, and you can never have too many! KITV filled up the entire 128 inputs in the first months of planning. We probably should have gone for 196 inputs initially!

What this kind of digital infrastructure give a station is plenty of headroom to grow. The Philips CCIR 656 router will pass 360 mB/sec. data rates, even though the present requirement for components serial digital SDTV is only 270 mB/sec. This is also added headroom for growth. It is widely believed that present day 360 mB/sec. routers will accommodate full HDTV (1200 mB/sec uncompressed data rate) at a modest 4:1 compression. In the

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[2] one switcher electronics assembly is required for each output channel

[3] In the Jupiter Machine Control system, even though all inputs are available, the software can be easily configured to preclude any source from ever being put direcly on-air, such as "color bars and tone".

present scheme of things, 1200 mB/sec. is not a realistic thing to pass around even a modest size studio plant.

Local origination – Production Concerns

The second island of digital is local origination, which is more or less the production control room, window-wall studio fronting on the corner of King Street and Archer Lane and all associated equipment (i.e. studio cameras, Chyrons, graphics, stills, effects, etc.).

This facility is used on a daily basis for production of local news programs, promotion production, production of local commercials, production of shows of local interest and some outside paid production.

KITV is a rarity in that it produces a significant number of hours each year of locally produced programming for telecast to the unique island State of Hawaii. Among these programs are "The Merrie Monarch" festival and the "Kiki Hula" contest, both of which have incredibly high viewership in the state.

KITV has chosen 480I, 16 x 9/4 x 3 aspect ratio SDTV as their initial digital local origination standard. This is effectuated through the employment of Philips LDK-10 switchable aspect ratio cameras and the DD-30 production switcher.

A major source of concern is how to handle the fact that for the moment all viewers have a 4 x 3 aspect ratio set and the station is capable of producing perfectly wonderful 16 x 9 aspect ratio pictures!

Digital Radio Frequency Transmission Issues

From a licensing and coverage point of view, the digital coverage of KITV actually consists of the coverage of KITV-DT, licensed to Honolulu, KHVO-DT, licensed to Hilo on the Big Island, and KMAU-DT[4], on Maui, the later two stations operated as satellite stations of KITV. KITV-DT covers the City of Honolulu and most of the island of Oahu from which it is not terrain shielded, which is where greater than 80% of the population of the state of Hawaii is located. KMAU-DT, from the 10,000 elevation of Haleakala, will cover most of the areas of Oahu which are terrain shielded from KITV-DT, as well as Maui, Molokai, Lanai and the Kona coast of the Big Island. KHVO will cover the coastal Hilo area.

These existing transmitter sites, converted to digital by the addition of modestly powered solid-state UHF DTV transmitters and broadband, carefully engineered directionalized wideband panel transmitting antennas achieves the goal of covering greater than 99% of the population of the state of Hawaii with Digital television at the outset of the Advanced Digital Television technology.

Real-world (terrain limited) coverage of the three ATV stations is shown in Figure 1, and the NTSC real-world coverage is shown in Figure 2.

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[4] KMAU-DT is not in operation as of this writing due to a controversy between the television broadcasters which have been transmitting from atop Haleakala, the extinct volcano on mauy isnce the early 1950's and the University of Hawaii's Astronomy department, which built optical observatories several hundred feet from the television stations much later in time.

Transmitter, Feeder System and Antenna Selection

The selection of transmitter, antenna and feeder system were treated as a simultaneous solution matrix in this case. DTV Transmitter availability was and is a questionable subject. The development of antenna patterns was done by Gary Cavell, Joe Davis and Tom Mann of Cavell, Mertz & Perryman in association with Ray Tattershall of LeBlanc & Royle/LDL of Canada and RFS of Australia. All three antennas are broadband UHF panel designs, installed as arrays which produce specific azimuthal and vertical radiation patterns engineered to cover the specific population centers which each site can "see".

Because of nature's layout of the islands, relatively high pattern gains can be used with very small vertical apertures. Two of the three antennas are only one panel high, the other is two panels high, thus avoiding beamscanning or other artifacts of antennae employing large vertical apertures for gain. The broadband nature and corporate feed system of the panel type of antenna also insures low group delay, which we believe to be a critical factor in error free DTV transmission.

Feeder systems employing continuous, semiflexible line with precision connectors was employed in all cases, and tuned line sections or elbows were used where required to provide a flat system. We believe that transmission line issues in DTV require an analysis of the compromise between power handling and efficiency versus bandwidth and group delay.

Where efficiency is not paramount, semiflexible, continuous line provides the easiest and best technical solution in many cases. When extremely high powers are required, efficiency becomes a much more serious issue and bandwidth and group delay may have to be compromised to safely accommodate high power levels.

When pressed with the threat of an actual order, most of the major North American transmitter manufacturers will tell you that they won't be able to deliver a production solid state transmitter of the 1 – 10 kW (Average DTV) power level until at least the 3rd quarter of 1998, and only then if the semiconductor manufacturers deliver the Silicon Carbide or LDMOS transistors on time. We didn't have the time to wait in KITV's case. The owners were intent on being on-the-air at the same time as then studio plant was inaugurated.

We considered tube transmitters, but felt an air cooled IOT would be impractical at the 10,000 foot altitude of the Haleakala transmitter site on Maui. The two lower powered transmitters at Honolulu and Hilo needed a power growth path if that proved to be necessary as viewers began buying DTV sets. Besides that, the Honolulu and Hilo transmitter sites simply had no room for a tube-type transmitter.

Another consideration was even more gut-level… what 1990's engineer, faced with a tube versus transistor choice for the amplifier device in a transmitter would voluntarily choose the tube alternative if the relative operating costs were similar? I suspect few but the die-hard tube creatures who still sometimes haunt transmitter sites. The US experience with solid state analog TV transmitters since 1988 has been remarkably good!

We had seen Itelco at the NAB for a number of years and they began to catch our attention two years ago with innovative design and a very high level of craftsmanship. At the 1997 NAB, we were intrigued by the liquid cooled solid-state design they offered and the fact that more conventional solid-state devices worked quite well when the junction temperatures were held 30 degrees (Centigrade) lower than air cooled devices could be. After a long series of conversations with Itelco's US President, Howard McClure, we decided to consider this transmitter line for the Hawaii DTV project.

The advantages were significant:

- Solid State design with popular and available BiPolar amplifier devices[5]
- Long life components
- Few of the problems associated with air cooling at high altitudes.
- Since cooling system is a closed loop of distilled water, there is less opportunity for the tropical climate (salt air, high humidity, warm) to shorten component life through corrosion
- Comparably efficient to tubes when all factors are considered.
- Relatively low operating cost due to solid-state design

Studio – Transmitter Link (STL) and Microwave considerations

The microwave studio/transmitter link between Honolulu, Maui and Hilo involves great heights at path ends, multiple long over the water paths and a grumpy environment. In conjunction with the Philips TokenMux digital multiplexer described elsewhere, the station required a 45mB/sec. Data path between the Studio site in Honolulu and the following locations: the Ala Moana Hotel Honolulu transmitter site, the Haleakala transmitter site on Maui, and the Saddle Ridge relay site on the Big Island. From the Saddle Ridge relay site to Hilo is an easy shot.

The 45 mB/sec data path we required is alternatively called a "DS-3 circuit" or in Europe, a "G-703 circuit", and can be provided in a microwave radio circuit (in single and bi-directional flavors), over short distances on copper, and by fiber optical circuits. It was desirable in the case of the Hawaii DTV project that both radio circuits and fiber circuits be able to be used, possibly redundant to each other at some point in the future, since fiber is prolific between the islands not only from GTE, the local telephone utility, but from other, independent sources as well.

The Philips TokenMux equipment gathers the analog NTSC video and stereo audio at the studio site as well as one or more channels of SDTV and/or HDTV and multiplexes them into a DS-3 bitstream. This bitstream is then handed off to a 45 mB/sec modem which is part of the Microwave system, which modulates the 45 mB/sec radio sets at 16QAM. On the receiving end, the receiving modem then hands the DS-3 stream back to the TokenMux (which may also pass-along an unmolested copy of the bitstream to another modem/radio set to go to a further transmitter site) which: 1) demodulates part of the stream in to NTSC video and audio to feed to the NTSC transmitter and further transcodes the stream into the ATSC 19.3 mB/sec. DTV stream to be fed to the digital transmitter.

Did it work? – Post Mortem

Remarkably [since it really is the first commercial digital television station, and on top of that, the only one at this writing

[5] The solid state transistor pairs used in the transmitters are available from at least 3 manufacturers.

which actually transmits real programming from real-world sources], it actually does all work! It wouldn't be fair to say that one facet was more difficult than any other, because they were all first-of-a-kind, but the biggest challenge was the TokenMux and microwave system, which was shepherded into existence (much against Mme. Pele[6]'s will) by David Bird of Philips for the TokenMux and Rich Sweitzer, General Manager and Director of Engineering at WNAC in Providence, R.I. and Howard McClure of Itelco for the microwave portion.

KITV is today transmitting simultaneous and independent NTSC analog and digital signals from one single master control room to over 99% of the population of the State of Hawaii.

The two program streams are independently derived at the studio site and independently conveyed to the transmitter sites. Under TokenMux supervisory control, differing program streams may be fed to different populations on different islands, making the microwave conveyance a unique and flexible tool for coverage of a unique place in this world!

-0-

Garrison C. Cavell is the senior partner in Cavell, Mertz and Perryman, Inc., Consulting Engineers, Management and Technology consultants headquartered in Fairfax, Virginia. Gary has been a broadcaster for more than 25 years, and has been a station owner and general manager as well as having been engaged in broadcast engineering as a consulting engineer, chief engineer and director of engineering.

A New Orleans native, Gary is a Senior member of the Society of Broadcast Engineers, the National Society of Professional Engineers and the Institute of Electrical and Electronic Engineers, of which he is currently President of the IEEE Broadcast Technology Society.

Joseph Davis is a partner in Cavell, Mertz and Perryman, Inc., Consulting Engineers, Management and Technology consultants headquartered in Fairfax, Virginia.

Joe is a native Virginian, and is a Registered Professional Engineer in the Commonwealth of Virginia and a member of the Society of Broadcast Engineers, the Association of Federal Communications Consulting Engineers and the Institute of Electrical and Electronic Engineers

Thomas L. Mann is a partner in partner in Cavell, Mertz and Perryman, Inc., Consulting Engineers, Management and Technology consultants, and runs the West Coast office in Los Angeles.

Tom is also a native Virginian, and was up until September of 1997 Vice President of Engineering and new Technology for Argyle Television, the licensee of KITV/KMAU/KHVO. He was Director of Engineering for the Walt Disney Company's KCAL-TV in Los Angeles, and Director of Operations and Engineering for NBC in Washington, D.C.

Tom has practiced engineering for more than 30 years, and is a member of the Society of Motion Picture and Television Engineers, the Royal Television Society, and the Institute of Electrical and Electronic Engineers, of which he is currently the Editor-in-Chief of the IEEE *Transactions on Broadcasting*.

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[6] For those who are not familiar with Hawaii, Mme. Pele is the volcano Goddess.

ENG/EFP: The Next Generation

Thursday, April 9, 1998
9:00 am - 12:00 pm

Chairperson:
Robert Hess
WBZ -TV, Boston, MA

Digital Video Microwave Systems for STL and ENG Applications and Test Results
Dr. John B. Payne
NUCOMM, Inc.
Hackettstown, NJ

***ABC & Loral Skynet DSNG Trial**
Tom Geiges
Loral Skynet
Holmdel, NJ

***HDTV Delivery Using Telecommunications Networks**
Gregory Coppa
CBS Inc.
New York, NY

***Trade-Offs and Considerations for Implementing Digital Television Networks**
Misko Popovic
INTELSAT
Washington, DC

Epoch-Making Technologies at the Nagano Winter Olympic Games Start of New Era of Hi-Vision
Noriaki Kumata
NHK Broadcast Engineering Department
Tokyo, Japan

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*Papers not available at the time of publication

DIGITAL VIDEO MICROWAVE SYSTEMS FOR STL & ENG APPLICATIONS & TEST RESULTS

Dr. John B. Payne, President
NUCOMM, Inc.
101 Bilby Rd
Hackettstown, NJ 07840
Ph. 908-852-3700, FAX 908-813-0399
e-mail: john@nucomm.com

Abstract

NAB97, the FCC ruling on deadlines for HDTV, and recent acts by Congress have signaled the dawn of a new era for digital video microwave for broadcast applications, including Fixed Point-to-Point (i.e. STL, TSL, ICRS, etc.) and ENG in both the United States and worldwide.

The broadcast industry is being challenged to explore new technologies to enhance their existing systems and is subsequently turning to digital video technology. A wide array of digital products are currently being offered for many aspects of the production system, such as digital cameras, editors, storage devices, and encoders. However, there has been little discussion about converting the STL and ENG microwave links from analog to digital transmission, which are a critical part of the total production system.

This paper presents an overview of how the digital video microwave technology can be applied to STL and ENG systems and discusses the tradeoffs of digital vs. analog video microwave systems. It also presents actual laboratory and field results of digital microwave STL and ENG tests conducted by NUCOMM.

The results show that applying digital video to STL and ENG microwave systems can conserve frequency spectrum and yield superior quality and performance equal to or better than analog systems under both fading and multi-path environments.

I. Introduction

NAB97, the FCC ruling on deadlines for HDTV, and recent acts by Congress have signaled the dawn of a new era for digital video microwave for broadcast applications including Fixed Point-to-Point (i.e. STL, TSL, ICRS, etc.) and ENG in both the United States and worldwide. The manufacturers of digital encoders / decoders (Codec), multiplexer (MUX), and modulators / demodulators (modem) equipment have little, if any, knowledge of the microwave link requirements. Furthermore, they appear to have no interest in integrating these systems. Therefore, the demand of digital video microwave Fixed Point-to-Point and ENG will require the microwave manufacturer to supply part or all of a turnkey package including the transmitter / receiver equipment, Codec, MUX and modem components. This offers an excellent opportunity, as well as, a challenge for the manufacturers of digital microwave equipment to move into a new and expanding market area.

As a result, it has become increasingly important for TV station engineers to know the advantages, disadvantages, and tradeoffs of digital vs. analog video microwave systems. The purpose of this paper is to:

- Present an overview of how the digital video microwave technology will be applied to STL and ENG systems,
- Present actual laboratory and field results of tests conducted by NUCOMM, using digital video microwave systems in STL and ENG applications.

Conclusions: Applying digital video to STL and ENG microwave systems can conserve frequency spectrum and yield superior quality and performance equal to or better than analog systems under both fading and multi-path types of environments.

II. Digital Video Microwave Architecture

The modulation and type of microwave radio required to transmit digital video information is considerably different from that used for analog video transmission. Figure 1 shows a simplified block diagram of a digital microwave system for single channel per carrier transmission. The encoder first digitizes and compresses the video and audio input signals, then multiplexes the compressed bit stream with the digital data inputs to produce a digital transport stream. The encoder output is in either serial or parallel form and the data rate is typically in the range of 1.5 to 34 Megabits per second (Mbps), depending on the amount of

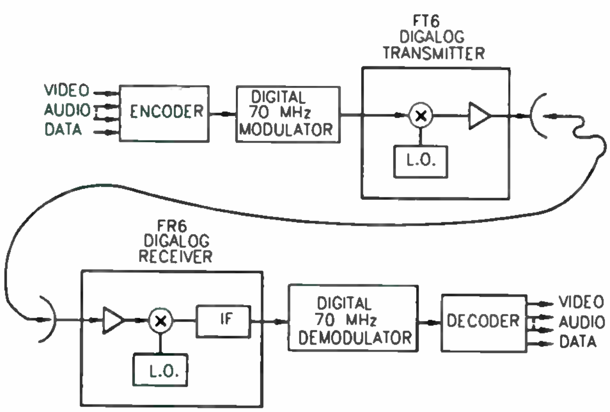

Figure 1:
Single Digital Video Heterodyne System

compression and forward error correction (FEC). The digital modulator converts the baseband signal to a 70 MHz RF signal. Typical digital modulation techniques that are widely available in commercial use are QPSK, multiple level PSK and QAM. They use a combination of phase and amplitude to modulate the 70 MHz signal. Some manufacturers integrate the encoder, multiplexer, and 70 MHz modulator into a single piece of equipment. The 70 MHz digital modulator output is upconverted (heterodyned) to the RF microwave frequency and amplified in a linear type RF amplifier. The RF microwave signal is sent directly to the antenna or diplexer with other microwave signals.

At the receiver, the reverse process takes place. The RF signal is received, down converted to 70 MHz, and demodulated to produce the compressed data stream. The decoder then decompresses the signal to generate the final video, audio, and data. The receiver, digital demodulator and decoder functionality's are generally combined into a single unit called an Integrated Receiver Decoder (IRD). These units typically accept either a 70 MHz or L-band RF signal.

The system is easily scalable to compress and transmit multiple channels on a single carrier. To support multi-channel transmission, only additional encoders and a multiplexer are added. The multiple encoder outputs are combined by a multiplexer that outputs a digital stream at a rate equal to the sum of the input data streams. Thus if two encoders output 15 Mbps and 10 Mbps respectively, then the multiplexer output will be 25 Mbps. The multiplexer output data stream is fed to the digital modulator that converts the signal to a QPSK or QAM phase and amplitude modulated 70 MHz signal. The microwave heterodyne transmitter upconverts the 70 MHz to the desired operating frequency for multiple channel per carrier transmission.

Today there are at least ten manufacturers of digital encoders and decoders. There are many more that are just starting development. They offer a wide range of compression systems depending on the requirements of the user. Some Codecs implement proprietary compression algorithms and others implement standards such as MPEG-2 and ETSI. Commercial use of digital video compression has only really emerged since the early 1990s. Prices for digital compression equipment have decreased significantly in the past year as more users have begun to embrace the technology. Prices will only come down over time as standards evolve and as competition continues to increase.

III. NTSC and HDTV Dual Channel Studio-to-Transmitter Link (STL)

Figure 2 shows how an NTSC and an HDTV transport stream can be simultaneously transmitted over a single microwave link from the studio to the transmitter. The NTSC composite signal is digitized and compressed to the desired output rate, 17 Mbps in this example. The 17 Mbps compressed NTSC signal is combined with the 19.39 Mbps HDTV transport stream to yield a multiplexed output rate of 36.39 Mbps. Using a QPSK modulator, the required transmission bandwidth is approximately 22 MHz. Using 16-QAM, the bandwidth is further reduced to 11 MHz.

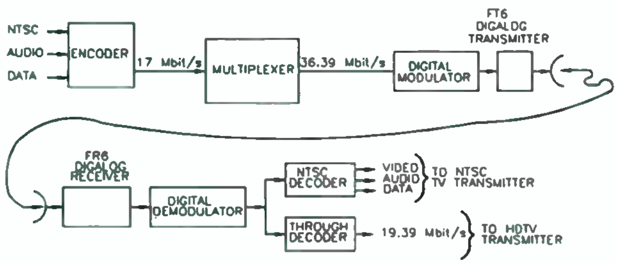

Figure 2:
Dual Channel Digital Video STL for NTSC and HDTV

At the receiver, the signal is demodulated and applied to two decoders. The NTSC decoder decompresses and outputs the composite video and audio signal to be applied to the NTSC transmitter. The other decoder demultiplexes to the HDTV transmitter. **NUCOMM offers a complete turnkey system for this application.**

IV. DIGALOG - Digital & Analog Microwave System

To meet the broadcasters' immediate need for continued transmission of analog signals today but to be ready for the transition to digital, NUCOMM has developed the DIGALOG FT6/FR6 Radio system for STL applications (and the DIGALOG MMPT6 for ENG applications). The DIGALOG Radio is highly configurable and can operate in either analog or digital mode. Figure 3 shows a block diagram of the DIGALOG FT6 transmitter. Depending on the existing system configuration, optional analog and digital modulators can be added to the two

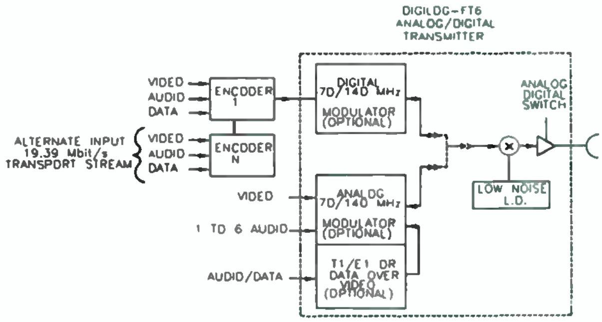

Figure 3:
Analog/Digital Microwave Transmitter DIGALOG FT6

rack high unit. If the input is already a 70/140 MHz modulated IF signal, the IF signal can be directly upconverted and amplified for transmission. By a single switch on the inside of the front panel, the power amplifier can either operate in analog mode for maximum power output or in digital mode for linear power output. The corresponding receiver (not shown here) is the DIGALOG FR6 Analog/Digital Microwave Receiver. Two IF bandwidths, 30 and 45 MHz, are provided in the IF amplifier. The 30 MHz bandwidth filter can be used for analog or low data rate digital operation. For data rates of 45 Mbps or higher, the 45 MHz bandwidth filter is switched in.

V. DATA RATE verses BANDWIDTH for a Digital Video Microwave System

Equation 1.1 below defines the bandwidth required to transmit an encoded bitstream at a given data rate. The transmission bandwidth is a function of input data rate (Za), modulation coding (M), FEC, and Spectrum Shape Factor (α).

$$\text{Bandwidth} = \frac{(1+\alpha)\sum Za\ \text{Mbps}}{FEC * M} \qquad (1.1)$$

where

$\sum Za$ = Sum of data rates from one or multiple encoders (Mbps)

FEC = Forward Error Correction
= VC * RS;

If no FEC is used, then FEC = 1

VC = Viterbi Coding:
Typical 1/2, 2/3, 5/6, 3/4, 7/8

RS = Reed-Solomon:
Typical 188/204, 192/208

M = Coding level of the modulator

α = Spectral Shaping Factor.

Table 1 shows the spectrum efficiency, in Bits/Hz, and carrier-to-noise level (C/N) for various modulation techniques. As M increases, the required transmission bandwidth for a given data rate decreases proportional to Bits/Hz (assuming FEC=1) and the required receive C/N level must increase for a given bit error rate (BER). This is the tradeoff for better transmission efficiency. The most robust and common form of digital modulation is QPSK, which has a spectrum efficiency of 1.66 bits per Hertz. In cases where more bandwidth reduction may be required, higher order modulation such as 16-QAM or 64-QAM can be used to give a spectrum efficiency of 3.33 bits per Hertz or 5.0 bits per Hertz respectively. However, as the coding number increases, the modulation is not as robust and becomes susceptible to RF interference, multi-path effects, etc. Also the system gain decreases substantially due to lower available output power and the requirement for higher receive carrier levels increases for a given bit error rate.

Table 1: Types of Modulation

Type of Modulation	M	Bits/Hz $\frac{M}{(1+\alpha)}$	C/N (dB)
PSK	1	.833	10
QPSK	2	1.66	10
8-PSK	3	2.50	14
16-QAM	4	3.33	17
64-QAM	6	5.00	23
256-QAM	8	6.66	28

Notes: 1-Normalized C/N corresponds to a BER of $1X10^{-6}$.
2-Assumes No FEC and α = .20

In an STL link where signal levels generally tend to be high and the transmission link reliable, the higher forms of modulation can usually be justified. However, in ENG links where multi-path and weak signals are the norm, a very robust modulation such as QPSK is needed. To fit the digital video data rate within the allocated bandwidth, the encoder data rate and FEC need to be adjusted according to the type of modulation technique used. Reducing the data rate with current encoders has little effect on the picture quality as will be shown from the test results given at the end of this paper.

Therefore, it becomes a judgment call on the part of the ENG management to assess the picture quality for given bandwidths. There may be no other option if the 2 GHz allocated bandwidths are further reduced by the FCC.

VI. Typical Analog vs. Digital Performance

Figure 4 shows the performance of an analog link and a comparable digital link for those

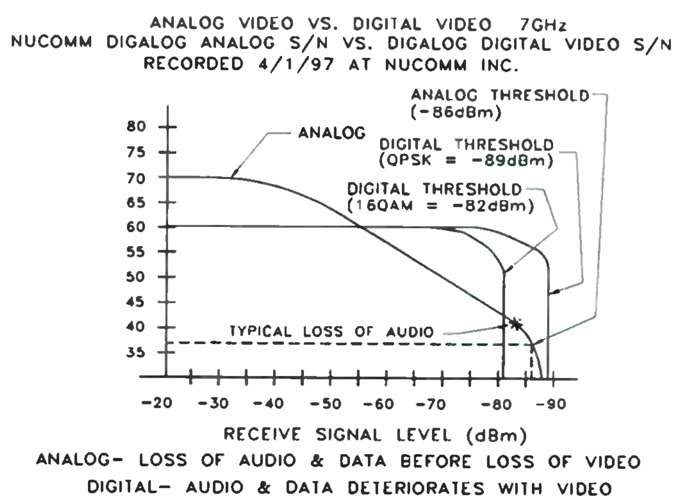

Figure 4:
Video S/N vs. Receiver Signal Level for Analog & Digital Systems

typical system configurations shown in Figures 1, 2, or 3. The analog link shows a video signal-to-noise (S/N) ratio of 70 dB for high receiver input signal levels. As the signal level drops, the video S/N becomes linearly proportional to the input signal level. When the receiver threshold is reached (typically -85 dBm at 7 GHz in current video receivers), the video S/N drops much more rapidly than the receiver input signal level. In a typical analog system, threshold is defined when the video S/N reaches 37 dB. At a receive level of about -82 dBm the audio channels become very noisy and unusable.

The digital link shows a lower S/N than the analog link for strong receive signal levels. This lower S/N is due to the limitations of the digitizer in the encoder. Typically, a 10 bit digitizer will give a S/N of about 60 dB. The advantage of the digital system is that even as the input signal level is reduced, the video S/N remains constant at 60 dB. This S/N is maintained until the error correction can no longer handle the error. The system then crashes and the result is that the video picture freezes. The point at which the S/N "fall off the cliff" is generally at or below the analog threshold point for QPSK. This "fall off the cliff" point depends primarily on the amount of error correction and the type of modulation used.

NUCOMM passed a 45 Mbps QPSK digital signal with error correction through the NUCOMM 7 GHz FT6/FR6 DIGALOG (Analog/Digital) transmitter and receiver. The digital threshold for QPSK (-89 dBm) was 4 dB lower than the systems analog threshold (-85 dBm). Using 16-QAM, the digital threshold (-82 dBm) was higher than the analog threshold by 4 dB. This 7 dB increase in threshold by using 16-QAM instead of QPSK has essentially enabled us to transmit twice the data rate within the same bandwidth.

VII. STL/Line of Sight Experimental Results

NUCOMM tested a digital video microwave system setup as shown in Figure 5. A 45 Mbps QPSK signal carrying five video and audio programs was received from a USSB satellite. This signal was downconverted to 70 MHz and then input to a NUCOMM 7 GHz DIGALOG radio operating in the digital mode. Using a variable attenuator, the transmitter output was reduced so that the input receive signal level was well below the receiver threshold. The 70 MHz receiver output was upconverted to L-band and fed to five IRDs. The output of each IRD was displayed on a color monitor. The IRDs have a built in BER counter that displays the BER as a function of signal strength. A signal strength reading of 100 means that there are no errors being detected. A reading of 10 means that there are many errors. Below this level the system crashes.

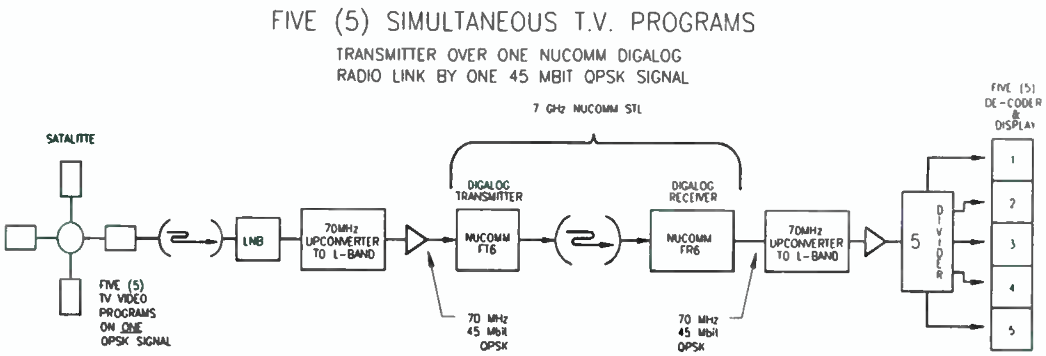

Figure 5: Test Setup for Measuring Digital Video Performance

Figure 6 below shows the measured S/N and BER correction as the microwave received signal strength changes. The analog threshold was measured at -85 dBm, as compared to the digital threshold at -89 dBm. That is 4 dB better than in the analog mode. Just as important is the fact that all five video pictures and audio sounds remained perfect until the digital threshold was reached. The difference in signal level between a perfect picture and a frozen picture was 1 dB.

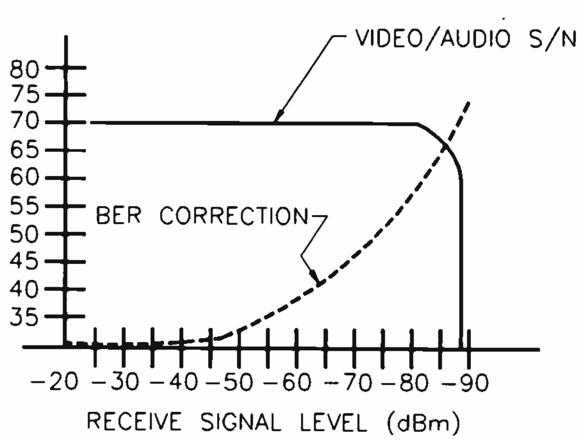

Figure 6:
Experimental Test Setup

VIII. Digital ENG 2 GHz Field Test with Extensive Multi-Path

When the subject of digital video being applied to 2 GHz ENG microwave systems is suggested, the immediate response by many ENG operators is that digital video will never work for ENG. They are typically faced with non-engineered paths where shots are made using multiple bounces in high multi-path environments. It is claimed that these conditions will cause the picture to freeze thus loosing the shot, and some picture is better than no picture.

To start to answer some of these questions, NUCOMM has recently completed field testing of its 2 GHz ENG digital video microwave system in New York City. This city was chosen because it represents one of the most severe and challenging environments for ENG operation. The results as given in this section were startlingly successful to all that participated.

I want to acknowledge and thank the New York City FOX station WNYW-TV and in particular Rich Paleski for providing the ENG truck, and their Empire State Building Central Receive site, as well as studio recording equipment and personnel. Rich is a seasoned New York City ENG engineer well acquainted with the many difficult multi-bounce and multi-path ENG shots in that city. I would also like to recognize and thank the WEGENER Corporation that supplied the encoder and IRD equipment.

Figure 7 shows the equipment configuration for this test. The analog transmitter was a NURAD 10 Watt model PT1 that was padded down for an output of 3 Watts. The digital transmitter was a NUCOMM DIGALOG FT6. Its power output was 1.5 Watts. The power amplifier on the mast at the antenna could saturation and would cause excessive spectrum spreading. The antenna was connected directly to the transmitters through 50 feet of Andrew ½ inch flexible coax and had a measured loss of 3 dB.

The antenna was a NURAD silhouette antenna mounted on a pan and tilt. To ensure stress testing the digital encoder, a difficult 2.5 minute video clip of a pre-recorded hockey game on Betacam-SP was used as source material because it included fast camera panning, fast action, high color contrast, and saturated colors.

At the Empire State Building the output from a steerable Super Quad antenna was divided to feed both the NURAD analog receiver and the NUCOMM DIGALOG FR6 digital receiver simultaneously. The 70 MHz output from the digital receiver was upconverted to L-Band for input into the IRD decoder. Each of the composite outputs from both the analog and digital receive systems was transmitted back to the studio over an analog fiber link where the

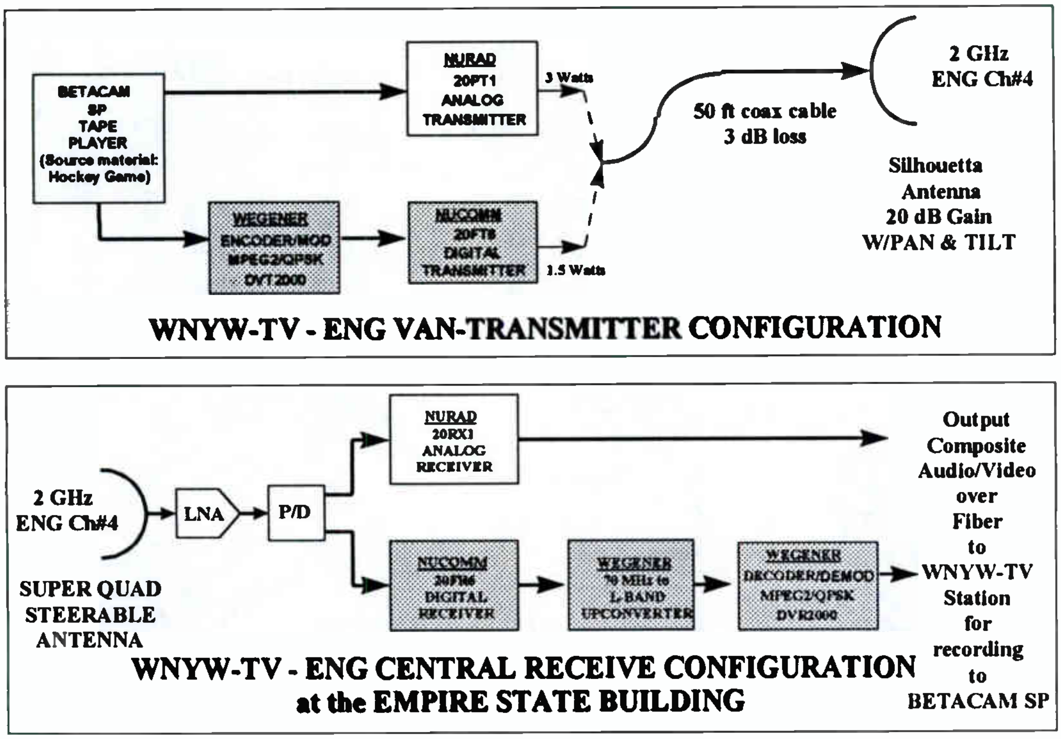

Figure 7
Experimental Test Setup

outputs were recorded on Betacam SP tape. Both transmitters operated on the same 2 GHz channel. Operating on two different frequencies could have negated the tests since multi-path effects would be different.

The first three case tests compared the audio and video quality of the 2 GHz analog FM signal with the quality of the digital MPEG-2 compressed and QPSK modulated signal under the following three environments: (1) direct line-of-sight transmission, (2) moderate multi-path transmission, and (3) extreme multi-path transmission. In each case, the ENG truck was located at E. 90th St. and 5th Ave. The receive site was on the Empire State Building located at E. 33-34th St. and 5th Ave, as shown in Figure 8. The antennas on both the transmitter and receiver sites were steered accordingly to establish the appropriate transmission environment.

The procedure for setting up each test was to first establish the analog shot geometry and picture quality. The resultant analog video picture was recorded for 2.5 minutes. Then, without moving the antennas, the digital transmitter was connected and the test repeated. Each digital case tested data rates ranging from 9 to 15 Mbps and two FEC rates of 3/4 and 7/8.

Test Case 1: Direct Line-of-Sight Transmission

The first test was a line-of-sight shot to make sure that the system was working properly. Both analog and digital transmissions produced overall good pictures for each configuration. Although the analog signal was strong, there were still some multi-paths and ghosting artifacts. The digital picture showed no sign of multi-path or ghosting.

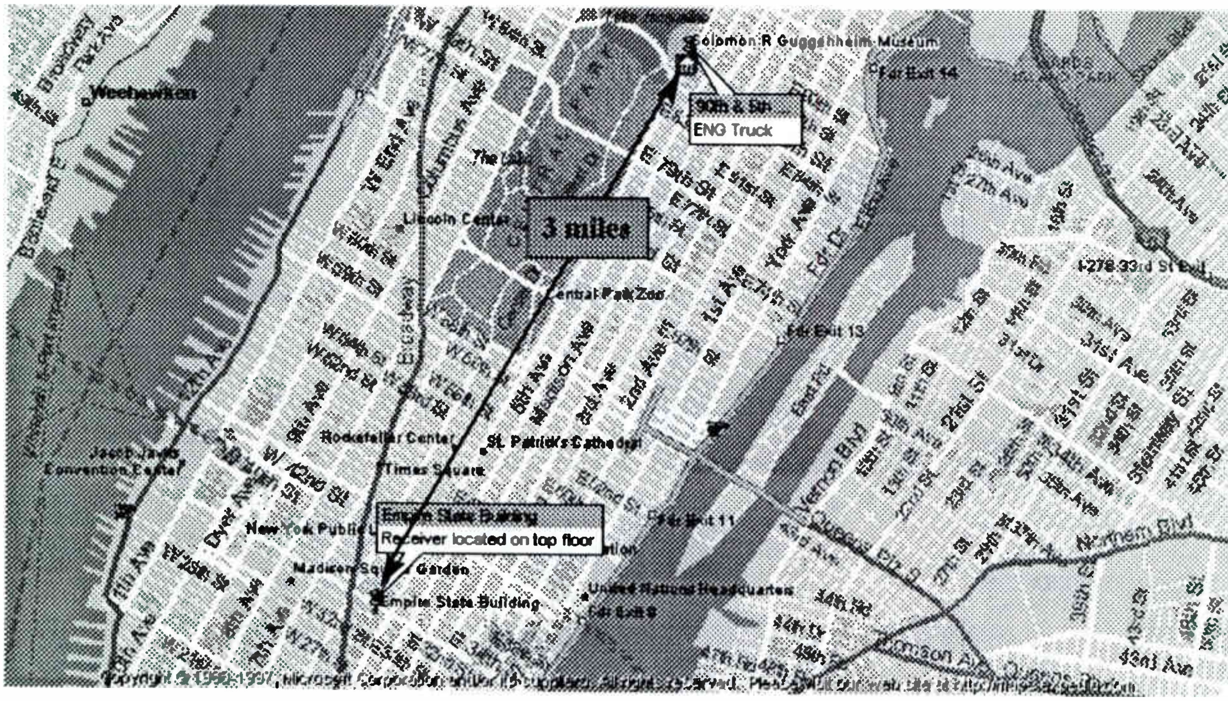

Figure 8
Experimental Test Sight

Test Case 2: Moderate Multi-path Transmission

The second case, moderate multi-path transmission, is representative of typical ENG operating conditions in major urban cities such as New York City where buildings are commonplace obstructions to obtaining direct line-of-sight transmission. The ENG truck was still in the same location as in the first case, but the antenna was moved to 45 degrees off true North so that at least one bounce was introduced in the path. The received signal measure was lower than the first case but still quite strong. The resulting analog signal showed noticeable ghosting artifacts and color shifting. The quality of this analog signal was considered a borderline usable picture for broadcasting. The digital signal, on the other hand, had no problem locking up and performed perfectly with no ghost or indication of multi-path in the picture.

Test Case 3: Extreme Multi-path Transmission

The third case tested extreme multi-path interference comprised of multiple reflections and scattering from buildings and possibly even moving vehicles. Here the ENG truck antenna aimed in the general direction toward the west side of Central Park. The resulting transmitted analog signal was severely degraded to the point where it was not at all usable and was so bad that a frame synchronizer had to be used to receive the picture. The analog video had significant ghosting artifacts and the audio had severe breakup. The studio reported that the picture quality was too poor to broadcast. When the digital signal used an FEC of 3/4, the IRD decoder had no problem locking on the signal and it produced a perfect picture.

The hockey picture shown below in Figure 10 is a split image of the same video picture transmitted in both analog and digital modes.

Picture Quality of Analog vs. Digital (9 Mbps, 3/4 FEC) Transmission for Case 3: Extreme multi-path

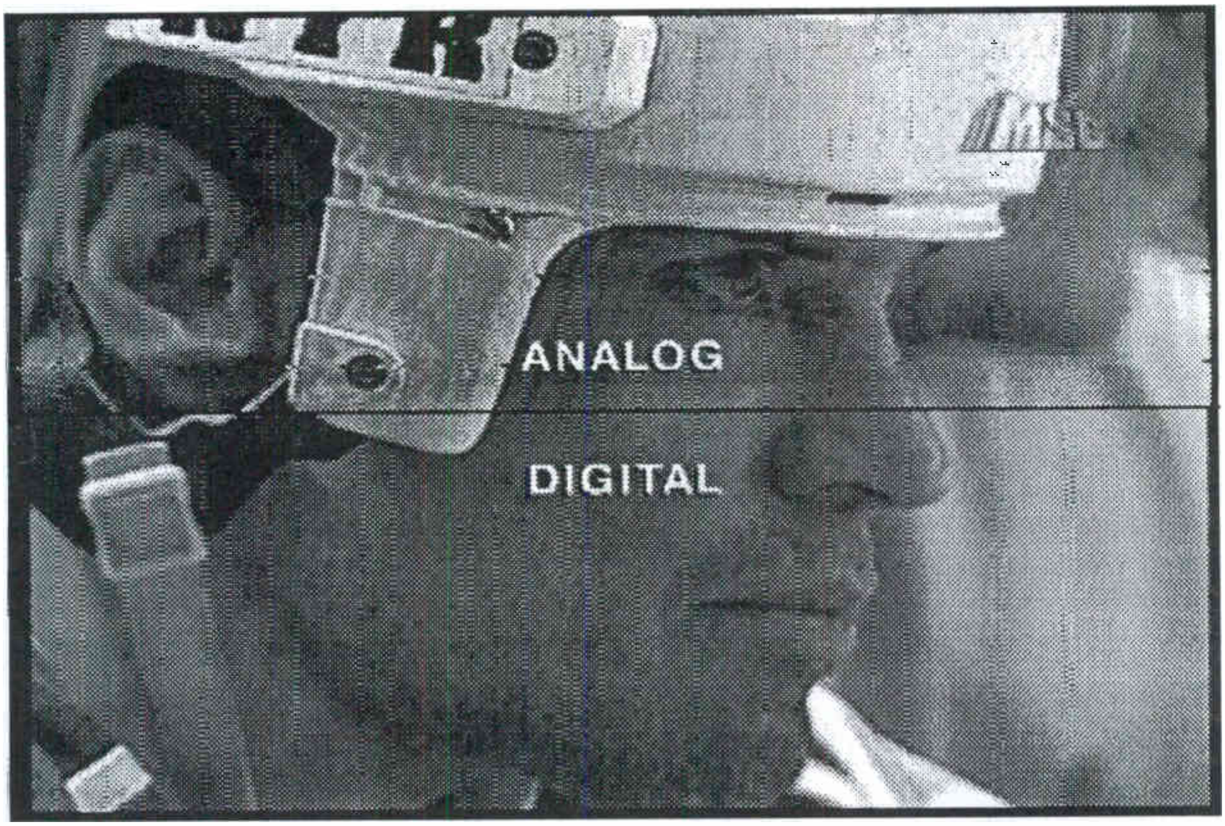

Figure 10
Case 3 Test Results

The analog signal, shown in the upper half, contains very noticeable ghosting artifacts and color distortions. On the other hand, the digital signal located in the lower half, is a clean well defined picture with no multi-path or noise.

In the presence of extreme multi-path, a 7/8 FEC was clearly not enough and the resulting errors can be observed by occasional slow picture motion, checker-boarding and drop outs. As predicted, an FEC rate of at least 3/4 was required to adequately recover from random errors induced by multi-path interference, and in our tests, an FEC rate of 3/4 seemed sufficient to recover from most errors. This test shows the importance of forward error correction. The signal path fully tested the capabilities of the digital signal to operate in the multi-path environment.

Table 2 summarizes the results for all the tested data rates and FEC rates for each of the three cases: case 1 (direct line-of-sight), case 2 (moderate multi-path), case 3 (extreme multi-path).

Table 2: Summary of Test Results

Mode	Bit Rate	FEC	Allocated Band-width	Case 1 Rcv	Case 2 Rcv	Case 3 Rcv
Analog FM			17 MHz	-25 dBm	-56 dBm	-70 dBm
Digital	9 Mbps	3/4	8.5 MHz	-28 dBm	-60 dBm	-74 dBm
	10.5 Mbps	7/8	8.5 MHz	-29 dBm	-60 dBm	-74 dBm
	12.5 Mbps	3/4	12 MHz	-28 dBm	-60 dBm	-73 dBm
	15 Mbps	7/8	12 MHz	-28 dBm	-60 dBm	-73 dBm
	8 Mbps	2/3	8.5 MHz			-72 dBm
	6 Mbps	1/2	8.5 MHz			-72 dBm
	4.5 Mbps	3/4	4.5 MHz			-74 dBm
	4.5 Mbps	½	6.5 MHz			-72 dBm

The results of these tests pleasantly surprised all concerned and clearly showed that digital ENG video transmitted in the 2 GHz band consistently produced a picture equal to and in most cases superior to the analog transmission system.

IX. Conclusions

We have presented an overview of how the digital video microwave technology can be applied to STL and ENG systems. We have also presented results for STL and ENG tests conducted by NUCOMM. Applying digital video compression, QPSK modulation, and forward error correction for STL and ENG systems can conserve frequency spectrum and yield superior quality and performance equal to and better than analog systems under both fading and multi-path types of environments. The digital ENG field tests specifically showed that an encoding rate of 9 Mbps yielded sufficient audio and video quality, and that a forward error correction rate of 3/4 provided adequate error protection for all the test cases including extreme multi-path interference, even when using demanding source material, such as a hockey game sequence. This combination (9 Mbps, 3/4 FEC) not only resulted in superior video and audio quality but also required only half the allocated analog FM transmission bandwidth – that is approximately 8.5 MHz. As it is expected that the 2 GHz BAS spectrum will be reduced from the current 120 MHz to 70 or 85 MHz spectrum, these tests show that the reduced spectrum can be supported using digital microwave transmission. Although QPSK worked well for both the STL and ENG tests, there are other digital modulation techniques available that can be used for these applications. For STL, higher order modulation than QPSK can be applied since their transmission links are fixed and tend to be very reliable. Nevertheless, further tests using 16-QAM and even higher order modulation codes for both STL and ENG still need to be conducted in order to fully assess its performance capabilities in the real environment.

The tests were performed without the use of adaptive equalization in the digital demodulators. Equalization was purposely not used so as to measure the uncorrected multi-path effects on such a system. The use of adaptive equalization can only further improve the performance of digital video systems.

Epoch-making Technologies at the Nagano Winter Olympic Games
- Start of a new era of Hi-Vision -

Noriaki kumata
NHK Broadcasting Engineering Department
Tokyo Japan

ABSTRACT

For Hi-Vision (High-definision television) broadcasting, NHK (the Japan Broadcasting Corporation) will mobilize nine broadcasting vehicles, 100 cameras and 92 VTRs to cover a total of 50 events in seven race categories at the 18th Winter Olympic Games at Nagano. This paper reports on HV triple speed super slow system, HV-WL cameras, HV ultra small cameras, HV-VTR integrated cameras and other latest inventions especially developed in time for the Games, as well as a 1.5Gbps optical multiplex HV transmission system for sending signals between Nagano and Tokyo and a 42-inch HV plasma display system. Also introduced are the "Bird-cam," "Virtual showdown," and "ice zone microphone" developed by NHK for NTSC broadcasting.

INTRODUCTION

Japan is the only country in the world with regular Hi-Vision broadcasting using a satellite. With as many as 600,000 Hi-Vision receivers already in the homes of people across the country, we are in the final stretch of program production under the slogan "Hi-Vision for home viewers."

All the 17 hours a day allocated for Hi-Vision will be devoted to Olympic broadcasting during the period of the Nagano Games with live broadcasting expected to account for as much as 42 percent of those hours. A total of nine specially equipped vehicles will be mobilized for live broadcasting. Those involved in production are confident that with extensive Hi-Vision hardware, in terms of both quantity and mobility, we can cover the excitement and thrill of the Olympic events as well as conventional NTSC broadcasting.

Some of the newest additions to Hi-Vision equipment include HV-NTSC compatible cameras, high-speed camera system three times as fast as conventional ones to be positioned near the K-point at the jump event sites, WL cameras allowed into the field for the first time during the opening ceremony, and ultra-small cameras to be positioned in a tiny space too small for conventional cameras very close to where the jumpers take off from the slope. The portable HV-VTR integrated camera will significantly boost the mobility of cameramen when shooting outdoor events where the foothold is unstable, shooting from a narrow space, or performing on-board ENG coverage.

For transmission, signals will travel via three types of optical transmission systems operated by domestic carriers. Of these, the most notable is the 400-km baseband transmission system based on the recently developed light wavelength multiplexing technology. A total of 40 42-inch HV plasma display units have been successfully tested throughout Japan, raising the expectations that the general public will be able to enjoy Olympic Games on large television screens in their homes which often lack space.

New equipment and technologies will be fully employed in all aspects of Olympic broadcasting from program production to viewing at home, ushering in a new era of Hi-Vision starting with the Nagano Winter Games.

In addition to these Hi-Vision devices, some new devices will also be used for NTSC broadcasting at the Winter Games. The Bird-cam positioned at a height above the ground moves along each competitor for the first 100 meters in the alpine skiing races, making the viewer feel like a bird looking down on the races below. The ice zone microphones embedded in the ice of the skate rink pick up the violent crashing on the ice amid the roar of the spectators as the edges of the athletes' boots strike the ice surface. The virtual showdown system simulates competitions by showing two players side by side competing in the same event but in different groups, showing them race toward the goal with their time clocked to 1/100 second. These new inventions, collectively called "new technology," will also be partially used to send international signals.

HI-VISION BROADCASTING

At the moment, Hi-Vision programs in Japan are being broadcast by seven networks--NHK, NTV, TBS, CX, ANB, TX, WOWOW--for 17 hours a day on experimental basis, with no income in preparation for full-scale operation. Under this format, it is difficult for NHK to secure enough funds for HDTV program production. Therefore, to cut the cost of equipment and manpower, NHK resorts to simulcasting with its ground broadcasting to 28 million subscribers and satellite broadcasting to 8.6 million subscribers.

Many of these latest Hi-Vision devices can operate either at 59.94 Hz or 60 Hz or both as required by SMPTE 240. With the improved performance of 59.94-60 Hz frame converters, we can use 59.94 Hz in post-production as well as in pre-production, converting the frequency to 60 Hz just before sending the signals to the broadcasting satellite.

Operations for the Nagano Winter Games will be handled jointly by NHK, TBS, CX, and ANB. Simulcasting of international signals and Hi-Vision signals will not take place although the Games will be held in Japan, but seven new cameras, not for play-by-play coverage, will be provided to ORTO98 for shooting the events at five different venues for Hi-Vision broadcasting as well.

The new HV-NTSC compatible camera mentioned above, first used at the Atlanta Summer Olympic Games, is produced by three manufacturers in Japan with two companies building broadcasting vehicles equipped with this camera. Here, we will not report the details of program production using these cameras and vehicles as they were introduced in the "Hi-Vision Program Production at the Atlanta Olympic Games" at NAB97. Instead, we explain below how these devices will be used, as well as about signal transmission and PDP display, now that Hi-Vision broadcasting has reached the level of NTSC broadcasting in mobility and special performances, contributing to richer program content.

a) Triple-speed super slow system

This Hi-Vision slow-motion system captures and records 180 frames per second, three times as many as conventional cameras. Its camera has four CCDs, two for the green channel. These two CCDs for the green are driven at 1.5 times the normal speed while reading, while the system's electronic shutters alternately close and

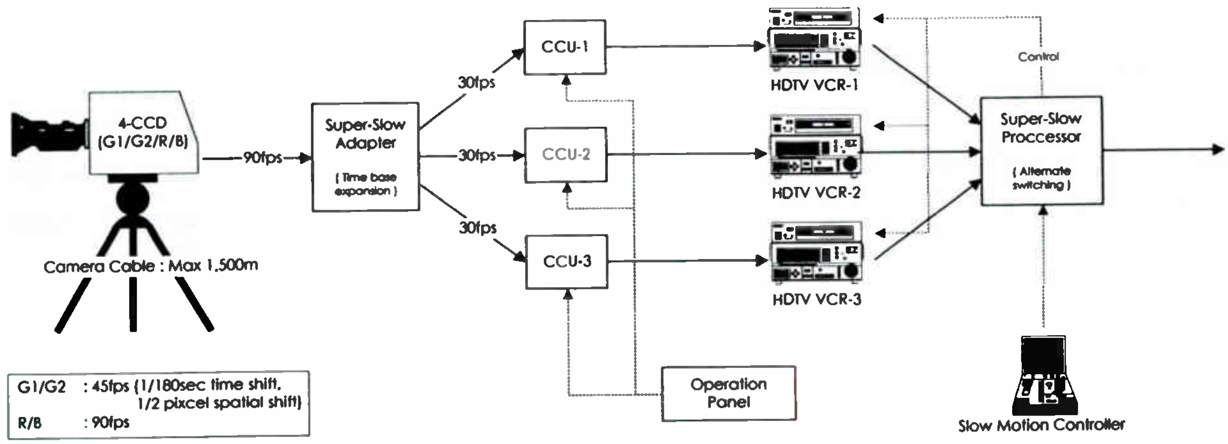

Fig.1 Principle of HV triple-speed super slow system

open to output images with three times as many fields. The slow adapter of the receiver divides this output into three HV signals, which are then fed into three CCUs for recording on three digital VTRs. For playback, these HV signals from the three VTRs, operated under the precise speed controlling of the slow controller, are combined into single HV signals by the super slow processor to produce smooth slow-motion pictures. During the Nagano Winter Games, this camera will be positioned near the K-point at the jump sites to capture the continuous flow of each jumper's movements of landing from airborne posture. Specifications of the camera are sensitivity F4 at 2000 lx, S/N 48 dB, and resolution 800 TV lines.

b) Wireless camera

The world's first wireless HV cameras will be used in the opening ceremony and inside the braking zone of the jump events at the Nagano Olympic Games. Hi-Vision signals are first converted into TCI signals before frequency modulation of the carrier in the 42 GHz band for transmission. The transmission antenna is the 5-cm horn type. Return video, tally, inter-communication, camera control signals are sent back at 10 GHz. The transmission bandwidth is 80 MHz and the transmission output is 100 mW. Other specifications are as follows: total weight, 12.4 kg including the battery (operated by a cameraman and an antenna operator); power consumption, 73 W; power source, three 60 Ah

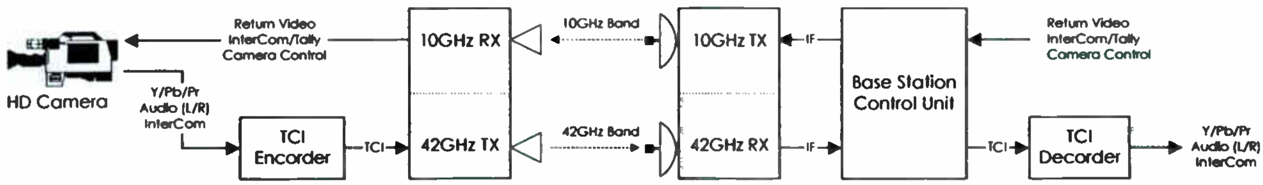

Fig.2 Configuration of the 42GHz band Wireless Camera system for HDTV

lithium ion batteries (which keep the camera operating for one hour at 10 degrees below zero).

c) Ultra-small camera and compact remote control camera

The ultra-small camera is made up of a color stripe filter and a 2/3-inch FIT-type CCD with 2 million pixels. The use of C-mount compact lenses contributes to the small size of the camera head. Only the CCD-driven horizontal pulse section is incorporated in the camera head, and its box-like design makes mounting on the printed circuit board easier. The gate array and vertical pulse section are housed in the CCU, connected to the camera head through a 10-meter cable 7 mm in diameter. Main specifications are as follows: camera sensitivity F5.6 at 2000 lx, S/N 54 dB, critical resolution 700 TV lines, head size 40Wx30Hx60D, and head weight 150 g. During the Nagano Games, this ultra-small camera will be used for the jump events, to be positioned very close to the point from where each jumper takes off, capturing the movement from approach down the slope to immediately before takeoff.

The compact remote control camera has a separated optical block. This remote control camera for conventional broadcasting was used to cover the gymnastics events at the 1988 Seoul Olympic Games, and we now have their new version for Hi-Vision broadcasting. The camera will be installed on the ceiling of the ice hockey rink A to capture the power and speed of the games down below.

d) VTR-integrated camera

This VTR-integrated camera has been developed to catch up with the NTSC equipment in terms of mobility, operability, and reliability, and to exceed the Uni-Hi system in picture quality. Its camera unit has three 2/3-inch FIT-type CCDs with 2 million pixels. Advanced process ICs used in the signal processing block for extensive digital processing have made it possible to reduce both the size and power consumption of the camera. Video signals (996 Mbps) from the camera are first rate-converted by a pre-filter from 4:2:2 to 3:1:1 and compressed to 1/1.6. Next, these signals are further compressed to 1/4.4 by discrete cosine transformation (DCT) inside the frame and variable length coding (VLC). Together with rate conversion, the signals are reduced 1/7 (140 Mpbs) the original level. With parity, SYNC and voice signals added, entire signals (180 Mbps) are then recorded on the VTR with the minimum recording waveform of 0.49 microns.

The camera system together with its lens, view-finder, tape, and battery weighs about 8 kg. Also equipped with a playback mechanism, it is expected to be the main equipment used for on-site coverage. For the Nagano Winter Games, a total of eight VTR-integrated cameras will be used for coverage and planning.

e) 1.5-Gbps light multiplex transmission

The distance between Nagano and Tokyo is 222 km along the Shinkansen super express train tracks, but is 320 km along the north route and 400 km along the south route, offering communications operators affiliated with electric power companies a diversity of transmission routes. Input SDI signals are each allocated wavelengths, and light-strength modulation (external modulation) is carried out in the baseband in order to keep the spectrums from spreading. Light wavelength multiplexing by optical couplers is carried out for up to 8 channels, then the signals are amplified by the photo amplifier for outputting.

In conventional optical transmission, signals have to be amplified every 40 km or so by optical-electric and electric-optical conversion. In this system, there is no such conversion. The light signals are transmitted without such changes over entire distance owing to in-line-type optical amplifiers installed along these transmission lines (six for the north route and seven for the south route). On the receiving side, the signals are first separated by a distribution coupler. Next, these divided signals go through an optical resonance oscillator comprising half mirror and piezoelectric device to select target wavelengths.

f) 42-inch PDP display

One of the major research objectives at NHK Science and Technical Research Laboratories has been the development of the so-called wall television that recreates a powerful sense of presence with a large screen and high image resolution beyond the limits of conventional CRTs. Years of effort have now brought us close to the practical application of a 42-inch PDP (plasma display panel) which provides satisfactory operating life, moving picture quality, and resolution. With the Nagano Winter Games just around the corner, a total of 40 of these panels will be installed across the country including five units at IBC.

The specifications of this 42-inch PDP have been determined based on the following considerations: the maximum size of the glass mask which serves as the original pattern plate, the requirement that it has as many TV lines (650) as Hi-Vision receivers currently on the market, and good matching of clock frequency, cell arrangement and the number of pixels. Extensive application of photo processing during the manufacturing phase has resulted in uniform shape, greater drive margin and brightness, and higher reliability.

As Japanese homes still lag far behind Western counterparts in terms of floor space, PDPs measuring only 7 cm in thickness are expected to boost the demand for Hi-Vision.

NEW TECHNOLOGY

NHK had set up a project team to develop new technology for the Nagano Winter Games as early as eight months before the start of the Lillehammer Winter Games, aiming to surpass Lillehammer in new broadcasting technology. Some of the project's target technologies were abandoned halfway through while some others were not accepted by ORTO98, but there have been a number of successful developments, including the Bird-cam to be used by ORTO98 for the women's downhill skiing, the ice zone microphone to cover all events on ice except for curling, and the virtual showdown system which was already tested in the speed skating events for NHK Unilateral.

1) Bird-cam

Skiers in the downhill skiing events often reach speeds exceeding 100 km/h, too fast for a conventional moving camera. The Bird-cam accelerates to the speed of 90 km as it free falls along the wire strung over a 100-meter distance from the start of the women's downhill to the second flag. To be used for ORTO98, it will provide the audience with thrilling bird-eye views of the skiers as they hurtle down the slope.

The unpowered glider of the Bird-cam free falls down the piano wire, which is strung between two poles 8 meters in height erected at the start and end of this 100-meter distance, at the moment each skier starts off. The camera and its pan head are radio-controlled and the glider speed is adjusted by a pre-installed program. Manual braking is also available in case of emergency.

The glider is 145 cm long, has a 128 cm wingspan and weighs 32 kg. It comes with three 1/2-inch CCDs camera and a 10-GHz FPU. The largest obstacle to its development was to find a way to stabilize the glider as it free falls. A gyro-sensor built in the body detects rolling, and automatically stabilizes the glider. The initial model had both fixed and movable wings, but the fixed wings were removed as they were found only to increase the instability. After reaching the end pole, the Birdcom is manually hooked to a separate recovery loop and wound up the slope by a oil pressure motor. The start cycle is 2 minutes.

2) Ice zone microphone

ORTO98 will employ this ice zone microphone for speed skating, figure skating, ice hockey, and bobsleigh races. The microphone picks up the sound produced when the edges of the skaters' boots strike the ice surface amid the roar of the spectators.

At NHK Science and Technical Research Laboratories, researchers started by freezing all types of microphones in ice in a freezer, and found that the dynamic type offers superior stability and durability. Fig.3 shows the design. The microphone picks up the sound as electric current is generated when the moving coil with a plus ring to have much inertia cuts the field of a magnet attached to the microphone case which vibrates when struck by the sound.

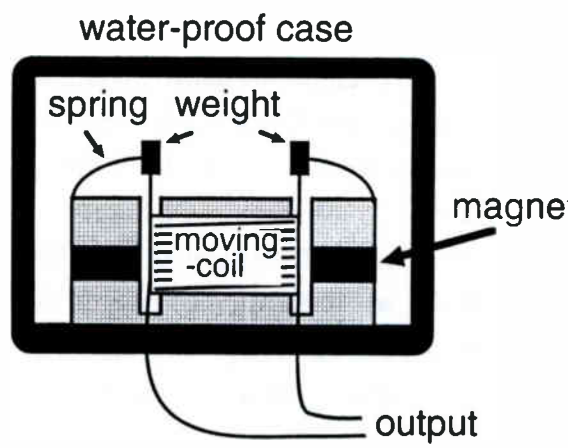

Fig.3 The structure of ice zone microphone

The microphone embedded in the ice must have flat-frequency-responce and be waterproof, resistant to load pressure, and thin. This ice zone microphone is just 1 cm thick, as the ice for the skating events is only 3-5 cm thick. It can withstand loads of up to 8 tons, enough to endure the weight of the re-icing vehicle.

At the 400-meter speed skating track , forty ice zone microphones are being embedded in the ice at 10-meter intervals. Through automatic mixing, computers turn on the microphones closest to the skaters by pre-programming the fact that two skaters compete at a time and move in same direction.

3) Virtual showdown system

A motion control camera recreates the movement as operated by a cameraman at a speed skating race. The system captures the movements of two skaters competing in the same event but in different groups and combines them as if there were competing side by side in the same race, showing their speeds to one-hundredth of a second. The system will be used at the NHK Unilateral.

The system consists of three units: (1) A motion control camera for recording information on the camera pan head and lens movements every 8 ms and recreating the camera movements entered by the cameraman; (2) A sync. retriger resets the television sync. signal upon receiving the start pulse from SEIKO. This sync. retriger is needed as the television sync. signal is independent from the start timing, which produces errors up to 1/30 second ; (3) A video synthesizer using black NAM.

By combining two skaters from different groups, viewers can see who is good at negotiating corners and who are the strong starters. It also possible to virtually superimpose two top skaters at the same start line.

SUMMARY

Poor mobility and low sensitivity have long been cited as the shortcomings of Hi-Vision equipment. With these new cameras described in this paper among the latest developments, however, Hi-Vision devices are fast catching up with NTSC equipment in terms of operability and price. While the introduction of new broadcasting vehicles brought Hi-Vision equipment to the level of NTSC twelve years ago, the development of these ultra small camera, wireless camera and other improvements puts Hi-Vision only three years behind NTSC, bridging the nine-year gap in just two years.

Hi-Vision is as good as 35-mm film in performance, but home users are the main target of Hi-Vision broadcasting. Viewers at home receive Hi-Vision as a TV with "a little wider screen and slightly prettier images." Such a rapid pace of equipment development would not have happened if the focus had been on producing programs for 100-inch or larger screens.

The excitement over the Nagano Winter Games is expected to double the demand for Hi-Vision receivers. Hi-Vision is now almost as good as NTSC in program production, although there are still some limitations on its transmission hardware. The future direction of broadcast digitization is still unclear, but the number of receivers in use is now close to one million, and so the market is expected to expand significantly.

New equipment and technologies will be fully employed in all aspects of Olympic broadcasting from program production to viewing at home, ushering in a new era of Hi-Vision starting with the Nagano Winter Games. I am indeed grateful that, as one of the Hi-Vision program producers, I had a oppotunity to be involved in all of these activities.

ACKNOWLEDGMENT

I would like to thank all those people at NHK Science and Technical Research Laboratories and NHK Broadcasting Engineering Department for their willing cooperation and advice.

Consumer Electronics for Digital Broadcasting

Thursday, April 9, 1998
9:00 am - 12:00 pm

Chairperson:
Andy Butler
PBS, Alexandria, VA

Target Practice: Compressing Your Signal, Expanding Your Reach
Dominick Stasi
Your Choice TV
Englewood, CO

***Set Top Shoot-out**
Rob Glidden
Quadramix, Inc.
Moroga, CA

Applying Smart Cards for Access to DTV Broadcast
Steven Humphreys
SCM Microsystems
Los Gatos, CA

***Using Market Forces to Implement DTV**
Brad Dick
Broadcast Engineering Magazine
Overland Park, KS

A Programmable Architecture For Digital Television
Neil Mitchell
Philips Semiconductors
TriMedia Products Group
Sunnyvale, CA

*Papers not available at the time of publication

Target Practice
Compressing your signal, Expanding your reach
By

Dom Stasi
Vice President
Engineering and Network Operations
Your Choice TV

&

Mike Asmussen
Sr. Partner
Skjei Telecom, Inc.

ABSTRACT
Not only is digital technology an explicit enabler of higher quality picture and sound, but it offers many *implicit* benefits to the television broadcaster as well. Notable among them are more efficient (and potentially profitable) use of bandwidth, ease of storage, and the ability to manipulate content in the domains of both space and time.

But opportunity without its share of pitfalls is the rarest of things. And fundamental to the paradox, especially from the engineering community's point of view, is the realization that never before in the, albeit brief, history of our medium has so much control over a signal's attributes been placed in the hands of the technologist.

As we enter an era that will see the first widespread broadcast of digital video signals, traditional broadcasters stand to benefit from the experiences of cable television programmers. Free of the rigid constraints of spectrum and modulation so characteristic of the broadcast sector, cable programmers and operators have already gained substantive experience (and not a few black eyes) in the development and distribution of multi-channel, digital video to the consumer. Further, one emerging cable programmer, Your Choice TV, is gaining insights that should be of particular interest to video broadcasters. Through their unique blend of digital technology, consumer interaction, and time-shifted, broadcast programming fare, Your Choice TV provides a vantage of insight from which traditional broadcasters might get a peek at what lies beyond their own horizon.

This paper will draw from that cable programmer's five-year experience to illustrate what are certain to emerge as both apparent opportunities and potential pitfalls for the broadcaster as he embarks upon the most fundamental technological change in his industry's history.

Introduction
Your Choice TV has established a program group currently comprising six networks. And while many programs are independently-acquired, the networks are characterized by their unique blend of time-shifted cable and broadcast programming. Your Choice TV is using its bandwidth to offer viewers a second (...third, forth, so on) chance to watch their favorite network programs when and if they missed the first run due to problems of schedule, work, soccer, etc. Additionally, Your Choice TV is experimenting with a technology-based scheme that, if

successfully implemented, will optimize the use of bandwidth through the application of progressively targeted advertising.

Though intended to be platform independent, the Your Choice TV networks are carried over both C and Ku-band domsats primarily for distribution to CATV and DBS outlets. This represents the first instance of current broadcast programs being digitally compressed and retransmitted in close succession to their analog broadcast airing. Time shift is usually four hours, thus also providing consumers their first chance to (wittingly or otherwise) compare quality against that with which they are eminently familiar.

Since this time-shift philosophy closely parallels a digital option recently held out to broadcast stations by the FCC, it appears that several precedents have been set by Your Choice TV. As a service hard-wired through (arguably) intelligent set tops, we are aggressively building the ability to gauge real-time audience response. Several lessons are being learned.

Video Quality

Let's first discuss that which in the analogue realm is determined in production and distribution, only to be too-often degraded in exhibition: video quality.

Cameras, lighting, tape formats and generational descent have been the traditional arbiters of picture quality. They remain so. But despite the rigorous control usually inherent to the selection and use of origination equipment and the production values particular to each program, broadcasters have been historically frustrated by the inadequacies of each viewer's unique analogue reception circumstance. With the advent of digital video from source to sink all of that stands to change. We usher in an era where co-channel will be rejected, "snow" will melt, and ghosting, along with the rest of those bugaboos which have always haunted viewers, will largely disappear. That's the good news, and for the first time broadcasters will have a real *opportunity* to predict and control signal degradation, in effect limiting it to those impairments inherent to production and format. There will be no technology-based reason why an adequately encoded digital video signal shouldn't deliver "studio quality" pictures to the antenna terminals of even the most multi-path compromised urban receiver, as well as the fringiest rural one with virtual impunity.

The devil, however, is in the details, and the *pitfall* in all this lies in our newly-expanded ability to customize signal characteristics. The myriad parameters of video encoding as defined by the MPEG suite of protocols provides video engineers with what has traditionally been characteristic of linear electronics: the continuously-variable control. With digital video, we suddenly have at our disposal the ability to control sampling and data rates, compression levels, resolution limits, and entropy. Further, we can to do so not only over extraordinarily broad limits, but at every interface in the origination and distribution chain.

In *retransmitting* programs with which the viewer is eminently familiar, such as in the Your Choice TV case, we are entrusted to uphold the standards set by the broadcast network and judged against the expectations viewers have developed based each one's particular analogue reception environment.

In a digital video scheme, the quality standards placed at the engineer's disposal ironically are less a matter of equipment-cost than they are one of discretion: how much is good enough? how much is too much? That should make it an easy question to answer.

In a compression environment, however, the engineer's discretion will be subject to the influence imposed by the accountant's ledger sheet: how many channels (and therefor revenue streams) can we squeeze into a given bandwidth? That makes it a tough question to answer. That makes it a question which, in an environment such as that which caharacterizes digital video, an environment virtually free of imposed regulation, yet rife with *opportunity* to compromise quality for quantity, will too often be answered by the accountant rather than the engineer.

Coding for Compression

With powerful processing and transforms, our ability to compress video signals is extensive. Coding gain and data rates can be varied over virtually infinite limits, and are clearly independent variables. Which leaves video quality to become the dependent variable, changing in inverse proportion to coding gain, and coding gain roughly following compression ratio.

Of the MPEG compression standards that exist today, MPEG-2 is that which will prevail and which has found its way into what is fast becoming widespread commercial network distribution application. However, in order to discuss MPEG-2, those aspects of its genealogy specific to MPEG-1 should be mentioned.

The isolation of redundancy between consecutive frames is the greatest contributing factor in the reduction of data rates when endeavoring to compress full motion video material. The MPEG compression techniques make optimal use of this powerful process. Additionally, MPEG signals are packetized, allowing switched distribution through asynchronous transport protocols and the manipulation of encoded video as digital files.

In standard 4:2:2 sampling the chroma is sampled at 1/2 the H resolution of luminance, while vertical resolution is unchanged. MPEG-1 subsamples all components, ignores alternate fields, and produces a 2:1:0 sampling lattice.

To briefly review, MPEG coding reduces redundancy through the use of motion estimation, bi-directional, and both intraframe and interframe coding. One might also limit resolution in order to achieve a lower data rate. The protocol is best characterized by its use of I, B, and P frames. These are additional sorts of video frames that allow motion estimation through the comparison of pixel locations between previous and subsequent frames. I frames are complete pictures sent every 10 frames or so, which allow for switching.

B frames are bi-directional predictors which are transmitted and stored at the decoder.

P frames make use of motion vectors and allow the received image to be shifted spatially in reception. All of these features reduce data rate, but add to the memory requirements in the decoder.

Sampled frames are downloaded and buffered in groups that allow for non-sequential processing. Interframe attributes such as motion or redundancy are identified in this manner. The first picture in the group will be intra coded and applied to the DCT. Once transformed and quantized, it is inverse quantized, inverse transformed and buffered for reference. This forms an *I* frame. A *P* frame is then processed by reference to the stored *I* frames and motion information is detected and subtracted from the frame being processed to generate a predicted picture. This frame is transform coded to form a motion compensated, compressed *P* frame, which is stored in encoder memory. The encoder now accesses intermediate frames skipped when the *P* frame was created. The encoder calculates forward motion vectors from the locally buffered *I* frame to the *B* picture, and reverse from the *P* frame created immediately prior. Intermediate *B* frames are processed and encoded next, and the process repeats.

In the decoder, the process is synchronized by the recognition of *I* frames, and the inverse transform applied.

MPEG-2 evolved from the precepts of MPEG-1, and is in fact more a set of rules within whose context compression designers may work than it is a rigid standard. But despite that MPEG-2 is a set of rules, observing those rules affords designers and users of the standard an extraordinary range of freedom, not previously available in analog systems.

MPEG-2 proffers a layered coding structure. Such an approach allows for scaleable levels of quality. These are determined by both the programmer and the receiver or set-top manufacturer. These are the *opportunity/pitfall* controls. Both encoder and decoder are capable of vast latitude, even within the layers of a protocol. Programmers are free to send full resolution, high bit rate data, while the viewer is free to choose a less complex receiver.

Error concealment, signal to noise ratio, resolution, and entropy propagation are all within the control of MPEG-2 system designers and users. As such, a digital TV signal might assume many forms and quality levels between production and exhibition.

In the production environment where bandwidth is rarely a factor, contribution quality video is nearly always 4:2:2 encoded to 720 pixel resolution, utilizing at the very least an 18Mbit data rate. That same signal might reach a digital set top decoder, however, in 4:2:0 format, decoded at 2.5Mbits with a horizontal video resolution of 360 pixels. Though it's a mere shell of its glorious former self, it nonetheless falls well within the parameters of MPEG-2.

MPEG-2 was developed to be neither a performance specification, nor an indication of quality. It is a set of syntactical rules, much like those used to make language convey information properly and understandably. And, as with language, there is ample room within its syntax to allow the user to create Hamlet or Disco Duck.

Concatenation Error
Digital signals are generally considered immune to the generational degradations characteristic of analogue video. But, if viewed in their own context, it becomes apparent that digital video is at least as susceptible to non-linear degradations as its linear counterpart. This is especially true of compressed signals. In the compression environment concatenation error, the buildup of square noise through the subsequent codification of artifacts, becomes an undesirable characteristic of low bit rate signals. When coding gain is stressed in an effort to compress at unrealistically high ratios, entropy loss becomes discernable as pixel-level quantizing errors. Unable to discriminate between artifacts and pixels-of-interest, subsequent encoding stages will generally encode the artifact, stealing critical data from desired picture elements and causing further entropy-loss. This is exacerbated by the unpredictable nature of such errors.

Random noise in received signals is also a limiting factor when those signals are intended for compression and retransmission. Again, the random nature of gausian-response noise is highly unpredictable, and like its square-response counterpart, will "fool" the encoder into codifying it, thus squandering the ever-rarer encoding data bits that define cascaded compression systems.

Contribution Quality Coding
That the scalability of MPEG-2 allows for variable quality levels at appropriate points in a distribution network is an indispensable attribute given the realities of network processing. In most distribution schemes, digital signals are decoded or transcoded several times along their journey to the user. If quality is to be maintained, source material must be coded at substantially higher levels than video at points in the distribution chain closer to the receive sink. Video coded at low bit rates will introduce random-like square response noise to video signals. This noise will be coded, consuming precious bits to the detriment of entropy in subsequent stages if an expected coding gain is to be maintained. Lost entropy cannot be reproduced in later stages. In the case of an underdesigned network, where unrealistic coding gains are assumed, this cycle is repeated causing each subsequent stage to introduce its own artifacts. The aggregate effect of this concatenation error can become more compelling than the signal itself.

Thus, video program material stored as MPEG-2 files at a transmission server may be coded at a much lower level than that intended for distribution at the programmer's server. In fact, video material coded at 3.0 Mbps at a CATV system headend file server might produce perfectly adequate video quality at the subscribers' set tops, since such coded files need only be robust enough to overcome the normal degradations of a CATV cascade. However, the earlier in the network's hierarchy the signal is coded, the more error correction it must contain, the less entropy loss it can support, and the lower its noise floor must be if it is to produce signals with adequate margin to be coded at 3.0Mbps at the headend. Such are "contribution quality" digital video signals.

Once a decision is reached regarding the level of entropy loss that can be sustained, the various coding levels (hence memory that must be defined in hardware) that will be accommodated at each point in the distribution chain, can be determined.

Your Choice TV transports some 150 hours per day of programming. Coding that many hours of video at contribution quality bit rates would require massive storage capabilities in an origination server. TCI and Your Choice TV therefor decided on a combination of tape and non-linear origination architecture.

In the non-linear case, signals are encoded and stored at main level, main profile, 4:2:2 MPEG-2 coded and distributed in memory for 1:1 redundancy. Coding levels are scaleable at between 5 and 18 megabits depending on application and equipment configuration. Initially, primary video files will be coded and stored at 10 megabits while backup files will be encoded at 5Mbps.

Encoding into the servers represents the first data reduction operation of the chain. Each encoding station will accept analog NTSC, analog component, S-VHS or SDI. For purposes of this discussion, it is here, where SMPTE 259M serial digital video, incoming at 270Mbit is real time encoded and compressed to 18 megabit, 4:2:2, "studio" MPEG-2 files. Studio MPEG is the highest "contribution" quality compressed signal encountered.

It is anticipated that in the backup mode, entropy loss and the inevitable encoding of quantizing square-noise will propagate degraded signals to subsequent network stages. It is believed that video encoded at 5Mbits so early in the chain will not provide adequate margin to concatenation errors and result in subscriber signals exhibiting objectionable artifacts. Tests are being conducted to determine the precise contribution level coding that best suits origination. It is projected that a coding level of 8Mbits, 4:2:2 will provide the best compromise, allowing both adequate storage efficiencies to accommodate contribution quality signals, and storage efficiencies appropriate to both primary and backup video files.

Recalled video files are routed to MPEG-2 decoders and decompressed. Each video output from the decompressors will again be 270Mbit SDI (SMPTE 259M compliant) component digital video. Audio is AES digital encoded at 48Khz and embedded in the serial digital bit stream. This group of signals is routed to the standards converter of the DigiCipher-II encoder.

Transport Quality Coding
At the DigiCipher-II encoder multiple channels of serial digital video, each representing one Your Choice TV video service, are input to the standards converter. MPEG-2 encoded and compressed at the television service processor, these signals now exist at data rates that vary in proportion to the operating bandwidth of the satellite transponder they'll occupy, divided by the number of services multiplexed. For a galaxy-VII Ku band, transponder, signaling data rates and coding budgets would break down as follows:

CODING BUDGET
8:1 Video Compression:

Satellite Data Rate..................	39.02^6 bps
FEC Data Rate	12.05^6
Information Rate	26.97^6
Messaging & Timing..............	1.62^6
Number of Video Svcs.	8
Data Rate per Svc....................	3.16^6
Audio Data Rate per Svc.	0.40^6
Average Video Data Rate......	**2.77^6**
Required E_b/N_0 Performance.	**3.5dB**

7:1 Video Compression:

Data Rate per Service	3.62^6 bps
Audio Data Rate.....................	0.40^6
Average Video Data Rate......	**3.22^6**

6:1 Video Compression:

Data Rate per Service	4.225^6 bps
Audio data Rate	0.40^6
Average Video Data Rate	**3.83^6**

5:1 Video Compression:

Data Rate per Service..............	5.07^6 bps
Audio data Rate	0.4^6
Average Video Data Rate.......	**4.67^6**

Satellite Distribution

A digital receive signal's required E_b/N_0 of 3.5 compares very favorably to the 9.5dB C/N ratios we're accustomed to dealing with in the analog domain. But what we gain in earth station performance through digital coding and modulation, we lose in required margin above receiver threshold. Shorter wavelengths make Ku frequencies susceptible to atmospheric attenuations (rain fade). Our 50 watt Ku downlinks will vary in receive EIRP as much as -13dB in the presence of heavy rains. This is far greater than anything we've experienced on the CATV C-band earth station network. While it's not common practice, a C-band receive station can be designed with margins above rf threshold of perhaps 2 or 3dB and never experience a discernible atmospheric fade.

Economics and good operating practice dictate that robust digital coding, shorter, Ku, wavelengths and higher EIRP levels should accrue to the advantage of receive station antenna aperture.

Antenna gain varies as the log of frequency. A parabolic receive antenna of given aperture will exhibit substantially higher gain at Ku than a comparable device at C-band. The following equation shows the relationship:

$$G_a = (20_{\log 10} f)+(20_{\log 10} d)+7.5\text{dB}$$

where: Ga = Antenna Gain
f = Frequency in Ghz
d = Antenna Diameter in Feet
Efficiencies of 55% are assumed

Addressable Advertising

Digital video compression and video distribution technologies will allow business opportunities which up until now have been denied the *broad*caster. Prominent among these will be the realization of addressable advertising. Along with more powerful computing systems and sophisticated software, digital video is opening up new opportunities in television advertising. Addressable advertising offers the opportunity for sponsors to gain the same sort of targetibility that has been achievable to date only through direct mail, and to do so with the captivating visual entertainment quality that only television offers. Addressable advertising ensures that the message of the advertiser is delivered to finer segments of the viewing audience than what is supported today so that the advertiser's message is not wasted. An interesting paradox of the broadcast industry is that the more successful a program or station is, the more diffuse its advertising focus must become.

Your Choice TV is creating a nationwide system to support the delivery of *addressable advertising* in conjunction with its current programming service, and is moving ahead with the deployment of the infrastructure to support this new paradigm.

However, some practical concerns do exist. Issues yet to be fully resolved include the complexity of managing the delivery of this advanced form of content, and the substantive technical challenges involved.

Why Now?

The advertising model that has been in place for the last several decades is changing. There has been much focus in the press lately about interactive advertising over the Internet. Although this is indeed an emerging opportunity, and a medium in which new approaches and techniques might be tried, the Internet's consumer reach is still limited when compared to that of broadcast television. Additionally, the advertising industry is still uncertain as to how the Internet will evolve, and even more uncertain whether the consumer will ever warmly embrace the use of the computer as a true entertainment device. Nonetheless, and though still in its infancy, the Internet is becoming a testing ground for some advertisers experimenting with targeting and interactive advertising. It also indicates that, given the chance, many sponsors and agencies are interested and willing to move towards more advanced forms of advertising, regardless of the medium. Additionally, a number of changes are taking place in the television industry, including the number of television channels to which viewers are exposed, resulting in the disaggregation of the mass market. Additionally, advertising will play a diminishing role as pay per view becomes a more prevalent delivery scheme. These two factors are causing sponsors to take notice that change is in the wind in television-based advertising.

Technological advancements continue to cause significant changes to the television industry, and specifically to advertising-approaches supported over this medium. The key developments in the industry that have opened the door to new possibilities in advertising are the development and spread of digital video, intelligent set-top boxes, and the flexibility the combination of the two introduce. Digital video compression allows for the ease of storage, transmission, insertion, and tracking of video content. Digital set-top boxes will allow for accumulation, storage of actual viewing actions in the home, and data retrieval. This provides an accurate and reliable measurement of viewing behavior due to its non-intrusive nature. Also, intelligent set-top boxes and televisions will allow for more sophisticated control features and more interactive, dynamic communications with the viewer. These capabilities, in combination with the ability to embed addressing within the programming, open the door for truly addressable advertising. Finally, the quantification and algorithmic translation of what has traditionally been heuristic and qualitative targeting techniques, along with the ability to sort through and correlate huge volumes of marketing data with previous viewing habit data is allowing the real-time optimization and targeting of advertising. Computer applications are being developed to manage the targeting, scheduling, and distribution of addressable advertising.

Feeder Channels

Your Choice TV's approach to advertising is two-fold: 1) allow for the support of the desired features supported in today's advertising model, and 2) add to it the advantages possible with new technologies. To this end, nationally distributed ads are supported and local promotions and advertising will ultimately be supported, as digital insertion into digital video streams becomes a reality. Both local cable systems and local program network affiliates will be in a position to provide regionalized or locally directed material during breaks set-aside for local insertion.

The more compelling approach is addressable advertising via nationally delivered, group-addressed feeder channels. With digital compression, bandwidth is available for delivering multiple channels of advertisements. Viewing measurement is made easy, due to the nature of PPV and supplemented through custom tracking applications residing on the set-top box.

Your Choice TV is using an evolving approach to achieve the desired goals of addressable advertising. First, national advertising will be supported, with ever-increasingly resolution on the optimized placement of this advertising. This placement is based on near-real time viewer feedback through PPV buy data and collected set-top box data. It also utilizes viewer demographics based on household location at a zip code (Zip+4) level. Your Choice TV has strategic relationships in place and through these relationships, the necessary tools are being fielded to support this automated optimization and ad placement process.

Next, as necessary technology is deployed, Your Choice TV can further segment and target ads, pushing targeting to the regional level, local level, and ultimately the household level. The "feeder channel" technique is being developed to deliver addressed ads. The feeder channel concept enables the selection of the specific advertisement that is best suited for a household from several ads airing simultaneously on different channels. Then, the household's digital set-top box is commanded to the appropriate feeder channel at the appropriate time to receive that ad. Your Choice TV is leading the industry effort and working closely with equipment vendors to develop the technology and infrastructure to support this concept. One of Your Choice TV's technology partners is developing the tool that can aggregate information from numerous sources, including geographic and demographic data, online marketing research databases, and information on exposure to past advertisements and use this information to automate the feeder channel selection and delivery process.

Making Addressable Advertising Work

Currently in the advertising industry, addressability in its simplest form is handled via the selection of the program in which an ad is to be placed. Refined addressability is achieved through local or regional airing of advertisements, at the affiliate level for broadcast television, or the head-end for cable television. Ultimately, the goal is for targeting of a specific commercial to a specific viewer in a household. However, until the infrastructure exists to support dedicated bandwidth directly to each home, and the mechanism exists to identify the specific viewer in the home, some simplifications must be made. The use of feeder channels accomplishes this simplification, yet allows for the future flexibility to evolve to more specific addressing techniques as technology allows.

The following generalized description is by no means meant to completely characterize the placement algorithm, but attempts to depict the process as automated, easily repeatable, and taking into account both the targeting concerns of the advertisers and the known characteristics of the Your Choice TV viewing base. The general targeting process consists of characterizing the audience into various targeting categories and groups associated with each category. This is initially being done down to the zip+4 zip code locale of each household using various demographic and geographic information from marketing databases and other proprietary sources. Then, programs are assigned certain evaluative attributes, as well as are advertising campaigns. These attributes include program rating, genres, and numerous other proprietary factors. For the various targeting groups, viewership of each program airing is estimated based on past historical PPV purchase data. This projected viewership information is used to determine a quantitative assessment as to how well an ad matches each program's attributes and the expected target market of the program.

Once the highest rated advertisements for a given program and the expected viewership is determined, the scheduling of these ads takes place across the available feeder channels. Then, set-top boxes are provided with a plan as to which feeder channel they are to tune to based on the above process. At the appropriate point in the program, the set-top box then seamlessly switches to the feeder channel containing the ad that results in the best match for the household, based on the group in which it has been placed. At the end of the airing of the ad, the set-top box seamlessly switches back to the original program channel. Following the airing of a program, feedback via PPV purchase data and data collected from set-top boxes is used to validate the actual viewership of the program and ad.

Summary

Examining these system architectures and the various data rates employed, it becomes apparent that their determination should be as much a product of economics as of engineering. As was stated at this paper's beginning: bandwidth is precious. But let me add now, that it's precious only to the extent that it can be marketed. As subscribers become sensitized to higher and higher levels of video quality, largely through their exposure to high resolution, progressively scanned, noise-free computer displays, they make sub-cognitive comparisons to their television picture.

The advertising industry is in agreement that addressable advertising holds great promise. However, it targeted

advertising is to succeed, then overcoming the inherent fear of intrusion experienced by the consumer who knows he is being targeted or monitored is critical.
Technological changes in the television industry will affect advertisers at least as profoundly as they will programmers. The impact of enhanced addressibility on their advertising effectiveness will define their place in this fast-evolving environment.

Further, as subscribers migrate toward ever-larger-screen television displays, they grow accustomed to the levels of resolution delivered through well maintained, broadband analog systems. Reducing resolution in a digital system to increase effective bandwidth will yield predictable results, results that are understood intuitively, and are not unlike what can be expected in an analog system. Digital pictures, however, are more complex than that. With MPEG coding we've been handed more latitude to change our product than has ever before been ceded to the technical sector. And our choices are rife with pitfalls that the system designer must understand cognitively as well as intuitively. Nowhere is a lack of such understanding more apparent than in the misguided pursuit of coding gain.

A review of the data rates and levels of coding gain we have discussed in this paper, should leave us all with a sense of control. The feeling, however, is too often one of discomfort as well. We can stuff more and more channels into a given space by forcing higher coding gains, greater entropy losses, and more frequent artifacts to become part of our picture. The too-frequent comparison is made to VHS tape and its consumer popularity. The comparison is grossly invalid.

So, rather than a headlong stampede to create more and more channels, at higher and higher compression ratios, a more ordered, creative approach to broadening our fare might be advisable. With the advent of new and unique forms of content, such as adressable advertising, both the business and technical concerns of this new age can be met. The creative use of digital bandwidth can satisfy both the engineer and the accountant.

APPLYING SMART CARDS FOR ACCESS TO DTV BROADCAST

Luc Vantalon
SCM Microsystems, Inc.
Los Gatos, California

ABSTRACT

The Society of Cable Telecommunication Engineers Inc. (SCTE) adopted in April '97, the NRSS conditional access module specifications. This module is designed for removing from end-user television equipment all security functions. It thereby enables consumers to choose their own preferred digital terminal, by breaking the link between cable operators and manufacturers. This paper will demonstrate that this major market advantage has been achieved without undermining the security of the digital set-top box and related services. In fact the NRSS module has the potential to deliver even tighter security.

INTRODUCTION

Currently the smart card is the most optimized integrated, secure device. It performs cryptographic protocols to either authenticate or encrypt digital content. Worldwide recognition has resulted in a high degree of integration. A small chip embedded in a plastic card provides excellent, portable availability of this secure device. Its serial interface reduces its susceptibility to piracy. However when large amounts of data must be secured in real time, the drawbacks of the smart card's high degree of integration make it useless.

With new digital pay-TV applications, a live MPEG2 compressed information stream (70 Mbit/s) must be decrypted, compared to the 115kbit/s maximum capability of smart cards today. Many current and emerging interactive services (pay-per-channel, pay-per-view, video-on-demand, home-shopping, home-banking, interactive-games) need to be managed simultaneously by the Conditional Access System (CAS), dramatically increasing program memory requirements.

The best solution consists of using smart cards only for user data storage, authentication and session key decryption, while routing the MPEG2 decryption and key exchange protocols to a side system. This system can be either the main end-user equipment or a specific module that consolidates the whole security application in a restricted, closed device with standardized interfaces.

Architecture of the NRSS module

Figure 1 depicts a standard NRSS module. It can be thought of as a physical adaptation security layer that reduces the amount of data to be processed by the smart card. It is an isolated, autonomous device which communicates only through its two standardized interfaces.

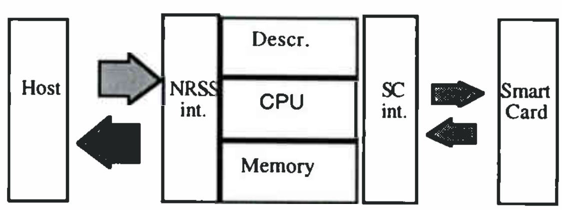

Figure 1: NRSS module architecture

The NRSS interface, physically based on the PC Card standard (PCMCIA), has the necessary bandwidth to carry in and out the 70Mbit/s MPEG2 Transport Stream. It includes internal filtering features that extract the security data from the stream. A bi-directional service channel supports a standardized communication protocol with the host.

The smart card interface is an ISO7816 standardized interface which can drive subscription and banking cards. This interface primarily supports the encrypted/decrypted key exchanges at up to 115kbit/s.

Between these two interface layers, the module includes a complete cryptographic platform that supports the end-user component of the conditional access application. From the relevant data filtering to the pay-per-view inquiry, numerous tasks must be performed before starting the final decryption. Volatile and non-volatile memories store the application's data and program. A powerful symmetric cipher (or descrambler) performs the final decryption of the MPEG2 stream. The conditional access application controls all resources and scheduling.

CAS cryptography overview

As shown on Figure 2, an advanced cryptographic system for a Digital TV CAS is based on three different algorithms. A Public-Key algorithm (1) is first used to deliver case by case authorizations using some Entitlement Management Messages (EMM) to each subscriber. This authorization consists of providing a 'secret key' that could be valid for a month (pay-per-channel subscription) or just for few hours (pay-per-view purchasing). For satellite networks, EMM's are inserted into the MPEG2 stream; for cable networks, EMM's are usually sent through an out-of-band forward channel.

When the smart card has decrypted the 'secret key', it can decrypt the Control Word (CW) carried by some Entitlement Control Messages (ECM). Unlike EMM's, which target just one subscriber or a group of subscribers, ECMs are data elements which that are attached to the content and to the broadcast. CW's are short-term session keys that can change as often as every 5 seconds. They are encrypted by a symmetric cipher (2).

Once a CW has been decrypted, it is transferred from the smart card to the module's descrambler (D3). A synchronization protocol initiated by the head-end handles CW's in pairs (CW_{odd} and CW_{even}) to eliminate any interruption of the descrambling process during the CW decryption.

Some other conditional access systems are using just one cipher block for keys encryption/decryption (E1=E2; D1=D2), but every time they process the two level of keys.

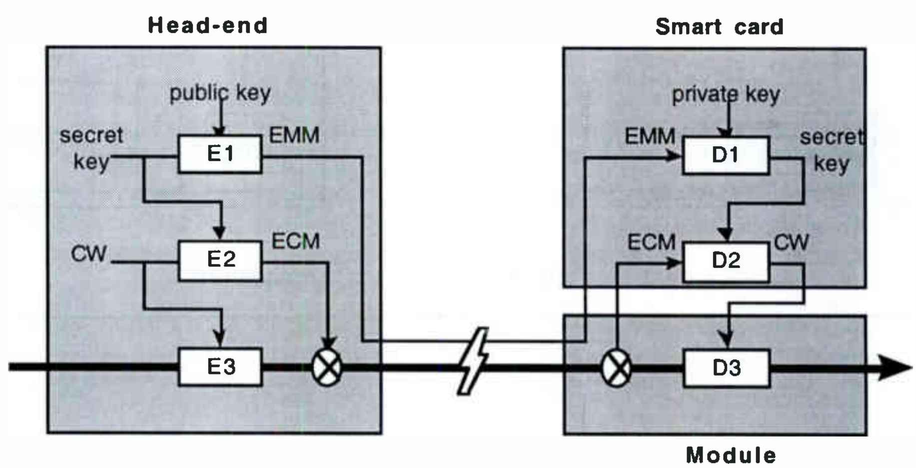

Figure 2: Typical Conditional Access protocols on cable networks

Security of the set-top terminal

Regardless of the complexity of a cryptographic system, the security of a Digital TV conditional access system is one of its weakest element. History has shown that pirates always attack the set-top terminal.

The commonly-used symmetric ciphers for the last stage (descrambler) are based on proven algorithms. The length and validity of the secret key are defined by cryptologists well aware of the existing computer capabilities, to remove any probability of a successful brute-force attack. As is common on a network application, threats are most likely to occur through the key distribution procedure. As shown in Figure 3, both the terminal with or without the NRSS module have an ISO 7816 connector where the decrypted control words could be monitored. The short lifetime of those keys limits their accessibility. Nevertheless a pirate organization could pick them up from the connector and locally re-broadcast the content. This scheme is not very practical since it results in a whole district watching the same movie; but it clearly illustrates the weaknesses inherent in allowing such low speed, clear access to the core decryption process.

In Figure 3, note that the terminal architecture of an NRSS module has another connector (PC Card) with a clear signal available (MPEG2 compressed Transport Stream). Since this is a byte stream running at 70Mbit/s within the terminal itself, clearly it is extremely difficult to intercept even over a span of a few inches! Unless a pirate uses a fully-equipped cable head-end to broadcast the stolen content, even state-of-the-art digital technology such as IEEE1394 (Firewire) could not deliver this data stream farther than a few meters.

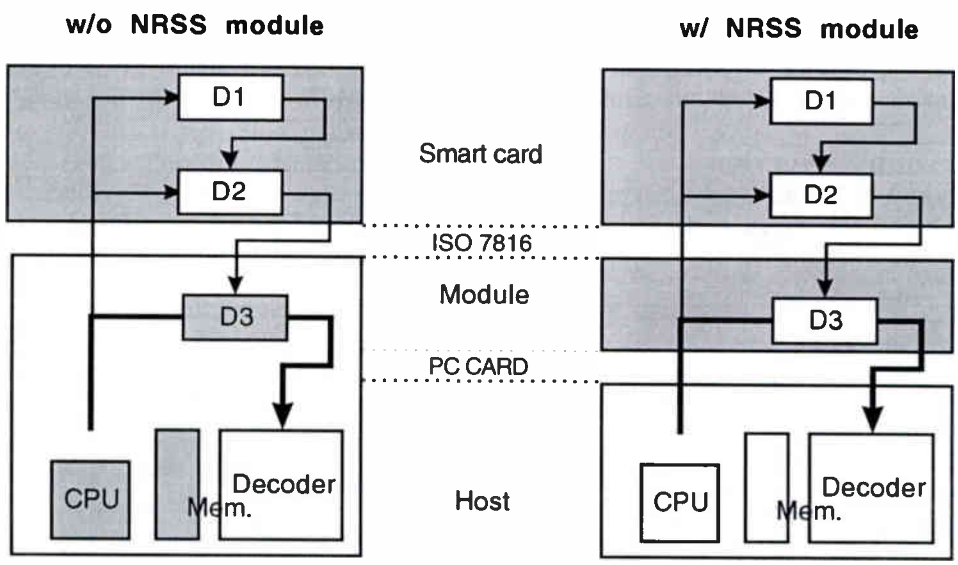

Figure 3: The Set-top terminal architecture

From the network security point of view, this process consists of checking that only the paying, authorized subscriber, is able to access the content; the availability of the MPEG2 transport stream in the clear on a connector does not change anything.

Another weakness of the non-NRSS terminal is that a piece of the security software is running in the terminal. This software is capable of extracting the relevant security information, including addresses and keys. This permits a basic cryptanalysis of the systems to be performed, leading to a potential security breakdown. When an NRSS module is used, the program owns and controls its dedicated resources (CPU and memory), which are defined by the CAS supplier. When a module is not used, the security application shares the resources of a generic platform that supports many other applications, and therefore will always be more vulnerable to compromise. For example, most current set-top terminal implementations are based on erasable non-volatile memories (FLASH) to facilitate downloading of new software versions. Having this sensitive software stored in the terminal fundamentally reduces the system's integrity and security.

In conclusion, the smart card remains at the core of the security system. Since the MPEG2 descrambler and the complete security software cannot be integrated onto the card, this configuration creates vulnerability to content theft at the interfaces. 'Spying' the terminal at these vulnerable points gives the hacker an opportunity to reverse engineer the system's cryptography, leading to potential piracy. Currently, the NRSS module is the best solution to emulate the idealized - but not existing - 'super smart card', providing low-cost and ease-of-use benefits with very high security.

Improvements using a NRSS module

The NRSS module has a high bandwidth standardized interface, an MPEG2 optimized descrambler, a dedicated CPU and scaleable memory. These features make it very competitive as a surrogate for a 'super smart card'. It is important to note, however, that its numerous I/Os and multi-chip implementation does give less protection than a completely integrated smart card would. This is why the smart card still must remain the core of the security system. The NRSS module is the second and last physical layer of security.

Since this technology component is 100% dedicated to the security issue, it can be adjusted to the security level required by any CAS supplier. This is a benefit which a set-top terminal <u>not</u> using an NRSS module can not have.

As smart card capabilities improve, the next generation of the architecture will consist of creating a secure channel between the smart card and the module. This will be based on an authentication, key exchange and encryption protocol that requests extra decryption resources and secret storage of keys at the module level. The session key can be initiated either by the module or the card, depending on various criteria like export rules, random pattern generator performance, etc.
Today a typical conditional access module uses 128-256kB of non-volatile memory and 32-64kB of volatile memory. These sizes correspond currently available low-cost components, and can be upgraded easily. The security drawback of this segmentation can be reduced by cryptographic protection. In this case the CPU of the module is both the sender and the receiver so it is the only device with access to the secret key, and can change it frequently. However the memory unit is random access memory, so the encryption should be based on a Electronic Code Book (ECB) no larger than the memory word.

Another concern is to control the program code stored in the external non-volatile memory. This is particularly vulnerable when downloading a new software version or during application execution. A secure one-way hash function can be included into the module to perform this verification.

At SCM Microsystems, we are committed to providing NRSS modules that combine the flexibility of a set-top terminal and the security of a smart card. Our current solutions greatly reduce equipment-specific constraints, while providing the CAS supplier with a homogeneous platform which supports their own smart cards and software requirements.

Copyright protection extension

The focus of copyright protection is on protecting digital content from illegal copying once it is descrambled within a set-top terminal or DVD player, and exposed to external devices (PCs or digital VCRs) through a high-performance serial bus such as IEEE 1394. Although the output of the NRSS module is not a clear digital signal, since it has to be demultiplexed and decompressed before being watched, it is not unreasonable to anticipate that this issue will have to be addressed in the module's definition in the near future.

The actual security functions available within the NRSS module are sufficient to provide significant levels of authentication and key exchange protocols between the module and any consumer video equipment. Should a common cipher system be defined by the consumer electronics industry or some representative consortium, it can be added easily into the module like a post-processor of the CAS security application.

A typical system should work without any assistance from a central server, meaning that the session key should be valid as long as the two electronic devices are connected. This will require extending the key length compared to the current MPEG2 descrambler. Knowing that the bit rate to be processed remains the same, this greater key length will highly increase the complexity of the integrated function. It should be noted that current processing capabilities will not support the integration of this function due to the power consumption level which would be required. The power dissipation of such a PC Card device would elevate the operating temperature well over standard consumer levels. A practical alternative solution is to copyright-encrypt the valuable content before entering the conditional access protocols at the head-end level.

CONCLUSION

Starting from our first discussion of the limitations of smart cards to fully manage a digital TV CAS, we demonstrated that the NRSS module provides the minimum feature set, combined with a smart card, to provide an efficient solution without changing current cryptographic systems. Integrating all the security functions within a closed, standard interface based sub-system, results in greatly improved set-top terminal security, without compromising ease-of-use. As a fully security oriented device, the NRSS module can be an important piece of the CAS supplier's strategy, since they can adopt this solution without compromising. Additionally, the module can support any type of further security extension like the copyright protection.

However, like all systems that protect valuable content, sought by large numbers of people, it is reasonable to believe that the current technology one day will be surpassed by more advanced pirating tools. Before that time, the NRSS module will have an absolute advantage that a set-top terminal will never have: it is RENEWABLE!.

Note

SCM Microsystems' strategic focus consists of securing digital information by providing standard platforms that support network security applications at the end-user level. For the digital TV market, SCM Microsystems was the world's first to demonstrate and promote Conditional Access Modules (SwapAccess™) based on the European DVB Common Interface standard which is closely related to the NRSS standard. SCM Microsystems today leads in the production, deployment and advancement of this technology.

Working with the main CAS suppliers to port their applications onto its modules, SCM Microsystems has been compiling unique experience in the intrinsic hardware security of this architecture, independent of any specific direct broadcast and interactive services security applications. Our experience base is contributing to the development of a product road-map highlighting the cryptographic functions that have to be (1) specific to different geographic areas (descrambler), (2) specific to different suppliers (key decryption) and (3) the ones that can be generic and shared by all the applications (copy protection, data/code encryption, etc).

A PROGRAMMABLE ARCHITECTURE FOR DIGITAL TELEVISION

Neil Mitchell
Philips Semiconductors, TriMedia Product Group
Sunnyvale, CA

ABSTRACT

As the expansion of technology in the consumer electronics market enables new innovative products, strong growth for the next generation of consumer electronics is expected. One of the first markets to revolutionize will be digital video. Implementing intelligent web-aware digital televisions (DTVs) that deliver movie-quality pictures and sound will require a fundamental change in hardware architectures.

This paper will discuss the benefits and limitations of current platforms for DTV and explain the advantages of a programmable media processor as an alternative. Existing technologies only address digital media processing exclusively for the PC or exclusively for consumer applications. Recognizing this dilemma, Philips enables cross-platform solutions that deliver new and unique types of content and complex media.

INTRODUCTION

The Standard

Prior to 1995 there was much discussion and disagreement on what High Definition Television (HDTV) should be. On one hand, the consumer electronics industry attempted to get its recommendations adopted via the "Grand Alliance" (AT&T, Zenith Electronics Corporation, General Instrument Corporation, Massachusetts Institute of Technology, Philips Consumer Electronics, Thomson Consumer Electronics and the David Sarnoff Research Center) while the computer industry eyed up a huge new potential market. Both attempted, and to some degree succeeded, in affecting the outcome of the decision making process. Finally, in September 1995 the Advanced Television System Committee (ATSC) put forward the standard for High Definition Television (HDTV) transmission and display[1].

This was a compromise as it defined the transmission and display resolutions allowable for high definition and not many of the details to mandate exactly how HDTV should be transmitted, received and displayed. The premise for this decision was to "let the market decide" on these details. Rather than select a single high definition resolution, the ATSC committee selected 18 different formats that included a selection of progressive scan and interlaced scan at varying resolutions. The transmission and display resolutions also spanned from what is seen today as "Standard Definition Television" (SDTV) through to the highest resolution of 1920 pixels by 1080 lines.

Reed Hunt, then the FCC chairman, wanted the buy in from both the consumer electronics industry, the broadcast industry and a new player to television, the computer industry. The 18 ATSC formats were his tool to achieve this. There was no way a single standard could be adopted without repudiating one or more of the important players. So the 18 ATSC formats enabled all parties to shift HDTV into high gear without mandating something that would exclude any players and hence slow the whole process.

Table 1 - The 18 ATSC Transmission and Display Picture Formats

Vertical Lines	Pixels	Aspect Ratio	Picture Rate
1080	1920	16:9	60I, 30P, 24P
720	1280	16:9	60P, 30P, 24P
480	704	16:9 & 4:3	60P, 60I, 30P, 24P
480	640	4:3	60P, 60I, 30P, 24P

I=Interlace Scan, P=Progressive

Display Technology and The Electronics Behind Them

Although this did give the desired momentum to start the HDTV bandwagon rolling, it still to this day has not placated all the players. From day one, the computer industry has been in favor of progressive scan formats, while the consumer industry continues to favor interlaced formats. Each industry has naturally favored the format in which it has most experience and knowledge. Although progressive displays can result in less "flicker," there is a huge penalty to pay (in silicon cost, described later).

At the higher resolutions within the ATSC table of formats, even the interlaced formats produce crystal sharp images without the cost associated with supporting progressive processing. The lines are so fine that the "interlacing" effects that can be seen with regular NTSC TV (shimmering of the whole screen) are undetectable to the normal human eye. However, the computer industry headed by Microsoft, Intel and Compaq (The "DTV Team") made a sub-selection of the 18 ATSC formats and have been pushing these as the standards broadcasters should accept. This subset is known as HD0[2].

Table 2 - DTV Team HD0

Vertical Lines	Pixel	Aspect Ratio	Picture Rate
720	1280	16:9	24P
480	720	16:9 & 4:3	60P, 30P, 24P
480	640	16:9 & 4:3	60P, 30P, 24P

The drawback of supporting progressive displays is that in the digital electronics world, to drive the same frame rate display with progressive scan over interlaced scanning requires twice the amount of bandwidth and twice the amount of processing power. This translates to twice the amount of silicon or silicon complexity and buses that are twice as fast or twice as large. In turn, significantly greater costs and risks are added in order to produce such devices. The added cost is something the consumer industry has not been willing to swallow. Although this industry does not mind the inclusion of progressive displays it does not want to impose on the public a requirement for progressive displays, which are in themselves more expensive.

Everyone admits that progressive displays will play an important roll in the emerging HDTV market. There are already Plasma displays entering the market that are, by nature, progressive scan devices (but are currently in the $15,000 price range). The consumer industry is certainly in favor of these. But the disagreement has been that the computer industry has, until recently, refused to compromise on its stance against interlaced displays. This recent development has Intel announcing support for both progressive and interlaced support.

With the inherent cost advantage of interlaced displays over progressive displays, it is vital to the consumer industry for these devices to be supported. This argument is born out further by the price the electronics proposed to drive the displays in the two different industries. High-end projection televisions for home theater systems are currently in the thousands of dollars although the electronics will drive them to the sub $500 range. This cost will drop just as there are low end TVs in the $200 range today. The PC industry is struggling to get systems down to under $1000 and this price range often does not include a display at all.

In favor of the computer industry, its lack of support for the highest resolution for digital broadcasts does result in a cost advantage by reducing memory needs in the end-user's appliance. The resolution is approximately half that of 1080I for the next step down (720P) with half the amount of memory required.

The DTV team claims comparisons of 1080I versus 720P prove little difference in perceived quality. The DTV team also claims that many digital televisions devices will not be able to display1080I, but instead there will be set-top boxes (STBs) that receive whatever format is transmitted and "down-convert" for display on a lower resolution (and possibly a conventional) television. This argument leads to the DTV Team's main point, which is why transmit 1080I if 720P will do in most cases, especially if much of the time higher resolutions are to be down-converted to 720P/I or 480P/I before being displayed?

One of the main reasons for the computer industry to convince broadcasters to transmit 720P instead of 1080I is to take some of that bandwidth away from audio and video and allow the transmission of other data ("data services"). Those in the consumer industry are in favor of data services as well, but want the option of transmitting 1080I for some content. Even at 1080I there is some small headroom for data. Another option is to have a complete channel assigned to data services with multiple other channels assigned to audio and video.

Microsoft also has HD1 and HD2 waiting in the wings. HD1 includes 17 of the 18 ATSC formats. (See table 3)

Table 3 - DTV Team HD1

Vertical Lines	Pixel	Aspect Ratio	Picture Rate
1080	1920	16:9	24P
720	1280	16:9	24P, 60P
480	720	16:9 & 4:3	60P, 30P, 24P
480	640	16:9 & 4:3	60P, 30P, 24P

With HD1, the 1280 x 720 60P format is introduced. In the digital TV appliance the amount of processing required for 720 x 1280 60P is similar to that required for full 1080I (1920 x 1080 60I).

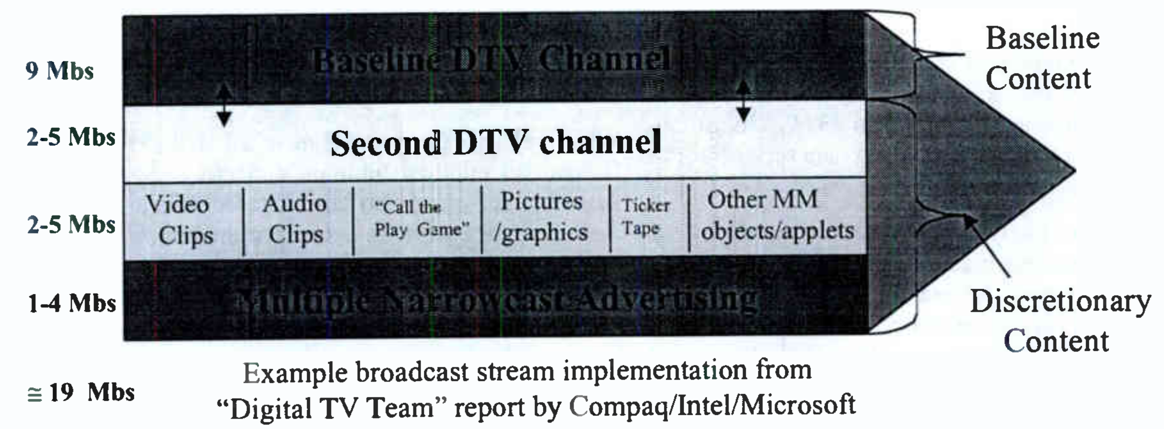

Figure 1 - Discretionary Content

There is still the argument that using 720P over 1080I uses less memory. This is true. However, semiconductor companies such as Philips Semiconductors are introducing mechanisms to compact the amount of memory required in MP@HL video decoding. A full 1080I video decode will be done in half the amount of memory normally required for a HDTV high level video decode.

Microsoft has even suggested an HD2 superset to the 18 ATSC formats. This is the first mode that steps outside of the ATSC standard and introduces 1920 x 1080 50P. This will require significantly more performance than any of the 18 ATSC formats.

The main message here is that within the 18 ATSC formats there is going to be a variety of these different modes broadcast, including 1920 x 1080 60I. There will also be data services provided in the remaining bandwidth and possibly in separate data channels. There will be displays in the market accepting and displaying full 1920 x 1080 60I as well as PC appliances accepting and/or displaying less. There will even be some consumer appliances receiving up to 1080I and down-converting for viewing on lesser displays and even on traditional TVs via SVHS input. I will explain in a short while how manufacturers cope with all this complexity.

HDTV ROLLOUT

There will be consumer DTV receivers available this year, but what about the availability of HD signals? Susan Ness FCC Commissioner discussed the commitments from broadcasters in a speech back in April 1997[3]. "On the commercial side, 24 network affiliates in the top ten markets have volunteered to do everything within their power to be on the air in digital by Christmas 1998. These commitments -- which include multiple stations in nine of the top ten markets -- evidence the strong desire of broadcasters to go digital." Ness went on to add, "Although the 18-month commitments were voluntary, the four major commercial network O & O's and affiliates are required by rule to commence digital broadcasting in the top ten markets within 24 months, and in the top 30 markets within 2½ years, in time for the 1999 Christmas shopping season." Thus, by the end of the 1990s, over fifty percent of U.S. households will have access to digital broadcasting if they so desire. This rollout has been closely watched, as many believed this was far too optimistic.

One bottleneck anticipated was the building of HDTV transmission towers. However, recently Cowles/SIMBA[4] reported "We're very encouraged that things are on schedule," said Bruce Allan, VP/GM of Harris' broadcast division. "We do not foresee any major issues that would preclude a timely rollout of digital television."

One of the more significant occurrences that will further the momentum being gained for HD transmissions is that of DirectTV. The company announced it will transmit HDTV signals this year throughout the U.S.[5]. This immediately expands the markets served from those Susan Ness discussed to the whole country this year rather than waiting until 1999.

WHAT ARE DATA SERVICES?

Data services can be infinite. All the discussion about resolutions is really a discussion about how the bandwidth provided to the broadcasters will be used. Many of the broadcasters have committed to some forms of HDTV audio and video transmissions. ABC, HBO and PBS are just examples. HBO has committed to 1080I transmissions for at least some of its content, as have many others. In addition to audio and video, broadcasters are also expected to slowly initiate transmission of other data. PBS has committed to data services this year.

This data could come for free or could allow a new revenue stream to multiple system operators (MSOs), broadcasters or even others. The consumer industry sees new subscription TV channels as new revenue streams and embraces data services. Either way, HDTV will allow broadcasters to decide how they split each 6MHz channel (which translates to 19.4Mb/s). Each could carry a single 1080I stream with a little left over for data services, or a lower resolution with other accompanying data services, or even multiple sub 1080I channels.

What are these services likely to be? There will no doubt be a lot of "content sensitive material" transmitted as data along with audio and video. An example includes transmission of sports players' statistics so that when a viewer is watching a game s/he can quickly click on a button to see profile and background information. These services may come for free or they may be subscription services that allow broadcasters more revenue sources.

Another example is in the area of advertising. This is one area where TV is hugely successful today. During a football game, several advertisements may be transmitted simultaneously at a lower resolution than the game itself. This in itself is not a data service. However, the receiver may use subscriber information to select the most appropriate advertisement to show the subscriber. Is it the expensive executive car ad, the ad for beer or one for ladies' perfume?

At the same time an icon at the top right of the screen may beckon the viewer to seek more information. If the user responds the information about that product could be displayed using the on-screen display. The next step in data services may be to combine this nationally broadcast data with locally inserted data such as the local dealer list for the viewer who responds to the car ad. This can be taken even further by allowing the DTV device to be linked to the Internet. The viewer could then request with a single key press to connect to a local dealers' Internet site and view the car, availability and even pick a particular model and select to see it viewed with specific options and add-ons. This is possible with a computer today. There is no reason it can not be done with a lower cost consumer device.

However, there is one huge difference between computer and consumer device Internet access. With a computer, a user accesses a web site and then when s/he comes across a site that uses Java™, Shockwave or some other foreign interpreter, has to change to a different web site, load the new interpreter, go back to the original web site after installing it and then access the site. This process must be completely automated for the consumer device. The user accesses the web site, the DTV appliance notices that it does not understand the object coming to it, finds out where to get it, obtains it, installs it and then accesses the web site as if nothing had happened. This works well on many platforms but as the algorithm downloaded becomes more complex, a more able subsystem has to be ready to execute that algorithm. What if the site has MPEG1, MPEG2, MPEG4, streaming video, etc.? This has impact on the DTV appliance in the home! It takes more than a simple RISC CPU to achieve this.

What is clear is that processing of all 18 different ATSC formats is a requirement in the end user equipment and manufacturers must also cater to these new, as yet undefined data services. Access to the Internet is certain to happen although the form may vary. As inferred above it may not be in the form of a Netscape/Microsoft browser but in a form tied into the content being viewed at the time by the subscriber.

THE CLIENT APPLIANCE — OPTIONS

There is huge debate about whether the HDTV appliance will be a PC, a STB, a TV, a network computer or some other flavor. It is likely that there will be TV- type appliances for distant viewing and PC appliances for closer more highly interactive interaction. Both types of appliances will have access to the Internet in one form or other.

There are also a number of options necessary for the client device in the user's home depending on the style of interaction and viewing style. All these appliances, if they are to receive and display audio and video, will have to have a basic set of functions. The details of such functions are beyond the scope of this paper. However there are a number of ways to implement these core functions. Many of the interfaces must be in hardware but the core functions of the system can be implemented in a variety of ways:

- All in hardware (with a small micro to control it all)
- Largely in software on a media processor core
- All in software (e.g. x86 and/or media processor)

Let us examine these options and why the differences are important to the broadcast, MSO, satellite, content and distribution industries.

All of these three can be applicable to simple audio and video implementations of the client DTV appliance. However, some of them have issues that must be well understood.

All Hardware

A solution, which lies mostly in hardware, can be seen today in many of the satellite systems that exist in people's homes today. To keep the cost as low as possible they are implemented with a conventional microprocessor. This severely limits what these systems can do apart from basic audio and video display. With many of the components hardwired such as an MPEG2 decoder and audio decoder, the system can really only handle the media it was originally designed for and lacks expandability offered by the other two solutions. In the world of data services, a hardwired solution will be severely hampered and will be unable to process much of the content sent to it.

All Software

In an ideal world this would be the preferred solution since its allows plenty of flexibility. Developers can implement software-only systems today with the all the functionality of a hardwired system as described above. However, this scenario raises two questions: Is software a cost effective solution in the implementation chosen and does a software-only solution impact any other parts of the system?

There are two types of software solutions that come to mind. One is the PC type system that PC vendors have been in favor of. This uses the expandable nature of the PC to enable the audio and video interfaces to be added, often via a PCI bus card. The majority of the audio and video decoding then takes place on the x86 microprocessor such as the Pentium/PentiumII. The problem here is that the Pentium is neither fast nor powerful enough for MPEG2 MP@ML (standard definition) video decoding without compromising quality and frame dropping. This is okay for the PC world today but is not appropriate for a consumer viewable device that is to be used as the main viewing platform in the home. The Pentium 200MHz is barely capable of MPEG1 decoding let alone MPEG2 MP@ML. The PentiumII at 266MHz gets closer, but then we are still talking about standard definition MPEG2 video decoding.

The jump in performance required from standard definition MPEG2 video decoding to 1080I is a factor of six. The jump to Microsoft's HD0 is lower than six but is still is over twice that of standard definition. Perhaps a new Intel architecture in the wings is designed to handle this factor of two and not the factor of six for 1080I. Obviously, this is some of the motivation behind HD0.

This scenario still has some issues. While such a PC system is doing video decoding, can it also do the audio decoding in software simultaneously? Also, what happens to the rest of the PC system? The system is saturated with the task of audio and video processing and certainly has nothing left over for normal PC type functions like editing a spreadsheet. These proposed systems would have to be the latest and greatest higher end PC technologies and not the sub $1000 PCs that are doing so well for the industry today.

Then there are the issues of what size display this should be viewed on and what the cost of that display should be. The PC industry will likely pick a smaller display as the viewing distance is closer and hence the display device will be less expensive (at least initially). This is one huge advantage for the PC world. Additionally, if the PC user is willing to put up with a few dropped frames and possibly some audio artifacts once in a while, then the PC world will have a solution for the user's den . . . but not for the living room.

By adding some hardware to support the necessary decoding, the PC will have to increase in cost. Suddenly we have a hybrid that could be called a PC or could be called something else.

Another option is to add a media processor to the PC either on a PCI card or on the motherboard. This becomes interesting as this off loads the x86 and still offers a software-orientated solution. An example of this is the Philips Semiconductors TriMedia™ media processor. A media processor may also help drive a move to the often mentioned "convergence appliance" that is a hybrid PC and consumer device. The secret here is to make sure the chosen solution is able to handle the media processing requirements with little overhead imposed on the x86 system.

The second "all software" approach is one offered by a stand-alone media processor. There have been many attempts to market a media processor that is cost effective for the consumer and PC market place. Some have come and gone while others have certain limitations such as requiring a host processor to share the load. Philips' TriMedia processor can be used with or without a host CPU and is applicable in this market. However, devices on the market today are capable of standard definition video decoding and not high definition (in a single chip). Future devices will be able to handle all audio, video and data service processing in a single cost-effective device.

Some media processors achieve this tremendous processing potential through a VLIW core (Very Long Instruction Word). For example, each cycle 5 CustomOps (Intel MMX type operations) can be executed at a rate of approximately five instructions operating on four objects 100 million times per second with special operations allowing a performance of 4BOPs (Billions of Operation Per Second). This is all achieved by programming the device with a conventional C or C++ compiler. Next generation devices will operate faster and be able to handle HDTV audio & video and data processing.

Largely in Software on a Media Processor Core

This type of solution attempts to offer the advantages of a VLIW media processor with a small amount of flexible acceleration hardware. This way the desired performance can be reached without impacting the programmable nature of the device while minimizing the cost of the device and system it is applied to. Naturally the secret is in the implementation of the flexible acceleration and how this interacts seamlessly with the VLIW core.

Media processors may be applied to PCs as it does not require much host overhead or it can be used very effectively in a stand-alone consumer device such as a STB, Digital TV or network computer. Many are able to handle the required audio and video while also carrying out the traditional housekeeping and organizing functions that a conventional RISC can do as well.

WHY THE HOME CLIENT IMPLEMENTATION IS IMPORTANT

Broadcasters, MSOs, Telcos (if they jump back into the market) and content providers should think about the implementation of the client box. They all accept that PCs will be there to receive digital TV and the data services that come with it. However, not everyone has a PC and the penetration of TV-type devices will continue to far exceed that of PCs for the foreseeable future. The question then remains, "If we assume PCs are a player, what is the consumer, long-distance, viewing device going to look like and how should it be implemented?"

The Importance of a Cost Effective, Flexible Open System

Just as PCs have an open architecture, it is vital for the consumer device to have a similar open system. This means any company can create software for it easily and effectively with as little economical, political or technical resistance. Ideally a small variety of operating systems could also play a role. There are those that oppose WindowsCE (WinCE) and those that favor it. It is likely that it will play a role due to Microsoft's partnership with TCI Cablevision. However, one thing is certain -- WinCE will not be the only operating system in this market. Other players such as PSOS and VRTX may take part and be more readily adopted by the consumer electronics industry as they seek alternatives that prevent the market from being predominantly owned by one operating system.

Consumer companies are concerned that even WinCE comes with much baggage that bloats the memory required in a consumer device and drives up costs. Also, once designed in and as the operating system grows, consumer manufacturers will then not have an alternative and become dependent on a closed architecture. By having a choice of operating systems, consumer companies can avoid such a situation and evaluate various open architectures that best fit their needs.

The broadcasters, MSOs, content providers and others need to understand the pros and cons of selecting each of the three architectures previously described (hardware, software and a combination of both).

Issues with Hardwired Solutions

Hardwired solutions have been adopted, as they were initially the only solutions available. Also they were relatively cost effective. With the never-ending march of semiconductor technology and the availability of small geometry process technologies, media processors are now a cost-effective viable option.

Hardwired solutions do not allow for systems that are flexible. There is the uncertainty of the video and audio-processing requirements as has been described here within and there is also the huge unknown about data service processing. With a low performance 32-bit microprocessor

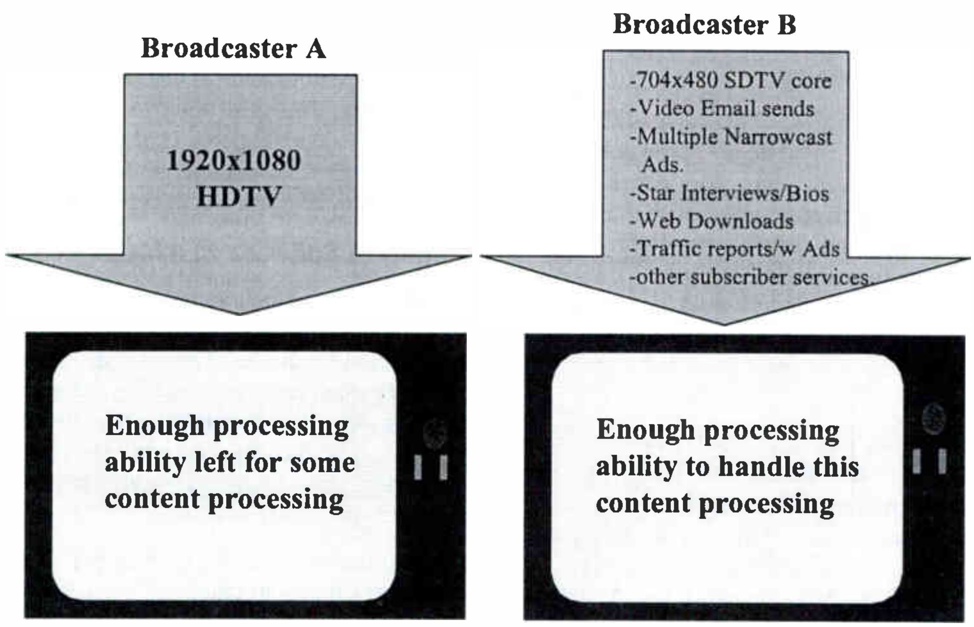

Figure 2 - Switching channels means more than Audio & Video Switching

or something incapable of processing audio and video, a hardwired solution is less able to bend itself in the direction of the media being sent to it. Why should the architecture of the box limit the data broadcasters and content providers want to send it?

Issues with PC Solutions

The major advantage of the PC has been its flexibility. The problem with this has been that more functions get added to the PC and the price of the system remains largely unchanged. The recent influx of sub $1000 PCs has come by using lower performance CPUs, less memory, smaller hard disk drives and often not including a monitor. The type of system required for good audio and video is not even on the market yet, but when it does, it will be with the higher end PCs in the $3000 range with a small display (by consumer electronics standards). PCs will be able to handle the variety of content but at a high price.

Another issue with PCs and even WinCE appliances is reliability. The PC architecture by its very nature is open to crashes and system errors of the type that are not acceptable in consumer appliances.

The Best of Both Worlds?

Figure 3 - TriMedia ATSC DTV Reference Platform

There is no such thing as an ideal solution. However, programmable media processor solutions take the consumer cost infrastructure and apply it to audio, video and data processing in a manner that is flexible like a PC. But the cost is only slightly more than a hardwired solution. Media processors can be applied to TVs, STBs, PCs and convergence appliances in a way that brings the software flexibility and upgradability of the PC to the consumer world of STBs that cost from $200-$500.

Why is this level of flexibility important? There is a wide range of possible video, audio and data signals that could be received by the client device. All these signals must be managed seamlessly as if the platform was always designed to handle whatever new data is sent to it. What happens when the user switches channels? One broadcaster may be transmitting 1080I while another transmits a lesser ATSC resolution with added data services. The digital TV appliance must react to this channel change and be able to cope with the new data that it has just encountered, even if it is data of a type it has never seen before.

EXAMPLE OF A DTV PLATFORM

A fully functional high-definition television platform based on cost-effective media processors will be ready for market entry this year. One example might consist of a board that is applicable to TV, STB or PC incarnations of a digital television, libraries required for the platform to function as a complete DTV system, example software, documentation and the necessary software tools such as compilers, debuggers and development aids.

An ideal platform should be capable of handling all 18 ATSC formats in terms or reception and display and require no additional microprocessor support. The TriMedia device handles all media and control processing, proving that VLIW architectures can address both DSP orientated processing as well as control processing (Intel agrees with this with their next generation VLIW x86 device code named Merced).

As with any STB or digital TV, it is important to carry forward the flexibility concept into the system design. An ideal platform should also allow for a pluggable network interface module (NIM) containing a tuner, IF and the demodulation required for the particular system concerned. There will be separate NIMs for Satellite (QPSK), Cable (QAM & others) and ATSC Terrestrial broadcast reception (VSB-8). The TriMedia design uses a NIM based on Philips Semiconductors' single chip VSB demodulator that takes in a modulated stream direct from the IF and outputs a transport stream suitable for streaming directly into the TriMedia processor.

The majority of the rest of the system should be done in software with some hardware acceleration for the video decoding. This acceleration should be extremely flexible with multiple filtering schemes and memory reduction techniques. Platforms like this, including those based on Philips' TriMedia programmable media processors are planned for retail sales by Christmas 1998.

Philips has taken the approach to include all software required for the system to function as a stand-alone digital TV appliance. Philips also allows customers to write their own applications with development tools. Manufacturers should also look for solutions that include a number of extra modules such as soft-modems, which allow the basic system functionality to be dramatically changed by simply swapping in and out well-defined software modules with their well-defined system interfaces (APIs). The ATSC demux could be swapped out and replaced with a different demux for the required TV appliance, examples may include DSS demux for satellite, DVB demux for European satellite and program stream demux for DVD. This paradigm should exist for the rest of the system, too. For example, the Dolby AC3 surround sound module could be swapped for a European MPEG 2 audio. When the Japanese agree on their new audio standard, this could be implemented quickly in software, validated and can then also be provided as part of a Japanese targeted solution (remember this is with the same hardware).

Thus, a platform like this would be applicable to worldwide markets and can grow as each of those markets evolves.

Manufacturers can achieve this shape shifting not only before the product is shipped to the customer's home but also while it is in a customer's home. The MSOs/Telcos can obtain a new form of revenue by offering downloadable/upgradable modules that are software independent. This may be Java applets that some architecture may be able to download and execute. However, with a media processor as the base of the system, other functions can be downloaded that middleware

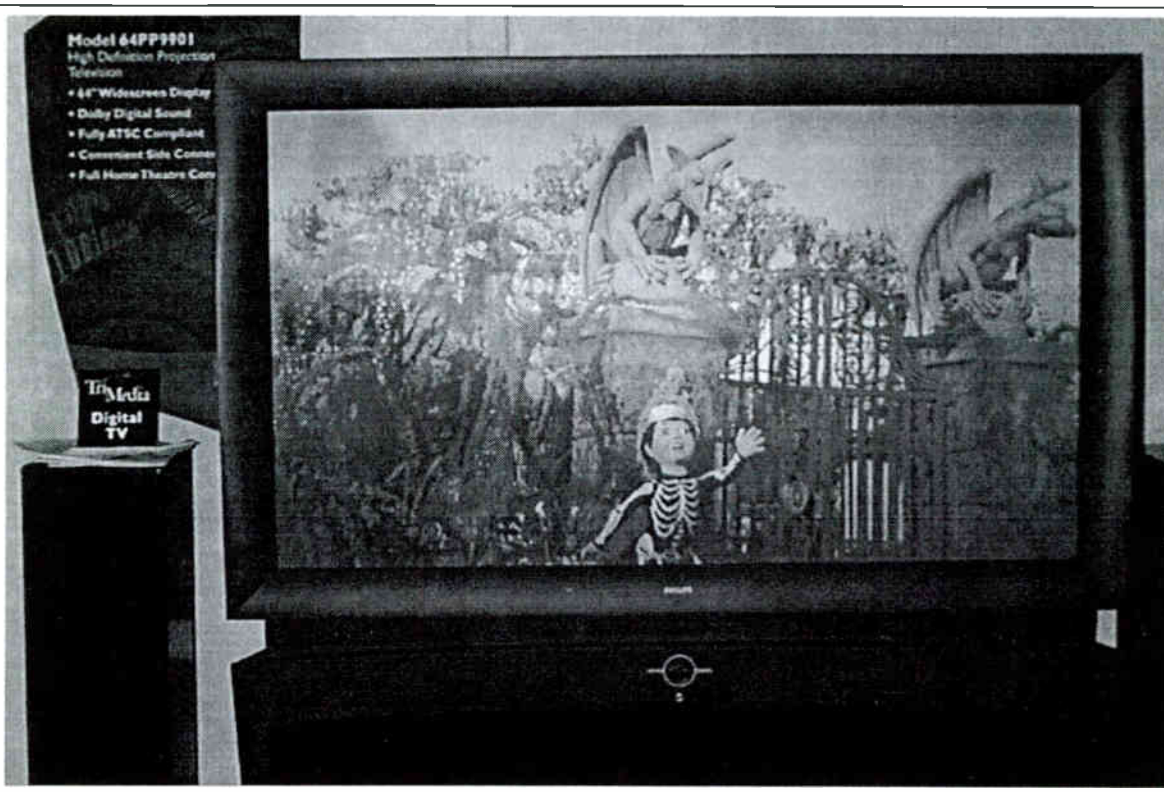

Figure 4 - Philips Consumer Electronics HDTV

systems like Java just can not provide.

These modules can be downloaded over the broadcast medium, across the Internet or any other network. The user may request it via a modem connection of some description and once the request is verified and the financial transaction (if any) is made, the software can be downloaded to the digital TV appliance and run. These services can be more than Java applets, they could be email, conventional Internet browser, video/audio e-mail and even video conferencing (that is already running on TriMedia systems in products today). What about fundamental system upgrades? Want to upgrade to an MPEG4 capable box? There is no need for the MSO to send a truck out to each home or request customers to return their STB for upgrade. Simply download the new algorithm onto the media processor across the existing network and run it.

LEGACY TV

Traditional TV broadcasts are going to be around for a long time. The most optimistic view is that of the FCC with the requirement for all broadcasters to cease NTSC transmissions in the year 2006. Until then, even digital TV appliances should be capable of receiving NTSC signals. After an NTSC signal has been received, a variety of things could happen depending on the digital TV appliance concerned. Some systems may not be able to do much other than pass a signal through and display it. This is true of hardwired solutions.

With a media processor the NTSC signal can readily be digitized and processed to improve the quality of the resultant audio and video. This is important, as expectations of consumers will rise as more and more of them see HDTV. Also when a consumer has a DTV system in their home, especially one capable of 1080I, they want to see an improvement in their regular TV reception as well. Even if the consumer has a STB capable of receiving all 18 ATSC formats and that down-converts them for display on a regular TV, the digital pictures will be better quality than when they tune to a raw NTSC source. Hardwired solutions offer no advantage here. Media processors can be applied to the processing of NTSC signals when the user is not watching HDTV signals. The whole media engine can be used for both spatial and temporal techniques for picture quality improvement.

Philips Semiconductors has a proven algorithm already running on the TriMedia processor that uses a "Natural Motion" algorithm to take the NTSC source (or film source) and de-interlace/line double and do frame "in-betweening." In-betweening is the creation of extra frames not in the original source received by the DTV appliance. The easiest way may be to just show a copy of the previous frame, but this makes motions jerky and really does not provide significant advantage over non-in-betweened sequences. However, by using motion-estimation/compensation on the two received frames, an in-between frame can be intelligently generated that takes into account the movement of objects from one frame to the next.

The resultant sequences has sharper frames (due to the de-interlacing/line doubling) and smoother motion as there are twice the number of frames displayed with motion smoothly added to the DTV appliance generated frames.

CONCLUSION

The key to HDTV's success depends on a variety of factors. One of them will be the ability to drive down the cost of the system to the consumer level. Initial consumer electronics products are going to range from low-cost set top converter products (a.k.a. set top boxes) and expensive 60+" projection HDTVs at 1920x1080 interlaced (known as 1080I) resolution. The consumer industry has initially decided to push the large expensive displays, which are 108I capable—these being the most appropriate to appreciate the benefit of the higher resolutions. Nearly every major consumer electronics company has displayed HDTVs capable of receiving and displaying 1080I.

At the same time there will be lower cost STBs from both the computer and consumer industries. Some STBs will be able to receive all 18 ATSC formats and display them on a lower resolution screen or conventional TV. Due to the digital quality of the received signal, the resultant displayed image will be better than if receiving a conventional NTSC signal. Also many U.S. TVs in customers' homes are able to display twice the number of lines that the conventional NTSC circuitry outputs. Some include basic line doubling and some allow SVHS input from other sources (like DSS STBs) in order to make use of the higher capabilities of the tube.

And, don't forget the audio. ATSC reception comes with Dolby AC3 surround sound. Even if displaying on a standard TV, the receivers get surround sound, too. There will be a market for these boxes due to the better-than-NTSC reception capabilities and the surround sound audio.

There will be some HDTVs in the market awaiting the highest resolution of the ATSC formats 1080I. At the same time, there will be systems capable of receiving and processing the data content that is in the process of being defined today. This will be true regardless of the implementation of this system (STB, TV or PCs).

Programmable media processors are here today and they have the cost structure and performance required to make DTV and data services common place. This is true due to the combination of:

- cost effective application to digital TV

- ability to apply picture improvement techniques to existing NTSC signals
- future proofing of DTV appliances in the home (via downloading of software)
- ability for a single platform to address different markets and,
- solutions that are here today with complete reference platforms.

Broadcasters, MSOs, Telcos, satellite and content companies should be aware that media engines will be a fundamental component in end-user digital and analog TV systems starting from the end of 1998. Additionally, media processors can be a DTV OEM's most powerful weapon in continually dealing with DTV market uncertainty in the fastest, most cost-effective manner without PC architecture baggage. At the same time media processors can be used in a PC to off load the x86 to allow the x86 to do its thing and let the audio and video processing be done elsewhere.

Media processors are applicable to TVs, STBs and PCs for digital audio, video and data processing regardless of the reception method (terrestrial, cable or satellite). They are available today and are being designed into DTV appliances for market entry this year.

TRADEMARKS

TriMedia is a Trademark of Philips Electronics North America Corporation. All other trademarks are property of their respective owners.

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Data Broadcasting for Radio

Thursday, April 9, 1998
9:00 am - 12:00 pm

Chairperson:
David Layer
NAB, Washington, DC

***Update on NRSC HSSC Subcommittee**
David Layer
NAB
Washington, DC

***CUE Networks - Data Broadcasting for the 21st Century**
Gordon Kaiser
Cue Network Corporation
Irvine, CA

Application of High-Speed Digital FM Broadcast Subcarriers to the Intelligent Transportation System
Jim Chadwick
MITRE Company
McLean, VA

***WavePhore Networks - A Glimpse of the Future**
Mo Gardner
WavePhore Networks, Inc.
Salt Lake City, UT

***Advanced Data Subcarrier Technology - The Seiko System**
Lee Balzer
Seiko Communications Systems Inc.
Beaverton, OR

***Data Broadcasting over FM Subcarrier in Japan**
Tsuyoshi Kawabuchi
Tokyo FM Broadcasting Co, Ltd.
Tokyo, Japan

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*Papers not available at the time of publication

Application of High-Speed Digital FM Broadcast Subcarriers to the Intelligent Transportation System

Jim Chadwick[1]
Chief Engineer
Center for Advanced Aviation System Development
The MITRE Corporation

The Intelligent Transportation System

The Intelligent Transportation System (ITS) is the application of advanced computer and communications technologies to reduce traffic congestion, improve surface transportation safety, lower environmental impact from exhaust emissions and increase the utilization of mass transit systems. The program was initiated in 1989 and has moved through several phases in the last nine years. A system architecture was developed and many operational field experiments were done to validate concepts and to assess technologies. ITS is now in early stages of implementation in several cities as part of a "Model Deployment" program.

ITS is not a large Federally run system like air traffic control or defense. Rather, it is being built in a way very similar to the highly successful Interstate Highway System, using both appropriated and Highway Trust Fund money. States and local governments provide funding as well. Implementation and operation is being done by state and local jurisdictions and by private companies.

Since ITS involves communications between a fixed transportation infrastructure and moving vehicles, radio communications are an essential part of the system architecture. To the maximum possible extent, existing commercial telecommunications services are being used to support a set of 29 user services. In some cases, however, dedicated communications channels will be required to meet availability, cost and liability requirements.

A large fraction of the user services depend on the availability of detailed, in-vehicle traffic data in near real time. Because this data forms the basis for many other ITS functions, its early availability is essential. The large amount of data that needs to be sent, combined with its one-way nature and the fact that the FM broadcast infrastructure will support rapid deployment, lead quickly to the decision to use FM broadcast subcarriers as a distribution channel. Recent advances in the mobile performance and capacity of digital subcarrier systems make them even more attractive as a means of data distribution for ITS.

[1] Jim Chadwick, a co-inventor of the STIC digital FM subcarrier system, is with the MITRE Corporation, 1820 Dolley Madison Blvd., Mail Stop W-371, McLean, VA 22102-3481; (703) 883-7010/FAX (703) 883-1242; email: chadwick@mitre.org

ITS Communications Requirements

In almost all communications systems, the inbound and outbound capacity requirements are symmetrical; that is, "what goes out must come back." ITS has no such requirement. In fact, the communications capacity in the outbound (infrastructure-to-vehicle) direction is more than an order of magnitude greater than the inbound (vehicle-to-infrastructure) requirement. This characteristic further favors the use of digital FM broadcast subcarriers in conjunction with a low-bandwidth return link that can be provided at much lower cost than a two-way system that meets the capacity needs of both directions.

Communications capacity requirements for ITS data distribution have been established as part of the system architecture development activities. About 3.5 kilobits/second (kb/s) is the user data rate that is needed for traffic data alone, to meet the needs of a large city like Los Angeles. An additional 2.0 kb/s supports other data services, while projected growth adds another 2.0 kb/s. Thus, an FM subcarrier system needs to provide a user data rate of 7.5 kb/s to meet the needs of ITS. For most error-correcting strategies, this translates to a channel rate of about 16 kb/s.

Because state and local government agencies will be the ultimate funding source for many ITS systems, cost is an important factor. Although it is recognized that in unusual cases it may not be possible, the goal is to use a subcarrier system that can provide data coverage for an entire city from a single FM station. A collateral concern in this area is subcarrier availability, given the growth in the use of analog subcarriers for a variety of applications. In this environment, having to acquire only one station's subcarrier is a distinct advantage.

As importantly, a single, standard subcarrier format that allows vehicles access to the data regardless of their location throughout North America is absolutely essential for ITS. The Federal Highway Administration (FHWA), where the Joint Program Office for ITS resides, has stated as policy the fact that it will only provide funding for a single standard ITS subcarrier system, which must meet these operational requirements:

- Not interfere with the FM station's entertainment channels
- Provide a user data rate of at least 7.5 kb/s
- Be capable of handling large blocks of information to reduce the latency of traffic data
- Be compatible with analog subcarriers at either 67 kHz or 92 kHz
- Be compatible with the Radio Broadcast Data System (RBDS)
- Provide low-error-rate data to moving vehicles in a variety of environments
- Be capable of providing coverage to typical cities in the U.S. from a single FM station

Digital FM Broadcast Subcarrier System Development

At the time when the advantages of FM broadcast subcarriers for ITS data distribution became evident, there was no suitable system available. FHWA tasked MITRE to develop a prototype system to show the feasibility of a high-speed digital system that could support the ITS requirements, and to do enough testing to validate its performance in a mobile environment. MITRE used computer-based design and analysis tools that it had originally acquired for its defense programs which greatly reduced the time and cost of the prototype's development - a "Peace Dividend". The system, called the Subcarrier Traffic Information Channel (STIC) was completed in only eight months in 1992. Initial tests were run at WQSM-FM in Fayetteville, NC in early 1993 and STIC was completely successful in meeting the design requirements.

STIC uses differential QPSK modulation (DQPSK), interleavers to protect message integrity in the presence of burst errors, and powerful error detection and correction algorithms. These features provide extremely robust system performance in the presence of severe multipath and fading that is characteristic of the FM channel in a mobile environment. The system is capable of providing a complete update of all affected traffic links (street segments, intersections) in a city the size of Los Angeles every minute.

These test were followed by a series of measurements in the Washington, DC area using WXTR-FM's transmitter in Maryland. This data further validated STIC's performance in a more urban environment. In all, MITRE built 2 subcarrier generators and 8 receivers in two different layout configurations to support these tests and to provide equipment to a few other experiments being done by various operational field trials around the country.

In late 1993, the National Radio Systems Committee (NRSC) formed a subcommittee on high-speed digital subcarriers whose goal was to develop a U.S. standard for these systems. Proposals were solicited for systems to test in early 1994. Three proponents responded: MITRE, Seiko and Digital DJ.

MITRE's STIC system was designed from the beginning to meet the ITS standards. Seiko's system design was tailored to deliver paging messages to wristwatch receivers where battery life was the driving parameter. Digital DJs DARC system is used to broadcast music title and type so that equipped receivers can respond to user preferences. Each of these systems performs the function for which they were designed better than the other two; not surprising since the applications are very different.

In mid-1995 the NRSC subcommittee adopted its goals and objectives and established plans to do formal testing in two stages. The first would be laboratory bench tests where the mobile environment would be represented by a fading channel simulator. The second set of tests would be conducted under actual field operating conditions in moving vehicles.

The laboratory tests were completed in 1996 and documented in an NRSC report. These tests showed that only the STIC system met the ITS requirements, and in fact was alone in performing well over the full range of test conditions. The field tests were completed in mid-1997 with the same results. FHWA then stated that STIC would be the standard system for ITS. The NRSC subcommittee is still working on an NRSC standard.

Extending the Design

The NRSC subcommittee recognized fairly early in its work that different stations use different analog subcarrier frequencies, principally 67 kHz and 92 kHz. The systems that were submitted for testing in 1996 all used a subcarrier frequency of about 76 kHz, with a bandwidth that overlapped the 67 kHz subcarrier sidebands. In order to accommodate the needs of different stations and thus increase the pool of available outlets, a decision was made to require that any NRSC standard system be able to operate on two different subcarrier frequencies so that either (though not both) analog subcarrier frequency could be used by a station at the same time as the digital system.

Both MITRE and Seiko agreed to furnish a "dual frequency" version of their system for testing. Digital DJ declined to do any further work and was thus dropped from consideration as an NRSC standard system. The two dual frequency systems were tested in the Fall of 1997, and each performed essentially the same as their corresponding single-frequency version. The second subcarrier frequency is 85.5 kHz.

After the test results were known, an effort was made to try to merge the MITRE and Seiko systems into a hybrid that might support both paging and ITS. After a thorough analysis by MITRE, it was concluded that such a design would compromise the ability of the hybrid to do the two separate functions to a degree that was beyond the potential benefit derived from having a common system. FHWA endorsed this decision and reiterated its policy that STIC was the ITS standard.

There are two areas where future versions of the STIC system could broaden its applications. The first is a strategy that allows the full interleaver to be bypassed for so-called "low latency" messages such as Differential GPS corrections. The second change would reduce the power consumption of the STIC receiver to allow hand-held ITS devices to get traffic data.

Summary

Three different designs of a high-speed digital FM broadcast subcarrier system have been built. Each performs its intended function very well, but none of the systems can be considered capable of providing optimum performance for all other functions. The STIC system has been adopted by FHWA as the standard for ITS, since it alone meets the performance requirements. This policy does not in any way preclude the use of the Seiko and Digital DJ systems for other services. Dual subcarrier frequency versions of the STIC and Seiko systems are available to allow broadcasters flexibility in the use of analog subcarriers along with a digital one.

Acknowledgments

The author would like to thank Mr. James Marshall of Radio Dynamics, Inc. and a co-inventor of STIC, for his extensive support of the STIC development and testing programs, and Mr. James Arnold of FHWA for his invaluable contributions toward the adoption of an ITS standard system. We also wish to thank the FHWA ITS Joint Program Office and its Director, Dr. Christine Johnson, for supporting the development of STIC under MITRE's system engineering contract.

Communications & Connectivity98
Countdown 2000: A Telecommunications Technology Update

Wednesday, April 8, 1998
2:00 - 5:00 pm

Chairperson:
David Layer
NAB, Washington, DC

***A Broadcasters' Guide for Selecting and Using Wide Area Network Connectivity**
Al Kovalick
Hewlett-Packard Company
Santa Clara, CA

***Bringing e-business to TV - The Appropriate Application of Telecommunications Technology to Broadcasting**
William Beckman
IBM Corporation
Somers, NY

Satellite Based Internet Opportunities for Broadcasters
Gerson Souto
INTELSAT
Washington, DC

***The Impact of Next Generation Network Technology on the Broadcasting Industry**
Dominic Orr
Alteon Networks
San Jose, CA

***Server Applications in Telecom Networks**
Joachim Paech
Deutsche Telekom, AG
Freiburg, Germanu

***Convergence Through the Eyes of the Consumer**
Krystol Cameron
Simply TV
New York, NY

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*Papers not available at the time of publication

Satellite-Based Internet Opportunities for Broadcasters

Matthew Halsey
William Karunaratne
Bertrand Rojat
Surendra Satija
Gerson Souto
INTELSAT
Washington DC

Abstract

Satellites have traditionally been a medium embraced by the video community because of the inherent advantages that satellites offer for broadcast applications. These same advantages can be capitalized upon in the Internet market, and provide broadcasters with the opportunity to offer enhanced and advanced services to their customers.

This paper explores how broadcast-oriented Internet technologies and applications, such as multicasting, can be used to further expand Direct to Home (DTH) services to support Internet and interactive DTH services. Satellites offer many advantages for the provision of push applications, through the use of multicast protocols, Internet Protocol (IP) based networks, Internet Service Providers (ISPs) and Internet content providers. These applications reflect the convergence of technologies, where video, audio and data streams can all be provided under the same service platform.

Applications and new services will be described in terms of today's geostationary satellites as well as how they will be affected by the advent of Ka-band satellites in the near future, for the provision of broadband services directly to the end user.

Direct to Home (DTH) Services Today

Thanks to Direct Broadcasting Satellites (DBS), DTH video services are available today on a worldwide basis. DTH services had an early start through the use of analog video technology. Today, digital video compression enables the provision of high quality content with more programs per unit of bandwidth, in comparison to analog technology. Digital video platforms, initially deployed with proprietary systems, now are being deployed on a worldwide basis using MPEG-2/DVB international standards.

A typical DTH end-user needs an antenna, RF electronics, and an Integrated Receiver-Decoder (IRD) to demodulate and decode the incoming digital transport stream, from the service provider to the end-user (forward link), into a useful signal that can used by TV sets. The basic video DTH service provides a combination of free to air and pay-TV channels. Depending on the DBS system being used, either single or multiple service providers deliver a number of video channels as *Digital Video Bouquets*. These video bouquets can be subscribed by the end user as packages or on a pay-per-view basis. A conditional access (CA) system is a key component of such DTH service platforms, since it enables the service provider to control and authorize end-users for the reception of the desired content, and provides support for billing systems.

Multimedia DTH Services

As mentioned before, DTH services are available today primarily for the distribution of video content. However, the growth in the video DTH market (Table 1) and the advent of multimedia services, as a result of the convergence of video, audio and data technologies, present additional market opportunities to the various players of the value-chain.

The value chain for DTH services includes content providers for video, audio and data, content editors and packagers, broadcasters, service providers, equipment manufacturers, retailers, and the satellite capacity providers.

In this regard, other applications such as datacasting, webcasting, home shopping and banking, interactive entertainment, Internet access, etc., can be coupled to a traditional video DTH service platform, and be provided to the end-user as value-added services.

The provision of these value-added services, driven by interactivity requirements, demands a return link between the end-user and the service provider. This return link can be provided either through a terrestrial or satellite path, depending on the region and market to be served. Figure 1 illustrates the concept of multimedia DTH services with a terrestrial return link. A terrestrial telephone line is usually used to support this return link, and several systems available in the market place provide this type of configuration. As part of an effort to develop a fully integrated DTH service via satellite, INTELSAT in conjunction with MEDIA4 is developing a prototype sub-meter (e.g. 0.7 m dish) multimedia terminal, which will efficiently use satellite return links. This is a very important step towards the deployment of planned broadband satellite systems operating at Ka-band, as will be discussed later in this paper.

In a multimedia DTH platform as described above, set-top-boxes can enable users to receive both video and data programs, and simultaneously feed their TVs and PCs. However, depending on the services to be provided, the set-top box might be dedicated to video and audio content, while a card inside the PC might be dedicated to the data content. The choice of one or another approach is a decision that the service provider and other players in the value-chain have to make as a function of the market, number of subscribers and type of content to be delivered.

REGION	1996	1997	1998	1999	2000	2001
North America	4.8	7.4	9.7	12.1	14.7	17.5
Europe	0.2	0.9	2.6	5.1	8.0	10.2
Africa/Middle East	0.1	0.6	1.5	2.5	4.5	6.6
Latin America	0.1	0.6	1.5	2.5	4.5	6.6
Asia	0.1	1.2	3.5	6.4	12.3	20.8
TOTAL	5.3	10.7	18.8	28.6	44.0	61.7

Table 1 - DTH Subscribers (*In Million of Households*) [1]

MPEG-2/DVB standards can be used with both approaches to deliver multimedia content, either by multiplexing video, audio and data streams, or by packaging these streams in dedicated transponders.

In any case, the end-user will have the ability to watch a variety of video and audio channels and to make specific program requests on a pay-per-view basis, while browsing on the Internet, sending an E-mail, etc.

Satellite Multicast Service Platform

The introduction of multimedia services into a DTH video platform to provide value added programming to the end user results in a very attractive solution for high speed Internet access and interactive services by satellite. Furthermore, it leverages upon existing and/or potential subscriber bases associated with the video DTH services. However, the inherent interactivity associated with multimedia services results, in many cases, in an enormous amount of point-to-point connections between the end-users and the broadcasting center / service provider. These connections make inefficient use of precious bandwidth available for the DTH platform, thus impacting the number of simultaneous subscribers that can be supported per unit of bandwidth. Therefore, limited network capacity can lead to network congestion, reduced quality of service, and service overbooking. In other words, systems scalability becomes an important and critical issue.

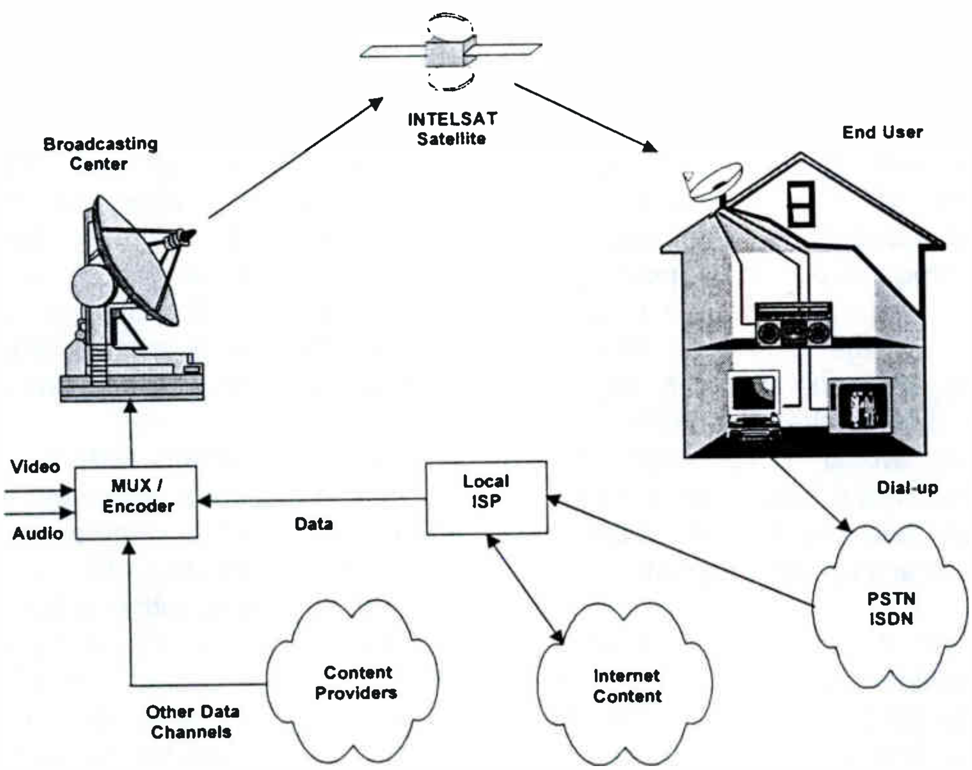

Figure 1 - Multimedia DTH Services with Terrestrial Return Link

Video DTH services scale very well in terms of number of subscribers. Even pay-per-view video DTH services scale well, since typically pay-per-view is handled through a Near Video On Demand (NVOD) system. Such NVOD systems allow multiple video channels with the same content to be staggered in time and be broadcast via the satellite. The end user can request one of these channels (e.g. movies) and receive the transmission of the next available channel, typically in 15 or 30 minute intervals.

With multimedia DTH services, the business case for the service provider needs to account for the cost associated with usage of satellite bandwidth for interactive applications, and network costs to connect the broadcasting center to data content providers and the Internet. These are critical elements in the deployment of a multimedia DTH service platform. In this regard, in order to leverage on the video subscriber base while providing multimedia services with good quality of service, service providers need to reduce the end user requirements for interactivity without restricting the end user's use of any of the above mentioned applications. One way of accomplishing this task is by "pushing" as much content as possible to the end user. If the packaging of the content is appropriate to the target audience, most of the information that end users would have requested through the return link would already have been delivered to them without the need for any specific request.

In light of the above discussion, service providers need to capitalize on the broadcasting capability of satellites and multicast (broadcast to registered subscribers) most of the multimedia content through the same video DTH platform. Such multicast satellite service platform can consist of real-time content streams (e.g. video, audio and data) as well as non real-time content (e.g. web pages). For example, thousands of web sites can be multicast per day (e.g. "the Best of the Web") using only a few Mbit/s of a forward link carrier. This is a very cost-effective solution, since it builds upon the broadcasting model in order to scale well to a very large number of subscribers. Another important aspect of this solution is that the interactivity capability is not removed from the system. Its requirement by the end user, however, is expected to drop considerably, enabling the development of a "sound" business case.

Recognizing the potential of such multimedia services, INTELSAT is currently developing with A&T Systems an integrated satellite multicast service platform connecting content providers, broadcasters, ISPs and end-users [2]. This satellite multicast service platform, illustrated in Figure 2, builds primarily upon content currently existent on the Internet. However, other sources of content might be used, as they become available. Broadcasters act as data warehouses, caching and replicating content which is then multicasted to various registered recipients (kiosks).

The concept considers ISPs to be the distribution channels of content to the end users. However, the concept can be easily extended to a DTH environment, using either a terrestrial or a satellite return link, as discussed earlier in the paper. INTELSAT considers this development to be a major building block for the worldwide deployment of multimedia DTH services.

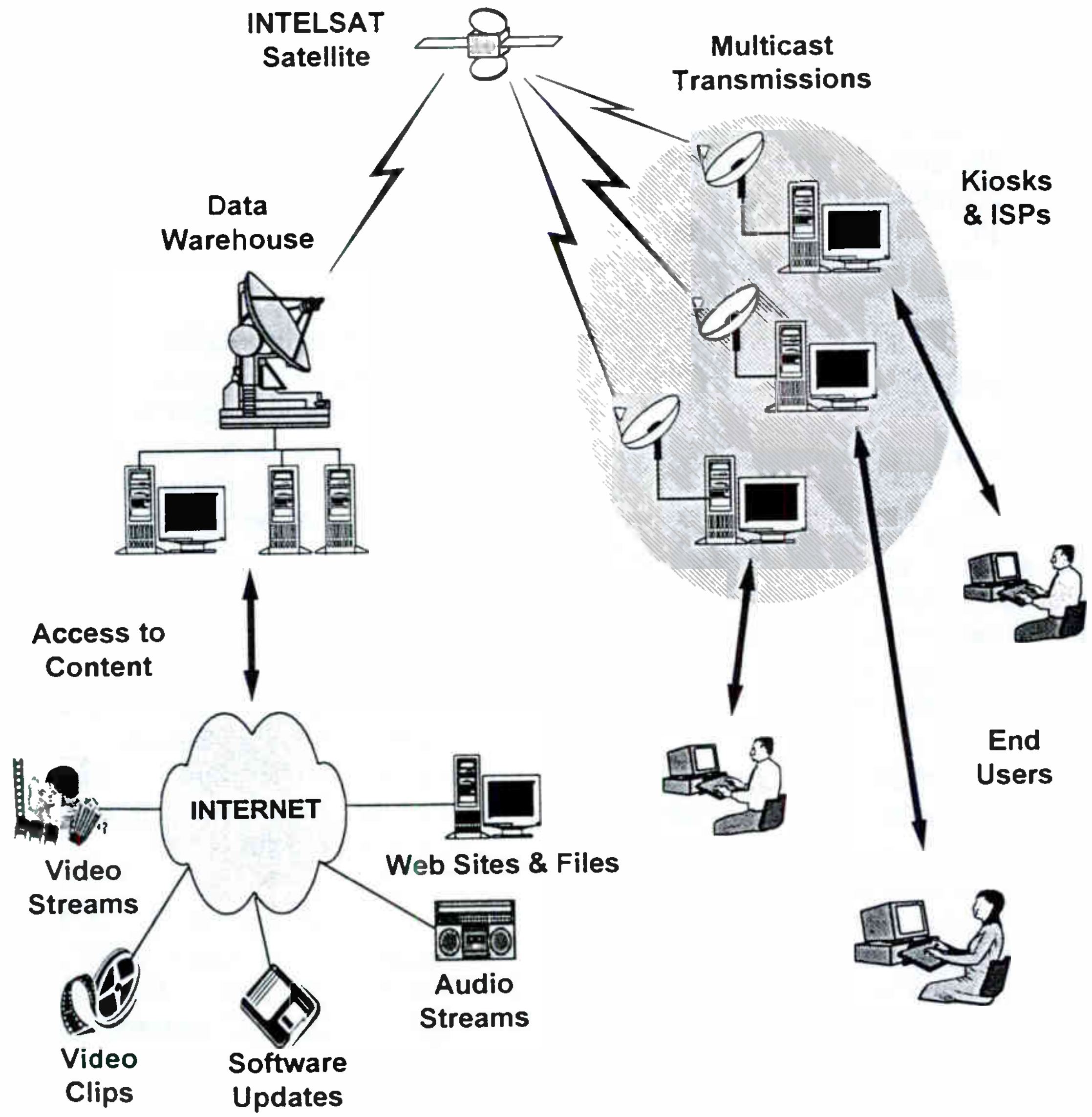

Figure 2 - Satellite Multicast Platform for Multimedia Services

Broadband Services

As applications demand more and more bandwidth and end users want to interact more with other end users (e.g. via videoconferencing, web phones), instead of interacting with the network (e.g. servers), multimedia service platforms need to evolve to accommodate such requirements. This is where the planned Ka-band systems come into picture.

Ka-band has been the choice for the evolution of multimedia satellite service platforms primarily because of the spectrum availability to accommodate the demand for more bandwidth. Secondly, it enables to use of the same sub-meter (e.g. 0.7 m dish) DTH terminals for broadband applications. In addition, the use of satellite on-board processing technologies will allow end users to communicate with other end users without the need for any intermediate gateway station, as illustrated in Figure 3. However, the building blocks described in the previous sections are still necessary for the deployment of a multimedia DTH service platform, in order to optimize the use of the bandwidth, as the interactive requirements increase in the future.

Conclusion

This paper presents Satellite-Based Internet Opportunities for Broadcasters, focusing on the deployment of multimedia DTH services. Multimedia DTH systems can leverage upon video DTH systems to provide value-added services to existing or potential video subscribers.

The introduction of interactivity in association with multimedia services requires special attention to the deployment of such services. In this regard, the need for multimedia services to be provided in conjunction with a multicast satellite service platform has been pointed out. This approach will allow the service provider to build upon the broadcasting model, which makes very efficient use of the satellite bandwidth, and scales very well to a large number of subscribers. As interactivity and bandwidth requirements increase, end user to end user connectivity will drive the deployment of Ka-band satellite systems.

Finally, multimedia DTH services can be provided today through the use of DBS systems, presenting a huge market opportunity for broadcasters to capitalize on the continuing growth of the DTH market worldwide.

References

[1] MEDIA4 Inc.,"Joint Provision of DTHTV and Data Services - Final Competitive Market Analysis", INTELSAT Contract INTEL-1815, November 17, 1997

[2] L. Buchsbaum, T. Oishi, G. Souto and J. Stevenson, "INTELSAT's Multicast Internet Caching & Replication System", submitted to the INET98 Conference.

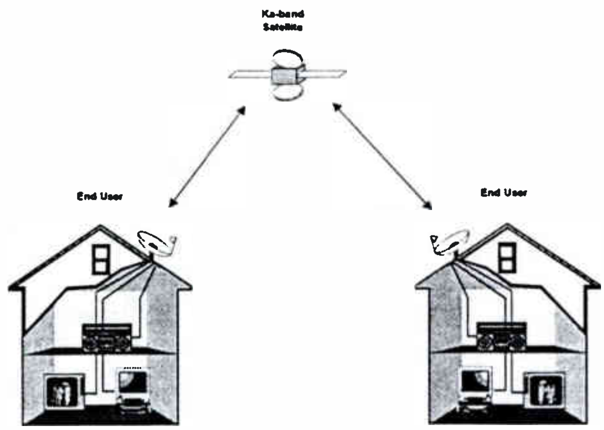

Figure 3 - Broadband Satellite Systems for Multimedia DTH Services

PROGRAM AND SYSTEM INFORMATION PROTOCOL FOR TERRESTRIAL BROADCAST AND CABLE

Editorial Notice

At press time for these Proceedings, errors that were discovered in Table 6.5 of ATSC Standard A/65 (Program and System Information Protocol for Terrestrial Broadcast and Cable) were brought to our attention by Art Allison, Sr. Engineer, NAB. This is the revised table submitted to the ATSC editors for their consideration as a replacement table for table 6.5. Contact the ATSC to determine the final contents of this table.

Table 6.5 Modulation Modes

modulation_mode	meaning
0x00	[Reserved]
0x01	**Analog** — The virtual channel is modulated using standard analog methods for analog television.
0x02	**SCTE_mode_1** — The virtual channel has a symbol rate of 5.057 Msps, transmitted in accordance with *Digital Transmission Standard for Cable Television*, Ref. [12] (Mode 1). Typically, mode 1 will be used for 64-QAM.
0x03	**SCTE_mode_2** — The virtual channel has a symbol rate of 5.361 Msps, transmitted in accordance with *Digital Transmission Standard for Cable Television*, Ref. [12] (Mode 2). Typically, mode 2 will be used for 256-QAM.
0x04	**ATSC (8 VSB)** — The virtual channel uses the 8-VSB modulation method conforming to the ATSC Digital Television Standard.
0x05	**ATSC (16 VSB)** — The virtual channel uses the 16-VSB modulation method conforming to the ATSC Digital Television Standard.
0x06 -0x7F	[Reserved for future use by ATSC]
0x80	Modulation parameters are defined by a private descriptor
0x81-0xFF	[User Private]

Doc. A/65
23 Dec 1997

PROGRAM AND SYSTEM INFORMATION PROTOCOL FOR TERRESTRIAL BROADCAST AND CABLE

ADVANCED TELEVISION SYSTEMS COMMITTEE

PROGRAM AND SYSTEM INFORMATION PROTOCOL FOR TERRESTRIAL BROADCAST AND CABLE

ATSC STANDARD

Table of Contents

PROGRAM AND SYSTEM INFORMATION PROTOCOL FOR TERRESTRIAL BROADCAST AND CABLE

ATSC STANDARD

1. SCOPE

1.1 Purpose

This document defines a Standard for System Information (SI) and Program Guide (PG) data compatible with digital multiplex bit streams constructed in accordance with ISO/IEC 13818-1 (MPEG-2 Systems). The document defines the standard protocol for transmission of the relevant data tables contained within packets carried in the Transport Stream multiplex. The protocol defined herein will be referred to as **Program and System Information Protocol (PSIP).** Prior to being approved as an ATSC Standard, this document was designated T3/S8-193 and later, after approval by T3, as Doc. T3-442.

This standard was prepared by the Advanced Television Systems Committee (ATSC) Technology Group on Distribution (T3). The document was approved by T3 on 22 October 1997 for submission by letter ballot to the membership of the full ATSC. The document was approved by the members of the ATSC on 23 December 1997.

For an informative description of the purpose, concepts, and tables defined in this protocol, first time readers are encouraged to start with Annex D.

1.2 Application

This document describes tables that shall be applicable to terrestrial (over-the-air) and cable signals. Some PSIP tables apply to terrestrial broadcast, some apply to cable, and others apply to both.

1.2.1 Terrestrial Broadcast

The following PSIP data shall be included in all ATSC-compliant Transport Streams to be transmitted via terrestrial broadcast:

NOTE: The user's attention is called to the possibility that compliance with this standard may require use of an invention covered by patent rights. By publication of this standard, no position is taken with respect to the validity of this claim, or of any patent rights in connection therewith. The patent holder has, however, filed a statement of willingness to grant a license under these rights on reasonable and nondiscriminatory terms and conditions to applicants desiring to obtain such a license. Details may be obtained from the publisher.

- The Terrestrial Virtual Channel Table (TVCT) defining, at a minimum, MPEG-2 programs embedded in the Transport Stream in which the TVCT is carried.
- The Master Guide Table (MGT) defining the type, packet identifiers, and versions for all the other PSIP tables in this Transport Stream, except for the System Time Table (STT).
- The Rating Region Table (RRT) defining the TV parental guideline system referenced by any content advisory descriptor carried within the Transport Stream.
- The System Time Table (STT), defining the current date and time of day.
- A service_location_descriptor for each digital virtual channel in the VCT.
- The first four Event Information Tables (EIT-0, EIT-1, EIT-2 and EIT-3) describing 12 hours of events (TV programs), each with a coverage of 3 hours, and including all of the virtual channels listed in the TVCT.

1.2.2 Cable

The following PSIP data shall be included in all ATSC-compliant Transport Streams to be transmitted via cable:

- The Cable Virtual Channel Table (CVCT) defining, at a minimum, the virtual channel structure for the collection of MPEG-2 programs embedded in the Transport Stream in which the CVCT is carried.
- The Master Guide Table (MGT) defining the type, packet identifiers, and versions for all of the other PSIP tables included in this Transport Stream except for the System Time Table (STT).
- The Rating Region Table (RRT) defining the TV parental guideline system referenced by any content advisory descriptor carried within the Transport Stream.
- The System Time Table (STT), defining the current date and time of day.

1.3 Organization

The sections of this document are organized as follows:

- **Section 1** — Provides this general introduction.
- **Section 2** — Lists references and applicable documents.
- **Section 3** — Provides a definition of terms and a list of acronyms and abbreviations used in this document.
- **Section 4** — Describes the data structure of the PSIP tables.
- **Section 5** — Describes the overall table hierarchy.
- **Section 6** — Describes formats for all of the PSIP tables.
- **Section 7** — Describes PSIP STD model.

- **Annex A**— Describes the daylight savings time control.
- **Annex B** — Describes the assignment of major_channel_number values for terrestrial broadcast in the U.S.
- **Annex C** — Describes the standard Huffman tables for text compression.
- **Annex D** — Provides an overview of PSIP for terrestrial broadcast with application examples.
- **Annex E** — Describes the typical sizes of PSIP tables.
- **Annex F** — Provides an overview of Huffman-based text compression.

2. REFERENCES

The following documents are applicable to this Standard:

1. ATSC Standard A/52 (1995), Digital Audio Compression (AC-3) (*normative*).
2. ATSC Standard A/53 (1995), ATSC Digital Television Standard (*normative*).
3. ATSC Standard A/55 (1996), Program Guide for Digital Television (*informative*).
4. ATSC Standard A/56 (1996), System Information for Digital Television (*informative*).
5. ATSC Standard A/57 (1996), Program/Episode/Version Identification (*normative*).
6. ISO 639, Code for the Representation of Names of Languages, 1988 (*informative*).
7. ISO CD 639.2, Code for the Representation of Names of Languages: alpha-3 code, Committee Draft, dated December 1994 (*normative*).
8. ISO/IEC 10646-1:1993, Information technology — Universal Multiple-Octet Coded Character Set (UCS) — Part 1: Architecture and Basic Multilingual Plane (*normative*).
9. ISO/IEC 8859, Information Processing — 8-bit Single-Octet Coded Character Sets, Parts 1 through 10 (*normative*).
10. ITU-T Rec. H.222.0 | ISO/IEC 13818-1:1996, Information Technology — Generic coding of moving pictures and associated audio — Part 1: Systems (*normative*).
11. ITU-T Rec. H.262 | ISO/IEC 13818-2:1996, Information Technology — Generic coding of moving pictures and associated audio — Part 2: Video (*normative*).
12. Digital Video Transmission Standard for Cable Television, SCTE DVS-031, Rev. 2, 29 May 1997 (*informative*).
13. EIA 708 *Specification for Advanced Television Closed Captioning (ATVCC)*, Electronic Industry Association.
14. EIA 752 *Specification for Transport of Transmission Signal Identifier (TSID) Using Extended Data Service)*, Electronic Industry Association.

3. DEFINITIONS

3.1 Compliance Notation

As used in this document, *"shall"* or *"will"* denotes a mandatory provision of the standard. *"Should"* denotes a provision that is recommended but not mandatory. *"May"* denotes a feature whose presence does not preclude compliance, that may or may not be present at the option of the implementer.

3.2 Acronyms and Abbreviations

The following acronyms and abbreviations are used within this specification:

ATSC	Advanced Television Systems Committee
bslbf	bit serial, leftmost bit first
CAT	Conditional Access Table
CRC	Cyclic Redundancy Check
CVCT	Cable Virtual Channel Table
DTV	Digital Television
EPG	Electronic Program Guide
EIT	Event Information Table
ETM	Extended Text Message
ETT	Extended Text Table
GA	Grand Alliance
GPS	Global Positioning System
PSIP	Program and System Information Protocol
MGT	Master Guide Table
MPAA	Motion Picture Association of America
MPEG	Moving Picture Experts Group
NVOD	Near Video On Demand
PAT	Program Association Table
PCR	Program Clock Reference
PES	Packetized Elementary Stream
PID	Packet Identifier
PMT	Program Map Table
PTC	Physical Transmission Channel
SCTE	Society of Cable Telecommunications Engineers
SI	System Information
STD	System Target Decoder
STT	System Time Table
rpchof	remainder polynomial coefficients, highest order first
RRT	Rating Region Table
TS	Transport Stream
TVCT	Terrestrial Virtual Channel Table

UTC	Coordinated Universal Time[1]
uimsbf	unsigned integer, most significant bit first
VCT	Virtual Channel Table. Used in reference to either TVCT or CVCT.
unicode	Unicode™

3.3 Definition of Terms

The following terms are used throughout this document:

descriptor: A data structure of the format: descriptor_tag, descriptor_length, and a variable amount of data. The tag and length fields are each 8 bits. The length specifies the length of data that begins immediately following the descriptor_length field itself. A descriptor whose descriptor_tag identifies a type not recognized by a particular decoder shall be ignored by that decoder. Descriptors can be included in certain specified places within PSIP tables, subject to certain restrictions (see Table 6.16). Descriptors may be used to extend data represented as fixed fields within the tables. They make the protocol very flexible since they can be included only as needed. New descriptor types can be standardized and included without affecting receivers that have not been designed to recognize and process the new types.

digital channel: A set of one or more digital elementary streams. See *virtual channel.*

event: A collection of elementary streams with a common time base, an associated start time, and an associated end time. An event is equivalent to the common industry usage of "television program."

instance: See *table instance.*

logical channel: See *virtual channel.*

physical channel: A generic term to refer to the each of the 6-8 MHz frequency bands where television signals are embedded for transmission. Also known as the physical transmission channel (PTC). One analog virtual channel fits in one PTC but multiple digital virtual channels typically coexist in one PTC.

physical transmission channel: See *physical channel.*

program element: A generic term for one of the elementary streams or other data streams that may be included in a program. For example: audio, video, data, etc.

program: A collection of program elements. Program elements may be elementary streams. Program elements need not have any defined time base; those that do have a common time base are intended for synchronized presentation. The term *program* is also commonly used in the context of a "television program" such as a scheduled daily news broadcast. In this specification the term "event" is used to refer to a "television program" to avoid ambiguity.

section: A data structure comprising a portion of an *ISO/IEC 13818-1* defined table, such as the Program Association Table (PAT), Conditional Access Table (CAT), or Program Map Table (PMT). All sections begin with the table_id and end with the CRC_32 field, and their starting points

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[1] Since unanimous agreement could not be achieved by the ITU on using either the English word order, CUT, or the French word order, TUC, a compromise to use neither was reached.

within a packet payload are indicated by the pointer_field mechanism defined in the *ISO/IEC 13818-1* International Standard.

stream: An ordered series of bytes. The usual context for the term *stream* is the series of bytes extracted from Transport Stream packet payloads which have a common unique PID value (e.g., video PES packets or Program Map Table sections).

table: PSIP is a collection of tables describing virtual channel attributes, event features, and others. PSIP tables are compliant with the private section syntax of ISO/IEC 13818-1.

table instance: Tables are identified by the table_id field. However, in cases such as the RRT and EIT, several instances of a table may be defined simultaneously. All instances have the same PID and table_id but different table_id_extension.

virtual channel: A virtual channel is the designation, usually a number, that is recognized by the user as the single entity that will provide access to an analog TV program or a set of one or more digital elementary streams. It is called "virtual" because its identification (name and number) may be defined independently from its physical location. Examples of virtual channels are: digital radio (audio only), a typical analog TV channel, a typical digital TV channel (composed of one audio and one video stream), multi-visual digital channels (composed of several video streams and one or more audio tracks), or a data broadcast channel (composed of one or more data streams). In the case of an analog TV channel, the virtual channel designation will link to a specific physical transmission channel. In the case of a digital TV channel, the virtual channel designation will link both to the physical transmission channel and to the particular video and audio streams within that physical transmission channel.

3.4 Section and Data Structure Syntax Notation

This document contains symbolic references to syntactic elements. These references are typographically distinguished by the use of a different font (e.g., restricted), may contain the underscore character (e.g., sequence_end_code) and may consist of character strings that are not English words (e.g., dynrng).

The formats of sections and data structures in this document are described using a C-like notational method employed in *ISO/IEC 13818-1*.

4. DATA STRUCTURE

This section describes the data structure common to all PSIP tables. It also lists valid table_id and PID values for every table that belongs to PSIP.

4.1 Table Format

Tables defined in this Standard are structured in the same manner used for carrying *ISO/IEC 13818-1* -defined PSI tables, shown in Table 4.1. The structure conforms to the generic private section syntax defined in ISO/IEC 13818-1

Table 4.1 Table format used in PSIP

	Bits	Format
typical_PSI_table( ){		
table_id	8	uimsbf
section_syntax_indicator	1	'1'
private_indicator	1	'0'
zero	2	'00'
section_length	12	uimsbf
table_id_extension	16	uimsbf
reserved	2	'11'
version_number	5	uimsbf
current_next_indicator	1	bslbf
section_number	8	uimsbf
last_section_number	8	uimsbf
protocol_version	8	uimsbf
actual_table_data	*	
CRC_32	32	rpchof
}		

4.2 Table ID Ranges and Values

Table 4.2 defines Table ID ranges and values.

Table 4.2 ID Ranges and Values

Table ID Value (hex)	Tables	PID	Ref.
	ISO/IEC 13818-1 Sections:		
0x00	PROGRAM ASSOCIATION TABLE (PAT)	0	Ref. [10]
0x01	CONDITIONAL ACCESS TABLE (CAT)	1	Ref. [10]
0x02	TS PROGRAM MAP TABLE (PMT)	per PAT	Ref. [10]
0x03-0x3F	[ISO Reserved]		
	User Private Sections:		
0x40-0x7F	[User Private for other systems]		
0x80-0xBF	[User Private]		
	Other documents:		
0xC0-0xC6	[Used in other systems]		
	PSIP Tables:		
0xC7	MASTER GUIDE TABLE (MGT)	0x1FFB	Sec.6.2
0xC8	TERRESTRIAL VIRTUAL CHANNEL TABLE (TVCT)	0x1FFB	Sec.6.3.1
0xC9	CABLE VIRTUAL CHANNEL TABLE (CVCT)	0x1FFB	Sec.6.3.2
0xCA	RATING REGION TABLE (RRT)	0x1FFB	Sec.6.4
0xCB	EVENT INFORMATION TABLE (EIT)	per MGT	Sec.6.5
0xCC	EXTENDED TEXT TABLE (ETT)	per MGT	Sec.6.6
0xCD	SYSTEM TIME TABLE (STT)	0x1FFB	Sec.6.1
0xCE-0xDF	[Reserved for future ATSC use]		
0xE0-0xE5	[Used in other systems]		
0xE6-0xFE	[Reserved for future ATSC use]		
0xFF	**Inter-message Filler**		

Tables defined in this PSIP Standard, and any created as user extensions to it are considered "private" with respect to *ISO/IEC 13818-1*. Table types 0x40 through 0xBF are user defined (outside the scope of this PSIP Standard).

4.3 Extensibility

The PSIP protocol describes a number of tables conveying system information and content guide data structures. The Standard is designed to be extensible via the following mechanisms:

1. **Reserved Fields:** Fields in this Standard marked reserved shall be reserved for use either when revising this Standard, or when another standard is issued that builds upon this one. See Section 4.4 below.

2. **Standard Table Types:** As indicated in Table 4.1, table_id values in the range 0xCE-0xDF and 0xE6-0xFE shall be reserved for use either when revising this PSIP Standard, or when another standard is issued that builds upon this one.

3. **User Private Table Types:** As indicated in Table 4.1, table_id values in the range 0x40 through 0xBF shall be reserved for "user private" use.
4. **User Private Descriptors:** Privately defined descriptors may be placed at designated locations throughout the tables described in this Standard. Ownership of one or more user private descriptors may be indicated by the presence of an MPEG registration_descriptor() preceding the descriptor(s).
5. **Protocol Version Field:** Initially this field is set to 0, but after approval, future structural modifications shall be accommodated by defining different protocol version numbers.

4.4 Reserved Fields

reserved — Fields in this PSIP Standard marked "reserved" shall not be assigned by the user, but shall be available for future use. Decoders are expected to disregard reserved fields for which no definition exists that is known to that unit. Each bit in the fields marked "reserved" shall be set to one until such time as they are defined and supported.

user_private — Indicates that the bit or bit field is not defined within the scope of this Standard. The owner of the bit, and hence the entity defining its meaning, is derived via its context within a message.

zero — Indicates that the bit or bit field shall have the value zero.

5. TABLE HIERARCHY AND STRUCTURE REQUIREMENTS

The Program and System Information Protocol (PSIP) is a collection of hierarchically arranged tables for describing system information and program guide data. These tables are packetized and multiplexed according to the transport protocol detailed in ISO/IEC 13818-1.

The base PID (base_PID) is an explicitly defined value (0x1FFB) used to identify the packets for the following tables for terrestrial and cable systems: The System Time Table (STT), the Master Guide Table (MGT), the Rating Region Table (RRT), and the Virtual Channel Table (VCT). Several Event Information Tables (EIT) are also part of the PSIP data structures, with their PIDs explicitly defined in the MGT. Figure 5.1 illustrates the relations between these elements.

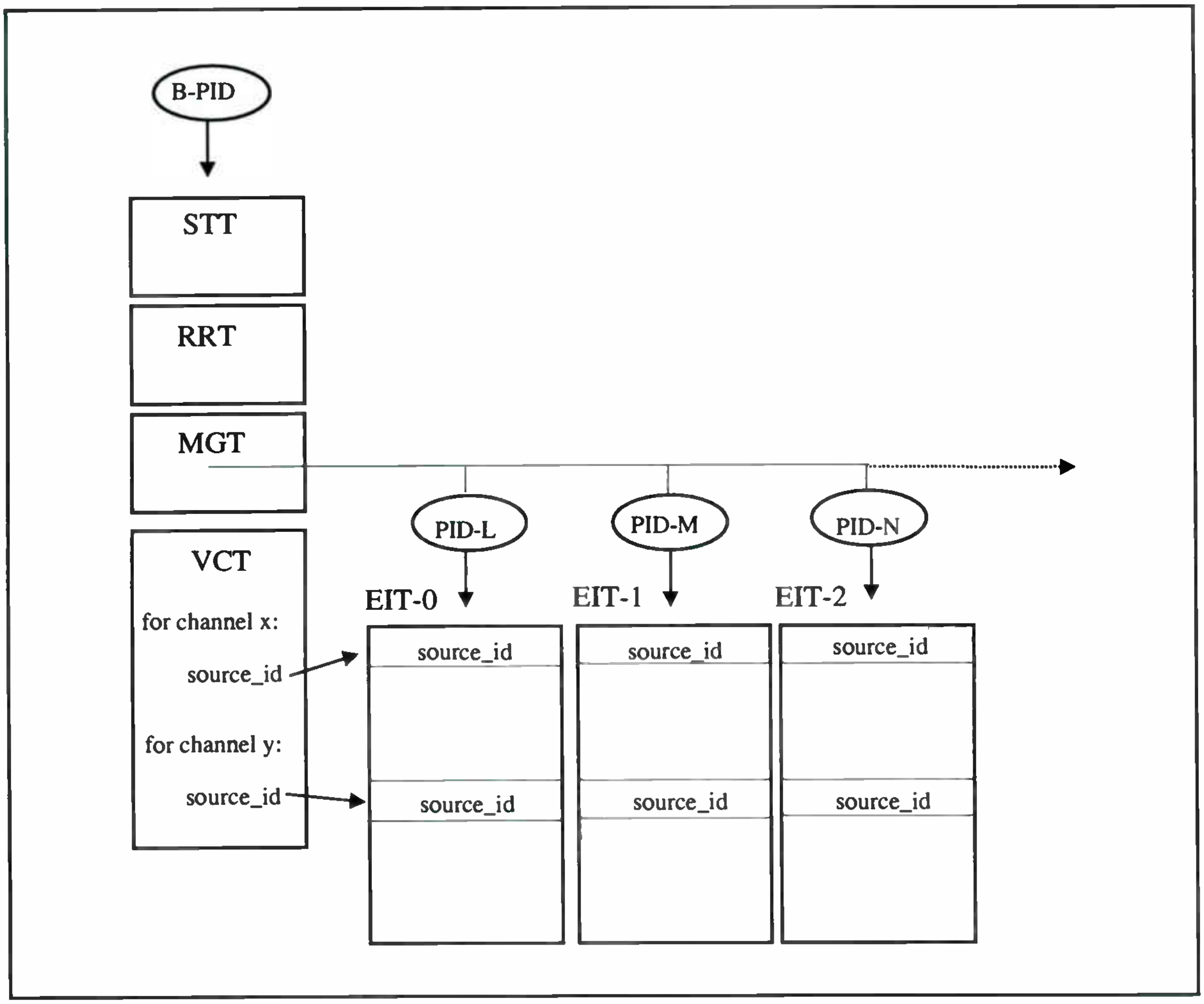

Figure 5.1 Table hierarchy for the Program and System Information Protocol (PSIP)

As the name indicates, the System Time Table (STT) carries time information needed for any application requiring synchronization. The Rating Region Table (RRT) defines rating tables valid for different regions or countries. The Master Guide Table (MGT) defines sizes, PIDs, and version numbers for all of the relevant tables. The Virtual Channel Table (VCT) actually exists in two versions: one for terrestrial and a second one for cable applications. Its purpose is to tabulate virtual channel attributes required for navigation and tuning. The terrestrial and cable versions are similar in structure, with the latter redefining the semantics of some fields pertinent to cable operations.

Each of the Event Information Tables (EITs) lists TV programs (events) for the virtual channels described in the VCT. The EITs are sequentially and chronologically organized from EIT-0 to EIT-127. The first table (EIT-0), corresponds to the currently valid list of events. The second table (EIT-1) corresponds to the next time window, and so on.

During remultiplexing, EIT tables which originally existed in separate Transport Streams may be multiplexed into a common Transport Stream or *vice versa*. For this reason, it is very convenient to synchronize the start times and durations of the EITs. Consequently, the next three synchronization rules shall be followed when EIT tables are prepared.

Requirement 1: *Each EIT shall have a duration of 3 hours.*

Requirement 2: *Start times for EITs are restricted to 0:00 (midnight), 3:00, 6:00, 9:00, 12:00 (noon), 15:00, 18:00 and 21:00. All of these times are UTC.*

Requirement 3: *EIT-0 lists all of the available events for the current 3-hour time segment. EIT-1 lists all of the available events for the next 3-hour time segment, and likewise, non-overlapping sequential time windows are allocated for all of the other EITs.*

For example, a broadcast group operating in the Eastern time zone of the U.S. at 15:30 EDT (19:30 UTC) is required to carry EIT-0 describing events from 14:00 to 17:00 EDT (18:00 to 21:00 in UTC time) plus EIT-1, EIT-2, and EIT-3 covering the next 9-hour interval between 17:00 to 2:00 EDT. At 17:00 EDT, the first table, EIT-0, will be obsolete while EIT-1 will still be valid. At this time, simply by shifting the listed PID values in the MGT, EIT-1 becomes EIT-0 and EIT-2 becomes EIT-1. Updating tables then becomes a process of shifting the list of PIDs in the MGT and their corresponding version numbers. However, updates and/or corrections to the information in the EITs may be performed at any time since the decoder monitors the MGT continuously, where the most current copy of the version number is maintained. Updates and/or corrections to the EIT (other than shifting) shall be signaled by increasing the version number by one.

Besides listing the PIDs for all of the EITs, the Master Guide Table (MGT) also lists a set of PIDs for Extended Text Tables (ETTs). These tables carry relatively long text messages for describing events and virtual channels. Each EIT has either zero or one associated ETT. Similarly, The VCT has either zero or one associated ETT. Figure 5.2 illustrates the concept.

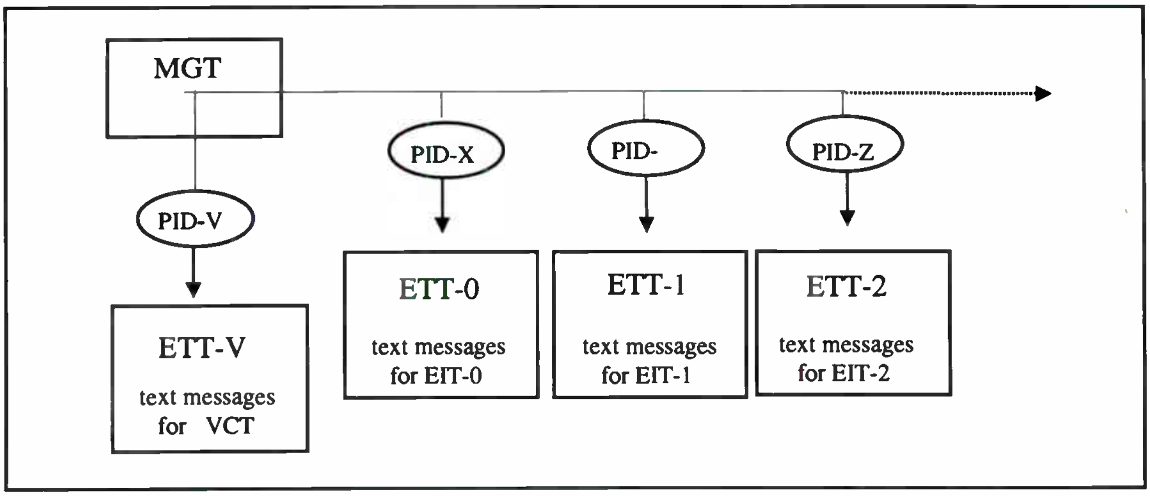

Figure 5.2 Extended Text Tables (ETTs) defined to carry text messages for describing virtual channels and events.

5.1 Requirements for terrestrial broadcast

The rules governing the transport of PSIP tables for terrestrial broadcast are:

Requirement 4: *Every digital Transport Stream in terrestrial broadcast shall include the STT, the RRT, the TVCT, the MGT, and the first four Event Information Tables (EIT-0, EIT-1, EIT-2 and EIT-3). All of the other EITs and the whole collection of ETTs are optional.*

Requirement 5: *The PSIP tables shall describe all of the digital channels multiplexed in the Transport Stream. For convenience, the tables may optionally include information about analog channels as well as other digital channels available in different Transport Streams.*

5.2 Requirements for cable

The rules governing the transport of PSIP tables for cable are:

Requirement 6: *The required tables for a cable system are: the STT, the RRT, the CVCT, and the MGT.*

Requirement 7: *The PSIP tables shall describe all of the digital channels multiplexed in the Transport Stream. For convenience, the tables may optionally include information about analog channels as well as other digital channels available in different Transport Streams.*

6. SPECIFICATIONS

This chapter describes the bit stream syntax and semantics for the System Time Table (STT), Master Guide table (MGT), Virtual Channel Table (VCT), Rating Region Table (RRT), Event Information Table (EIT), Extended Text Table (ETT), core descriptors, and the multiple string structure.

6.1 System Time Table (STT)

The System Time Table provides the current date and time of day information.

The following constraints apply to the Transport Stream packet carrying the STT:

- PID for STT shall have the value 0x1FFB (base_PID)
- transport_scrambling_control bits shall have the value '00'
- adaptation_field_control bits shall have the value '01'

The bit stream syntax for the System Time Table is shown in Table 6.1.

Table 6.1 Bit Stream Syntax for the System Time Table

Syntax	Bits	Format
system_time_table_section () {		
table_id	8	0xCD
section_syntax_indicator	1	'1'
private_indicator	1	'1'
zero	2	'00'
section_length	12	uimsbf
table_id_extension	16	0x0000
reserved	2	'11'
version_number	5	'00000'
current_next_indicator	1	'1'
section_number	8	0x00
last_section_number	8	0x00
protocol_version	8	uimsbf
system_time	32	uimsbf
GPS_UTC_offset	8	uimsbf
daylight_savings	16	uimsbf
for (I = 0;I< N;I++) {		
descriptors()	var	
}		
CRC_32	32	rpchof
}		

table_id — This is an 8-bit field, which shall be set to 0xCD, identifying this table as the System Time Table.

section_syntax_indicator — This 1-bit field shall be set to '1'. It denotes that the section follows the generic section syntax beyond the section length field.

private_indicator — This 1-bit field shall be set to '1'.

section_length — 12-bit field specifying the number of remaining bytes in this section immediately following the section_length field up to the end of the section. The value of the section_length shall be no larger than 1021.

table_id_extension — This 16-bit field shall be set to 0x0000.

version_number — This 5-bit field shall have a value of zero.

current_next_indicator — This 1-bit indicator is always set to '1' for an STT section; the STT sent is always currently applicable.

section_number — The value of this 8-bit field shall always be 0x00 (this table is only one section long).

last_section_number — The value of this 8-bit field shall always be 0x00.

protocol_version — An 8-bit unsigned integer field whose function is to allow, in the future, this table type to carry parameters that may be structured differently than those defined in the current protocol. At present, the only valid value for protocol_version is zero. Non-zero values of protocol_version may only be processed by decoders designed to accommodate the later versions as they become standardized.

system_time — A 32-bit unsigned integer quantity representing the current system time as the number of GPS seconds since 12 am, January 6th, 1980. The count of GPS seconds and leap second count shall be accurate and correct to within plus or minus four seconds, as timed at the arrival in the decoder of the Transport Stream packet carrying the last byte of the CRC.

GPS_UTC_offset — An 8-bit unsigned integer that defines the current offset in whole seconds between GPS and UTC time standards. To convert GPS time to UTC, the GPS_UTC_offset is subtracted from GPS time. Whenever the International Bureau of Weights and Measures decides that the current offset is too far in error, an additional leap second may be added (or subtracted), and the GPS_UTC_offset will reflect the change.

daylight_savings — Daylight Savings Time Control bytes. Refer to Annex A for the use of these two bytes.

CRC_32 — This is a 32-bit field that contains the CRC value that ensures a zero output from the registers in the decoder defined in Annex A of ISO/IEC 13818-1 "MPEG-2 Systems" after processing the entire System Time Table section.

6.2 Master Guide Table (MGT)

The MGT lists version numbers, length in bytes, and PIDs for all of the PSIP tables with the exception of the STT which works independently from the other tables.

The Master Guide Table is carried in a single section with table ID 0xC7, and obeys the syntax and semantics of the Private Section as described in Section 2.4.4.10 and 2.4.4.11 of ISO/IEC 13818-1. The following constraints apply to the Transport Stream packet carrying the MGT:

- PID for MGT shall have the value 0x1FFB (base_PID)

- transport_scrambling_control bits shall have the value '00'
- adaptation_field_control bits shall have the value '01'
- payload_unit_start_indicator of the Transport Stream packet carrying the table_id field of the MGT section shall be 1 (first Transport Stream packet of the section)
- pointer_field of the Transport Stream packet carrying the table_id field of the MGT section shall have the value 0x00 (section starts immediately after the pointer_field)

The bit stream syntax for the Master Guide Table is shown in Table 6.2.

Table 6.2 Bit Stream Syntax for the Master Guide Table

Syntax	Bits	Format
master_guide_table_section () {		
table_id	8	0xC7
section_syntax_indicator	1	'1'
private_indicator	1	'1'
zero	2	'00'
section_length	12	uimsbf
table_id_extension	16	0x0000
reserved	2	'11'
version_number	5	uimsbf
current_next_indicator	1	'1'
section_number	8	0x00
last_section_number	8	0x00
protocol_version	8	uimsbf
tables_defined	16	uimsbf
for (i=0;i<tables_defined;i++) {		
table_type	16	uimsbf
reserved	3	'111'
table_type_PID	13	uimsbf
reserved	3	'111'
table_type_version_number	5	uimsbf
number_bytes	32	uimsbf
reserved	4	'1111'
table_type_descriptors_length	12	uimsbf
for (k=0;k<N;k++)		
descriptor()	var	
}		
reserved	4	'1111'
descriptors_length	12	uimsbf
for (I = 0;I< N;I++)		
descriptor()	var	
CRC_32	32	rpchof
}		

table_id — This is an 8-bit field which shall be set to 0xC7, identifying this table as the Master Guide Table.

section_syntax_indicator — This 1-bit field shall be set to '1'. It denotes that the section follows the generic section syntax beyond the section length field.

private_indicator — This 1-bit field shall be set to '1'.

section_length — 12-bit field specifying the number of remaining bytes in this section immediately following the section_length field up to the end of the section. The value of the section_length shall be no larger than 4093.

table_id_extension — This 16-bit field shall be set to 0x0000.

version_number — This 5-bit field is the version number of MGT. The version number shall be incremented by 1 modulo 32 when any field in the table_types defined in the loop below or the MGT itself changes.

current_next_indicator — This 1-bit indicator is always set to '1' for the MGT section; the MGT sent is always currently applicable.

section_number — The value of this 8-bit field shall always be 0x00 (this table is only one section long).

last_section_number — The value of this 8-bit field shall always be 0x00.

protocol_version — An 8-bit unsigned integer field whose function shall be to allow, in the future, this table type to carry parameters that may be structured differently than those defined in the current protocol. At present, the only valid value for protocol_version is zero. Non-zero values of protocol_version may only be processed by decoders designed to accommodate the later versions as they become standardized.

tables_defined — This 16-bit unsigned integer in the range 0 to 65535 represents the number of tables in the following loop.

table_type — This 16-bit unsigned integer specifies the type of table, based on Table 6.3.

Table 6.3 Table Types

table_type	Meaning
0x0000	Terrestrial VCT with current_next_indicator=1
0x0001	Terrestrial VCT with current_next_indicator=0
0x0002	Cable VCT with current_next_indicator=1
0x0003	Cable VCT with current_next_indicator=0
0x0004	channel ETT
0x0005-0x00FF	[Reserved for future ATSC use]
0x0100-0x017F	EIT-0 to EIT-127
0x0180-0x01FF	[Reserved for future ATSC use]
0x0200-0x027F	event ETT-0 to event ETT-127
0x0280-0x0300	[Reserved for future ATSC use]
0x0301-0x03FF	RRT with rating_region 1-255
0x0400-0x0FFF	[User private]
0x1000-0xFFFF	[Reserved for future ATSC use]

table_type_PID — This 13-bit field specifies the PID for the table_type described in the loop.

table_type_version_number— This 5-bit field reflects the version number of the table_type described in the loop. The value of this field shall be the same as the version_number entered in the corresponding fields of tables and table instances. The version number for the next VCT (current_next_indicator = 0) shall be one unit more (modulo 32) than the version number for the current VCT (current_next_indicator = 1). For example, the value of this field for EIT-3 will be the same as that of the version_number that appears in the actual EIT-3.

number_bytes — This 32-bit unsigned integer field indicates the total number of bytes used for the table_type described in the loop.

table_type_descriptors_length — Total length of the descriptors for the table_type described in the loop (in bytes).

descriptors_length — Total length of the MGT descriptor list that follows (in bytes).

CRC_32 — This is a 32-bit field that contains the CRC value that ensures a zero output from the registers in the decoder defined in Annex A of ISO/IEC 13818-1 "MPEG-2 Systems" after processing the entire Master Guide Table section.

6.3 Virtual Channel Table (VCT)

The Virtual Channel Table (VCT) contains a list of attributes for virtual channels carried in the Transport Stream. Any changes in the virtual channel structure shall be conveyed with a new version number. The basic information contained in the VCT table body includes Transport Stream ID, channel number (major and minor), short channel name, carrier frequency, program number, access controlled flag, location field for extended text messages, and service type. Additional information may be carried by descriptors which may be placed in the descriptor loop after the basic information.

The Virtual Channel Table may be segmented into as many as 256 sections. One section may contain information for several virtual channels, but the information for one virtual channel shall not be segmented and put into two or more sections. Thus for each section, the first field after protocol_version shall be num_channels_in_section.

6.3.1 Terrestrial Virtual Channel Table

The Terrestrial Virtual Channel Table is carried in private sections with table ID 0xC8, and obeys the syntax and semantics of the Private Section as described in Section 2.4.4.10 and 2.4.4.11 of ISO/IEC 13818-1. The following constraints apply to the Transport Stream packets carrying the VCT sections:

- PID for Terrestrial VCT shall have the value 0x1FFB (base_PID)
- transport_scrambling_control bits shall have the value '00'
- adaptation_field_control bits shall have the value '01'

The bit stream syntax for the Terrestrial Virtual Channel Table is shown in Table 6.4.

table_id — An 8-bit unsigned integer number that indicates the type of table section being defined here. For the terrestrial_virtual_channel_table_section(), the table_id shall be 0xC8.

section_syntax_indicator— The section_syntax_indicator is a one-bit field which shall be set to '1' for the terrestrial_virtual_channel_table_section().

private_indicator — This 1-bit field shall be set to '1'.

section_length — This is a twelve bit field, the first two bits of which shall be '00'. It specifies the number of bytes of the section, starting immediately following the section_length field, and including the CRC. The value in this field shall not exceed 1021.

transport_stream_id — The 16-bit MPEG-2 Transport Stream ID, as it appears in the Program Association Table (PAT) identified by a PID value of zero for this multiplex. The transport_stream_id distinguishes this Terrestrial Virtual Channel Table from others that may be broadcast in different PTCs.

Table 6.4 Bit Stream Syntax for the Terrestrial Virtual Channel Table

Syntax	Bits	Format
terrestrial_virtual_channel_table_section () {		
table_id	8	0xC8
section_syntax_indicator	1	'1'
private_indicator	1	'1'
zero	2	'00'
section_length	12	uimsbf
transport_stream_id	16	uimsbf
reserved	2	'11'
version_number	5	uimsbf
current_next_indicator	1	bslbf
section_number	8	uimsbf
last_section_number	8	uimsbf
protocol_version	8	uimsbf
num_channels_in_section	8	uimsbf
for(i=0; i<num_channels_in_section;i++) {		
short_name	7*16	unicode™ BMP
reserved	4	'1111'
major_channel_number	10	uimsbf
minor_channel_number	10	uimsbf
modulation_mode	8	uimsbf
carrier_frequency	32	uimsbf
channel_TSID	16	uimsbf
program_number	16	uimsbf
ETM_location	2	uimsbf
access_controlled	1	bslbf
hidden	1	bslbf
reserved	6	'111111'
service_type	6	uimsbf
source_id	16	uimsbf
reserved	6	'111111'
descriptors_length	10	uimsbf
for (i=0;i<N;i++) {		
descriptors()		
}		
}		
reserved	6	'111111'
additional_descriptors_length	10	uimsbf
for(j=0; j<N;j++) {		
additional_descriptors()		
}		
CRC_32	32	rpchof
}		

version_number— This 5 bit field is the version number of the Virtual Channel Table. For the current VCT (current_next_indicator = 1), the version number shall be incremented by 1 whenever the definition of the current VCT changes. Upon reaching the value 31, it wraps around to 0. For the next VCT (current_next_indicator = 0), the version number shall be one unit more than that of the current VCT (also in modulo 32 arithmetic). In any case, the value of the version_number shall be identical to that of the corresponding entries in the MGT.

current_next_indicator— A one-bit indicator, which when set to '1' indicates that the Virtual Channel Table sent is currently applicable. When the bit is set to '0', it indicates that the table sent is not yet applicable and shall be the next table to become valid.

section_number— This 8 bit field gives the number of this section. The section_number of the first section in the Terrestrial Virtual Channel Table shall be 0x00. It shall be incremented by one with each additional section in the Terrestrial Virtual Channel Table.

last_section_number— This 8 bit field specifies the number of the last section (that is, the section with the highest section_number) of the complete Terrestrial Virtual Channel Table.

protocol_version — An 8-bit unsigned integer field whose function is to allow, in the future, this table type to carry parameters that may be structured differently than those defined in the current protocol. At present, the only valid value for protocol_version is zero. Non-zero values of protocol_version may only be processed by decoders designed to accommodate the later versions as they become standardized.

num_channels_in_section— This 8 bit field specifies the number of virtual channels in this VCT section. The number is limited by the section length.

short_name— The name of the virtual channel, represented as a sequence of one to seven 16-bit character codes coded in accordance with the Basic Multilingual Plane (BMP) of Unicode™, as specified in ISO 10646-1. If the name of the virtual channel is shorter than seven Unicode™ characters, one or more instances of the null character value 0x0000 shall be used to pad the string to its fixed 14-byte length.

major_channel_number— A 10-bit number that represents the "major" channel number associated with the virtual channel being defined in this iteration of the "for" loop. Each virtual channel must be associated with a major and a minor channel number. The major channel number, along with the minor channel number, act as the user's reference number for the virtual channel. The major_channel_number shall be between 1 and 99. For major_channel_number assignments in the U.S., refer to Annex B.

minor_channel_number— A 10-bit number in the range 0 to 999 that represents the "minor" or "sub-" channel number. This field, together with major_channel_number, performs as a two-part channel number, where minor_channel_number represents the second or right-hand part of the number. When the service_type is analog television, minor_channel_number shall be set to 0. Services whose service_type is either ATSC_digital_television or ATSC_audio_only shall use minor numbers between 1 and 99. For other types of services, such as data broadcasting, valid minor virtual channel numbers are between 1 and 999

modulation_mode — An 8-bit unsigned integer number that indicates the modulation mode for the transmitted carrier associated with this virtual channel. Values of modulation_mode are defined by this standard in Table 6.5. For digital signals, the standard values for modulation mode (values below 0x80) indicate transport framing structure, channel coding, interleaving, channel modulation, forward error correction, symbol rate, and other transmission-related parameters, by means of a reference to an appropriate standard. Values of modulation_mode 0x80 and above are outside the scope of ATSC. These may be used to specify non-standard modulation modes in private systems. A value of 0x80 for modulation_mode indicates that modulation parameters are specified in a private descriptor.

Table 6.5 Modulation Modes

modulation_mode	meaning	terrestrial broadcast	cable
0x00	[Reserved]		
0x01	**analog** — The virtual channel is modulated using standard analog methods for analog television.		
0x02	**SCTE_mode_1** — The virtual channel has a symbol rate of 5.057 Msps, transmitted in accordance with *Digital Transmission Standard for Cable Television*, Ref. [12] (Mode 1). Typically, mode 1 will be used for 64-QAM.	Not valid	
0x03	**SCTE_mode_2** — The virtual channel has a symbol rate of 5.361 Msps, transmitted in accordance with *Digital Transmission Standard for Cable Television*, Ref. [12] (Mode 2). Typically, mode 2 will be used for 256-QAM.	Not valid	
0x04	**ATSC (8 VSB)** — The virtual channel uses the 8-VSB modulation method conforming to the ATSC Digital Television Standard.		Not valid
0x05 -0x7F	[Reserved for future use by ATSC]		
0x80	Modulation parameters are defined by a private descriptor		
0x81-0xFF	[User Private]		

carrier_frequency— A 32-bit unsigned integer that represents the carrier frequency associated with the analog or digital transmission associated with this virtual channel, in units of one Hz. For VSB-modulated signals, the given carrier_frequency represents the location of the pilot tone; for analog signals, it represents the frequency of the picture carrier. In the case of a digital terrestrial broadcast signal that is transmitted at multiple carrier frequencies (via one or more translators), the carrier_frequency may be specified as zero. In such cases, the receiver is expected to associate the Transport Stream identified by the given transport_stream_id with the frequency tuned to acquire it.

For the ATSC Digital Television Standard, where the PTC bandwidth is 6 MHz, the pilot tone is located 310 kHz above the lower edge of the physical transmission channel, or 2.690 MHz below the specified center of the band. Similarly, for analog NTSC transmitted in the US, the picture carrier is 1.25 MHz above the lower edge of the 6 MHz physical transmission channel.

channel_TSID— A 16-bit unsigned integer field in the range 0x0000 to 0xFFFF that represents the MPEG-2 Transport Stream ID associated with the Transport Stream carrying the MPEG-2 program referenced by this virtual channel. The receiver may use the channel_TSID to verify that a TS acquired at the referenced carrier frequency is actually the desired multiplex. Analog signals may have a TSID provided that it is different from any DTV Transport Stream identifier; that is, it shall be truly unique if present.[2] A value of 0xFFFF for channel_TSID shall be specified for analog channels that do not have a valid TSID.

[2] A method to include such a unique 16-bit "Transmission Signal ID" in the NTSC VBI is specified in the EIA-752 specification.

program_number — A 16-bit unsigned integer number that associates the virtual channel being defined here with the MPEG-2 PROGRAM ASSOCIATION and TS PROGRAM MAP tables. For virtual channels representing analog services, a value of 0xFFFF shall be specified for **program_number**.

ETM_location — This 2-bit field specifies the existence and the location of an Extended Text Message (ETM), based on Table 6.6.

Table 6.6 ETM location

ETM_location	Meaning
0x00	No ETM
0x01	ETM located in the PTC carrying this PSIP
0x02	ETM located in the PTC specified by the channel_TSID
0x03	[Reserved for future ATSC use]

access_controlled — A 1-bit Boolean flag that indicates, when set, that the events associated with this virtual channel may be access controlled. When the flag is set to 0, event access is not restricted.

hidden — A 1-bit Boolean flag that indicates, when set, that the virtual channel is not accessed by the user by direct entry of the virtual channel number. Hidden virtual channels are skipped when the user is channel surfing, and appear as if undefined, if accessed by direct channel entry. Typical applications for hidden channels are test signals and NVOD services.

service_type— A 6-bit enumerated type field that identifies the type of service carried in this virtual channel, based on Table 6.7.

Table 6.7 Service Types

service_type	Meaning
0x00	[Reserved]
0x01	**analog_television** — The virtual channel carries analog television programming
0x02	**ATSC_digital_television** — The virtual channel carries television programming (audio, video and data) conforming to the ATSC Digital Television Standard
0x03	**ATSC_audio_only** — The virtual channel conforms to the ATSC Digital Television Standard, and has one or more standard audio and data components but no video.
0x04	**ATSC_data_broadcast_service** — Conforming to the ATSC data broadcast standard under development by T3/S13.
0x05-0x3F	[Reserved for future ATSC use]

source_id— A 16-bit unsigned integer number that identifies the programming source associated with the virtual channel. In this context, a *source* is one specific source of video, text, data, or audio programming. Source ID value zero is reserved. Source ID values in the range 0x0001 to 0x0FFF shall be unique within the Transport Stream that carries the VCT, while values 0x1000 to 0xFFFF shall be unique at the regional level. Values for source_ids 0x1000 and above shall be issued and administered by a Registration Authority designated by the ATSC.

descriptors_length — Total length (in bytes) of the descriptors for this virtual channel that follows.

additional_descriptors_length — Total length (in bytes) of the VCT descriptor list that follows.

CRC_32 — This is a 32-bit field that contains the CRC value that ensures a zero output from the registers in the decoder defined in Annex A of ISO/IEC 13818-1 "MPEG-2 Systems" after processing the entire Terrestrial Virtual Channel Table section.

6.3.2 Cable Virtual Channel Table

The Cable Virtual Channel Table is carried in private sections with table ID 0xC9, and obeys the syntax and semantics of the Private Section as described in Section 2.4.4.10 and 2.4.4.11 of ISO/IEC 13818-1. The following constraints apply to the Transport Stream packets carrying the VCT sections:

- PID for Cable VCT shall have the value 0x1FFB (base_PID)
- transport_scrambling_control bits shall have the value '00'
- adaptation_field_control bits shall have the value '01'

The bit stream syntax for the Cable Virtual Channel Table is shown in Table 6.8. The semantics for the CVCT are the same as the TVCT except for those fields explicitly defined below.

table_id — An 8-bit unsigned integer number that indicates the type of table section being defined here. For the cable_VCT_section, the table_id shall be 0xC9.

major_channel_number — A 10-bit number in the range 1 to 999 that represents the "major" virtual channel number associated with the virtual channel being defined in this iteration of the "for" loop. Each virtual channel must be associated with a major and a minor virtual channel number. The major virtual channel number, along with the minor virtual channel number, act as the user's reference number for the virtual channel.

minor_channel_number — A 10-bit number in the range 0 to 999 that represents the "minor" or "sub-" virtual channel number. This field, together with major_channel_number, performs a two-part virtual channel number, where minor_channel_number represents the second or right-hand part of the number

Table 6.8 Bit Stream Syntax for the Cable Virtual Channel Table

Syntax	Bits	Format
cable_virtual_channel_table_section () {		
table_id	8	0xC9
section_syntax_indicator	1	'1'
private_indicator	1	'1'
zero	2	'00'
section_length	12	uimsbf
transport_stream_id	16	uimsbf
reserved	2	'11'
version_number	5	uimsbf
current_next_indicator	1	bslbf
section_number	8	uimsbf
last_section_number	8	uimsbf
protocol_version	8	uimsbf
num_channels_in_section	8	uimsbf
for(i=0; i<num_channels_in_section;i++) {		
short_name	7*16	unicode™ BMP
reserved	4	'1111'
major_channel_number	10	uimsbf
minor_channel_number	10	uimsbf
modulation mode	8	uimsbf
carrier_frequency	32	uimsbf
channel_TSID	16	uimsbf
program_number	16	uimsbf
ETM_location	2	uimsbf
access_controlled	1	bslbf
hidden	1	bslbf
path_select	1	bslbf
out_of_band	1	bslbf
reserved	4	'1111'
service_type	6	uimsbf
source_id	16	uimsbf
reserved	6	'111111'
descriptors_length	10	uimsbf
for (i=0;i<N;i++) {		
descriptors()		
}		
}		
reserved	6	'111111'
additional_descriptors_length	10	uimsbf
for(j=0; j<N;j++) {		
additional_descriptors()		
}		
CRC_32	32	rpchof
}		

path_select — A 1-bit field that associates the virtual channel with a transmission path. For the cable transmission medium, path_select identifies which of two physical input cables carries the Transport Stream associated with this virtual channel. Table 6.9 defines path_select.

Table 6.9 Path Select

path_select	Meaning
0	path 1
1	path 2

out_of_band — A Boolean flag that indicates, when set, that the virtual channel defined in this iteration of the "for" loop is carried on the cable on an out-of-band physical transmission channel whose frequency is indicated by carrier_frequency. When clear, the virtual channel is carried within a standard tuned multiplex at that frequency.

source_id — A 16-bit unsigned integer number that identifies the programming source associated with the virtual channel. In this context, a *source* is one specific source of video, text, data, or audio programming. Source ID value zero is reserved to indicate that the programming source is not identified. Source ID values in the range 0x0001 to 0x0FFF shall be unique within the Transport Stream that carries the VCT, while values 0x1000 to 0xFFFF shall be unique at the regional level. Values for source_ids 0x1000 and above shall be issued and administered by a Registration Authority designated by the ATSC.

6.4 Rating Region Table (RRT)

The Rating Region Table (RRT) carries rating information for multiple geographical regions. Each RRT instance, identified by rating_region (the 8 least significant bits of table_id_extension), conveys the rating system information for one specific region. The size of each RRT instance shall not be more than 1024 bytes (including section header and trailer), and it shall be carried by only one MPEG-2 private section.

The following constraints apply to the Transport Stream packets carrying the RRT sections.

- PID shall have the value 0x1FFB (base_PID)
- transport_scrambling_control bits shall have the value '00'
- adaptation_field_control bits shall have the value '01'

The bit stream syntax for the Rating Region Table is shown in Table 6.10.

table_id — This is an 8-bit field, which shall be set to 0xCA, identifying this table as the Rating Region Table (RRT).

section_syntax_indicator — This 1-bit field shall be set to '1'. It denotes that the section follows the generic section syntax beyond the section length field.

private_indicator — This 1-bit field shall be set to '1'.

section_length — 12-bit field specifying the number of remaining bytes in this section immediately following the section_length field up to the end of the section. The value of the section_length shall be no larger than 1021.

Table 6.10 Bit Stream Syntax for the Rating Region Table

Syntax	Bits	Format
rating_region_table_section () {		
table_id	8	0xCA
section_syntax_indicator	1	'1'
private_indicator	1	'1'
zero	2	'00'
section_length	12	uimsbf
table_id_extension{		
reserved	8	0xFF
rating_region	8	uimsbf
}		
reserved	2	'11'
version_number	5	uimsbf
current_next_indicator	1	'1'
section_number	8	uimsbf
last_section_number	8	uimsbf
protocol_version	8	uimsbf
rating_region_name_length	8	uimsbf
rating_region_name_text()	var	
dimensions_defined	8	uimsbf
for(i=0; i<dimensions_defined;i++) {		
dimension_name_length	8	uimsbf
dimension_name_text()	var	
reserved	3	'111'
graduated_scale	1	bslbf
values_defined	4	uimsbf
for (j=0;j<values_defined;j ++) {		
abbrev_rating_value_length	8	uimsbf
abbrev_rating_value_ text()	var	
rating_value_length	8	uimsbf
rating_value_ text()	var	
}		
}		
reserved	6	'111111'
descriptors_length	10	uimsbf
for (i=0;i<N;i++) {		
descriptors()	var	
}		
CRC_32	32	rpchof
}		

rating_region — An 8-bit unsigned integer number that defines the rating region to be associated with the text in this rating_region_table_section(). The value of this field is the identifier of this rating region, and thus this field may be used by the other tables (e.g. MGT) for referring to a specific rating region table. Values of rating_region are defined in Table 6.11.

Table 6.11 Rating Regions

rating_region	Rating Region Name
0x00	Forbidden
0x01	**US (50 states + possessions)**
0x02-0xFF	[Reserved]

version_number — This 5-bit field is the version number of the Rating Region table identified by combination of the fields table_id and table_id_extension. The version number shall be incremented by 1 modulo 32 when any field in this instance of the Rating Region Table changes. The value of this field shall be the same as that of the corresponding entry in MGT.

current_next_indicator — This 1-bit indicator is always set to '1'.

section_number — The value of this 8-bit field shall always be 0x00.

last_section_number — The value of this 8-bit field shall always be 0x00.

protocol_version — The value of this 8-bit field shall always be 0x00.

rating_region_name_length — An 8-bit unsigned integer number that defines the total length (in bytes) of the rating_region_name_text() field to follow.

rating_region_name_text() — A data structure containing a multiple string structure which represents the rating region name, e.g. "U.S. (50 states + possessions)", associated with the value given by rating_region. Text strings are formatted according to the rules outlined in Section 6.8. The display string for the rating region name shall be limited to 32 characters or less.

dimensions_defined — This 8-bit field (1-255) specifies the number of dimensions defined in this rating_region_table_section().

dimension_name_length — An 8-bit unsigned integer number that defines the total length in bytes of the dimension_name_text() field to follow.

dimension_name_text() — A data structure containing a multiple string structure which represents the dimension name being described in the loop. One dimension in the U.S. rating region, for example, is used to describe the MPAA list. The dimension name for such a case may be defined as "MPAA". Text strings are formatted according to the rules outlined in Section 6.8. The dimension name display string shall be limited to 20 characters or less.

graduated_scale — This 1-bit flag indicates whether or not the rating values in this dimension represent a graduated scale, i.e., higher rating values represent increasing levels of rated content within the dimension. Value 1 means yes, while value 0 means no.

values_defined — This 4-bit field (1-15) specifies the number of values defined for this particular dimension.

abbrev_rating_value_length — An 8-bit unsigned integer number that defines the total length (in bytes) of the abbrev_rating_value_text() field to follow.

abbrev_rating_value_text() — A data structure containing a multiple string structure which represents the abbreviated name for one particular rating value. The abbreviated name for rating value 0 shall be set to a null string, i.e., "". Text strings are formatted according to the rules outlined in Section 6.8. The abbreviated value display string shall be limited to 8 characters or less.

rating_value_length — An 8-bit unsigned integer number that defines the total length (in bytes) of the rating_value_text() field to follow.

rating_value_text() — A data structure containing a multiple string structure which represents the full name for one particular rating value. The full name for rating value 0 shall be set to a null string, i.e., "". Text strings are formatted according to the rules outlined in Section 6.8. The rating value display string shall be limited to 150 characters or less.

descriptors_length — Length (in bytes) of all of the descriptors that follow this field.

CRC_32 — This is a 32-bit field that contains the CRC value that ensures a zero output from the registers in the decoder defined in Annex A of ISO/IEC 13818-1 "MPEG-2 Systems" after processing the entire Rating Region Table section.

6.5 Event Information Table (EIT)

The Event Information Table (EIT) contains information (titles, start times, etc.) for events on defined virtual channels. An event is, in most cases, a typical TV program, however its definition may be extended to include particular data broadcasting sessions and other information segments. Up to 128 EITs may be transmitted and each of them is referred to as EIT-k, with k = 0, 1, … 127.

Each EIT-k can have multiple instances, each of which contains information for one virtual channel, and each of which is identified by the combination of table_id and source_id. Each EIT-k instance may be segmented into as many as 256 sections. One section may contain information for several events, but the information for one event shall not be segmented and put into two or more sections. Thus the first field after protocol_version for each section shall be num_events_in_section.

The PSIP shall have at least four EITs and no more than 128 EITs, each of which provides the event information for a certain time span. Any event programmed for a time interval that extends over one or more EITs shall be described in each of these EITs, with the same event_id. For instance, an event that starts at 17:30 UTC and lasts until 19:30 UTC will appear in two EITs with the same event_id, the EIT covering 15:00-18:00 (UTC) as well as the EIT covering 18:00-21:00 (UTC). For a particular virtual channel, an event_id identifies uniquely each of the events programmed for the 3-hour interval of an EIT.

Each virtual channel defined in the VCT shall have a corresponding instance of EIT-k, unless the virtual channel belongs to a group sharing the same source_id. Virtual channels sharing a source_id appear in applications such as NVOD. In such a case, the entire group will have a unique instance of EIT-k identified precisely by the source_id. If a virtual channel has no event in the time span covered by EIT-k, its corresponding EIT instance shall have only one section, and the field num_events_in_section shall be set to zero.

Events shall be in the order of their starting times, i.e., the start time of the first event shall be ahead of that of the second event, and the start time of the last event in section one shall be equal or less than that of the first event in section two with the equality holding only when both events are the same..

The Event Information Table is carried in private sections with table ID 0xCB, and obeys the syntax and semantics of the Private Section as described in Section 2.4.4.10 and 2.4.4.11 of ISO/IEC 13818-1. The following constraints apply to the Transport Stream packets carrying the EIT sections:

- PID for EIT-k shall have the same value as specified in the MGT, and shall be unique among the collection of table_type_PID values listed in the MGT.
- transport_scrambling_control bits shall have the value '00'.
- adaptation_field_control bits shall have the value '01'.

The bit stream syntax for the Event Information Table is shown in Table 6.12.

table_id — This is an 8-bit field which shall be set to 0xCB, identifying this section as belonging to the Event Information Table.

section_syntax_indicator — This 1-bit field shall be set to '1'. It denotes that the section follows the generic section syntax beyond the section length field.

private_indicator — This 1-bit field shall be set to '1'.

section_length — 12-bit field specifying the number of remaining bytes in this section immediately following the section_length field up to the end of the section, including the CRC_32 field. The value of this field shall not exceed 4093.

source_id — This 16-bit field specifies the source_id of the virtual channel carrying the events described in this section.

version_number — This 5-bit field is the version number of EIT-i. The version number shall be incremented by 1 modulo 32 when any field in the EIT-i changes. Note that the version_number for EIT-i has no relation with that for EIT-j when j is not equal to i. The value of this field shall be identical to that of the corresponding entry in the MGT.

current_next_indicator — This 1-bit indicator is always set to '1' for EIT sections; the EIT sent is always currently applicable.

section_number — This 8-bit field gives the number of this section.

last_section_number — This 8-bit field specifies the number of the last section.

protocol_version — An 8-bit unsigned integer field whose function is to allow, in the future, this table type to carry parameters that may be structured differently than those defined in the current protocol. At present, the only valid value for protocol_version is zero. Non-zero values of protocol_version may only be processed by decoders designed to accommodate the later versions as they become standardized.

Table 6.12 Bit Stream Syntax for the Event Information Table

Syntax	Bits	Format
event_information_table_section () {		
table_id	8	0xCB
section_syntax_indicator	1	'1'
private_indicator	1	'1'
reserved	2	'11'
section_length	12	uimsbf
source_id	16	uimsbf
zero	2	'00'
version_number	5	uimsbf
current_next_indicator	1	'1'
section_number	8	uimsbf
last_section_number	8	uimsbf
protocol_version	8	uimsbf
num_events_in_section	8	uimsbf
for (j = 0; j< num_events_in_section;j++) {		
reserved	2	'11'
event_id	14	uimsbf
start_time	32	uimsbf
reserved	2	'11'
ETM_location	2	uimsbf
length_in_seconds	20	uimsbf
title_length	8	uimsbf
title_text()	var	
reserved	4	'1111'
descriptors_length	12	
for (i=0;i<N;i++) {		
descriptor()		
}		
}		
CRC_32	32	rpchof
}		

num_events_in_section — Indicates the number of events in this EIT section. Value 0 indicates no events defined in this section.

event_id — This field specifies the identification number of the event described. This number will serve as a part of the event ETM_id (identifier for event extended text message).

start_time — A 32-bit unsigned integer quantity representing the start time of this event as the number of GPS seconds since 12 am, January 6th, 1980.

ETM_location — This 2-bit field specifies the existence and the location of an Extended Text Message (ETM), based on Table 6.13

Table 6.13 ETM_location

ETM_location	Meaning
0x00	No ETM
0x01	ETM located in the PTC carrying this PSIP
0x02	ETM located in the PTC carrying this event
0x03	[Reserved for future ATSC use]

length_in_seconds — Duration (in seconds) of this event.

title_length — This field specifies the length (in bytes) of the title_text(). Value 0 means that no title exists for this event.

title_text() — The event title in the format of a multiple string structure (see Section 6.8).

descriptors_length — Total length (in bytes) of the event descriptor list that follows.

CRC_32 — This is a 32-bit field that contains the CRC value that ensures a zero output from the registers in the decoder defined in Annex A of ISO-13818-1 "MPEG-2 Systems" after processing the entire Event Information Table section.

6.6 Extended Text Table

The Extended Text Table (ETT) contains Extended Text Message (ETM) streams, which are optional and are used to provide detailed descriptions of virtual channels (channel ETM) and events (event ETM). An ETM is a multiple string data structure (see Section 6.8), and thus, it may represent a description in several different languages (each string corresponding to one language). If necessary, the description may be truncated to fit allocated display space.

Within a Transport Stream, the Extended Text Message is carried on a private section with table ID 0xCC. Each description is distinguished by its unique 32-bit ETM_id immediately after the field protocol_version. This allows the receiver to search for a single description quickly without having to parse the payload of a large table.

The ETT section for a virtual channel or an event is carried in the home physical transmission channel (the physical transmission channel carrying that virtual channel or event) with PID specified by the field table_type_PID in corresponding entries in the MGT. This specific PID is exclusively reserved for the ETT stream.

The following constraints apply to the Transport Stream packets carrying the ETT sections.

- PID for ETT shall have the same value as the field table_type_PID in corresponding entries in the MGT, and shall be unique among the collection of table_type_PID values listed in the MGT.
- transport_scrambling_control bits shall have the value '00'
- adaptation_field_control bits shall have the value '01'

The bit stream syntax for the Extended Text Table is shown in Table 6.14.

Table 6.14 Bit Stream Syntax for the Extended Text Table

Syntax	Bits	Format
extended_text_table_section () {		
table_id	8	0xCC
section_syntax_indicator	1	'1'
private_indicator	1	'1'
reserved	2	'11'
section_length	12	uimsbf
table_id_extension	16	0x00
reserved	2	'11'
version_number	5	0x00
current_next_indicator	1	'1'
section_number	8	0x00
last_section_number	8	0x00
protocol_version	8	uimsbf
ETM_id	32	uimsbf
extended_text_message ()	var	
CRC_32	32	rpchof
}		

table_id — Identifies this section as belonging to a Extended Text Table. (0xCC)

section_syntax_indicator — This 1-bit field shall be set to '1'. It denotes that the section follows the generic section syntax beyond the section length field.

private_indicator — This 1-bit field shall be set to '1'.

section_length — 12-bit field specifying the number of remaining bytes in the section immediately following the section_length field up to the end of the section. The value of the section_length shall be no larger than 4093.

table_id_extension — This 16-bit field shall be set to 0x00.

version_number — For the channel ETT, this 5-bit field indicates the version number of the channel ETT. The version number shall be incremented by 1 modulo 32 when any ETM in the channel ETT changes. For event ETT, this 5-bit field indicates the version number of event ETT-i, where i, as in the EIT case, is the index of time span. The version number shall be incremented by 1 modulo 32 when any ETM in the event ETT-i changes. Note that the version_number for event ETT-i has no relation with that for event ETT-j when j is not equal to i. The value of this field shall be identical to that of the corresponding entry in the MGT.

current_next_indicator — This 1-bit indicator is always set to '1' for ETT sections; the ETT sent is always currently applicable.

section_number — The value of this 8-bit field shall always be 0x00 (this table is only one section long).

last_section_number — The value of this 8-bit field shall always be 0x00.

protocol_version — An 8-bit unsigned integer field whose function is to allow, in the future, this table type to carry parameters that may be structured differently than those defined in the current protocol. At present, the only valid value for protocol_version is zero. Non-zero values of protocol_version may only be processed by decoders designed to accommodate the later versions as they become standardized.

ETM_id — Unique 32-bit identifier of this extended text message. This identifier is assigned by the rule shown in Table 6.15.

Table 6.15 ETM ID

	MSB					
						LSB
Bit	**31**	**16**	**15**	**2**	**1**	**0**
channel ETM_id	source_id		0	0	0	0
event ETM_id	source_id		event_id		1	0

extended_text_message() — The extended text message in the format of a multiple string structure (see Section 6.8).

CRC_32 — This is a 32-bit field that contains the CRC value that ensures a zero output from the registers in the decoder defined in Annex A of ISO-13818-1 "MPEG-2 Systems" after processing the entire Transport Stream ETT section.

6.7 Core Descriptors

Table 6.16 lists all of the core descriptors and their descriptor tags. Asterisks mark the tables where the descriptors may appear. The range of MPEG-2 defined or reserved descriptor tags is between 0x02 and 0x3F.

Table 6.16 List of Descriptors for PSIP Tables.

Descriptor Name	Descriptor tag	Terrestrial				Cable		
		PMT	MGT	VCT	EIT	PMT	MGT	VCT
stuffing descriptor	0x80	*	*	*	*	*	*	*
AC-3 audio descriptor	0x81	*			*	*		
program identifier descriptor	0x85	*				*		
caption service descriptor	0x86	*			*	*		
content advisory descriptor	0x87	*			*	*		
extended channel name descriptor	0xA0			*				*
service location descriptor	0xA1			*				
time-shifted service descriptor	0xA2			*				*
component name descriptor	0xA3					*		
user private	0xC0-0xFF		*	*	*		*	*

6.7.1 AC-3 Audio Descriptor

The AC-3 audio descriptor, as defined in Ref. [1] and constrained in Annex B of Ref. [2] , may be used in the PMT and/or in EITs.

6.7.2 Program Identifier Descriptor

The program_identifier_descriptor, as defined in Ref. [5], may be used in the PMT.

6.7.3 Caption Service Descriptor

The caption service descriptor provides closed captioning information, such as closed captioning type and language code for events with closed captioning service. This descriptor shall not appear on events with no closed captioning service.

The bit stream syntax for the closed captioning service descriptor is shown in Table 6.17.

Table 6.17 Bit Stream Syntax for the Caption Service Descriptor

Syntax	Bits	Format
caption_service_descriptor () {		
descriptor_tag	8	0x86
descriptor_length	8	uimsbf
reserved	3	'111'
number_of_services	5	uimsbf
for (i=0;i<number_of_services;i++) {		
language	8*3	uimsbf
cc_type	1	bslbf
reserved	1	'1'
if (cc_type==line21) {		
reserved	5	'11111'
line21_field	1	bslbf
}		
else		
caption_service_number	6	uimsbf
easy_reader	1	bslbf
wide_aspect_ratio	1	bslbf
reserved	14	'11111111111111'
}		
}		

descriptor_tag — An 8-bit field that identifies the type of descriptor. For the caption_service_descriptor() the value is 0x86.

descriptor_length — An 8-bit count of the number of bytes following the descriptor_length itself.

number_of_services — An unsigned 5-bit integer in the range 1 to 16 that indicates the number of closed caption services present in the associated video service. Note that if the video service does not carry television closed captioning, the caption_service_descriptor() shall not be present either in the Program Map Table or in the Event Information Table.

Each iteration of the "for" loop defines one closed caption service present as a sub-stream within the 9600 bit per second closed captioning stream. Each iteration provides the sub-stream's language, attributes, and (for advanced captions) the associated Service Number reference. Refer to Ref. [13] for a description of the use of the Service Number field within the syntax of the closed caption stream.

language — A 3-byte language code per ISO 639.2/B (Ref. [7]) defining the language associated with one closed caption service. The ISO_639_language_code field contains a three-character code as specified by ISO 639.2/B. Each character is coded into 8 bits according to ISO 8859-1 (ISO Latin-1) and inserted in order into the 24-bit field.

cc_type — A flag that indicates, when set, that an advanced television closed caption service is present in accordance with Ref. [13]. When the flag is clear, a line-21 closed caption service is present. For line 21 closed captions, the line21_field field indicates whether the service is carried in the even or odd field.

line21_field — A flag that indicates, when set, that the line 21 closed caption service is associated with the field 2 of the NTSC waveform. When the flag is clear, the line-21 closed caption service is associated with field 1 of the NTSC waveform. The line21_field flag is defined only if the cc_type flag indicates line-21 closed caption service.

caption_service_number — A 6-bit unsigned integer value in the range zero to 63 that identifies the Service Number within the closed captioning stream that is associated with the language and attributes defined in this iteration of the "for" loop. See Ref. [13] for a description of the use of the Service Number. The caption_service_number field is defined only if the cc_type flag indicates closed captioning in accordance with Ref. [13].

easy_reader — A Boolean flag which indicates, when set, that the closed caption service contains text tailored to the needs of beginning readers. Refer to Ref. [13] for a description of "easy reader" television closed captioning services. When the flag is clear, the closed caption service is not so tailored.

wide_aspect_ratio — A Boolean flag which indicates, when set, that the closed caption service is formatted for displays with 16:9 aspect ratio. When the flag is clear, the closed caption service is formatted for 4:3 display, but may be optionally displayed centered within a 16:9 display.

6.7.4 Content Advisory Descriptor

The Content Advisory Descriptor is used to indicate, for a given event, ratings for any or all of the rating dimensions defined in the RRT (Rating Region Table). Ratings may be given for any or all of the defined regions, up to a maximum of 8 regions per event. An Event without a Content Advisory Descriptor indicates that the rating value for any rating dimension defined in any rating region is zero. The absence of ratings for a specific dimension is completely equivalent to having a zero-valued rating for such a dimension. The absence of ratings for a specific region implies the absence of ratings for all of the dimensions in the region. The absence of a Content Advisory Descriptor for a specific event implies the absence of ratings for all of the regions for the event.

The bit stream syntax for the Content Advisory Descriptor is shown in Table 6.18.

descriptor_tag — This 8-bit unsigned integer shall have the value 0x87, identifying this descriptor as content_advisory_descriptor.

descriptor_length — This 8-bit unsigned integer specifies the length (in bytes) immediately following this field up to the end of this descriptor.

rating_region_count — A 6-bit unsigned integer value in the range 1 to 8 that indicates the number of rating region specifications to follow.

rating_region — An unsigned 8-bit integer that specifies the rating region for which the data in the bytes to follow is defined. The rating_region associates ratings data given here with data defined in a Ratings Region Table tagged with the corresponding rating region.

rated_dimensions — An 8-bit unsigned integer field that specifies the number of rating dimensions for which content advisories are specified for this event. The value of this field shall not be greater than the value specified by the field dimensions_defined in the corresponding RRT section.

Table 6.18 Bit Stream Syntax for the Content Advisory Descriptor

Syntax	Bits	Format
content_advisory_descriptor () {		
descriptor_tag	8	0x87
descriptor_length	8	uimsbf
reserved	2	'11'
rating_region_count	6	
for (i=0; i<rating_region_count; i++) {		
rating_region	8	uimsbf
rated_dimensions	8	uimsbf
for (j=0;j<rated_dimensions;j++) {		
rating_dimension_j	8	uimsbf
reserved	4	'1111'
rating_value	4	uimsbf
}		
rating_description_length	8	uimsbf
rating_description_text()	var	
}		
}		

rating_dimension_j — An 8-bit unsigned integer field specifies the dimension index into the RRT instance for the region specified by the field rating_region. These dimension indices shall be listed in numerical order, i.e., the value of rating_dimension_j+1 shall be greater than that of rating_dimension_j.

rating_value — A 4-bit field represents the rating value of the dimension specified by the field rating_dimension_j for the region given by rating_region.

rating_description_length — An 8-bit unsigned integer value in the range zero to 80 that represents the length of the rating_description_text() field to follow.

rating_description_text() — The rating description in the format of a multiple string structure (see Section 6.8). The rating_description display string shall be limited to 16 characters or less. The rating description text shall represent the program's rating in an abbreviated form suitable for on-screen display. The rating description text collects multidimensional text information into a single small text string. If "xxx" and "yyy" are abbreviated forms for rating values in two dimensions, then "xxx-yyy" and "xxx (yyy)" are examples of possible strings represented in rating_description_text().

6.7.5 Extended Channel Name Descriptor

The extended channel name descriptor provides the long channel name for the virtual channel containing this descriptor.

The bit stream syntax for the extended channel name descriptor is shown in Table 6.19.

Table 6.19 Bit Stream Syntax for the Extended Channel Name Descriptor

Syntax	Bits	Format
extended_channel_name_descriptor () {		
descriptor_tag	8	0xA0
descriptor_length	8	uimsbf
long_channel_name_text()	var	
}		

descriptor_tag — This 8-bit unsigned integer shall have the value 0xA0, identifying this descriptor as extended_channel_name_descriptor().

descriptor_length — This 8-bit unsigned integer specifies the length (in bytes) immediately following this field up to the end of this descriptor.

long_channel_name_text() — The long channel name in the format of a multiple string structure (see Section 6.8).

6.7.6 Service Location Descriptor

This descriptor specifies the stream types, PID and language code for each elementary stream. This descriptor shall appear in the TVCT, and must be valid for the current event in the corresponding virtual channel.

The bit stream syntax for the service location descriptor is shown in Table 6.20.

Table 6.20 Bit Stream Syntax for the Service Location Descriptor

Syntax	Bits	Format
service_location_descriptor () {		
descriptor_tag	8	0xA1
descriptor_length	8	uimsbf
reserved	3	'111'
PCR_PID	13	uimsbf
number_elements	8	uimsbf
for (i=0;i<number_elements;i++) {		
stream_type	8	uimsbf
reserved	3	'111'
elementary_PID	13	uimsbf
ISO_639_language_code	8*3	uimsbf
}		
}		

descriptor_tag — This 8-bit unsigned integer shall have the value 0xA1, identifying this descriptor as service_location_descriptor().

descriptor_length — This 8-bit unsigned integer specifies the length (in bytes) immediately following this field up to the end of this descriptor.

PCR_PID — This is a 13 bit field indicating the PID of the Transport Stream packets which shall contain the PCR fields valid for the program specified by program_number. If no PCR is associated with a program definition for private streams then this field shall take the value of 0x1FFF.

number_elements — This 8-bit unsigned integer indicates the number of PIDs used for this program.

stream_type — This 8-bit unsigned integer field specifies the type of the elementary stream according to Table 6.21.

Table 6.21 Stream Type Assignments

Value	Description
0x00	ITU-T I ISO/IEC Reserved
0x01-0x7F	As specified in Table 2.29 (Stream type assignments) of Ref. [10]
0x80	[Used in other systems]
0x81	ATSC A/53 audio
0x82-0x84	[Used in other systems]
0x85	UPID (Ref.[5])
0x86-0xBF	Reserved
0xC0-0xFF	User Private

elementary_PID — Packet Identifier for the elementary stream.

ISO_639_language_code — This 3-byte (24 bits) field, based on ISO 639.2/B, specifies the language used for the elementary stream. In case of no language specified for this elementary stream, e.g. video, each byte shall have the value 0x00.

6.7.7 Time-Shifted Service Descriptor

This descriptor links one virtual channel with one or more virtual channels that carry the same programming on a time-shifted basis. The typical application is for Near Video On Demand (NVOD) services.

The bit stream syntax for the time_shifted_service_descriptor() is shown in Table 6.22.

Table 6.22 Bit Stream Syntax for the Time Shifted Service Descriptor

Syntax	Bits	Format
time_shifted_service_descriptor () {		
descriptor_tag	8	0xA2
descriptor_length	8	uimsbf
reserved	3	'111'
number_of_services	5	uimsbf
for (i=0;i<number_of_services;i++) {		
reserved	6	'111111'
time_shift	10	uimsbf
reserved	4	'1111'
major_channel_number	10	uimsbf
minor_channel_number	10	uimsbf
}		
}		

descriptor_tag — This 8-bit unsigned integer shall have the value 0xA2, identifying this descriptor as time_shifted_service_descriptor().

descriptor_length — This 8-bit unsigned integer specifies the length (in bytes) immediately following this field up to the end of this descriptor.

number_of_services — A 5-bit number in the range 1 to 20 that indicates the number of time-shifted services being defined here.

time_shift — A 10-bit number in the range 1 to 720 that represents the number of minutes the time-shifted service indicated by major_channel_number and minor_channel_number is time-shifted from the virtual channel associated with this descriptor.

major_channel_number — A 10-bit number in the range 1 to 999 that represents the "major" channel number associated with a time-shifted service.

minor_channel_number — A 10-bit number in the range 0 to 999 that, when non-zero, represents the "minor" or "sub-" channel number of the virtual channel that carries a time-shifted service.

6.7.8 Component Name Descriptor

Table 6.23 defines the component_name_descriptor(), which serves to define an optional textual name tag for any component of the service.

Table 6.23 Bit Stream Syntax for the Component Name Descriptor

Syntax	Bits	Format
component_name_descriptor() {		
descriptor_tag		80xA3
descriptor_length		8uimsbf
component_name_string()		v
	ar	
}		

descriptor_tag — This 8-bit unsigned integer shall have the value 0xA3, identifying this descriptor as component_name_descriptor.

descriptor_length — This 8-bit unsigned integer specifies the length (in bytes) immediately following this field up to the end of this descriptor.

component_name_string() — The name string in the format of a multiple string structure (see Section 6.8).

6.7.9 Stuffing Descriptor

For certain applications it is necessary to define a block of N bytes as a placeholder. The N bytes themselves are not to be processed or interpreted. The stuffing_descriptor() is specified for this purpose. The stuffing_descriptor() is simply a descriptor type for which the contents, as indicated by the descriptor_length field, are to be disregarded. The tag type for the stuffing descriptor is 0x80. The stuffing_descriptor() may appear where descriptors are allowed in any table defined in the PSIP.

6.8 Multiple String Structure

This is a general data structure used specifically for text strings. Text strings appear as event titles, long channel names, the ETT messages, and RRT text items. The bit stream syntax for the Multiple String Structure is shown in Table 6.24.

number_strings — This 8-bit unsigned integer field identifies the number of strings in the following data.

ISO_639_language_code — This 3-byte (24 bits) field, based on ISO 639.2/B, specifies the language used for the i^{th} string.

number_segments — This 8-bit unsigned integer field identifies the number of segments in the following data. A specific mode is assigned for each segment.

Table 6.24 Bit Stream Syntax for the Multiple String Structure

Syntax	Bits	Format
multiple_string_structure () {		
number_strings	8	uimsbf
for (i= 0;i< number_strings;i++) {		
ISO_639_language_code	8*3	uimsbf
number_segments	8	uimsbf
for (j=0;j<number_segments;j++) {		
compression_type	8	uimsbf
mode	8	uimsbf
number_bytes	8	uimsbf
for (k= 0;k<number_bytes;k++)		
compressed_string_byte [k]	8	bslbf
}		
}		
}		

compression_type — This 8-bit field identifies the compression type for the jth segment. Allowed values for this field are shown in Table 6.25.

Table 6.25 Compression Types

compression_type	compression method
0x00	No compression
0x01	Huffman coding using standard encode/decode tables defined in Table C.4 and C.5 in Annex C.
0x02	Huffman coding using standard encode/decode tables defined in Table C.6 and C.7 in Annex C.
0x03 to 0xAF	reserved
0xB0 to 0xFF	user private

mode — An 8-bit value representing the text mode to be used to interpret characters in the segment to follow. See Table 6.26 for definition. Mode values in the range zero through 0x3E select 8-bit Unicode™ character code pages. Mode value 0x3F selects 16-bit Unicode™ character coding. Mode values 0x40 through 0xDF are reserved for future use by ATSC. Mode values 0xE0 through 0xFE are user private. Mode value 0xFF indicates the text mode is not applicable. Decoders shall ignore string bytes associated with unknown or unsupported mode values.

number_bytes — This 8-bit unsigned integer field identifies the number of bytes that follows.

compressed_string_byte[k] — The kth byte of the jth segment.

Table 6.26 Modes

Mode	Meaning	Language(s) or Script
0x00	Select ISO/IEC 10646-1 Page 0x00	ASCII, ISO Latin-1 (Roman)[3]
0x01	Select ISO/IEC 10646-1 Page 0x01	European Latin (many)[4]
0x02	Select ISO/IEC 10646-1 Page 0x02	Standard Phonetic
0x03	Select ISO/IEC 10646-1 Page 0x03	Greek
0x04	Select ISO/IEC 10646-1 Page 0x04	Russian, Slavic
0x05	Select ISO/IEC 10646-1 Page 0x05	Armenian, Hebrew
0x06	Select ISO/IEC 10646-1 Page 0x06	Arabic[5]
0x07-0x08	Reserved	-
0x09	Select ISO/IEC 10646-1 Page 0x09	Devanagari[6], Bengali
0x0A	Select ISO/IEC 10646-1 Page 0x0A	Punjabi, Gujarati
0x0B	Select ISO/IEC 10646-1 Page 0x0B	Oriya, Tamil
0x0C	Select ISO/IEC 10646-1 Page 0x0C	Telugu, Kannada
0x0D	Select ISO/IEC 10646-1 Page 0x0D	Malayalam
0x0E	Select ISO/IEC 10646-1 Page 0x0E	Thai, Lao
0x0F	Reserved	-
0x10	Select ISO/IEC 10646-1 Page 0x10	Tibetan, Georgian
0x11-0x1F	Reserved	-
0x20	Select ISO/IEC 10646-1 Page 0x20	Miscellaneous
0x21	Select ISO/IEC 10646-1 Page 0x21	Misc. symbols, arrows
0x22	Select ISO/IEC 10646-1 Page 0x22	Mathematical operators
0x23	Select ISO/IEC 10646-1 Page 0x23	Misc. technical
0x24	Select ISO/IEC 10646-1 Page 0x24	OCR, enclosed alpha-num.
0x25	Select ISO/IEC 10646-1 Page 0x25	Form and chart components
0x26	Select ISO/IEC 10646-1 Page 0x26	Miscellaneous dingbats
0x27	Select ISO/IEC 10646-1 Page 0x27	Zapf dingbats
0x28-0x2F	Reserved	-
0x30	Select ISO/IEC 10646-1 Page 0x30	Hiragana, Katakana
0x31	Select ISO/IEC 10646-1 Page 0x31	Bopomopho, Hangul elem.
0x32	Select ISO/IEC 10646-1 Page 0x32	Enclosed CJK Letters, ideo.
0x33	Select ISO/IEC 10646-1 Page 0x33	Enclosed CJK Letters, ideo.
0x34-0x3E	Reserved	-
0x3F	Select 16-bit ISO/IEC 10646-1 mode	all
0x40-0xDF	Reserved	
0xE0-0xFE	User private	
0xFF	Not applicable	

[3] The languages supported by ASCII plus the Latin-1 supplement include Danish, Dutch, English, Faroese, Finnish, Flemish, German, Icelandic, Irish, Italian, Norwegian, Portuguese, Spanish and Swedish. Many other languages can be written with this set of characters, including Hawaiian, Indonesian, and Swahili.

[4] When combined with page zero (ASCII and ISO Latin-1), covers Afrikaans, Breton, Basque, Catalan, Croatian, Czech, Esperanto, Estonian, French , Frisian, Greenlandic, Hungarian, Latin, Latvian, Lithuanian, Maltese, Polish, Provencal, Rhaeto-Romanic, Romanian, Romany, Sami, Slovak, Slovenian, Sorbian, Turkish, Welsh, and many others.

[5] Also Persian, Urdu, Pashto, Sindhi, and Kurdish.

[6] Devanagari script is used for writing Sanskrit and Hindi, as well as other languages of northern India (such as Marathi) and of Nepal (Nepali). In addition, at least two dozen other Indian languages use Devanagari script.

7. PSIP STD MODEL

7.1 Buffer Model for Terrestrial Broadcast

Table 7.1 lists the maximum cycle time for all PSIP tables, except EITs and ETTs. Table 7.2 lists the maximum transmission rate for PSIP packet streams according to their PIDs. The recommended maximum cycle time for EIT-0 is 500 ms.

Table 7.1 Maximum cycle time for the STT, MGT, VCT and RRT

Table	STT	MGT	VCT	RRT
Cycle time (ms)	1000	150	400	60000

Table 7.2 Maximum rate for each PSIP packet stream

PID	base_PID	EIT_PID	ETT_PID
Rate (bps)	250,000	250,000	250,000

For terrestrial broadcast applications the following constraints apply:

- In terrestrial broadcast applications, the PSIP elementary streams identified by Transport Stream packets with PID 0x1FFB (base_PID), EIT PIDs and ETT PIDs shall adhere to an STD model with the following parameters:
- sb_leak_rate shall be 625 (indicating a leak rate of 250,000 bps)
- sb_size shall be 1024 (indicating a smoothing buffer size of 1024 bytes)

7.2 Buffer Model for Cable

Transmission rates for cable will be standardized by the SCTE.

ANNEX A

(Normative)

DAYLIGHT SAVINGS TIME CONTROL

In order to convert GPS into local time, the receiver needs to store a time offset (from GPS to local time) in local memory and an indicator as to whether daylight savings is observed. These two quantities can be obtained from the user interface (indicating time zone and daylight savings observance) or from the conditional access system, if present, and stored in non-volatile receiver memory.

Since there is a common time (GPS) transmitted in the PSIP, there needs to be a mechanism to indicate when the receiver should switch into (or out of) daylight savings time at the appropriate local time. Once all the receivers have transitioned at their local times, the entire system can be shifted into daylight savings time. This is accomplished by appropriate setting of the daylight_savings in the STT. The structure of daylight savings time control is shown in Table A.1, and the basic use of daylight savings fields through the year is shown in Table A.2.

Table A.1 Structure of Daylight Savings Time Control

Syntax	Bits	Format
daylight_savings () {		
DS_status	1	bslbf
reserved	2	'11'
DS_day_of_month	5	uimsbf
DS_hour	8	uimsbf
}		

DS_status — This bit indicate the status of daylight savings.

DS_status = '0': Not in daylight savings time.

DS_status = '1': In daylight savings time.

DS_day_of_month — This 5-bit unsigned integer field indicates the local day of the month on which the transition into or out of daylight savings time is to occur (1-31).

DS_hour — This 8-bit unsigned integer field indicates the local hour at which the transition into or out of daylight savings time is to occur (0-18). This usually occurs at 2 a.m. in the U.S.

Table A.2 Basic Use of Daylight Savings Fields Through the Year

Conditions	DS status	DS_day of_month	DS_hour
At the beginning of the year (January) daylight savings is off. This is the status of the fields until:	0	0	0
• When the transition into daylight savings time is within less than one month, the DS_day_of_month field takes the value day_in, and the DS_hour field takes the value hour_in. The DS_status bit is 0 indicating it is not yet daylight savings time. (The transition is to occur on the day_in day of the month at hour=hour_in; for example, if the transition were on April 15 at 2 a.m., then day_in=15 and hour_in=2)	0	day_in	hour_in
• After all time zone daylight transitions (within the span of the network) have occurred, the DS_status bit takes the value 1, indicating that daylight savings time is on. The DS_day_of_month field and the DS_hour field take the value 0. (In the U.S., this transition has to occur no later than 7 p.m. Pacific Time on the day day_in). This is the status of the fields until:	1	0	0
When the transition out of daylight savings time is within less than one month, the DS_day_of_month field takes the value day_out, and the DS_hour field takes the value hour_out. The DS_status bit is 1 indicating it is still daylight savings time. (The transition is to occur on the day_out day of the month at hour=hour_out; for example, if the transition were on October 27 at 2 a.m., then day_out=27 and hour_out=2)	1	day_out	hour_out
• After all time zones (within the span of the network) have shifted out of daylight savings time, the DS_status bit takes the value 0, indicating that daylight savings time is off. The DS_day_of_month field and the DS_hour field take the value 0. (In the U.S., this transition has to occur no later than 7 p.m. Pacific Time on the day day_out). This finishes the cycle.	0	0	0

ANNEX B

(Normative)

ASSIGNMENT OF MAJOR CHANNEL NUMBER VALUES FOR TERRESTRIAL BROADCAST IN THE U.S.

The assignmentn of major_channel_number values in the U.S. is based on the rules below.

- For broadcasters with existing NTSC licenses, the major_channel_number for the existing NTSC channels, as well as the Digital TV channels, controlled by the broadcaster, shall be set to the current NTSC RF channel number. E.g. Assume a broadcaster who has an NTSC broadcast license for RF channel 13 is assigned RF channel 39 for Digital ATSC broadcast. That broadcaster will use major_channel_number 13 for identification of the analog NTSC channel on RF channel 13, as well as the digital channels it is controlling on RF channel 39.
- For a new broadcaster without an existing NTSC license, the major_channel_number for the Digital TV channels controlled by the broadcaster shall be set to the FCC assigned RF channel number for ATSC Digital TV broadcast. E.g. Assume a broadcaster who currently has no NTSC broadcast license applies and receives a license for Digital ATSC broadcast on RF channel 49. That broadcaster will use major_channel_number 49 for identification of the digital channels that it is controlling on RF channel 49.
- The two provisions above assign major_channel_number values 2 through 69 uniquely to broadcasters with license to broadcast NTSC and/or Digital ATSC signals.
- Values for major_channel_number from 70 to 99 may be used to identify groups of digital services carried in an ATSC multiplex that the broadcaster wishes to be identified by a different major channel number. Values 70 through 99 must be unique in each potential receiving location or the receiver will not be able to correctly select such services. For example a local broadcaster transmitting community college lectures in its bit stream may want to use a major_channel_number different than its own major_channel_number for the virtual channel carrying the lectures. The assessment of the feasibility of using this capability, as well as the coordination process for assignment of these major_channel_number values is beyond the scope of this document.

ANNEX C

(Normative)

STANDARD HUFFMAN TABLES FOR TEXT COMPRESSION[7]

This Annex describes the compression method adopted for the transmission of English-language text strings in PSIP. The method distinguishes two types of text strings: titles and program descriptions. For each of these types, Huffman tables are defined based on 1st-order conditional probabilities. Section C.2 defines standard Huffman encode and decode tables optimized for English-language text such as that typically found in program titles. Section C.3 defines Huffman encode and decode tables optimized for English-language text such as that typically found in program descriptions. Receivers supporting the English language are expected to support decoding of text using either of these two standard Huffman compression tables.

The encode tables provide necessary and sufficient information to build the Huffman trees that need to be implemented for decoding. The decode tables described in Tables C.5 and C.7 are a particular mapping of those trees into a numerical array suitable for storage. This array can be easily implemented and used with the decoding algorithm. However, the user is free to design its own decoding tables as long as they follow the Huffman trees and rules defined in this Annex.

C1. CHARACTER SET DEFINITION

This compression method supports the full ISO/IEC 8859-1 (Latin-1) character set, although only characters in the ASCII range (character codes 1 to 127) can be compressed. The following characters have special definitions:

Table C.1 Characters with Special Definitions

Character	Value (Decimal)	Meaning
String Terminate (ASCII Null)	0	The *Terminate* character is used to terminate strings. The Terminate character is appended to the string in either compressed or uncompressed form. The first encoded character in a compressed string is encoded/decoded from the Terminate sub-tree. In other words, when encoding or decoding the first character in a compressed string, assume that the previous character was a Terminate character.
Order-1 Escape (ASCII ESC)	27	Used to escape from first-order context to uncompressed context. The character which follows the Escape character is uncompressed.

[7] Tables C.4 through C.7 are © 1997 General Instrument Corporation. Unlimited use in conjunction with this ATSC standard is granted on a royalty-free basis by General Instrument Corporation. All other rights are reserved.

C1.1 First Order Escape

The order-1 Huffman trees are *partial*, that is, codes are not defined for every possible character sequence. For example, the standard decode tables do not contain codes for the character sequence *qp*. When uncompressed text contains a character sequence which is not defined in the decode table, the order-1 escape character is used to escape back to the uncompressed context. Uncompressed symbols are coded as 8-bit ASCII (Latin I). For example, the character sequence *qpa* would be coded with *compressed q*, *compressed ESC*, *uncompressed p*, *compressed a*.

First-order escape rules for compressed strings:

- Any character which follows a first-order escape character is an uncompressed (8-bit) character. (Any character which follows an uncompressed escape character is compressed).
- Characters (128 .. 255) cannot be compressed.
- Any character which follows a character from the set (128 .. 255) is uncompressed.

C1.2 Decode Table Data Structures

Decode tables have two sections:

- **Tree Root Offset List:** Provides the table offsets, in *bytes* from the start of the decode table, for the roots of the 128 first-order decode trees. The list is contained in bytes (0 .. 255) of the decode table, and is defined by the first "for" loop in Table C.1.
- **Order-1 Decode Trees:** Each and every character in the range (0 .. 127) has a corresponding first-order decode tree. For example, if the previous character was "s", then the decoder would use the "s" first-order decode tree (decode tree #115) to decode the next character (ASCII "s" equals 115 decimal). These 128 decode trees are delimited by the second "for" loop in Table C.2.

Decode tables have the following format:

Table C.2 Decode Table Format

Syntax	Bits	Format
decode_table() {		
for (i==0; i<128; i++) {		
byte_offset_of_char_i_tree_root	16	uimsbf
}		
for (i==0; i<128; i++) {		
character_i_order_1_tree()	8*M	
}		
}		

Note that even though the ISO Latin-1 character set supports up to 256 characters, only the first 128 characters may be represented in compressed form.

C1.2.1 Tree Root Byte Offsets

byte_offset_of_character_i_tree_root—A 16-bit unsigned integer specifying the location, in bytes from the beginning of the decode table, of the root for the i^{th} character's order-1 tree.

C1.2.2 Order-1 Decode Trees

Order-1 decode trees are binary trees. The roots of the decode trees are located at the table offsets specified in the tree root offset list. The left and right children of a given node are specified as *word* offsets from the root of the tree (a *word* is equivalent to two bytes).

Decode trees have the following format:

Table C.3 Decode Tree Format

Syntax	Bits	Format
character_i_order_1_tree() {		
for (j==0; j<N; j++) {		
left_child_word_offset_or_char_leaf	8	uimsbf
right_child_word_offset_or_char_leaf	8	uimsbf
}		
}		

left_child_word_offset_or_character_leaf—An 8-bit unsigned integer number with the following interpretation: If the highest bit is cleared (i.e. bit 7 is zero), the number specifies the offset, in words, of the left child from the root of the order-1 decode tree; if the highest bit is set (bit 7 is one), the lower 7 bits give the code (e.g., in ASCII) for a leaf character.

right_child_word_offset_or_character_leaf—An 8-bit unsigned integer number with the following interpretation: If the highest bit is cleared (i.e. bit 7 is zero), the number specifies the offset, in words, of the right child from the root of the order-1 decode tree; if the highest bit is set (bit 7 is one), the lower 7 bits give the code (e.g., in ASCII) for a leaf character.

It can be seen from Table F.3 that each node (corresponding to one iteration of the for-loop) has a byte for the left child or character, and a byte for the right child or character.

Characters are *leaves* of the order-1 decode trees, and are differentiated from intermediate nodes by the byte's most significant bit. When the most significant bit is set, the byte is a character leaf. When the most significant bit is not set, the byte contains the tabular word offset of the child node.

C2. STANDARD COMPRESSION TYPE 1 ENCODE/DECODE TABLES

The following encode/decode tables are optimized for English-language program title text. These tables correspond to multiple_string_structure() with compression_type value 0x01, and a mode equal to 0xFF.

Table C.4 English-language Program Title Encode Table

Prior Symbol: 0 Symbol: 27 Code: 11001011
Prior Symbol: 0 Symbol: '$' Code: 1100101011
Prior Symbol: 0 Symbol: '2' Code: 011010010
Prior Symbol: 0 Symbol: '4' Code: 1100101010
Prior Symbol: 0 Symbol: '7' Code: 011010011
Prior Symbol: 0 Symbol: 'A' Code: 0111
Prior Symbol: 0 Symbol: 'B' Code: 1001
Prior Symbol: 0 Symbol: 'C' Code: 1011
Prior Symbol: 0 Symbol: 'D' Code: 11011
Prior Symbol: 0 Symbol: 'E' Code: 10001
Prior Symbol: 0 Symbol: 'F' Code: 11000
Prior Symbol: 0 Symbol: 'G' Code: 11100
Prior Symbol: 0 Symbol: 'H' Code: 11111
Prior Symbol: 0 Symbol: 'I' Code: 10000
Prior Symbol: 0 Symbol: 'J' Code: 01100
Prior Symbol: 0 Symbol: 'K' Code: 1100110
Prior Symbol: 0 Symbol: 'L' Code: 11101
Prior Symbol: 0 Symbol: 'M' Code: 1010
Prior Symbol: 0 Symbol: 'N' Code: 0011
Prior Symbol: 0 Symbol: 'O' Code: 011011
Prior Symbol: 0 Symbol: 'P' Code: 11110
Prior Symbol: 0 Symbol: 'Q' Code: 01101000
Prior Symbol: 0 Symbol: 'R' Code: 11010
Prior Symbol: 0 Symbol: 'S' Code: 000
Prior Symbol: 0 Symbol: 'T' Code: 010
Prior Symbol: 0 Symbol: 'U' Code: 0110101
Prior Symbol: 0 Symbol: 'V' Code: 1100111
Prior Symbol: 0 Symbol: 'W' Code: 0010
Prior Symbol: 0 Symbol: 'Y' Code: 1100100
Prior Symbol: 0 Symbol: 'Z' Code: 110010100
Prior Symbol: 1 Symbol: 27 Code: 1
Prior Symbol: 2 Symbol: 27 Code: 1
Prior Symbol: 3 Symbol: 27 Code: 1
Prior Symbol: 4 Symbol: 27 Code: 1
Prior Symbol: 5 Symbol: 27 Code: 1
Prior Symbol: 6 Symbol: 27 Code: 1
Prior Symbol: 7 Symbol: 27 Code: 1
Prior Symbol: 8 Symbol: 27 Code: 1
Prior Symbol: 9 Symbol: 27 Code: 1
Prior Symbol: 10 Symbol: 27 Code: 1
Prior Symbol: 11 Symbol: 27 Code: 1
Prior Symbol: 12 Symbol: 27 Code: 1
Prior Symbol: 13 Symbol: 27 Code: 1
Prior Symbol: 14 Symbol: 27 Code: 1
Prior Symbol: 15 Symbol: 27 Code: 1
Prior Symbol: 16 Symbol: 27 Code: 1
Prior Symbol: 17 Symbol: 27 Code: 1
Prior Symbol: 18 Symbol: 27 Code: 1
Prior Symbol: 19 Symbol: 27 Code: 1
Prior Symbol: 20 Symbol: 27 Code: 1
Prior Symbol: 21 Symbol: 27 Code: 1
Prior Symbol: 22 Symbol: 27 Code: 1
Prior Symbol: 23 Symbol: 27 Code: 1
Prior Symbol: 24 Symbol: 27 Code: 1
Prior Symbol: 25 Symbol: 27 Code: 1
Prior Symbol: 26 Symbol: 27 Code: 1
Prior Symbol: 27 Symbol: 27 Code: 1
Prior Symbol: 28 Symbol: 27 Code: 1
Prior Symbol: 29 Symbol: 27 Code: 1
Prior Symbol: 30 Symbol: 27 Code: 1
Prior Symbol: 31 Symbol: 27 Code: 1
Prior Symbol: ' ' Symbol: 27 Code: 10010100
Prior Symbol: ' ' Symbol: '&' Code: 010001
Prior Symbol: ' ' Symbol: ''' Code: 010000100
Prior Symbol: ' ' Symbol: '-' Code: 00000001
Prior Symbol: ' ' Symbol: '1' Code: 010000101
Prior Symbol: ' ' Symbol: '2' Code: 00000010
Prior Symbol: ' ' Symbol: '3' Code: 01000001
Prior Symbol: ' ' Symbol: '9' Code: 000000000
Prior Symbol: ' ' Symbol: 'A' Code: 10111
Prior Symbol: ' ' Symbol: 'B' Code: 0010
Prior Symbol: ' ' Symbol: 'C' Code: 1100
Prior Symbol: ' ' Symbol: 'D' Code: 11100
Prior Symbol: ' ' Symbol: 'E' Code: 011010
Prior Symbol: ' ' Symbol: 'F' Code: 10011
Prior Symbol: ' ' Symbol: 'G' Code: 00001
Prior Symbol: ' ' Symbol: 'H' Code: 10101
Prior Symbol: ' ' Symbol: 'I' Code: 111111
Prior Symbol: ' ' Symbol: 'J' Code: 111110
Prior Symbol: ' ' Symbol: 'K' Code: 010011
Prior Symbol: ' ' Symbol: 'L' Code: 11110
Prior Symbol: ' ' Symbol: 'M' Code: 0101
Prior Symbol: ' ' Symbol: 'N' Code: 10110
Prior Symbol: ' ' Symbol: 'O' Code: 011011
Prior Symbol: ' ' Symbol: 'P' Code: 11101
Prior Symbol: ' ' Symbol: 'Q' Code: 100100011
Prior Symbol: ' ' Symbol: 'R' Code: 10100
Prior Symbol: ' ' Symbol: 'S' Code: 1101
Prior Symbol: ' ' Symbol: 'T' Code: 1000
Prior Symbol: ' ' Symbol: 'U' Code: 1001001
Prior Symbol: ' ' Symbol: 'V' Code: 1001011
Prior Symbol: ' ' Symbol: 'W' Code: 0011
Prior Symbol: ' ' Symbol: 'X' Code: 0000000010
Prior Symbol: ' ' Symbol: 'Y' Code: 000001
Prior Symbol: ' ' Symbol: 'Z' Code: 00000011
Prior Symbol: ' ' Symbol: 'a' Code: 01100
Prior Symbol: ' ' Symbol: 'b' Code: 10010101
Prior Symbol: ' ' Symbol: 'c' Code: 01000000
Prior Symbol: ' ' Symbol: 'd' Code: 01000011
Prior Symbol: ' ' Symbol: 'e' Code: 0000000011
Prior Symbol: ' ' Symbol: 'f' Code: 10010000
Prior Symbol: ' ' Symbol: 'i' Code: 010010
Prior Symbol: ' ' Symbol: 'l' Code: 100100010
Prior Symbol: ' ' Symbol: 'o' Code: 0001
Prior Symbol: ' ' Symbol: 't' Code: 0111
Prior Symbol: '!' Symbol: 0 Code: 1
Prior Symbol: '!' Symbol: 27 Code: 01
Prior Symbol: '!' Symbol: ' ' Code: 00
Prior Symbol: '"' Symbol: 27 Code: 1
Prior Symbol: '#' Symbol: 27 Code: 1
Prior Symbol: '$' Symbol: 27 Code: 1
Prior Symbol: '$' Symbol: '1' Code: 0
Prior Symbol: '%' Symbol: 27 Code: 1
Prior Symbol: '&' Symbol: 27 Code: 0
Prior Symbol: '&' Symbol: ' ' Code: 1
Prior Symbol: ''' Symbol: 27 Code: 011
Prior Symbol: ''' Symbol: ' ' Code: 010
Prior Symbol: ''' Symbol: '9' Code: 0001
Prior Symbol: ''' Symbol: 'd' Code: 0000
Prior Symbol: ''' Symbol: 's' Code: 1
Prior Symbol: ''' Symbol: 't' Code: 001
Prior Symbol: '(' Symbol: 27 Code: 1
Prior Symbol: ')' Symbol: 27 Code: 1
Prior Symbol: '*' Symbol: 27 Code: 00
Prior Symbol: '*' Symbol: 'A' Code: 01
Prior Symbol: '*' Symbol: 'H' Code: 10
Prior Symbol: '*' Symbol: 'S' Code: 11
Prior Symbol: '+' Symbol: 27 Code: 1
Prior Symbol: ',' Symbol: 27 Code: 0
Prior Symbol: ',' Symbol: ' ' Code: 1
Prior Symbol: '-' Symbol: 27 Code: 01
Prior Symbol: '-' Symbol: ' ' Code: 111
Prior Symbol: '-' Symbol: '-' Code: 1101
Prior Symbol: '-' Symbol: '1' Code: 1000
Prior Symbol: '-' Symbol: 'A' Code: 001
Prior Symbol: '-' Symbol: 'M' Code: 000
Prior Symbol: '-' Symbol: 'R' Code: 1001
Prior Symbol: '-' Symbol: 'S' Code: 1010
Prior Symbol: '-' Symbol: 'T' Code: 1011
Prior Symbol: '-' Symbol: 'U' Code: 1100
Prior Symbol: '.' Symbol: 0 Code: 111
Prior Symbol: '.' Symbol: 27 Code: 101
Prior Symbol: '.' Symbol: ' ' Code: 0
Prior Symbol: '.' Symbol: '.' Code: 110
Prior Symbol: '.' Symbol: 'I' Code: 10010
Prior Symbol: '.' Symbol: 'S' Code: 1000
Prior Symbol: '.' Symbol: 'W' Code: 10011
Prior Symbol: '/' Symbol: 27 Code: 1
Prior Symbol: '0' Symbol: 0 Code: 01
Prior Symbol: '0' Symbol: 27 Code: 001
Prior Symbol: '0' Symbol: ' ' Code: 10
Prior Symbol: '0' Symbol: '-' Code: 000
Prior Symbol: '0' Symbol: '0' Code: 11
Prior Symbol: '1' Symbol: 0 Code: 010
Prior Symbol: '1' Symbol: 27 Code: 011
Prior Symbol: '1' Symbol: ' ' Code: 110
Prior Symbol: '1' Symbol: '0' Code: 111
Prior Symbol: '1' Symbol: '1' Code: 100
Prior Symbol: '1' Symbol: '2' Code: 101
Prior Symbol: '1' Symbol: '9' Code: 00
Prior Symbol: '2' Symbol: 0 Code: 11
Prior Symbol: '2' Symbol: 27 Code: 10
Prior Symbol: '2' Symbol: '0' Code: 01
Prior Symbol: '2' Symbol: '1' Code: 000
Prior Symbol: '2' Symbol: ':' Code: 001
Prior Symbol: '3' Symbol: 0 Code: 0
Prior Symbol: '3' Symbol: 27 Code: 11
Prior Symbol: '3' Symbol: '0' Code: 10
Prior Symbol: '4' Symbol: 27 Code: 0
Prior Symbol: '4' Symbol: '8' Code: 1
Prior Symbol: '5' Symbol: 27 Code: 1
Prior Symbol: '6' Symbol: 27 Code: 1
Prior Symbol: '7' Symbol: 27 Code: 0
Prior Symbol: '7' Symbol: '0' Code: 1
Prior Symbol: '8' Symbol: 27 Code: 0
Prior Symbol: '8' Symbol: ' ' Code: 1
Prior Symbol: '9' Symbol: 27 Code: 11
Prior Symbol: '9' Symbol: '0' Code: 01
Prior Symbol: '9' Symbol: '1' Code: 100
Prior Symbol: '9' Symbol: '3' Code: 101
Prior Symbol: '9' Symbol: '9' Code: 00
Prior Symbol: ':' Symbol: 27 Code: 0
Prior Symbol: ':' Symbol: ' ' Code: 1
Prior Symbol: ';' Symbol: 27 Code: 1
Prior Symbol: '<' Symbol: 27 Code: 1
Prior Symbol: '=' Symbol: 27 Code: 1
Prior Symbol: '>' Symbol: 27 Code: 1
Prior Symbol: '?' Symbol: 0 Code: 1
Prior Symbol: '?' Symbol: 27 Code: 0
Prior Symbol: '@' Symbol: 27 Code: 1
Prior Symbol: 'A' Symbol: 27 Code: 00010
Prior Symbol: 'A' Symbol: ' ' Code: 010
Prior Symbol: 'A' Symbol: '*' Code: 1101000
Prior Symbol: 'A' Symbol: '-' Code: 1101001
Prior Symbol: 'A' Symbol: '.' Code: 1101010
Prior Symbol: 'A' Symbol: 'B' Code: 110110
Prior Symbol: 'A' Symbol: 'b' Code: 110010
Prior Symbol: 'A' Symbol: 'c' Code: 01100

Prior Symbol: 'A' Symbol: 'd' Code: 001
Prior Symbol: 'A' Symbol: 'f' Code: 01101
Prior Symbol: 'A' Symbol: 'g' Code: 011110
Prior Symbol: 'A' Symbol: 'i' Code: 110011
Prior Symbol: 'A' Symbol: 'l' Code: 100
Prior Symbol: 'A' Symbol: 'm' Code: 111
Prior Symbol: 'A' Symbol: 'n' Code: 101
Prior Symbol: 'A' Symbol: 'p' Code: 110111
Prior Symbol: 'A' Symbol: 'r' Code: 0000
Prior Symbol: 'A' Symbol: 's' Code: 00011
Prior Symbol: 'A' Symbol: 't' Code: 011111
Prior Symbol: 'A' Symbol: 'u' Code: 11000
Prior Symbol: 'A' Symbol: 'v' Code: 1101011
Prior Symbol: 'A' Symbol: 'w' Code: 01110
Prior Symbol: 'B' Symbol: 27 Code: 00010
Prior Symbol: 'B' Symbol: 'A' Code: 000110
Prior Symbol: 'B' Symbol: 'C' Code: 0000
Prior Symbol: 'B' Symbol: 'S' Code: 000111
Prior Symbol: 'B' Symbol: 'a' Code: 111
Prior Symbol: 'B' Symbol: 'e' Code: 01
Prior Symbol: 'B' Symbol: 'i' Code: 1010
Prior Symbol: 'B' Symbol: 'l' Code: 1011
Prior Symbol: 'B' Symbol: 'o' Code: 110
Prior Symbol: 'B' Symbol: 'r' Code: 001
Prior Symbol: 'B' Symbol: 'u' Code: 100
Prior Symbol: 'C' Symbol: 27 Code: 00101
Prior Symbol: 'C' Symbol: ' ' Code: 10110
Prior Symbol: 'C' Symbol: 'A' Code: 0011100
Prior Symbol: 'C' Symbol: 'B' Code: 001111
Prior Symbol: 'C' Symbol: 'O' Code: 101110
Prior Symbol: 'C' Symbol: 'a' Code: 100
Prior Symbol: 'C' Symbol: 'e' Code: 101111
Prior Symbol: 'C' Symbol: 'h' Code: 01
Prior Symbol: 'C' Symbol: 'i' Code: 00110
Prior Symbol: 'C' Symbol: 'l' Code: 000
Prior Symbol: 'C' Symbol: 'o' Code: 11
Prior Symbol: 'C' Symbol: 'r' Code: 1010
Prior Symbol: 'C' Symbol: 'u' Code: 00100
Prior Symbol: 'C' Symbol: 'y' Code: 0011101
Prior Symbol: 'D' Symbol: 27 Code: 01001
Prior Symbol: 'D' Symbol: 'a' Code: 10
Prior Symbol: 'D' Symbol: 'e' Code: 111
Prior Symbol: 'D' Symbol: 'i' Code: 110
Prior Symbol: 'D' Symbol: 'o' Code: 00
Prior Symbol: 'D' Symbol: 'r' Code: 011
Prior Symbol: 'D' Symbol: 'u' Code: 0101
Prior Symbol: 'D' Symbol: 'y' Code: 01000
Prior Symbol: 'E' Symbol: 27 Code: 011
Prior Symbol: 'E' Symbol: 'C' Code: 1010
Prior Symbol: 'E' Symbol: 'a' Code: 111
Prior Symbol: 'E' Symbol: 'd' Code: 000
Prior Symbol: 'E' Symbol: 'l' Code: 1100
Prior Symbol: 'E' Symbol: 'm' Code: 0100
Prior Symbol: 'E' Symbol: 'n' Code: 1101
Prior Symbol: 'E' Symbol: 'q' Code: 101110
Prior Symbol: 'E' Symbol: 's' Code: 10110
Prior Symbol: 'E' Symbol: 'u' Code: 101111
Prior Symbol: 'E' Symbol: 'v' Code: 100
Prior Symbol: 'E' Symbol: 'x' Code: 001
Prior Symbol: 'E' Symbol: 'y' Code: 0101
Prior Symbol: 'F' Symbol: 27 Code: 011111
Prior Symbol: 'F' Symbol: ' ' Code: 011110
Prior Symbol: 'F' Symbol: 'L' Code: 01110
Prior Symbol: 'F' Symbol: 'a' Code: 10
Prior Symbol: 'F' Symbol: 'e' Code: 0110
Prior Symbol: 'F' Symbol: 'i' Code: 110
Prior Symbol: 'F' Symbol: 'l' Code: 000
Prior Symbol: 'F' Symbol: 'o' Code: 010
Prior Symbol: 'F' Symbol: 'r' Code: 111
Prior Symbol: 'F' Symbol: 'u' Code: 001
Prior Symbol: 'G' Symbol: 27 Code: 10110
Prior Symbol: 'G' Symbol: '.' Code: 101010
Prior Symbol: 'G' Symbol: 'A' Code: 101111
Prior Symbol: 'G' Symbol: 'a' Code: 1110
Prior Symbol: 'G' Symbol: 'e' Code: 110
Prior Symbol: 'G' Symbol: 'h' Code: 10100
Prior Symbol: 'G' Symbol: 'i' Code: 100
Prior Symbol: 'G' Symbol: 'l' Code: 101011
Prior Symbol: 'G' Symbol: 'o' Code: 01
Prior Symbol: 'G' Symbol: 'r' Code: 00
Prior Symbol: 'G' Symbol: 'u' Code: 1111
Prior Symbol: 'G' Symbol: 'y' Code: 101110
Prior Symbol: 'H' Symbol: 0 Code: 111010
Prior Symbol: 'H' Symbol: 27 Code: 111011
Prior Symbol: 'H' Symbol: 'a' Code: 110
Prior Symbol: 'H' Symbol: 'e' Code: 10
Prior Symbol: 'H' Symbol: 'i' Code: 1111
Prior Symbol: 'H' Symbol: 'o' Code: 0
Prior Symbol: 'H' Symbol: 'u' Code: 11100
Prior Symbol: 'I' Symbol: 0 Code: 1000
Prior Symbol: 'I' Symbol: 27 Code: 1001
Prior Symbol: 'I' Symbol: ' ' Code: 11110
Prior Symbol: 'I' Symbol: '.' Code: 111110
Prior Symbol: 'I' Symbol: ':' Code: 101110
Prior Symbol: 'I' Symbol: 'I' Code: 1100
Prior Symbol: 'I' Symbol: 'T' Code: 101111
Prior Symbol: 'I' Symbol: 'c' Code: 10110
Prior Symbol: 'I' Symbol: 'm' Code: 1010
Prior Symbol: 'I' Symbol: 'n' Code: 0
Prior Symbol: 'I' Symbol: 'r' Code: 111111
Prior Symbol: 'I' Symbol: 's' Code: 1101
Prior Symbol: 'I' Symbol: 't' Code: 1110
Prior Symbol: 'J' Symbol: 27 Code: 000
Prior Symbol: 'J' Symbol: 'a' Code: 01
Prior Symbol: 'J' Symbol: 'e' Code: 11
Prior Symbol: 'J' Symbol: 'o' Code: 10
Prior Symbol: 'J' Symbol: 'u' Code: 001
Prior Symbol: 'K' Symbol: 27 Code: 000
Prior Symbol: 'K' Symbol: 'a' Code: 0100
Prior Symbol: 'K' Symbol: 'e' Code: 001
Prior Symbol: 'K' Symbol: 'i' Code: 1
Prior Symbol: 'K' Symbol: 'n' Code: 0111
Prior Symbol: 'K' Symbol: 'o' Code: 0101
Prior Symbol: 'K' Symbol: 'u' Code: 0110
Prior Symbol: 'L' Symbol: 27 Code: 01001
Prior Symbol: 'L' Symbol: ' ' Code: 01000
Prior Symbol: 'L' Symbol: 'a' Code: 10
Prior Symbol: 'L' Symbol: 'e' Code: 011
Prior Symbol: 'L' Symbol: 'i' Code: 11
Prior Symbol: 'L' Symbol: 'o' Code: 00
Prior Symbol: 'L' Symbol: 'u' Code: 0101
Prior Symbol: 'M' Symbol: 27 Code: 1011111
Prior Symbol: 'M' Symbol: '"' Code: 10111100
Prior Symbol: 'M' Symbol: 'T' Code: 10111101
Prior Symbol: 'M' Symbol: 'a' Code: 11
Prior Symbol: 'M' Symbol: 'c' Code: 101110
Prior Symbol: 'M' Symbol: 'e' Code: 1010
Prior Symbol: 'M' Symbol: 'i' Code: 100
Prior Symbol: 'M' Symbol: 'o' Code: 00
Prior Symbol: 'M' Symbol: 'r' Code: 10110
Prior Symbol: 'M' Symbol: 'u' Code: 010
Prior Symbol: 'M' Symbol: 'y' Code: 011
Prior Symbol: 'N' Symbol: 27 Code: 1000
Prior Symbol: 'N' Symbol: ' ' Code: 110001
Prior Symbol: 'N' Symbol: 'B' Code: 1001
Prior Symbol: 'N' Symbol: 'F' Code: 110010
Prior Symbol: 'N' Symbol: 'N' Code: 110000
Prior Symbol: 'N' Symbol: 'a' Code: 1101
Prior Symbol: 'N' Symbol: 'e' Code: 0
Prior Symbol: 'N' Symbol: 'i' Code: 111
Prior Symbol: 'N' Symbol: 'o' Code: 101
Prior Symbol: 'N' Symbol: 'u' Code: 110011
Prior Symbol: 'O' Symbol: 27 Code: 010
Prior Symbol: 'O' Symbol: ' ' Code: 001
Prior Symbol: 'O' Symbol: 'd' Code: 01110
Prior Symbol: 'O' Symbol: 'f' Code: 11010
Prior Symbol: 'O' Symbol: 'l' Code: 1100
Prior Symbol: 'O' Symbol: 'n' Code: 10
Prior Symbol: 'O' Symbol: 'p' Code: 0001
Prior Symbol: 'O' Symbol: 'r' Code: 0110
Prior Symbol: 'O' Symbol: 's' Code: 01111
Prior Symbol: 'O' Symbol: 'u' Code: 111
Prior Symbol: 'O' Symbol: 'v' Code: 11011
Prior Symbol: 'O' Symbol: 'w' Code: 0000
Prior Symbol: 'P' Symbol: 27 Code: 111111
Prior Symbol: 'P' Symbol: ' ' Code: 1111100
Prior Symbol: 'P' Symbol: '.' Code: 011001
Prior Symbol: 'P' Symbol: 'G' Code: 111101
Prior Symbol: 'P' Symbol: 'R' Code: 111100
Prior Symbol: 'P' Symbol: 'a' Code: 00
Prior Symbol: 'P' Symbol: 'e' Code: 010
Prior Symbol: 'P' Symbol: 'i' Code: 0111
Prior Symbol: 'P' Symbol: 'l' Code: 1110
Prior Symbol: 'P' Symbol: 'o' Code: 110
Prior Symbol: 'P' Symbol: 'r' Code: 10
Prior Symbol: 'P' Symbol: 's' Code: 1111101
Prior Symbol: 'P' Symbol: 'u' Code: 01101
Prior Symbol: 'P' Symbol: 'y' Code: 011000
Prior Symbol: 'Q' Symbol: 27 Code: 00
Prior Symbol: 'Q' Symbol: 'V' Code: 01
Prior Symbol: 'Q' Symbol: 'u' Code: 1
Prior Symbol: 'R' Symbol: 27 Code: 10001
Prior Symbol: 'R' Symbol: 'a' Code: 101
Prior Symbol: 'R' Symbol: 'e' Code: 11
Prior Symbol: 'R' Symbol: 'h' Code: 10000
Prior Symbol: 'R' Symbol: 'i' Code: 00
Prior Symbol: 'R' Symbol: 'o' Code: 01
Prior Symbol: 'R' Symbol: 'u' Code: 1001
Prior Symbol: 'S' Symbol: 27 Code: 101110
Prior Symbol: 'S' Symbol: ' ' Code: 1110100
Prior Symbol: 'S' Symbol: '"' Code: 1011000
Prior Symbol: 'S' Symbol: '.' Code: 1011011
Prior Symbol: 'S' Symbol: 'a' Code: 1111
Prior Symbol: 'S' Symbol: 'c' Code: 11100
Prior Symbol: 'S' Symbol: 'e' Code: 000
Prior Symbol: 'S' Symbol: 'h' Code: 100
Prior Symbol: 'S' Symbol: 'i' Code: 1100
Prior Symbol: 'S' Symbol: 'k' Code: 101111
Prior Symbol: 'S' Symbol: 'l' Code: 1011001
Prior Symbol: 'S' Symbol: 'm' Code: 1110110
Prior Symbol: 'S' Symbol: 'n' Code: 1110111
Prior Symbol: 'S' Symbol: 'o' Code: 1010
Prior Symbol: 'S' Symbol: 'p' Code: 001
Prior Symbol: 'S' Symbol: 'q' Code: 1011010
Prior Symbol: 'S' Symbol: 't' Code: 01
Prior Symbol: 'S' Symbol: 'u' Code: 1101
Prior Symbol: 'S' Symbol: 'w' Code: 1110101
Prior Symbol: 'T' Symbol: 27 Code: 1111010
Prior Symbol: 'T' Symbol: '-' Code: 11110110
Prior Symbol: 'T' Symbol: 'N' Code: 11110111
Prior Symbol: 'T' Symbol: 'V' Code: 111100
Prior Symbol: 'T' Symbol: 'a' Code: 1010
Prior Symbol: 'T' Symbol: 'e' Code: 1011
Prior Symbol: 'T' Symbol: 'h' Code: 0
Prior Symbol: 'T' Symbol: 'i' Code: 1110
Prior Symbol: 'T' Symbol: 'o' Code: 110
Prior Symbol: 'T' Symbol: 'r' Code: 100
Prior Symbol: 'T' Symbol: 'u' Code: 111110
Prior Symbol: 'T' Symbol: 'w' Code: 111111
Prior Symbol: 'U' Symbol: 27 Code: 101
Prior Symbol: 'U' Symbol: '.' Code: 1001
Prior Symbol: 'U' Symbol: 'l' Code: 1000
Prior Symbol: 'U' Symbol: 'n' Code: 0
Prior Symbol: 'U' Symbol: 'p' Code: 11
Prior Symbol: 'V' Symbol: 0 Code: 000
Prior Symbol: 'V' Symbol: 27 Code: 0011
Prior Symbol: 'V' Symbol: ' ' Code: 01010
Prior Symbol: 'V' Symbol: 'C' Code: 01011
Prior Symbol: 'V' Symbol: 'a' Code: 011
Prior Symbol: 'V' Symbol: 'e' Code: 0100
Prior Symbol: 'V' Symbol: 'i' Code: 1
Prior Symbol: 'V' Symbol: 'o' Code: 0010
Prior Symbol: 'W' Symbol: 27 Code: 00011
Prior Symbol: 'W' Symbol: 'F' Code: 000100
Prior Symbol: 'W' Symbol: 'W' Code: 000101
Prior Symbol: 'W' Symbol: 'a' Code: 111
Prior Symbol: 'W' Symbol: 'e' Code: 110
Prior Symbol: 'W' Symbol: 'h' Code: 001
Prior Symbol: 'W' Symbol: 'i' Code: 01
Prior Symbol: 'W' Symbol: 'o' Code: 10
Prior Symbol: 'W' Symbol: 'r' Code: 0000
Prior Symbol: 'X' Symbol: 27 Code: 1
Prior Symbol: 'Y' Symbol: 27 Code: 001
Prior Symbol: 'Y' Symbol: 'a' Code: 000
Prior Symbol: 'Y' Symbol: 'e' Code: 01
Prior Symbol: 'Y' Symbol: 'o' Code: 1
Prior Symbol: 'Z' Symbol: 27 Code: 00
Prior Symbol: 'Z' Symbol: 'a' Code: 01
Prior Symbol: 'Z' Symbol: 'o' Code: 1
Prior Symbol: '[' Symbol: 27 Code: 1
Prior Symbol: '\' Symbol: 27 Code: 1
Prior Symbol: ']' Symbol: 27 Code: 1
Prior Symbol: '^' Symbol: 27 Code: 1

Prior Symbol: '_' Symbol: 27 Code: 1
Prior Symbol: '`' Symbol: 27 Code: 1
Prior Symbol: 'a' Symbol: 0 Code: 00010
Prior Symbol: 'a' Symbol: 27 Code: 1111010110
Prior Symbol: 'a' Symbol: ' ' Code: 10110
Prior Symbol: 'a' Symbol: '"' Code: 11110100
Prior Symbol: 'a' Symbol: ':' Code: 1111010111
Prior Symbol: 'a' Symbol: 'b' Code: 010010
Prior Symbol: 'a' Symbol: 'c' Code: 11111
Prior Symbol: 'a' Symbol: 'd' Code: 10100
Prior Symbol: 'a' Symbol: 'e' Code: 101011000
Prior Symbol: 'a' Symbol: 'f' Code: 10101101
Prior Symbol: 'a' Symbol: 'g' Code: 01000
Prior Symbol: 'a' Symbol: 'h' Code: 0100111
Prior Symbol: 'a' Symbol: 'i' Code: 10111
Prior Symbol: 'a' Symbol: 'j' Code: 101011001
Prior Symbol: 'a' Symbol: 'k' Code: 101010
Prior Symbol: 'a' Symbol: 'l' Code: 001
Prior Symbol: 'a' Symbol: 'm' Code: 0101
Prior Symbol: 'a' Symbol: 'n' Code: 110
Prior Symbol: 'a' Symbol: 'p' Code: 111100
Prior Symbol: 'a' Symbol: 'r' Code: 100
Prior Symbol: 'a' Symbol: 's' Code: 1110
Prior Symbol: 'a' Symbol: 't' Code: 011
Prior Symbol: 'a' Symbol: 'u' Code: 1111011
Prior Symbol: 'a' Symbol: 'v' Code: 00011
Prior Symbol: 'a' Symbol: 'w' Code: 1010111
Prior Symbol: 'a' Symbol: 'x' Code: 111101010
Prior Symbol: 'a' Symbol: 'y' Code: 0000
Prior Symbol: 'a' Symbol: 'z' Code: 0100110
Prior Symbol: 'b' Symbol: 0 Code: 11111
Prior Symbol: 'b' Symbol: 27 Code: 111101
Prior Symbol: 'b' Symbol: ' ' Code: 0110
Prior Symbol: 'b' Symbol: 'a' Code: 00
Prior Symbol: 'b' Symbol: 'b' Code: 01111
Prior Symbol: 'b' Symbol: 'e' Code: 1010
Prior Symbol: 'b' Symbol: 'i' Code: 1110
Prior Symbol: 'b' Symbol: 'l' Code: 010
Prior Symbol: 'b' Symbol: 'o' Code: 110
Prior Symbol: 'b' Symbol: 'r' Code: 1011
Prior Symbol: 'b' Symbol: 's' Code: 111100
Prior Symbol: 'b' Symbol: 'u' Code: 01110
Prior Symbol: 'b' Symbol: 'y' Code: 100
Prior Symbol: 'c' Symbol: 0 Code: 010110
Prior Symbol: 'c' Symbol: 27 Code: 1000011
Prior Symbol: 'c' Symbol: ' ' Code: 0100
Prior Symbol: 'c' Symbol: 'C' Code: 0010110
Prior Symbol: 'c' Symbol: 'G' Code: 1000010
Prior Symbol: 'c' Symbol: 'L' Code: 0010111
Prior Symbol: 'c' Symbol: 'a' Code: 011
Prior Symbol: 'c' Symbol: 'c' Code: 001010
Prior Symbol: 'c' Symbol: 'e' Code: 111
Prior Symbol: 'c' Symbol: 'h' Code: 101
Prior Symbol: 'c' Symbol: 'i' Code: 0011
Prior Symbol: 'c' Symbol: 'k' Code: 110
Prior Symbol: 'c' Symbol: 'l' Code: 010111
Prior Symbol: 'c' Symbol: 'o' Code: 1001
Prior Symbol: 'c' Symbol: 'r' Code: 10001
Prior Symbol: 'c' Symbol: 's' Code: 00100
Prior Symbol: 'c' Symbol: 't' Code: 000
Prior Symbol: 'c' Symbol: 'u' Code: 01010
Prior Symbol: 'c' Symbol: 'y' Code: 100000
Prior Symbol: 'd' Symbol: 0 Code: 011
Prior Symbol: 'd' Symbol: 27 Code: 101110
Prior Symbol: 'd' Symbol: ' ' Code: 11
Prior Symbol: 'd' Symbol: '.' Code: 101101110
Prior Symbol: 'd' Symbol: 'a' Code: 1010
Prior Symbol: 'd' Symbol: 'd' Code: 100000
Prior Symbol: 'd' Symbol: 'e' Code: 00
Prior Symbol: 'd' Symbol: 'g' Code: 100001
Prior Symbol: 'd' Symbol: 'i' Code: 1001
Prior Symbol: 'd' Symbol: 'l' Code: 1011010
Prior Symbol: 'd' Symbol: 'o' Code: 101111
Prior Symbol: 'd' Symbol: 'r' Code: 101100
Prior Symbol: 'd' Symbol: 's' Code: 0101
Prior Symbol: 'd' Symbol: 'u' Code: 101101111
Prior Symbol: 'd' Symbol: 'v' Code: 10001
Prior Symbol: 'd' Symbol: 'w' Code: 10110110
Prior Symbol: 'd' Symbol: 'y' Code: 0100
Prior Symbol: 'e' Symbol: 0 Code: 001
Prior Symbol: 'e' Symbol: 27 Code: 1010111100
Prior Symbol: 'e' Symbol: ' ' Code: 01
Prior Symbol: 'e' Symbol: '!' Code: 1010111101
Prior Symbol: 'e' Symbol: '"' Code: 10101100
Prior Symbol: 'e' Symbol: '-' Code: 1010111110
Prior Symbol: 'e' Symbol: ':' Code: 00010010
Prior Symbol: 'e' Symbol: 'a' Code: 1000
Prior Symbol: 'e' Symbol: 'b' Code: 10101101
Prior Symbol: 'e' Symbol: 'c' Code: 100111
Prior Symbol: 'e' Symbol: 'd' Code: 00011
Prior Symbol: 'e' Symbol: 'e' Code: 10100
Prior Symbol: 'e' Symbol: 'f' Code: 1001100
Prior Symbol: 'e' Symbol: 'g' Code: 1010100
Prior Symbol: 'e' Symbol: 'h' Code: 1010111111
Prior Symbol: 'e' Symbol: 'i' Code: 10101110
Prior Symbol: 'e' Symbol: 'j' Code: 000100000
Prior Symbol: 'e' Symbol: 'k' Code: 1010101
Prior Symbol: 'e' Symbol: 'l' Code: 10010
Prior Symbol: 'e' Symbol: 'm' Code: 1001101
Prior Symbol: 'e' Symbol: 'n' Code: 1110
Prior Symbol: 'e' Symbol: 'o' Code: 000101
Prior Symbol: 'e' Symbol: 'p' Code: 000001
Prior Symbol: 'e' Symbol: 'q' Code: 000100001
Prior Symbol: 'e' Symbol: 'r' Code: 110
Prior Symbol: 'e' Symbol: 's' Code: 1111
Prior Symbol: 'e' Symbol: 't' Code: 10110
Prior Symbol: 'e' Symbol: 'u' Code: 000100010
Prior Symbol: 'e' Symbol: 'v' Code: 000000
Prior Symbol: 'e' Symbol: 'w' Code: 10111
Prior Symbol: 'e' Symbol: 'x' Code: 00010011
Prior Symbol: 'e' Symbol: 'y' Code: 00001
Prior Symbol: 'e' Symbol: 'z' Code: 000100011
Prior Symbol: 'f' Symbol: 0 Code: 11100
Prior Symbol: 'f' Symbol: 27 Code: 1111001
Prior Symbol: 'f' Symbol: ' ' Code: 0
Prior Symbol: 'f' Symbol: 'a' Code: 11101
Prior Symbol: 'f' Symbol: 'e' Code: 110
Prior Symbol: 'f' Symbol: 'f' Code: 1011
Prior Symbol: 'f' Symbol: 'i' Code: 1001
Prior Symbol: 'f' Symbol: 'l' Code: 111101
Prior Symbol: 'f' Symbol: 'o' Code: 1010
Prior Symbol: 'f' Symbol: 'r' Code: 111111
Prior Symbol: 'f' Symbol: 's' Code: 111110
Prior Symbol: 'f' Symbol: 't' Code: 1000
Prior Symbol: 'f' Symbol: 'u' Code: 1111000
Prior Symbol: 'g' Symbol: 0 Code: 110
Prior Symbol: 'g' Symbol: 27 Code: 1110000
Prior Symbol: 'g' Symbol: ' ' Code: 01
Prior Symbol: 'g' Symbol: '"' Code: 1001100
Prior Symbol: 'g' Symbol: ':' Code: 11100010
Prior Symbol: 'g' Symbol: 'a' Code: 1000
Prior Symbol: 'g' Symbol: 'e' Code: 101
Prior Symbol: 'g' Symbol: 'g' Code: 1111010
Prior Symbol: 'g' Symbol: 'h' Code: 00
Prior Symbol: 'g' Symbol: 'i' Code: 11101
Prior Symbol: 'g' Symbol: 'l' Code: 1111011
Prior Symbol: 'g' Symbol: 'n' Code: 100111
Prior Symbol: 'g' Symbol: 'o' Code: 111001
Prior Symbol: 'g' Symbol: 'r' Code: 10010
Prior Symbol: 'g' Symbol: 's' Code: 11111
Prior Symbol: 'g' Symbol: 't' Code: 1001101
Prior Symbol: 'g' Symbol: 'u' Code: 111100
Prior Symbol: 'g' Symbol: 'y' Code: 11100011
Prior Symbol: 'h' Symbol: 0 Code: 11101
Prior Symbol: 'h' Symbol: 27 Code: 1110001
Prior Symbol: 'h' Symbol: ' ' Code: 1011
Prior Symbol: 'h' Symbol: 'a' Code: 1100
Prior Symbol: 'h' Symbol: 'b' Code: 11100110
Prior Symbol: 'h' Symbol: 'e' Code: 0
Prior Symbol: 'h' Symbol: 'i' Code: 100
Prior Symbol: 'h' Symbol: 'l' Code: 1110010
Prior Symbol: 'h' Symbol: 'n' Code: 101001
Prior Symbol: 'h' Symbol: 'o' Code: 1101
Prior Symbol: 'h' Symbol: 'r' Code: 10101
Prior Symbol: 'h' Symbol: 't' Code: 1111
Prior Symbol: 'h' Symbol: 'u' Code: 11100111
Prior Symbol: 'h' Symbol: 'w' Code: 1110000
Prior Symbol: 'h' Symbol: 'y' Code: 101000
Prior Symbol: 'i' Symbol: 0 Code: 00110101
Prior Symbol: 'i' Symbol: 27 Code: 00110110
Prior Symbol: 'i' Symbol: ' ' Code: 000100
Prior Symbol: 'i' Symbol: '!' Code: 001101000
Prior Symbol: 'i' Symbol: 'a' Code: 00011
Prior Symbol: 'i' Symbol: 'b' Code: 0011000
Prior Symbol: 'i' Symbol: 'c' Code: 1111
Prior Symbol: 'i' Symbol: 'd' Code: 0010
Prior Symbol: 'i' Symbol: 'e' Code: 1101
Prior Symbol: 'i' Symbol: 'f' Code: 00111
Prior Symbol: 'i' Symbol: 'g' Code: 1100
Prior Symbol: 'i' Symbol: 'i' Code: 00110010
Prior Symbol: 'i' Symbol: 'k' Code: 00110011
Prior Symbol: 'i' Symbol: 'l' Code: 0110
Prior Symbol: 'i' Symbol: 'm' Code: 11101
Prior Symbol: 'i' Symbol: 'n' Code: 10
Prior Symbol: 'i' Symbol: 'o' Code: 0100
Prior Symbol: 'i' Symbol: 'p' Code: 000101
Prior Symbol: 'i' Symbol: 'r' Code: 11100
Prior Symbol: 'i' Symbol: 's' Code: 0111
Prior Symbol: 'i' Symbol: 't' Code: 0101
Prior Symbol: 'i' Symbol: 'v' Code: 0000
Prior Symbol: 'i' Symbol: 'x' Code: 001101001
Prior Symbol: 'i' Symbol: 'z' Code: 00110111
Prior Symbol: 'j' Symbol: 27 Code: 10
Prior Symbol: 'j' Symbol: 'a' Code: 11
Prior Symbol: 'j' Symbol: 'o' Code: 0
Prior Symbol: 'k' Symbol: 0 Code: 01
Prior Symbol: 'k' Symbol: 27 Code: 00011
Prior Symbol: 'k' Symbol: ' ' Code: 111
Prior Symbol: 'k' Symbol: ':' Code: 00001
Prior Symbol: 'k' Symbol: 'T' Code: 000000
Prior Symbol: 'k' Symbol: 'a' Code: 001111
Prior Symbol: 'k' Symbol: 'e' Code: 10
Prior Symbol: 'k' Symbol: 'f' Code: 000100
Prior Symbol: 'k' Symbol: 'i' Code: 110
Prior Symbol: 'k' Symbol: 'l' Code: 000101
Prior Symbol: 'k' Symbol: 'o' Code: 000001
Prior Symbol: 'k' Symbol: 's' Code: 0010
Prior Symbol: 'k' Symbol: 'w' Code: 001110
Prior Symbol: 'k' Symbol: 'y' Code: 00110
Prior Symbol: 'l' Symbol: 0 Code: 1000
Prior Symbol: 'l' Symbol: 27 Code: 0111001
Prior Symbol: 'l' Symbol: ' ' Code: 010
Prior Symbol: 'l' Symbol: '"' Code: 01100010
Prior Symbol: 'l' Symbol: '-' Code: 11110011
Prior Symbol: 'l' Symbol: ':' Code: 01100011
Prior Symbol: 'l' Symbol: 'a' Code: 1110
Prior Symbol: 'l' Symbol: 'b' Code: 0110000
Prior Symbol: 'l' Symbol: 'c' Code: 01110000
Prior Symbol: 'l' Symbol: 'd' Code: 000
Prior Symbol: 'l' Symbol: 'e' Code: 110
Prior Symbol: 'l' Symbol: 'f' Code: 1111000
Prior Symbol: 'l' Symbol: 'i' Code: 001
Prior Symbol: 'l' Symbol: 'k' Code: 011001
Prior Symbol: 'l' Symbol: 'l' Code: 101
Prior Symbol: 'l' Symbol: 'm' Code: 1111010
Prior Symbol: 'l' Symbol: 'o' Code: 11111
Prior Symbol: 'l' Symbol: 'r' Code: 11110010
Prior Symbol: 'l' Symbol: 's' Code: 01101
Prior Symbol: 'l' Symbol: 't' Code: 011101
Prior Symbol: 'l' Symbol: 'u' Code: 01111
Prior Symbol: 'l' Symbol: 'v' Code: 1111011
Prior Symbol: 'l' Symbol: 'w' Code: 01110001
Prior Symbol: 'l' Symbol: 'y' Code: 1001
Prior Symbol: 'm' Symbol: 0 Code: 0100
Prior Symbol: 'm' Symbol: 27 Code: 010101
Prior Symbol: 'm' Symbol: ' ' Code: 001
Prior Symbol: 'm' Symbol: 'a' Code: 101
Prior Symbol: 'm' Symbol: 'b' Code: 0000
Prior Symbol: 'm' Symbol: 'e' Code: 11
Prior Symbol: 'm' Symbol: 'i' Code: 011
Prior Symbol: 'm' Symbol: 'm' Code: 0001
Prior Symbol: 'm' Symbol: 'o' Code: 1001
Prior Symbol: 'm' Symbol: 'p' Code: 1000
Prior Symbol: 'm' Symbol: 's' Code: 010111
Prior Symbol: 'm' Symbol: 'u' Code: 010110
Prior Symbol: 'm' Symbol: 'y' Code: 010100
Prior Symbol: 'n' Symbol: 0 Code: 000
Prior Symbol: 'n' Symbol: 27 Code: 01110011
Prior Symbol: 'n' Symbol: ' ' Code: 110
Prior Symbol: 'n' Symbol: '"' Code: 011101
Prior Symbol: 'n' Symbol: ':' Code: 1001010
Prior Symbol: 'n' Symbol: 'a' Code: 11100
Prior Symbol: 'n' Symbol: 'b' Code: 111010000

Prior Symbol: 'n' Symbol: 'c' Code: 01111
Prior Symbol: 'n' Symbol: 'd' Code: 001
Prior Symbol: 'n' Symbol: 'e' Code: 010
Prior Symbol: 'n' Symbol: 'f' Code: 1001011
Prior Symbol: 'n' Symbol: 'g' Code: 101
Prior Symbol: 'n' Symbol: 'h' Code: 111010101
Prior Symbol: 'n' Symbol: 'i' Code: 1000
Prior Symbol: 'n' Symbol: 'j' Code: 111010001
Prior Symbol: 'n' Symbol: 'k' Code: 1110110
Prior Symbol: 'n' Symbol: 'l' Code: 111010110
Prior Symbol: 'n' Symbol: 'm' Code: 111010111
Prior Symbol: 'n' Symbol: 'n' Code: 10011
Prior Symbol: 'n' Symbol: 'o' Code: 1110111
Prior Symbol: 'n' Symbol: 'r' Code: 111010100
Prior Symbol: 'n' Symbol: 's' Code: 0110
Prior Symbol: 'n' Symbol: 't' Code: 1111
Prior Symbol: 'n' Symbol: 'u' Code: 11101001
Prior Symbol: 'n' Symbol: 'v' Code: 0111000
Prior Symbol: 'n' Symbol: 'y' Code: 100100
Prior Symbol: 'n' Symbol: 'z' Code: 01110010
Prior Symbol: 'o' Symbol: 0 Code: 00101
Prior Symbol: 'o' Symbol: 27 Code: 01110001
Prior Symbol: 'o' Symbol: ' ' Code: 0101
Prior Symbol: 'o' Symbol: ''' Code: 01110000
Prior Symbol: 'o' Symbol: '.' Code: 0111011010
Prior Symbol: 'o' Symbol: '?' Code: 011101100
Prior Symbol: 'o' Symbol: 'a' Code: 1100010
Prior Symbol: 'o' Symbol: 'b' Code: 001001
Prior Symbol: 'o' Symbol: 'c' Code: 110000
Prior Symbol: 'o' Symbol: 'd' Code: 01111
Prior Symbol: 'o' Symbol: 'e' Code: 0111001
Prior Symbol: 'o' Symbol: 'f' Code: 1001
Prior Symbol: 'o' Symbol: 'g' Code: 00010
Prior Symbol: 'o' Symbol: 'h' Code: 0111010
Prior Symbol: 'o' Symbol: 'i' Code: 01110111
Prior Symbol: 'o' Symbol: 'k' Code: 1100011
Prior Symbol: 'o' Symbol: 'l' Code: 0100
Prior Symbol: 'o' Symbol: 'm' Code: 1000
Prior Symbol: 'o' Symbol: 'n' Code: 111
Prior Symbol: 'o' Symbol: 'o' Code: 0011
Prior Symbol: 'o' Symbol: 'p' Code: 01101
Prior Symbol: 'o' Symbol: 'r' Code: 101
Prior Symbol: 'o' Symbol: 's' Code: 11001
Prior Symbol: 'o' Symbol: 't' Code: 00011
Prior Symbol: 'o' Symbol: 'u' Code: 1101
Prior Symbol: 'o' Symbol: 'v' Code: 01100
Prior Symbol: 'o' Symbol: 'w' Code: 0000
Prior Symbol: 'o' Symbol: 'x' Code: 0010000
Prior Symbol: 'o' Symbol: 'y' Code: 0010001
Prior Symbol: 'o' Symbol: 'z' Code: 0111011011
Prior Symbol: 'p' Symbol: 0 Code: 1101
Prior Symbol: 'p' Symbol: 27 Code: 101110
Prior Symbol: 'p' Symbol: ' ' Code: 010
Prior Symbol: 'p' Symbol: ''' Code: 1100101
Prior Symbol: 'p' Symbol: 'a' Code: 1001
Prior Symbol: 'p' Symbol: 'd' Code: 101111
Prior Symbol: 'p' Symbol: 'e' Code: 111
Prior Symbol: 'p' Symbol: 'h' Code: 11000
Prior Symbol: 'p' Symbol: 'i' Code: 1010
Prior Symbol: 'p' Symbol: 'l' Code: 0110
Prior Symbol: 'p' Symbol: 'm' Code: 1100100
Prior Symbol: 'p' Symbol: 'o' Code: 00
Prior Symbol: 'p' Symbol: 'p' Code: 0111
Prior Symbol: 'p' Symbol: 'r' Code: 10001
Prior Symbol: 'p' Symbol: 's' Code: 10000
Prior Symbol: 'p' Symbol: 't' Code: 10110
Prior Symbol: 'p' Symbol: 'y' Code: 110011
Prior Symbol: 'q' Symbol: 27 Code: 0
Prior Symbol: 'q' Symbol: 'u' Code: 1
Prior Symbol: 'r' Symbol: 0 Code: 1001
Prior Symbol: 'r' Symbol: 27 Code: 01100101
Prior Symbol: 'r' Symbol: ' ' Code: 1111
Prior Symbol: 'r' Symbol: ''' Code: 0110011
Prior Symbol: 'r' Symbol: ',' Code: 110011101
Prior Symbol: 'r' Symbol: '.' Code: 0111100
Prior Symbol: 'r' Symbol: ':' Code: 110011100
Prior Symbol: 'r' Symbol: 'a' Code: 000
Prior Symbol: 'r' Symbol: 'b' Code: 01111101
Prior Symbol: 'r' Symbol: 'c' Code: 0111111
Prior Symbol: 'r' Symbol: 'd' Code: 11000
Prior Symbol: 'r' Symbol: 'e' Code: 101
Prior Symbol: 'r' Symbol: 'f' Code: 11001111
Prior Symbol: 'r' Symbol: 'g' Code: 0111101
Prior Symbol: 'r' Symbol: 'i' Code: 010
Prior Symbol: 'r' Symbol: 'k' Code: 110010
Prior Symbol: 'r' Symbol: 'l' Code: 0011
Prior Symbol: 'r' Symbol: 'm' Code: 011000
Prior Symbol: 'r' Symbol: 'n' Code: 01101
Prior Symbol: 'r' Symbol: 'o' Code: 1101
Prior Symbol: 'r' Symbol: 'p' Code: 01111100
Prior Symbol: 'r' Symbol: 'r' Code: 01110
Prior Symbol: 'r' Symbol: 's' Code: 1110
Prior Symbol: 'r' Symbol: 't' Code: 1000
Prior Symbol: 'r' Symbol: 'u' Code: 1100110
Prior Symbol: 'r' Symbol: 'v' Code: 01100100
Prior Symbol: 'r' Symbol: 'y' Code: 0010
Prior Symbol: 's' Symbol: 0 Code: 11
Prior Symbol: 's' Symbol: 27 Code: 0010011
Prior Symbol: 's' Symbol: ' ' Code: 01
Prior Symbol: 's' Symbol: ''' Code: 001011010
Prior Symbol: 's' Symbol: ',' Code: 001011011
Prior Symbol: 's' Symbol: '.' Code: 00100101
Prior Symbol: 's' Symbol: ':' Code: 0000001
Prior Symbol: 's' Symbol: '?' Code: 001011100
Prior Symbol: 's' Symbol: 'C' Code: 001011101
Prior Symbol: 's' Symbol: 'H' Code: 001011110
Prior Symbol: 's' Symbol: 'a' Code: 101010
Prior Symbol: 's' Symbol: 'c' Code: 101011
Prior Symbol: 's' Symbol: 'd' Code: 001011111
Prior Symbol: 's' Symbol: 'e' Code: 1011
Prior Symbol: 's' Symbol: 'f' Code: 00000000
Prior Symbol: 's' Symbol: 'h' Code: 00001
Prior Symbol: 's' Symbol: 'i' Code: 0011
Prior Symbol: 's' Symbol: 'k' Code: 000001
Prior Symbol: 's' Symbol: 'l' Code: 00101010
Prior Symbol: 's' Symbol: 'm' Code: 00000001
Prior Symbol: 's' Symbol: 'n' Code: 00101011
Prior Symbol: 's' Symbol: 'o' Code: 10100
Prior Symbol: 's' Symbol: 'p' Code: 001000
Prior Symbol: 's' Symbol: 'r' Code: 00100100
Prior Symbol: 's' Symbol: 's' Code: 0001
Prior Symbol: 's' Symbol: 't' Code: 100
Prior Symbol: 's' Symbol: 'u' Code: 0010100
Prior Symbol: 's' Symbol: 'y' Code: 00101100
Prior Symbol: 't' Symbol: 0 Code: 010
Prior Symbol: 't' Symbol: 27 Code: 11000010
Prior Symbol: 't' Symbol: ' ' Code: 101
Prior Symbol: 't' Symbol: ''' Code: 11000011
Prior Symbol: 't' Symbol: ':' Code: 110110000
Prior Symbol: 't' Symbol: '?' Code: 110110001
Prior Symbol: 't' Symbol: 'a' Code: 0000
Prior Symbol: 't' Symbol: 'b' Code: 100000
Prior Symbol: 't' Symbol: 'c' Code: 1101101
Prior Symbol: 't' Symbol: 'd' Code: 11000000
Prior Symbol: 't' Symbol: 'e' Code: 011
Prior Symbol: 't' Symbol: 'h' Code: 111
Prior Symbol: 't' Symbol: 'i' Code: 001
Prior Symbol: 't' Symbol: 'l' Code: 10001
Prior Symbol: 't' Symbol: 'm' Code: 100001
Prior Symbol: 't' Symbol: 'n' Code: 11011001
Prior Symbol: 't' Symbol: 'o' Code: 1001
Prior Symbol: 't' Symbol: 'r' Code: 11010
Prior Symbol: 't' Symbol: 's' Code: 0001
Prior Symbol: 't' Symbol: 't' Code: 110111
Prior Symbol: 't' Symbol: 'u' Code: 11001
Prior Symbol: 't' Symbol: 'w' Code: 11000001
Prior Symbol: 't' Symbol: 'y' Code: 110001
Prior Symbol: 'u' Symbol: 0 Code: 0011110
Prior Symbol: 'u' Symbol: 27 Code: 000100
Prior Symbol: 'u' Symbol: ' ' Code: 001110
Prior Symbol: 'u' Symbol: 'a' Code: 00110
Prior Symbol: 'u' Symbol: 'b' Code: 10011
Prior Symbol: 'u' Symbol: 'c' Code: 11100
Prior Symbol: 'u' Symbol: 'd' Code: 10000
Prior Symbol: 'u' Symbol: 'e' Code: 0010
Prior Symbol: 'u' Symbol: 'f' Code: 0011111
Prior Symbol: 'u' Symbol: 'g' Code: 11101
Prior Symbol: 'u' Symbol: 'i' Code: 00011
Prior Symbol: 'u' Symbol: 'k' Code: 0001010
Prior Symbol: 'u' Symbol: 'l' Code: 0000
Prior Symbol: 'u' Symbol: 'm' Code: 10010
Prior Symbol: 'u' Symbol: 'n' Code: 110
Prior Symbol: 'u' Symbol: 'p' Code: 10001
Prior Symbol: 'u' Symbol: 'r' Code: 01
Prior Symbol: 'u' Symbol: 's' Code: 101
Prior Symbol: 'u' Symbol: 't' Code: 1111
Prior Symbol: 'u' Symbol: 'z' Code: 0001011
Prior Symbol: 'v' Symbol: 27 Code: 0010
Prior Symbol: 'v' Symbol: 'a' Code: 000
Prior Symbol: 'v' Symbol: 'e' Code: 1
Prior Symbol: 'v' Symbol: 'i' Code: 01
Prior Symbol: 'v' Symbol: 'o' Code: 00111
Prior Symbol: 'v' Symbol: 's' Code: 00110
Prior Symbol: 'w' Symbol: 0 Code: 001
Prior Symbol: 'w' Symbol: 27 Code: 01010
Prior Symbol: 'w' Symbol: ' ' Code: 011
Prior Symbol: 'w' Symbol: ''' Code: 010010
Prior Symbol: 'w' Symbol: 'a' Code: 000
Prior Symbol: 'w' Symbol: 'b' Code: 010011
Prior Symbol: 'w' Symbol: 'c' Code: 010111
Prior Symbol: 'w' Symbol: 'e' Code: 1111
Prior Symbol: 'w' Symbol: 'i' Code: 1100
Prior Symbol: 'w' Symbol: 'l' Code: 010110
Prior Symbol: 'w' Symbol: 'n' Code: 1110
Prior Symbol: 'w' Symbol: 'o' Code: 1101
Prior Symbol: 'w' Symbol: 'r' Code: 01000
Prior Symbol: 'w' Symbol: 's' Code: 10
Prior Symbol: 'x' Symbol: 0 Code: 110
Prior Symbol: 'x' Symbol: 27 Code: 1010
Prior Symbol: 'x' Symbol: ' ' Code: 1011
Prior Symbol: 'x' Symbol: 'a' Code: 000
Prior Symbol: 'x' Symbol: 'e' Code: 001
Prior Symbol: 'x' Symbol: 'i' Code: 100
Prior Symbol: 'x' Symbol: 'p' Code: 111
Prior Symbol: 'x' Symbol: 't' Code: 01
Prior Symbol: 'y' Symbol: 0 Code: 10
Prior Symbol: 'y' Symbol: 27 Code: 111110
Prior Symbol: 'y' Symbol: ' ' Code: 0
Prior Symbol: 'y' Symbol: '!' Code: 1101101
Prior Symbol: 'y' Symbol: ''' Code: 110101
Prior Symbol: 'y' Symbol: '-' Code: 11110101
Prior Symbol: 'y' Symbol: 'a' Code: 1101110
Prior Symbol: 'y' Symbol: 'b' Code: 1111011
Prior Symbol: 'y' Symbol: 'c' Code: 11110100
Prior Symbol: 'y' Symbol: 'd' Code: 1100000
Prior Symbol: 'y' Symbol: 'e' Code: 11001
Prior Symbol: 'y' Symbol: 'i' Code: 1100001
Prior Symbol: 'y' Symbol: 'l' Code: 111111
Prior Symbol: 'y' Symbol: 'm' Code: 1101111
Prior Symbol: 'y' Symbol: 'n' Code: 1100010
Prior Symbol: 'y' Symbol: 'o' Code: 1100011
Prior Symbol: 'y' Symbol: 'p' Code: 1101000
Prior Symbol: 'y' Symbol: 's' Code: 1110
Prior Symbol: 'y' Symbol: 't' Code: 1101001
Prior Symbol: 'y' Symbol: 'v' Code: 1101100
Prior Symbol: 'y' Symbol: 'w' Code: 111100
Prior Symbol: 'z' Symbol: 0 Code: 110
Prior Symbol: 'z' Symbol: 27 Code: 100
Prior Symbol: 'z' Symbol: ' ' Code: 000
Prior Symbol: 'z' Symbol: 'a' Code: 01
Prior Symbol: 'z' Symbol: 'e' Code: 1010
Prior Symbol: 'z' Symbol: 'i' Code: 111
Prior Symbol: 'z' Symbol: 'y' Code: 001
Prior Symbol: 'z' Symbol: 'z' Code: 1011
Prior Symbol: '{' Symbol: 27 Code: 1
Prior Symbol: '|' Symbol: 27 Code: 1
Prior Symbol: '}' Symbol: 27 Code: 1
Prior Symbol: '~' Symbol: 27 Code: 1
Prior Symbol: 127 Symbol: 27 Code: 1

Table C.5 English-language Program Title Decode Table

0	1	79	220	158	3	237	34	316	155	395	4	474	155
1	0	80	1	159	100	238	7	317	155	396	155	475	160
2	1	81	230	160	3	239	44	318	155	397	226	476	4
3	58	82	1	161	122	240	7	319	155	398	5	477	243
4	1	63	232	162	3	241	70	320	155	399	6	478	228
5	60	84	1	163	148	242	7	321	155	400	7	479	185
6	1	85	234	164	3	243	84	322	155	401	8	480	1
7	62	86	1	165	152	244	7	323	155	402	9	481	244
8	1	87	240	166	3	245	124	324	155	403	213	482	160
9	64	88	1	167	164	246	7	325	155	404	10	483	155
10	1	89	242	168	3	247	138	326	155	405	214	484	2
11	66	90	1	169	200	248	7	327	155	406	11	485	3
12	1	91	244	170	3	249	140	328	155	407	217	486	155
13	68	92	2	171	222	250	7	329	155	408	12	487	155
14	1	93	6	172	3	251	142	330	155	409	166	488	155
15	70	94	2	173	230	252	7	331	155	410	233	489	155
16	1	95	18	174	3	253	144	332	155	411	203	490	1
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1826	4
1827	229
1828	243
1829	239
1830	155
1831	1
1832	225
1833	2
1834	3
1835	233
1836	11
1837	12
1838	167
1839	226
1840	236
1841	227
1842	242
1843	1
1844	155
1845	2
1846	3
1847	4
1848	233
1849	239
1850	238
1851	229
1852	225
1853	128
1854	5
1855	160
1856	6
1857	7
1856	8
1859	9
1860	243
1861	10
1862	5
1863	6
1864	155
1865	160
1866	225
1867	229
1868	233
1869	1
1870	128
1871	240
1872	2
1873	244
1874	3
1875	4
1876	160
1877	19
1878	227
1879	173
1880	228
1881	233
1882	238
1883	239
1884	240
1885	244
1886	246
1887	161
1888	225
1889	237
1890	1
1891	226
1892	2
1893	3
1894	4
1895	167
1896	5
1897	6
1898	247
1899	7
1900	155
1901	236
1902	8
1903	229
1904	9
1905	10
1906	11
1907	12
1908	13
1909	14
1910	243
1911	15
1912	16
1913	17
1914	128
1915	18
1916	5
1917	6
1918	229
1919	250
1920	160
1921	249
1922	155
1923	1
1924	128
1925	233
1926	2
1927	225
1928	3
1929	4
1930	155
1931	155
1932	155
1933	155
1934	155
1935	155
1936	155
1937	155
1938	155
1939	155

C3. STANDARD COMPRESSION TYPE 2 HUFFMAN ENCODE/DECODE TABLES

The following encode/decode tables are optimized for English-language program description text. These tables correspond to multiple_string_structure() with compression_type value 0x02, and mode equal to 0xFF.

Table C.6 English-language Program Description Encode Table

Prior Symbol: 0 Symbol: 27 Code: 1110000
Prior Symbol: 0 Symbol: '"' Code: 111001
Prior Symbol: 0 Symbol: 'A' Code: 010
Prior Symbol: 0 Symbol: 'B' Code: 0011
Prior Symbol: 0 Symbol: 'C' Code: 0111
Prior Symbol: 0 Symbol: 'D' Code: 11101
Prior Symbol: 0 Symbol: 'E' Code: 10010
Prior Symbol: 0 Symbol: 'F' Code: 10110
Prior Symbol: 0 Symbol: 'G' Code: 011011
Prior Symbol: 0 Symbol: 'H' Code: 10111
Prior Symbol: 0 Symbol: 'I' Code: 011000
Prior Symbol: 0 Symbol: 'J' Code: 1100
Prior Symbol: 0 Symbol: 'K' Code: 00101
Prior Symbol: 0 Symbol: 'L' Code: 10011
Prior Symbol: 0 Symbol: 'M' Code: 1111
Prior Symbol: 0 Symbol: 'N' Code: 00100
Prior Symbol: 0 Symbol: 'O' Code: 011001
Prior Symbol: 0 Symbol: 'P' Code: 000
Prior Symbol: 0 Symbol: 'R' Code: 1000
Prior Symbol: 0 Symbol: 'S' Code: 1010
Prior Symbol: 0 Symbol: 'T' Code: 1101
Prior Symbol: 0 Symbol: 'V' Code: 1110001
Prior Symbol: 0 Symbol: 'W' Code: 011010
Prior Symbol: 1 Symbol: 27 Code: 1
Prior Symbol: 2 Symbol: 27 Code: 1
Prior Symbol: 3 Symbol: 27 Code: 1
Prior Symbol: 4 Symbol: 27 Code: 1
Prior Symbol: 5 Symbol: 27 Code: 1
Prior Symbol: 6 Symbol: 27 Code: 1
Prior Symbol: 7 Symbol: 27 Code: 1
Prior Symbol: 8 Symbol: 27 Code: 1
Prior Symbol: 9 Symbol: 27 Code: 1
Prior Symbol: 10 Symbol: 27 Code: 1
Prior Symbol: 11 Symbol: 27 Code: 1
Prior Symbol: 12 Symbol: 27 Code: 1
Prior Symbol: 13 Symbol: 27 Code: 1
Prior Symbol: 14 Symbol: 27 Code: 1
Prior Symbol: 15 Symbol: 27 Code: 1
Prior Symbol: 16 Symbol: 27 Code: 1
Prior Symbol: 17 Symbol: 27 Code: 1
Prior Symbol: 18 Symbol: 27 Code: 1
Prior Symbol: 19 Symbol: 27 Code: 1
Prior Symbol: 20 Symbol: 27 Code: 1
Prior Symbol: 21 Symbol: 27 Code: 1
Prior Symbol: 22 Symbol: 27 Code: 1
Prior Symbol: 23 Symbol: 27 Code: 1
Prior Symbol: 24 Symbol: 27 Code: 1
Prior Symbol: 25 Symbol: 27 Code: 1
Prior Symbol: 26 Symbol: 27 Code: 1
Prior Symbol: 27 Symbol: 27 Code: 1
Prior Symbol: 28 Symbol: 27 Code: 1
Prior Symbol: 29 Symbol: 27 Code: 1
Prior Symbol: 30 Symbol: 27 Code: 1
Prior Symbol: 31 Symbol: 27 Code: 1
Prior Symbol: ' ' Symbol: 27 Code: 101000001
Prior Symbol: ' ' Symbol: '"' Code: 111111010
Prior Symbol: ' ' Symbol: '(' Code: 1111111100
Prior Symbol: ' ' Symbol: '-' Code: 11111111110
Prior Symbol: ' ' Symbol: '/' Code: 11111111111
Prior Symbol: ' ' Symbol: '1' Code: 0101011
Prior Symbol: ' ' Symbol: '2' Code: 0100010
Prior Symbol: ' ' Symbol: '3' Code: 1111111101
Prior Symbol: ' ' Symbol: '4' Code: 110010100
Prior Symbol: ' ' Symbol: '5' Code: 1111111110
Prior Symbol: ' ' Symbol: '7' Code: 1010000000
Prior Symbol: ' ' Symbol: 'A' Code: 10010
Prior Symbol: ' ' Symbol: 'B' Code: 010100
Prior Symbol: ' ' Symbol: 'C' Code: 111100
Prior Symbol: ' ' Symbol: 'D' Code: 1111010
Prior Symbol: ' ' Symbol: 'E' Code: 0100011
Prior Symbol: ' ' Symbol: 'F' Code: 0101010
Prior Symbol: ' ' Symbol: 'G' Code: 000010
Prior Symbol: ' ' Symbol: 'H' Code: 1111011
Prior Symbol: ' ' Symbol: 'I' Code: 11001011
Prior Symbol: ' ' Symbol: 'J' Code: 000011
Prior Symbol: ' ' Symbol: 'K' Code: 1100100
Prior Symbol: ' ' Symbol: 'L' Code: 010110
Prior Symbol: ' ' Symbol: 'M' Code: 101001
Prior Symbol: ' ' Symbol: 'N' Code: 001100
Prior Symbol: ' ' Symbol: 'O' Code: 10100001
Prior Symbol: ' ' Symbol: 'P' Code: 001101
Prior Symbol: ' ' Symbol: 'R' Code: 1111100
Prior Symbol: ' ' Symbol: 'S' Code: 01001
Prior Symbol: ' ' Symbol: 'T' Code: 1100110
Prior Symbol: ' ' Symbol: 'U' Code: 111111011
Prior Symbol: ' ' Symbol: 'V' Code: 111111100
Prior Symbol: ' ' Symbol: 'W' Code: 010000
Prior Symbol: ' ' Symbol: 'Y' Code: 111111101
Prior Symbol: ' ' Symbol: 'Z' Code: 1010000001
Prior Symbol: ' ' Symbol: 'a' Code: 011
Prior Symbol: ' ' Symbol: 'b' Code: 10111
Prior Symbol: ' ' Symbol: 'c' Code: 10011
Prior Symbol: ' ' Symbol: 'd' Code: 10000
Prior Symbol: ' ' Symbol: 'e' Code: 100010
Prior Symbol: ' ' Symbol: 'f' Code: 11101
Prior Symbol: ' ' Symbol: 'g' Code: 100011
Prior Symbol: ' ' Symbol: 'h' Code: 0001
Prior Symbol: ' ' Symbol: 'i' Code: 10101
Prior Symbol: ' ' Symbol: 'j' Code: 11001111
Prior Symbol: ' ' Symbol: 'k' Code: 11111010
Prior Symbol: ' ' Symbol: 'l' Code: 010111
Prior Symbol: ' ' Symbol: 'm' Code: 00000
Prior Symbol: ' ' Symbol: 'n' Code: 1010001
Prior Symbol: ' ' Symbol: 'o' Code: 0010
Prior Symbol: ' ' Symbol: 'p' Code: 10110
Prior Symbol: ' ' Symbol: 'q' Code: 110010101
Prior Symbol: ' ' Symbol: 'r' Code: 00111
Prior Symbol: ' ' Symbol: 's' Code: 11100
Prior Symbol: ' ' Symbol: 't' Code: 1101
Prior Symbol: ' ' Symbol: 'u' Code: 11111011
Prior Symbol: ' ' Symbol: 'v' Code: 11111100
Prior Symbol: ' ' Symbol: 'w' Code: 11000
Prior Symbol: ' ' Symbol: 'y' Code: 11001110
Prior Symbol: '!' Symbol: 27 Code: 1
Prior Symbol: '"' Symbol: 0 Code: 000
Prior Symbol: '"' Symbol: 27 Code: 10
Prior Symbol: '"' Symbol: ' ' Code: 11
Prior Symbol: '"' Symbol: '.' Code: 001
Prior Symbol: '"' Symbol: 'H' Code: 010
Prior Symbol: '"' Symbol: 'T' Code: 011
Prior Symbol: '#' Symbol: 27 Code: 1
Prior Symbol: '$' Symbol: 27 Code: 1
Prior Symbol: '%' Symbol: 27 Code: 1
Prior Symbol: '&' Symbol: 27 Code: 1
Prior Symbol: ''' Symbol: 27 Code: 00
Prior Symbol: ''' Symbol: ' ' Code: 010
Prior Symbol: ''' Symbol: 's' Code: 1
Prior Symbol: ''' Symbol: 't' Code: 011
Prior Symbol: '(' Symbol: 27 Code: 1
Prior Symbol: ')' Symbol: 27 Code: 1
Prior Symbol: ')' Symbol: ',' Code: 0
Prior Symbol: '*' Symbol: 27 Code: 1
Prior Symbol: '+' Symbol: 27 Code: 1
Prior Symbol: ',' Symbol: 27 Code: 00
Prior Symbol: ',' Symbol: ' ' Code: 1
Prior Symbol: ',' Symbol: '"' Code: 01
Prior Symbol: '-' Symbol: 27 Code: 10
Prior Symbol: '-' Symbol: ' ' Code: 1110
Prior Symbol: '-' Symbol: 'a' Code: 000
Prior Symbol: '-' Symbol: 'b' Code: 0010
Prior Symbol: '-' Symbol: 'c' Code: 110
Prior Symbol: '-' Symbol: 'd' Code: 0011
Prior Symbol: '-' Symbol: 'e' Code: 0100
Prior Symbol: '-' Symbol: 'f' Code: 0101
Prior Symbol: '-' Symbol: 'r' Code: 1111
Prior Symbol: '-' Symbol: 's' Code: 011
Prior Symbol: '.' Symbol: 0 Code: 1
Prior Symbol: '.' Symbol: 27 Code: 000
Prior Symbol: '.' Symbol: ' ' Code: 01
Prior Symbol: '.' Symbol: '"' Code: 0010
Prior Symbol: '.' Symbol: 'J' Code: 00110
Prior Symbol: '.' Symbol: 'S' Code: 00111
Prior Symbol: '/' Symbol: 27 Code: 0
Prior Symbol: '/' Symbol: ' ' Code: 1
Prior Symbol: '0' Symbol: 27 Code: 100
Prior Symbol: '0' Symbol: ' ' Code: 111
Prior Symbol: '0' Symbol: '0' Code: 00
Prior Symbol: '0' Symbol: '7' Code: 101
Prior Symbol: '0' Symbol: 's' Code: 01
Prior Symbol: '0' Symbol: 't' Code: 110
Prior Symbol: '1' Symbol: 27 Code: 111
Prior Symbol: '1' Symbol: ' ' Code: 10
Prior Symbol: '1' Symbol: '8' Code: 110
Prior Symbol: '1' Symbol: '9' Code: 0
Prior Symbol: '2' Symbol: 27 Code: 101
Prior Symbol: '2' Symbol: ' ' Code: 11
Prior Symbol: '2' Symbol: '.' Code: 0
Prior Symbol: '2' Symbol: '6' Code: 100
Prior Symbol: '3' Symbol: 27 Code: 10
Prior Symbol: '3' Symbol: ' ' Code: 0
Prior Symbol: '3' Symbol: '0' Code: 11
Prior Symbol: '4' Symbol: 27 Code: 10
Prior Symbol: '4' Symbol: ' ' Code: 11
Prior Symbol: '4' Symbol: '.' Code: 0
Prior Symbol: '5' Symbol: 27 Code: 11
Prior Symbol: '5' Symbol: ' ' Code: 10
Prior Symbol: '5' Symbol: '.' Code: 0
Prior Symbol: '6' Symbol: 27 Code: 1
Prior Symbol: '7' Symbol: 27 Code: 0
Prior Symbol: '7' Symbol: ',' Code: 10
Prior Symbol: '7' Symbol: '.' Code: 11
Prior Symbol: '8' Symbol: 27 Code: 1
Prior Symbol: '9' Symbol: 27 Code: 110
Prior Symbol: '9' Symbol: ' ' Code: 111
Prior Symbol: '9' Symbol: '5' Code: 00
Prior Symbol: '9' Symbol: '6' Code: 01
Prior Symbol: '9' Symbol: '8' Code: 10
Prior Symbol: ':' Symbol: 27 Code: 0
Prior Symbol: ':' Symbol: ' ' Code: 1
Prior Symbol: ';' Symbol: 27 Code: 0
Prior Symbol: ';' Symbol: ' ' Code: 1
Prior Symbol: '<' Symbol: 27 Code: 1
Prior Symbol: '=' Symbol: 27 Code: 1
Prior Symbol: '>' Symbol: 27 Code: 1
Prior Symbol: '?' Symbol: 27 Code: 0
Prior Symbol: '?' Symbol: ' ' Code: 1
Prior Symbol: '@' Symbol: 27 Code: 1
Prior Symbol: 'A' Symbol: 27 Code: 10010
Prior Symbol: 'A' Symbol: ' ' Code: 11
Prior Symbol: 'A' Symbol: 'd' Code: 10011
Prior Symbol: 'A' Symbol: 'f' Code: 101000
Prior Symbol: 'A' Symbol: 'l' Code: 00
Prior Symbol: 'A' Symbol: 'm' Code: 10101
Prior Symbol: 'A' Symbol: 'n' Code: 01

Prior Symbol: 'A' Symbol: 'r' Code: 1011
Prior Symbol: 'A' Symbol: 's' Code: 10000
Prior Symbol: 'A' Symbol: 't' Code: 10001
Prior Symbol: 'A' Symbol: 'u' Code: 101001
Prior Symbol: 'B' Symbol: 27 Code: 10010
Prior Symbol: 'B' Symbol: 'a' Code: 101
Prior Symbol: 'B' Symbol: 'e' Code: 111
Prior Symbol: 'B' Symbol: 'i' Code: 00
Prior Symbol: 'B' Symbol: 'l' Code: 10011
Prior Symbol: 'B' Symbol: 'o' Code: 110
Prior Symbol: 'B' Symbol: 'r' Code: 01
Prior Symbol: 'B' Symbol: 'u' Code: 1000
Prior Symbol: 'C' Symbol: 27 Code: 01110
Prior Symbol: 'C' Symbol: 'a' Code: 00
Prior Symbol: 'C' Symbol: 'h' Code: 10
Prior Symbol: 'C' Symbol: 'i' Code: 01111
Prior Symbol: 'C' Symbol: 'l' Code: 110
Prior Symbol: 'C' Symbol: 'o' Code: 111
Prior Symbol: 'C' Symbol: 'r' Code: 0101
Prior Symbol: 'C' Symbol: 'u' Code: 0110
Prior Symbol: 'C' Symbol: 'y' Code: 0100
Prior Symbol: 'D' Symbol: 27 Code: 1111
Prior Symbol: 'D' Symbol: 'a' Code: 01
Prior Symbol: 'D' Symbol: 'e' Code: 100
Prior Symbol: 'D' Symbol: 'i' Code: 00
Prior Symbol: 'D' Symbol: 'o' Code: 101
Prior Symbol: 'D' Symbol: 'r' Code: 1101
Prior Symbol: 'D' Symbol: 'u' Code: 1110
Prior Symbol: 'D' Symbol: 'y' Code: 1100
Prior Symbol: 'E' Symbol: 27 Code: 10
Prior Symbol: 'E' Symbol: 'a' Code: 0110
Prior Symbol: 'E' Symbol: 'd' Code: 000
Prior Symbol: 'E' Symbol: 'i' Code: 0111
Prior Symbol: 'E' Symbol: 'l' Code: 001
Prior Symbol: 'E' Symbol: 'n' Code: 1100
Prior Symbol: 'E' Symbol: 'r' Code: 111
Prior Symbol: 'E' Symbol: 's' Code: 010
Prior Symbol: 'E' Symbol: 'v' Code: 1101
Prior Symbol: 'F' Symbol: 27 Code: 00
Prior Symbol: 'F' Symbol: 'e' Code: 100
Prior Symbol: 'F' Symbol: 'l' Code: 101
Prior Symbol: 'F' Symbol: 'o' Code: 01
Prior Symbol: 'F' Symbol: 'r' Code: 11
Prior Symbol: 'G' Symbol: 27 Code: 000
Prior Symbol: 'G' Symbol: 'a' Code: 110
Prior Symbol: 'G' Symbol: 'e' Code: 01
Prior Symbol: 'G' Symbol: 'i' Code: 100
Prior Symbol: 'G' Symbol: 'l' Code: 001
Prior Symbol: 'G' Symbol: 'o' Code: 1011
Prior Symbol: 'G' Symbol: 'r' Code: 111
Prior Symbol: 'G' Symbol: 'u' Code: 1010
Prior Symbol: 'H' Symbol: 27 Code: 010
Prior Symbol: 'H' Symbol: 'a' Code: 00
Prior Symbol: 'H' Symbol: 'e' Code: 011
Prior Symbol: 'H' Symbol: 'i' Code: 110
Prior Symbol: 'H' Symbol: 'o' Code: 10
Prior Symbol: 'H' Symbol: 'u' Code: 111
Prior Symbol: 'I' Symbol: 27 Code: 011
Prior Symbol: 'I' Symbol: ' ' Code: 000
Prior Symbol: 'I' Symbol: '.' Code: 100
Prior Symbol: 'I' Symbol: 'l' Code: 001
Prior Symbol: 'I' Symbol: 'n' Code: 11
Prior Symbol: 'I' Symbol: 'r' Code: 101
Prior Symbol: 'I' Symbol: 's' Code: 010
Prior Symbol: 'J' Symbol: 27 Code: 1000
Prior Symbol: 'J' Symbol: '.' Code: 1001
Prior Symbol: 'J' Symbol: 'a' Code: 111
Prior Symbol: 'J' Symbol: 'e' Code: 1101
Prior Symbol: 'J' Symbol: 'i' Code: 1100
Prior Symbol: 'J' Symbol: 'o' Code: 0
Prior Symbol: 'J' Symbol: 'u' Code: 101
Prior Symbol: 'K' Symbol: 27 Code: 111
Prior Symbol: 'K' Symbol: 'a' Code: 100
Prior Symbol: 'K' Symbol: 'e' Code: 0
Prior Symbol: 'K' Symbol: 'i' Code: 101
Prior Symbol: 'K' Symbol: 'r' Code: 110
Prior Symbol: 'L' Symbol: 27 Code: 0110
Prior Symbol: 'L' Symbol: 'a' Code: 11
Prior Symbol: 'L' Symbol: 'e' Code: 00
Prior Symbol: 'L' Symbol: 'i' Code: 0111
Prior Symbol: 'L' Symbol: 'o' Code: 10
Prior Symbol: 'L' Symbol: 'u' Code: 010
Prior Symbol: 'M' Symbol: 27 Code: 11010
Prior Symbol: 'M' Symbol: 'a' Code: 0
Prior Symbol: 'M' Symbol: 'c' Code: 11011
Prior Symbol: 'M' Symbol: 'e' Code: 1111
Prior Symbol: 'M' Symbol: 'i' Code: 10
Prior Symbol: 'M' Symbol: 'o' Code: 1100
Prior Symbol: 'M' Symbol: 'u' Code: 1110
Prior Symbol: 'N' Symbol: 27 Code: 1100
Prior Symbol: 'N' Symbol: 'a' Code: 111
Prior Symbol: 'N' Symbol: 'e' Code: 0
Prior Symbol: 'N' Symbol: 'i' Code: 1101
Prior Symbol: 'N' Symbol: 'o' Code: 10
Prior Symbol: 'O' Symbol: 27 Code: 10
Prior Symbol: 'O' Symbol: ''' Code: 010
Prior Symbol: 'O' Symbol: 'l' Code: 110
Prior Symbol: 'O' Symbol: 'n' Code: 011
Prior Symbol: 'O' Symbol: 'r' Code: 111
Prior Symbol: 'O' Symbol: 's' Code: 00
Prior Symbol: 'P' Symbol: 27 Code: 10010
Prior Symbol: 'P' Symbol: 'a' Code: 0
Prior Symbol: 'P' Symbol: 'e' Code: 111
Prior Symbol: 'P' Symbol: 'h' Code: 10011
Prior Symbol: 'P' Symbol: 'i' Code: 1000
Prior Symbol: 'P' Symbol: 'l' Code: 1101
Prior Symbol: 'P' Symbol: 'o' Code: 101
Prior Symbol: 'P' Symbol: 'r' Code: 1100
Prior Symbol: 'Q' Symbol: 27 Code: 1
Prior Symbol: 'R' Symbol: 27 Code: 0000
Prior Symbol: 'R' Symbol: '.' Code: 0001
Prior Symbol: 'R' Symbol: 'a' Code: 01
Prior Symbol: 'R' Symbol: 'e' Code: 10
Prior Symbol: 'R' Symbol: 'i' Code: 001
Prior Symbol: 'R' Symbol: 'o' Code: 11
Prior Symbol: 'S' Symbol: 27 Code: 1011
Prior Symbol: 'S' Symbol: '.' Code: 0001
Prior Symbol: 'S' Symbol: 'a' Code: 100
Prior Symbol: 'S' Symbol: 'c' Code: 0010
Prior Symbol: 'S' Symbol: 'e' Code: 1110
Prior Symbol: 'S' Symbol: 'h' Code: 110
Prior Symbol: 'S' Symbol: 'i' Code: 0011
Prior Symbol: 'S' Symbol: 'o' Code: 1111
Prior Symbol: 'S' Symbol: 't' Code: 01
Prior Symbol: 'S' Symbol: 'u' Code: 1010
Prior Symbol: 'S' Symbol: 'v' Code: 00000
Prior Symbol: 'S' Symbol: 'y' Code: 00001
Prior Symbol: 'T' Symbol: 27 Code: 1010
Prior Symbol: 'T' Symbol: 'V' Code: 1000
Prior Symbol: 'T' Symbol: 'a' Code: 1001
Prior Symbol: 'T' Symbol: 'e' Code: 11010
Prior Symbol: 'T' Symbol: 'h' Code: 0
Prior Symbol: 'T' Symbol: 'i' Code: 1011
Prior Symbol: 'T' Symbol: 'o' Code: 111
Prior Symbol: 'T' Symbol: 'r' Code: 1100
Prior Symbol: 'T' Symbol: 'w' Code: 11011
Prior Symbol: 'U' Symbol: 27 Code: 10
Prior Symbol: 'U' Symbol: '.' Code: 0
Prior Symbol: 'U' Symbol: 'n' Code: 11
Prior Symbol: 'V' Symbol: 27 Code: 111
Prior Symbol: 'V' Symbol: ' ' Code: 10
Prior Symbol: 'V' Symbol: 'e' Code: 110
Prior Symbol: 'V' Symbol: 'i' Code: 0
Prior Symbol: 'W' Symbol: 27 Code: 010
Prior Symbol: 'W' Symbol: 'a' Code: 111
Prior Symbol: 'W' Symbol: 'e' Code: 110
Prior Symbol: 'W' Symbol: 'h' Code: 011
Prior Symbol: 'W' Symbol: 'i' Code: 10
Prior Symbol: 'W' Symbol: 'o' Code: 00
Prior Symbol: 'X' Symbol: 27 Code: 1
Prior Symbol: 'Y' Symbol: 27 Code: 0
Prior Symbol: 'Y' Symbol: 'o' Code: 1
Prior Symbol: 'Z' Symbol: 27 Code: 1
Prior Symbol: '[' Symbol: 27 Code: 1
Prior Symbol: '\' Symbol: 27 Code: 1
Prior Symbol: ']' Symbol: 27 Code: 1
Prior Symbol: '^' Symbol: 27 Code: 1
Prior Symbol: '_' Symbol: 27 Code: 1
Prior Symbol: '`' Symbol: 27 Code: 1
Prior Symbol: 'a' Symbol: 27 Code: 111001101
Prior Symbol: 'a' Symbol: ' ' Code: 101
Prior Symbol: 'a' Symbol: ''' Code: 111001110
Prior Symbol: 'a' Symbol: '.' Code: 1110010
Prior Symbol: 'a' Symbol: 'b' Code: 001011
Prior Symbol: 'a' Symbol: 'c' Code: 11001
Prior Symbol: 'a' Symbol: 'd' Code: 00111
Prior Symbol: 'a' Symbol: 'e' Code: 0011001
Prior Symbol: 'a' Symbol: 'f' Code: 001010
Prior Symbol: 'a' Symbol: 'g' Code: 00100
Prior Symbol: 'a' Symbol: 'h' Code: 001100010
Prior Symbol: 'a' Symbol: 'i' Code: 111000
Prior Symbol: 'a' Symbol: 'k' Code: 110000
Prior Symbol: 'a' Symbol: 'l' Code: 1101
Prior Symbol: 'a' Symbol: 'm' Code: 11101
Prior Symbol: 'a' Symbol: 'n' Code: 01
Prior Symbol: 'a' Symbol: 'o' Code: 001100011
Prior Symbol: 'a' Symbol: 'p' Code: 00000
Prior Symbol: 'a' Symbol: 'r' Code: 100
Prior Symbol: 'a' Symbol: 's' Code: 0001
Prior Symbol: 'a' Symbol: 't' Code: 1111
Prior Symbol: 'a' Symbol: 'u' Code: 110001
Prior Symbol: 'a' Symbol: 'v' Code: 001101
Prior Symbol: 'a' Symbol: 'w' Code: 111001111
Prior Symbol: 'a' Symbol: 'x' Code: 111001100
Prior Symbol: 'a' Symbol: 'y' Code: 00001
Prior Symbol: 'a' Symbol: 'z' Code: 00110000
Prior Symbol: 'b' Symbol: 27 Code: 101000
Prior Symbol: 'b' Symbol: ' ' Code: 0101
Prior Symbol: 'b' Symbol: '.' Code: 101001
Prior Symbol: 'b' Symbol: 'a' Code: 100
Prior Symbol: 'b' Symbol: 'b' Code: 101010
Prior Symbol: 'b' Symbol: 'd' Code: 1010110
Prior Symbol: 'b' Symbol: 'e' Code: 00
Prior Symbol: 'b' Symbol: 'i' Code: 1011
Prior Symbol: 'b' Symbol: 'l' Code: 0100
Prior Symbol: 'b' Symbol: 'o' Code: 110
Prior Symbol: 'b' Symbol: 'r' Code: 1110
Prior Symbol: 'b' Symbol: 's' Code: 1010111
Prior Symbol: 'b' Symbol: 'u' Code: 1111
Prior Symbol: 'b' Symbol: 'y' Code: 011
Prior Symbol: 'c' Symbol: 27 Code: 00010
Prior Symbol: 'c' Symbol: ' ' Code: 10000
Prior Symbol: 'c' Symbol: ',' Code: 010000
Prior Symbol: 'c' Symbol: '.' Code: 0100011
Prior Symbol: 'c' Symbol: 'D' Code: 0100110
Prior Symbol: 'c' Symbol: 'a' Code: 110
Prior Symbol: 'c' Symbol: 'c' Code: 010010
Prior Symbol: 'c' Symbol: 'e' Code: 011
Prior Symbol: 'c' Symbol: 'h' Code: 111
Prior Symbol: 'c' Symbol: 'i' Code: 0101
Prior Symbol: 'c' Symbol: 'k' Code: 1001
Prior Symbol: 'c' Symbol: 'l' Code: 10001
Prior Symbol: 'c' Symbol: 'o' Code: 101
Prior Symbol: 'c' Symbol: 'q' Code: 0100010
Prior Symbol: 'c' Symbol: 'r' Code: 00011
Prior Symbol: 'c' Symbol: 't' Code: 001
Prior Symbol: 'c' Symbol: 'u' Code: 0000
Prior Symbol: 'c' Symbol: 'y' Code: 0100111
Prior Symbol: 'd' Symbol: 27 Code: 1010001
Prior Symbol: 'd' Symbol: ' ' Code: 11
Prior Symbol: 'd' Symbol: ''' Code: 01111010
Prior Symbol: 'd' Symbol: ',' Code: 101011
Prior Symbol: 'd' Symbol: '.' Code: 0100
Prior Symbol: 'd' Symbol: ';' Code: 01111011
Prior Symbol: 'd' Symbol: 'a' Code: 1000
Prior Symbol: 'd' Symbol: 'd' Code: 01010
Prior Symbol: 'd' Symbol: 'e' Code: 00
Prior Symbol: 'd' Symbol: 'f' Code: 10100000
Prior Symbol: 'd' Symbol: 'g' Code: 10101011
Prior Symbol: 'd' Symbol: 'i' Code: 1011
Prior Symbol: 'd' Symbol: 'l' Code: 011111
Prior Symbol: 'd' Symbol: 'm' Code: 10100001
Prior Symbol: 'd' Symbol: 'n' Code: 1010100
Prior Symbol: 'd' Symbol: 'o' Code: 0110
Prior Symbol: 'd' Symbol: 'r' Code: 01110
Prior Symbol: 'd' Symbol: 's' Code: 1001
Prior Symbol: 'd' Symbol: 'u' Code: 101001
Prior Symbol: 'd' Symbol: 'v' Code: 0111100
Prior Symbol: 'd' Symbol: 'w' Code: 10101010
Prior Symbol: 'd' Symbol: 'y' Code: 01011
Prior Symbol: 'e' Symbol: 27 Code: 101110011
Prior Symbol: 'e' Symbol: ' ' Code: 111
Prior Symbol: 'e' Symbol: ''' Code: 10111010

Prior Symbol: 'e' Symbol: ')' Code: 100110000
Prior Symbol: 'e' Symbol: ',' Code: 000111
Prior Symbol: 'e' Symbol: '-' Code: 10011001
Prior Symbol: 'e' Symbol: '.' Code: 00110
Prior Symbol: 'e' Symbol: ';' Code: 10011010
Prior Symbol: 'e' Symbol: 'a' Code: 1000
Prior Symbol: 'e' Symbol: 'b' Code: 0001100
Prior Symbol: 'e' Symbol: 'c' Code: 10010
Prior Symbol: 'e' Symbol: 'd' Code: 0000
Prior Symbol: 'e' Symbol: 'e' Code: 10100
Prior Symbol: 'e' Symbol: 'f' Code: 10111011
Prior Symbol: 'e' Symbol: 'g' Code: 0001101
Prior Symbol: 'e' Symbol: 'h' Code: 100110001
Prior Symbol: 'e' Symbol: 'i' Code: 000100
Prior Symbol: 'e' Symbol: 'k' Code: 10011011
Prior Symbol: 'e' Symbol: 'l' Code: 0010
Prior Symbol: 'e' Symbol: 'm' Code: 100111
Prior Symbol: 'e' Symbol: 'n' Code: 010
Prior Symbol: 'e' Symbol: 'o' Code: 001110
Prior Symbol: 'e' Symbol: 'p' Code: 001111
Prior Symbol: 'e' Symbol: 'r' Code: 110
Prior Symbol: 'e' Symbol: 's' Code: 011
Prior Symbol: 'e' Symbol: 't' Code: 10101
Prior Symbol: 'e' Symbol: 'u' Code: 101110010
Prior Symbol: 'e' Symbol: 'v' Code: 101100
Prior Symbol: 'e' Symbol: 'w' Code: 101111
Prior Symbol: 'e' Symbol: 'x' Code: 000101
Prior Symbol: 'e' Symbol: 'y' Code: 101101
Prior Symbol: 'e' Symbol: 'z' Code: 10111000
Prior Symbol: 'f' Symbol: 27 Code: 1110111
Prior Symbol: 'f' Symbol: ' ' Code: 10
Prior Symbol: 'f' Symbol: '.' Code: 1110110
Prior Symbol: 'f' Symbol: 'a' Code: 1111
Prior Symbol: 'f' Symbol: 'e' Code: 000
Prior Symbol: 'f' Symbol: 'f' Code: 0101
Prior Symbol: 'f' Symbol: 'i' Code: 001
Prior Symbol: 'f' Symbol: 'l' Code: 111010
Prior Symbol: 'f' Symbol: 'o' Code: 110
Prior Symbol: 'f' Symbol: 'r' Code: 011
Prior Symbol: 'f' Symbol: 't' Code: 0100
Prior Symbol: 'f' Symbol: 'u' Code: 11100
Prior Symbol: 'g' Symbol: 27 Code: 1111010
Prior Symbol: 'g' Symbol: ' ' Code: 10
Prior Symbol: 'g' Symbol: '"' Code: 1111011
Prior Symbol: 'g' Symbol: ',' Code: 111110
Prior Symbol: 'g' Symbol: '-' Code: 0101010
Prior Symbol: 'g' Symbol: '.' Code: 01011
Prior Symbol: 'g' Symbol: 'a' Code: 1110
Prior Symbol: 'g' Symbol: 'e' Code: 00
Prior Symbol: 'g' Symbol: 'g' Code: 0101011
Prior Symbol: 'g' Symbol: 'h' Code: 011
Prior Symbol: 'g' Symbol: 'i' Code: 1101
Prior Symbol: 'g' Symbol: 'l' Code: 111100
Prior Symbol: 'g' Symbol: 'o' Code: 0100
Prior Symbol: 'g' Symbol: 'r' Code: 111111
Prior Symbol: 'g' Symbol: 's' Code: 11000
Prior Symbol: 'g' Symbol: 'u' Code: 11001
Prior Symbol: 'g' Symbol: 'y' Code: 010100
Prior Symbol: 'h' Symbol: 27 Code: 1011100
Prior Symbol: 'h' Symbol: ' ' Code: 100
Prior Symbol: 'h' Symbol: '"' Code: 10101000
Prior Symbol: 'h' Symbol: ',' Code: 10101001
Prior Symbol: 'h' Symbol: '-' Code: 10101011
Prior Symbol: 'h' Symbol: '.' Code: 101001
Prior Symbol: 'h' Symbol: 'a' Code: 011
Prior Symbol: 'h' Symbol: 'e' Code: 11
Prior Symbol: 'h' Symbol: 'i' Code: 00
Prior Symbol: 'h' Symbol: 'n' Code: 101011
Prior Symbol: 'h' Symbol: 'o' Code: 010
Prior Symbol: 'h' Symbol: 'r' Code: 101111
Prior Symbol: 'h' Symbol: 's' Code: 10101010
Prior Symbol: 'h' Symbol: 't' Code: 10110
Prior Symbol: 'h' Symbol: 'u' Code: 101000
Prior Symbol: 'h' Symbol: 'y' Code: 1011101
Prior Symbol: 'i' Symbol: 27 Code: 00011101
Prior Symbol: 'i' Symbol: ' ' Code: 0001111
Prior Symbol: 'i' Symbol: ',' Code: 100110100
Prior Symbol: 'i' Symbol: '.' Code: 10011000
Prior Symbol: 'i' Symbol: 'a' Code: 11010
Prior Symbol: 'i' Symbol: 'b' Code: 100110101
Prior Symbol: 'i' Symbol: 'c' Code: 1111

Prior Symbol: 'i' Symbol: 'd' Code: 10000
Prior Symbol: 'i' Symbol: 'e' Code: 1110
Prior Symbol: 'i' Symbol: 'f' Code: 100111
Prior Symbol: 'i' Symbol: 'g' Code: 10010
Prior Symbol: 'i' Symbol: 'k' Code: 10011011
Prior Symbol: 'i' Symbol: 'l' Code: 1100
Prior Symbol: 'i' Symbol: 'm' Code: 10001
Prior Symbol: 'i' Symbol: 'n' Code: 01
Prior Symbol: 'i' Symbol: 'o' Code: 11011
Prior Symbol: 'i' Symbol: 'p' Code: 000110
Prior Symbol: 'i' Symbol: 'r' Code: 0000
Prior Symbol: 'i' Symbol: 's' Code: 101
Prior Symbol: 'i' Symbol: 't' Code: 001
Prior Symbol: 'i' Symbol: 'v' Code: 00010
Prior Symbol: 'i' Symbol: 'x' Code: 00011100
Prior Symbol: 'i' Symbol: 'z' Code: 10011001
Prior Symbol: 'j' Symbol: 27 Code: 000
Prior Symbol: 'j' Symbol: 'a' Code: 001
Prior Symbol: 'j' Symbol: 'e' Code: 010
Prior Symbol: 'j' Symbol: 'o' Code: 1
Prior Symbol: 'j' Symbol: 'u' Code: 011
Prior Symbol: 'k' Symbol: 27 Code: 0000
Prior Symbol: 'k' Symbol: ' ' Code: 01
Prior Symbol: 'k' Symbol: '"' Code: 10000
Prior Symbol: 'k' Symbol: ',' Code: 10011
Prior Symbol: 'k' Symbol: '.' Code: 0001
Prior Symbol: 'k' Symbol: 'e' Code: 11
Prior Symbol: 'k' Symbol: 'i' Code: 101
Prior Symbol: 'k' Symbol: 'l' Code: 100100
Prior Symbol: 'k' Symbol: 'n' Code: 10001
Prior Symbol: 'k' Symbol: 's' Code: 001
Prior Symbol: 'k' Symbol: 'y' Code: 100101
Prior Symbol: 'l' Symbol: 27 Code: 0011100
Prior Symbol: 'l' Symbol: ' ' Code: 110
Prior Symbol: 'l' Symbol: '"' Code: 00111100
Prior Symbol: 'l' Symbol: ',' Code: 001101
Prior Symbol: 'l' Symbol: '-' Code: 00111101
Prior Symbol: 'l' Symbol: '.' Code: 00100
Prior Symbol: 'l' Symbol: 'a' Code: 000
Prior Symbol: 'l' Symbol: 'b' Code: 0011101
Prior Symbol: 'l' Symbol: 'c' Code: 00111111
Prior Symbol: 'l' Symbol: 'd' Code: 10111
Prior Symbol: 'l' Symbol: 'e' Code: 111
Prior Symbol: 'l' Symbol: 'f' Code: 010110
Prior Symbol: 'l' Symbol: 'i' Code: 011
Prior Symbol: 'l' Symbol: 'k' Code: 10110110
Prior Symbol: 'l' Symbol: 'l' Code: 100
Prior Symbol: 'l' Symbol: 'm' Code: 010111
Prior Symbol: 'l' Symbol: 'n' Code: 00111110
Prior Symbol: 'l' Symbol: 'o' Code: 1010
Prior Symbol: 'l' Symbol: 'p' Code: 00101
Prior Symbol: 'l' Symbol: 'r' Code: 10110111
Prior Symbol: 'l' Symbol: 's' Code: 01010
Prior Symbol: 'l' Symbol: 't' Code: 001100
Prior Symbol: 'l' Symbol: 'u' Code: 1011010
Prior Symbol: 'l' Symbol: 'v' Code: 101100
Prior Symbol: 'l' Symbol: 'y' Code: 0100
Prior Symbol: 'm' Symbol: 27 Code: 101010
Prior Symbol: 'm' Symbol: ' ' Code: 111
Prior Symbol: 'm' Symbol: '"' Code: 1010110
Prior Symbol: 'm' Symbol: '.' Code: 110101
Prior Symbol: 'm' Symbol: ';' Code: 1010111
Prior Symbol: 'm' Symbol: 'a' Code: 00
Prior Symbol: 'm' Symbol: 'b' Code: 10100
Prior Symbol: 'm' Symbol: 'e' Code: 01
Prior Symbol: 'm' Symbol: 'i' Code: 1100
Prior Symbol: 'm' Symbol: 'm' Code: 10110
Prior Symbol: 'm' Symbol: 'o' Code: 1000
Prior Symbol: 'm' Symbol: 'p' Code: 1001
Prior Symbol: 'm' Symbol: 's' Code: 10111
Prior Symbol: 'm' Symbol: 'u' Code: 11011
Prior Symbol: 'm' Symbol: 'y' Code: 110100
Prior Symbol: 'n' Symbol: 27 Code: 0100000
Prior Symbol: 'n' Symbol: ' ' Code: 10
Prior Symbol: 'n' Symbol: '"' Code: 0100011
Prior Symbol: 'n' Symbol: ',' Code: 111100
Prior Symbol: 'n' Symbol: '-' Code: 011011010
Prior Symbol: 'n' Symbol: '.' Code: 01100
Prior Symbol: 'n' Symbol: ';' Code: 011011011
Prior Symbol: 'n' Symbol: 'a' Code: 11111
Prior Symbol: 'n' Symbol: 'b' Code: 011011100

Prior Symbol: 'n' Symbol: 'c' Code: 01001
Prior Symbol: 'n' Symbol: 'd' Code: 110
Prior Symbol: 'n' Symbol: 'e' Code: 001
Prior Symbol: 'n' Symbol: 'f' Code: 01000101
Prior Symbol: 'n' Symbol: 'g' Code: 000
Prior Symbol: 'n' Symbol: 'i' Code: 01111
Prior Symbol: 'n' Symbol: 'j' Code: 011011101
Prior Symbol: 'n' Symbol: 'k' Code: 1111010
Prior Symbol: 'n' Symbol: 'l' Code: 01101100
Prior Symbol: 'n' Symbol: 'm' Code: 011011110
Prior Symbol: 'n' Symbol: 'n' Code: 01110
Prior Symbol: 'n' Symbol: 'o' Code: 1111011
Prior Symbol: 'n' Symbol: 'r' Code: 011011111
Prior Symbol: 'n' Symbol: 's' Code: 0101
Prior Symbol: 'n' Symbol: 't' Code: 1110
Prior Symbol: 'n' Symbol: 'u' Code: 0100001
Prior Symbol: 'n' Symbol: 'v' Code: 0110100
Prior Symbol: 'n' Symbol: 'y' Code: 0110101
Prior Symbol: 'n' Symbol: 'z' Code: 01000100
Prior Symbol: 'o' Symbol: 27 Code: 101010011
Prior Symbol: 'o' Symbol: ' ' Code: 001
Prior Symbol: 'o' Symbol: ',' Code: 01001111
Prior Symbol: 'o' Symbol: '-' Code: 01001110
Prior Symbol: 'o' Symbol: '.' Code: 0100110
Prior Symbol: 'o' Symbol: 'B' Code: 101010010
Prior Symbol: 'o' Symbol: 'a' Code: 100001
Prior Symbol: 'o' Symbol: 'b' Code: 110111
Prior Symbol: 'o' Symbol: 'c' Code: 100000
Prior Symbol: 'o' Symbol: 'd' Code: 110101
Prior Symbol: 'o' Symbol: 'e' Code: 1010101
Prior Symbol: 'o' Symbol: 'f' Code: 000
Prior Symbol: 'o' Symbol: 'g' Code: 1101000
Prior Symbol: 'o' Symbol: 'h' Code: 1101001
Prior Symbol: 'o' Symbol: 'i' Code: 1101101
Prior Symbol: 'o' Symbol: 'k' Code: 010010
Prior Symbol: 'o' Symbol: 'l' Code: 0101
Prior Symbol: 'o' Symbol: 'm' Code: 1100
Prior Symbol: 'o' Symbol: 'n' Code: 111
Prior Symbol: 'o' Symbol: 'o' Code: 10100
Prior Symbol: 'o' Symbol: 'p' Code: 01000
Prior Symbol: 'o' Symbol: 'r' Code: 011
Prior Symbol: 'o' Symbol: 's' Code: 10001
Prior Symbol: 'o' Symbol: 't' Code: 10010
Prior Symbol: 'o' Symbol: 'u' Code: 1011
Prior Symbol: 'o' Symbol: 'v' Code: 101011
Prior Symbol: 'o' Symbol: 'w' Code: 10011
Prior Symbol: 'o' Symbol: 'x' Code: 10101000
Prior Symbol: 'o' Symbol: 'y' Code: 1101100
Prior Symbol: 'p' Symbol: 27 Code: 011011
Prior Symbol: 'p' Symbol: ' ' Code: 000
Prior Symbol: 'p' Symbol: '-' Code: 1010010
Prior Symbol: 'p' Symbol: '.' Code: 101000
Prior Symbol: 'p' Symbol: 'a' Code: 001
Prior Symbol: 'p' Symbol: 'e' Code: 110
Prior Symbol: 'p' Symbol: 'h' Code: 1111
Prior Symbol: 'p' Symbol: 'i' Code: 1011
Prior Symbol: 'p' Symbol: 'l' Code: 010
Prior Symbol: 'p' Symbol: 'm' Code: 1010011
Prior Symbol: 'p' Symbol: 'o' Code: 0111
Prior Symbol: 'p' Symbol: 'p' Code: 11101
Prior Symbol: 'p' Symbol: 'r' Code: 100
Prior Symbol: 'p' Symbol: 's' Code: 01100
Prior Symbol: 'p' Symbol: 't' Code: 11100
Prior Symbol: 'p' Symbol: 'u' Code: 10101
Prior Symbol: 'p' Symbol: 'y' Code: 011010
Prior Symbol: 'q' Symbol: 27 Code: 0
Prior Symbol: 'q' Symbol: 'u' Code: 1
Prior Symbol: 'r' Symbol: 27 Code: 10011111
Prior Symbol: 'r' Symbol: ' ' Code: 111
Prior Symbol: 'r' Symbol: '"' Code: 1001110
Prior Symbol: 'r' Symbol: ')' Code: 100111100
Prior Symbol: 'r' Symbol: ',' Code: 100100
Prior Symbol: 'r' Symbol: '-' Code: 11001100
Prior Symbol: 'r' Symbol: '.' Code: 10001
Prior Symbol: 'r' Symbol: ';' Code: 100111101
Prior Symbol: 'r' Symbol: 'a' Code: 1101
Prior Symbol: 'r' Symbol: 'b' Code: 11001101
Prior Symbol: 'r' Symbol: 'c' Code: 100001
Prior Symbol: 'r' Symbol: 'd' Code: 11000
Prior Symbol: 'r' Symbol: 'e' Code: 101
Prior Symbol: 'r' Symbol: 'f' Code: 110011111

Prior Symbol: 'r' Symbol: 'g' Code: 100101
Prior Symbol: 'r' Symbol: 'i' Code: 010
Prior Symbol: 'r' Symbol: 'k' Code: 110010
Prior Symbol: 'r' Symbol: 'l' Code: 00100
Prior Symbol: 'r' Symbol: 'm' Code: 00101
Prior Symbol: 'r' Symbol: 'n' Code: 01100
Prior Symbol: 'r' Symbol: 'o' Code: 000
Prior Symbol: 'r' Symbol: 'p' Code: 11001110
Prior Symbol: 'r' Symbol: 'r' Code: 100110
Prior Symbol: 'r' Symbol: 's' Code: 0111
Prior Symbol: 'r' Symbol: 't' Code: 0011
Prior Symbol: 'r' Symbol: 'u' Code: 100000
Prior Symbol: 'r' Symbol: 'v' Code: 110011110
Prior Symbol: 'r' Symbol: 'y' Code: 01101
Prior Symbol: 's' Symbol: 27 Code: 10011100
Prior Symbol: 's' Symbol: ' ' Code: 0
Prior Symbol: 's' Symbol: '"' Code: 100111100
Prior Symbol: 's' Symbol: ''' Code: 100111101
Prior Symbol: 's' Symbol: ',' Code: 111011
Prior Symbol: 's' Symbol: '.' Code: 1000
Prior Symbol: 's' Symbol: ';' Code: 11101011
Prior Symbol: 's' Symbol: 'a' Code: 110011
Prior Symbol: 's' Symbol: 'b' Code: 100111110
Prior Symbol: 's' Symbol: 'c' Code: 10010
Prior Symbol: 's' Symbol: 'e' Code: 1101
Prior Symbol: 's' Symbol: 'h' Code: 11000
Prior Symbol: 's' Symbol: 'i' Code: 11100
Prior Symbol: 's' Symbol: 'k' Code: 100111111
Prior Symbol: 's' Symbol: 'l' Code: 1110100
Prior Symbol: 's' Symbol: 'm' Code: 111010100
Prior Symbol: 's' Symbol: 'n' Code: 111010101
Prior Symbol: 's' Symbol: 'o' Code: 11110
Prior Symbol: 's' Symbol: 'p' Code: 1001101
Prior Symbol: 's' Symbol: 's' Code: 11111
Prior Symbol: 's' Symbol: 't' Code: 101
Prior Symbol: 's' Symbol: 'u' Code: 110010
Prior Symbol: 's' Symbol: 'w' Code: 10011101
Prior Symbol: 's' Symbol: 'y' Code: 1001100
Prior Symbol: 't' Symbol: 27 Code: 11000011
Prior Symbol: 't' Symbol: ' ' Code: 111
Prior Symbol: 't' Symbol: ''' Code: 11000100
Prior Symbol: 't' Symbol: ',' Code: 0111100
Prior Symbol: 't' Symbol: '-' Code: 01111110
Prior Symbol: 't' Symbol: '.' Code: 01101
Prior Symbol: 't' Symbol: ';' Code: 110000100
Prior Symbol: 't' Symbol: 'a' Code: 0100
Prior Symbol: 't' Symbol: 'b' Code: 110000101
Prior Symbol: 't' Symbol: 'c' Code: 11000101
Prior Symbol: 't' Symbol: 'e' Code: 101
Prior Symbol: 't' Symbol: 'h' Code: 00
Prior Symbol: 't' Symbol: 'i' Code: 1101
Prior Symbol: 't' Symbol: 'l' Code: 0111101
Prior Symbol: 't' Symbol: 'm' Code: 01111111
Prior Symbol: 't' Symbol: 'n' Code: 0111110
Prior Symbol: 't' Symbol: 'o' Code: 100
Prior Symbol: 't' Symbol: 'r' Code: 11001
Prior Symbol: 't' Symbol: 's' Code: 0101
Prior Symbol: 't' Symbol: 't' Code: 01100
Prior Symbol: 't' Symbol: 'u' Code: 01110
Prior Symbol: 't' Symbol: 'w' Code: 1100000
Prior Symbol: 't' Symbol: 'y' Code: 1100011
Prior Symbol: 'u' Symbol: 27 Code: 1001100
Prior Symbol: 'u' Symbol: ' ' Code: 100000
Prior Symbol: 'u' Symbol: 'a' Code: 100111
Prior Symbol: 'u' Symbol: 'b' Code: 100001
Prior Symbol: 'u' Symbol: 'c' Code: 10001
Prior Symbol: 'u' Symbol: 'd' Code: 11100
Prior Symbol: 'u' Symbol: 'e' Code: 11101
Prior Symbol: 'u' Symbol: 'g' Code: 11110
Prior Symbol: 'u' Symbol: 'i' Code: 10010
Prior Symbol: 'u' Symbol: 'k' Code: 1001101
Prior Symbol: 'u' Symbol: 'l' Code: 0100
Prior Symbol: 'u' Symbol: 'm' Code: 111111
Prior Symbol: 'u' Symbol: 'n' Code: 110
Prior Symbol: 'u' Symbol: 'o' Code: 11111010
Prior Symbol: 'u' Symbol: 'p' Code: 0101
Prior Symbol: 'u' Symbol: 'r' Code: 00
Prior Symbol: 'u' Symbol: 's' Code: 011
Prior Symbol: 'u' Symbol: 't' Code: 101
Prior Symbol: 'u' Symbol: 'v' Code: 11111011
Prior Symbol: 'u' Symbol: 'y' Code: 1111100
Prior Symbol: 'v' Symbol: 27 Code: 00010
Prior Symbol: 'v' Symbol: 'a' Code: 001
Prior Symbol: 'v' Symbol: 'e' Code: 1
Prior Symbol: 'v' Symbol: 'i' Code: 01
Prior Symbol: 'v' Symbol: 'o' Code: 0000
Prior Symbol: 'v' Symbol: 's' Code: 000110
Prior Symbol: 'v' Symbol: 'y' Code: 000111
Prior Symbol: 'w' Symbol: 27 Code: 011101
Prior Symbol: 'w' Symbol: ' ' Code: 001
Prior Symbol: 'w' Symbol: '.' Code: 011100
Prior Symbol: 'w' Symbol: 'a' Code: 010
Prior Symbol: 'w' Symbol: 'e' Code: 1110
Prior Symbol: 'w' Symbol: 'h' Code: 000
Prior Symbol: 'w' Symbol: 'i' Code: 10
Prior Symbol: 'w' Symbol: 'l' Code: 011110
Prior Symbol: 'w' Symbol: 'm' Code: 011111
Prior Symbol: 'w' Symbol: 'n' Code: 11111
Prior Symbol: 'w' Symbol: 'o' Code: 110
Prior Symbol: 'w' Symbol: 'r' Code: 0110
Prior Symbol: 'w' Symbol: 's' Code: 11110
Prior Symbol: 'x' Symbol: 27 Code: 10
Prior Symbol: 'x' Symbol: ' ' Code: 0110
Prior Symbol: 'x' Symbol: ',' Code: 0111
Prior Symbol: 'x' Symbol: '-' Code: 1100
Prior Symbol: 'x' Symbol: 'a' Code: 111
Prior Symbol: 'x' Symbol: 'e' Code: 00
Prior Symbol: 'x' Symbol: 'i' Code: 010
Prior Symbol: 'x' Symbol: 't' Code: 1101
Prior Symbol: 'y' Symbol: 27 Code: 01010
Prior Symbol: 'y' Symbol: ' ' Code: 1
Prior Symbol: 'y' Symbol: ''' Code: 010010
Prior Symbol: 'y' Symbol: ',' Code: 0001
Prior Symbol: 'y' Symbol: '.' Code: 0111
Prior Symbol: 'y' Symbol: ';' Code: 011001
Prior Symbol: 'y' Symbol: '?' Code: 0100110
Prior Symbol: 'y' Symbol: 'a' Code: 0100111
Prior Symbol: 'y' Symbol: 'b' Code: 0110000
Prior Symbol: 'y' Symbol: 'd' Code: 000001
Prior Symbol: 'y' Symbol: 'e' Code: 0010
Prior Symbol: 'y' Symbol: 'f' Code: 0110001
Prior Symbol: 'y' Symbol: 'i' Code: 000010
Prior Symbol: 'y' Symbol: 'l' Code: 01000
Prior Symbol: 'y' Symbol: 'm' Code: 000000
Prior Symbol: 'y' Symbol: 'n' Code: 01011
Prior Symbol: 'y' Symbol: 'o' Code: 01101
Prior Symbol: 'y' Symbol: 's' Code: 0011
Prior Symbol: 'y' Symbol: 'w' Code: 000011
Prior Symbol: 'z' Symbol: 27 Code: 100
Prior Symbol: 'z' Symbol: ' ' Code: 1110
Prior Symbol: 'z' Symbol: '.' Code: 1111
Prior Symbol: 'z' Symbol: 'a' Code: 000
Prior Symbol: 'z' Symbol: 'e' Code: 001
Prior Symbol: 'z' Symbol: 'i' Code: 110
Prior Symbol: 'z' Symbol: 'l' Code: 010
Prior Symbol: 'z' Symbol: 'o' Code: 101
Prior Symbol: 'z' Symbol: 'z' Code: 011
Prior Symbol: '{' Symbol: 27 Code: 1
Prior Symbol: '|' Symbol: 27 Code: 1
Prior Symbol: '}' Symbol: 27 Code: 1
Prior Symbol: '~' Symbol: 27 Code: 1
Prior Symbol: 127 Symbol: 27 Code: 1

Table C.7 English-language Program Description Decode Table

0	1	79	242	158	3	237	134	316	155	395	197	474	52
1	0	80	1	159	16	238	6	317	155	396	198	475	53
2	1	81	248	160	3	239	146	318	155	397	177	476	54
3	44	82	1	161	26	240	6	319	155	398	10	477	55
4	1	83	250	162	3	241	170	320	155	399	238	478	155
5	48	84	1	183	40	242	6	321	155	400	203	479	155
6	1	85	252	164	3	243	184	322	155	401	11	480	3
7	48	88	1	165	42	244	6	323	155	402	212	481	4
8	1	87	254	166	3	245	220	324	155	403	12	482	128
9	50	88	2	167	52	246	6	325	155	404	196	483	174
10	1	89	0	168	3	247	236	326	155	405	200	484	200
11	52	90	2	169	74	248	6	327	155	406	210	485	212
12	1	91	4	170	3	249	238	328	155	407	13	486	1
13	54	92	2	171	90	250	6	329	155	408	14	487	2
14	1	93	22	172	3	251	240	330	155	409	15	488	155
15	56	94	2	173	94	252	6	331	155	410	199	489	160
16	1	95	32	174	3	253	242	332	155	411	202	490	155
17	58	96	2	175	100	254	6	333	155	412	206	491	155
18	1	97	34	176	3	255	244	334	155	413	208	492	155
19	60	98	2	177	110	256	20	335	155	414	215	493	155
20	1	99	44	178	3	257	21	336	155	415	16	494	155
21	62	100	2	179	112	258	155	337	155	416	194	495	155
22	1	101	50	180	3	259	214	338	155	417	17	496	155
23	64	102	2	181	114	260	201	339	155	418	204	497	155
24	1	103	56	182	3	261	207	340	155	419	236	498	2
25	66	104	2	183	116	262	215	341	155	420	229	499	243
26	1	105	60	184	3	263	199	342	155	421	231	500	160
27	68	106	2	185	118	264	1	343	155	422	18	501	244
28	1	107	64	188	3	265	162	344	155	423	205	502	155
29	70	108	2	187	120	266	206	345	155	424	19	503	1
30	1	109	68	188	3	267	203	346	155	425	20	504	155
31	72	110	2	189	122	268	2	347	155	426	195	505	155
32	1	111	70	190	3	269	3	348	155	427	21	506	172
33	74	112	2	191	124	270	197	349	155	428	22	507	155
34	1	113	74	192	3	271	204	350	155	429	23	508	155
35	76	114	2	193	126	272	198	351	155	430	237	509	155
36	1	115	76	194	3	273	200	352	155	431	24	510	155
37	78	116	2	195	128	274	4	353	155	432	25	511	155
38	1	117	84	196	3	275	196	354	155	433	242	512	1
39	80	118	2	197	180	276	5	355	155	434	26	513	160
40	1	119	88	198	3	277	194	356	155	435	211	514	155
41	82	120	2	199	206	278	6	357	155	436	27	515	162
42	1	121	88	200	3	279	195	356	155	437	28	516	7
43	84	122	2	201	240	280	210	359	155	438	228	517	8
44	1	123	90	202	4	281	7	360	155	439	29	518	226
45	86	124	2	203	26	282	211	361	155	440	193	519	228
46	1	125	92	204	4	283	8	362	56	441	227	520	229
47	88	126	2	205	88	284	202	363	57	442	30	521	230
48	1	127	94	206	4	285	212	364	173	443	233	522	160
49	90	128	2	207	110	288	9	365	175	444	240	523	242
50	1	129	96	208	4	287	205	366	183	445	226	524	225
51	92	130	2	209	142	288	208	367	218	446	247	525	1
52	1	131	98	210	4	289	10	366	168	447	31	526	2
53	94	132	2	211	172	290	193	369	179	446	243	527	243
54	1	133	118	212	4	291	11	370	181	449	230	528	227
55	96	134	2	213	216	292	12	371	1	450	32	529	3
56	1	135	132	214	4	293	13	372	2	451	33	530	4
57	98	136	2	215	224	294	14	373	155	452	34	531	5
58	1	137	148	216	4	295	15	374	180	453	232	532	155
59	100	138	2	217	244	296	16	375	241	454	239	533	6
60	1	139	162	218	5	297	17	376	162	455	35	534	4
61	102	140	2	219	36	298	18	377	213	456	36	535	128
62	1	141	178	220	5	299	19	378	214	457	37	536	202
63	104	142	2	221	64	300	155	379	217	458	38	537	211
64	1	143	188	222	5	301	155	380	3	459	39	538	162
65	106	144	2	223	118	302	155	381	4	460	40	539	1
66	1	145	200	224	5	303	155	382	5	461	41	540	155
67	222	146	2	225	174	304	155	383	207	462	42	541	2
68	1	147	210	226	5	305	155	384	6	463	244	542	3
69	224	148	2	227	206	306	155	385	201	464	43	543	160
70	1	149	222	228	5	307	155	386	249	465	44	544	155
71	234	150	2	229	208	308	155	387	234	466	45	545	160
72	1	151	234	230	6	309	155	388	235	467	46	546	3
73	236	152	2	231	6	310	155	389	245	468	47	547	4
74	1	153	242	232	6	311	155	390	246	469	225	548	155
75	238	154	2	233	52	312	155	391	7	470	48	549	183
76	1	155	252	234	6	313	155	392	8	471	49	550	244
77	240	156	3	235	96	314	155	393	9	472	50	551	160
78	1	157	8	236	6	315	155	394	178	473	51	552	176

553	243	634	245	715	229	796	155	877	2	958	236	1039	243
554	1	635	1	716	233	797	232	878	155	959	160	1040	12
555	2	636	2	717	245	798	233	879	155	960	4	1041	233
556	165	637	225	718	225	799	1	880	155	961	233	1042	13
557	2	638	239	719	1	800	242	881	239	962	242	1043	14
558	184	639	229	720	239	801	236	882	155	963	245	1044	15
559	155	640	233	721	2	802	2	883	155	964	5	1045	16
560	160	641	242	722	4	803	239	684	155	965	249	1046	229
561	1	642	3	723	5	804	3	885	155	966	225	1047	17
562	174	643	4	724	160	805	229	886	155	967	6	1048	18
563	2	644	6	725	201	806	4	887	155	968	239	1049	160
564	182	645	7	726	243	807	5	888	155	969	7	1050	29
565	155	846	155	727	155	808	155	889	155	970	229	1051	30
566	1	647	233	728	174	809	155	890	155	971	8	1052	189
567	160	648	249	729	242	810	3	891	155	972	9	1053	232
568	160	649	242	730	1	811	4	892	155	973	10	1054	245
569	1	650	245	731	2	812	155	893	155	974	15	1055	155
570	155	651	1	732	3	813	174	894	155	975	16	1056	1
571	176	652	2	733	238	814	1	895	155	976	241	1057	173
572	174	653	3	734	239	815	233	896	24	977	174	1058	187
573	1	654	236	735	5	816	2	897	25	978	196	1059	235
574	155	655	239	736	155	817	225	898	232	979	249	1060	250
575	160	656	225	737	174	818	229	899	239	980	172	1061	2
576	174	657	4	738	233	819	239	900	248	981	1	1062	167
577	1	656	232	739	229	820	9	901	155	982	227	1063	230
578	160	659	5	740	1	821	10	902	167	983	2	1064	226
579	155	660	5	741	245	822	246	903	247	984	155	1065	231
580	155	661	6	742	2	823	249	904	250	985	242	1066	3
581	155	662	249	743	225	824	1	905	1	986	3	1067	4
582	155	663	242	744	3	825	174	906	2	987	4	1068	5
583	1	684	245	745	4	826	227	907	3	988	160	1069	6
584	172	665	155	746	229	827	233	908	4	989	236	1070	233
585	174	666	229	747	3	828	245	909	229	990	245	1071	248
586	155	667	239	748	225	829	155	910	174	991	5	1072	7
587	155	668	1	749	233	830	229	911	5	992	6	1073	172
588	2	669	2	750	242	831	239	912	230	993	233	1074	239
589	3	670	233	751	155	832	2	913	226	994	7	1075	240
590	155	671	225	752	1	833	3	914	6	995	235	1076	8
591	160	672	3	753	2	834	225	915	246	996	8	1077	237
592	181	673	4	754	3	835	4	916	235	997	244	1078	246
593	182	674	6	755	4	836	232	917	245	998	9	1079	249
594	184	675	7	756	155	837	5	918	233	999	229	1080	9
595	1	676	225	757	233	838	6	919	7	1000	10	1081	247
596	155	677	233	758	245	839	244	920	240	1001	239	1082	10
597	160	678	238	759	1	840	7	921	249	1002	225	1083	11
598	155	679	246	760	229	841	8	922	231	1003	232	1084	174
599	160	680	228	761	2	842	232	923	8	1004	11	1085	12
800	155	681	236	762	239	843	7	924	9	1005	12	1088	227
601	155	882	243	763	225	844	229	925	228	1006	13	1087	13
602	155	683	1	764	225	845	247	926	10	1007	14	1088	229
603	155	684	2	765	5	846	214	927	227	1008	19	1089	244
604	155	885	242	766	155	847	225	928	11	1009	20	1090	14
605	155	686	3	767	227	846	155	929	237	1010	167	1091	15
606	155	687	4	768	239	849	233	930	12	1011	187	1092	228
607	160	688	155	769	1	850	242	931	243	1012	230	1093	16
608	155	689	5	770	245	851	1	932	13	1013	237	1094	236
609	155	690	2	771	229	852	2	933	14	1014	247	1095	17
610	8	691	3	772	2	853	3	934	15	1015	231	1096	225
611	9	692	229	773	3	854	4	935	236	1016	246	1097	18
612	230	693	236	774	233	855	239	936	16	1017	1	1098	19
613	245	694	155	775	4	856	5	937	244	1018	2	1099	20
614	243	695	239	776	229	857	6	938	17	1019	155	1100	21
615	244	696	1	777	3	858	174	939	18	1020	238	1101	22
616	155	697	242	778	155	859	1	940	242	1021	3	1102	238
617	228	698	5	779	233	860	155	941	160	1022	4	1103	243
618	1	699	6	780	1	861	238	942	19	1023	236	1104	23
619	237	700	245	781	225	862	233	943	20	1024	5	1105	24
620	2	701	239	782	239	863	2	944	21	1025	245	1106	242
621	3	702	155	783	2	864	229	945	238	1026	6	1107	160
622	4	703	236	784	3	865	155	946	22	1027	172	1108	25
623	242	704	233	785	4	866	160	947	23	1028	228	1109	26
624	5	705	1	786	167	867	1	948	11	1029	249	1110	27
625	6	706	225	787	238	868	3	949	12	1030	242	1111	28
626	236	707	242	788	236	869	4	950	228	1031	7	1112	9
627	238	708	2	789	242	870	155	951	243	1032	8	1113	10
628	7	709	229	790	243	871	232	952	155	1033	9	1114	174
629	160	710	3	791	1	872	229	953	174	1034	174	1115	155
630	5	711	4	792	155	873	225	954	226	1035	10	1116	236
631	6	712	3	793	2	874	239	955	1	1036	239	1117	1
632	155	713	4	794	225	875	1	956	2	1037	11	1118	245
633	236	714	155	795	6	876	233	957	3	1038	225	1119	2

1120	244	1201	155	1282	244	1363	249	1444	18	1525	243	1606	5
1121	230	1202	174	1283	172	1364	5	1445	242	1526	14	1607	6
1122	3	1203	250	1284	4	1365	6	1446	19	1527	15	1608	7
1123	225	1204	1	1285	5	1366	235	1447	20	1528	16	1609	8
1124	229	1205	235	1286	230	1367	239	1446	21	1529	225	1610	244
1125	233	1206	2	1287	237	1368	7	1449	238	1530	239	1611	174
1126	4	1207	160	1288	246	1369	8	1450	22	1531	17	1612	245
1127	242	1208	3	1289	6	1370	9	1451	23	1532	233	1613	9
1128	239	1209	4	1290	174	1371	10	1452	24	1533	18	1614	10
1129	5	1210	240	1291	240	1372	172	1453	25	1534	19	1615	242
1130	6	1211	5	1292	7	1373	11	1454	14	1535	229	1616	225
1131	7	1212	6	1293	8	1374	12	1455	15	1536	20	1617	243
1132	160	1213	230	1294	243	1375	227	1456	173	1537	160	1618	11
1133	8	1214	246	1295	9	1376	174	1457	237	1538	21	1619	12
1134	14	1215	7	1296	10	1377	13	1458	249	1539	22	1620	13
1135	15	1216	228	1297	228	1378	238	1459	155	1540	23	1621	233
1136	173	1217	237	1298	11	1379	233	1460	174	1541	24	1622	14
1137	231	1218	231	1299	12	1380	14	1461	1	1542	160	1623	15
1138	155	1219	8	1300	249	1381	225	1462	243	1543	22	1624	239
1139	167	1220	225	1301	13	1382	15	1463	2	1544	162	1625	229
1140	249	1221	239	1302	239	1383	243	1464	3	1545	167	1626	16
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1142	236	1223	9	1304	225	1385	17	1466	244	1547	235	1628	232
1143	2	1224	10	1305	15	1386	244	1467	240	1546	237	1629	17
1144	172	1225	11	1306	16	1387	18	1468	4	1549	238	1630	18
1145	242	1226	236	1307	233	1386	231	1469	239	1550	155	1631	19
1146	3	1227	12	1308	236	1389	229	1470	5	1551	247	1632	17
1147	174	1228	229	1309	17	1390	19	1471	233	1552	1	1633	18
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1149	245	1230	13	1311	229	1392	228	1473	232	1554	3	1635	246
1150	4	1231	244	1312	18	1393	21	1474	160	1555	187	1636	155
1151	5	1232	14	1313	19	1394	22	1475	225	1556	249	1637	235
1152	239	1233	243	1314	20	1395	23	1476	236	1557	240	1638	249
1153	6	1234	15	1315	21	1396	160	1477	7	1558	4	1639	1
1154	7	1235	16	1316	12	1397	24	1478	242	1559	5	1640	160
1155	233	1236	17	1317	13	1398	26	1479	8	1560	236	1641	226
1156	225	1237	238	1318	167	1399	27	1460	229	1561	6	1642	2
1157	8	1238	18	1319	187	1400	194	1481	9	1562	7	1643	225
1158	9	1239	19	1320	155	1401	155	1482	10	1563	8	1644	3
1159	232	1240	3	1321	1	1402	173	1483	11	1564	245	1645	237
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1162	229	1243	225	1324	226	1405	1	1486	155	1567	172	1648	233
1163	12	1244	229	1325	2	1406	174	1487	245	1568	227	1649	5
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1166	13	1247	2	1328	3	1409	229	1490	169	1571	11	1652	231
1167	14	1248	8	1329	245	1410	231	1491	187	1572	233	1653	6
1168	167	1249	9	1330	239	1411	232	1492	246	1573	12	1654	236
1169	172	1250	236	1331	240	1412	249	1493	230	1574	239	1655	240
1170	243	1251	249	1332	4	1413	233	1494	1	1575	243	1656	7
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1172	1	1253	238	1334	233	1415	4	1496	173	1577	13	1658	9
1173	2	1254	1	1335	6	1416	227	1497	226	1578	14	1659	10
1174	155	1255	172	1336	7	1417	225	1498	240	1579	229	1660	11
1175	249	1256	155	1337	8	1418	5	1499	2	1580	15	1661	243
1176	245	1257	174	1338	9	1419	246	1500	167	1581	16	1662	12
1177	174	1258	2	1339	160	1420	6	1501	3	1582	17	1663	244
1178	3	1259	3	1340	225	1421	228	1502	4	1583	244	1664	238
1179	238	1260	4	1341	229	1422	7	1503	5	1584	18	1665	13
1180	4	1261	243	1342	10	1423	226	1504	245	1585	19	1666	242
1181	242	1262	5	1343	11	1424	240	1505	227	1586	20	1667	14
1182	5	1263	233	1344	25	1425	8	1506	172	1587	21	1668	15
1163	6	1264	6	1345	26	1426	9	1507	231	1588	20	1669	16
1184	244	1265	160	1346	173	1427	243	1508	242	1589	21	1670	5
1185	7	1266	7	1347	187	1428	244	1509	6	1590	187	1671	229
1188	8	1267	229	1348	226	1429	247	1510	235	1591	226	1672	243
1187	9	1268	22	1349	234	1430	239	1511	7	1592	173	1673	249
1188	239	1269	23	1350	237	1431	10	1512	236	1593	237	1674	155
1189	225	1270	167	1351	242	1432	11	1513	237	1594	1	1675	1
1190	160	1271	173	1352	250	1433	12	1514	238	1595	155	1676	239
1191	10	1272	238	1353	230	1434	13	1515	249	1596	167	1677	2
1192	233	1273	227	1354	236	1435	236	1516	8	1597	227	1678	3
1193	11	1274	235	1355	1	1436	14	1517	174	1598	172	1679	225
1194	12	1275	242	1356	2	1437	15	1518	9	1599	236	1680	4
1195	229	1276	155	1357	3	1438	16	1519	10	1600	238	1661	233
1196	20	1277	226	1358	155	1439	245	1520	228	1601	2	1682	10
1197	21	1278	1	1359	245	1440	237	1521	11	1602	247	1683	11
1198	172	1279	2	1360	4	1441	17	1522	12	1603	3	1684	174
1199	226	1280	245	1361	167	1442	230	1523	244	1604	4	1685	155
1200	248	1281	3	1362	246	1443	160	1524	13	1605	249	1686	236

1687	237
1688	1
1689	2
1690	243
1691	238
1692	242
1693	3
1694	229
1695	4
1696	232
1697	160
1698	225
1699	5
1700	239
1701	6
1702	7
1703	8
1704	233
1705	9
1706	5
1707	6
1708	160
1709	172
1710	173
1711	244
1712	233
1713	1
1714	2
1715	225
1716	229
1717	3
1718	155
1719	4
1720	17
1721	160
1722	191
1723	225
1724	226
1725	230
1726	237
1727	228
1728	233
1729	247
1730	167
1731	1
1732	2
1733	187
1734	3
1735	4
1736	236
1737	5
1738	155
1739	238
1740	6
1741	239
1742	7
1743	172
1744	229
1745	243
1746	8
1747	9
1748	10
1749	174
1750	11
1751	12
1752	13
1753	14
1754	15
1755	16
1756	6
1757	7
1758	160
1759	174
1760	225
1761	229
1762	236
1763	250
1764	155
1765	239
1766	233
1767	1
1768	2
1769	3
1770	4
1771	5
1772	155
1773	155
1774	155
1775	155
1776	155
1777	155
1778	155
1779	155
1780	155
1781	155

ANNEX D

(Informative)

AN OVERVIEW OF PSIP FOR TERRESTRIAL BROADCAST WITH APPLICATION EXAMPLES

The Program and System Information Protocol (PSIP) is a small collection of tables designed to operate within every Transport Stream for terrestrial broadcast of digital TV. Its purpose is to describe the information at the system and event levels for all virtual channels carried in a particular Transport Stream. Additionally, information for analog channels as well as digital channels from other Transport Streams may be incorporated. The relational hierarchy for the component tables is explained through typical application examples in this document.

PSIP is the result of combining and compacting two existing optional ATSC protocols: A/55 and A/56. Although these protocols were individually efficient and accomplished their purpose, their mutual implementation was difficult due to their structural differences and their overlapping definitions. PSIP solves this problem. The tables defined in PSIP use packet identifiers (PIDs) that are different from those specified by the optional A/55 and A/56 standards. This provision has been included to enable the operation of existing equipment designed or manufactured to support A/55 and/or A/56.

D1. INTRODUCTION

Under the adopted ATSC standard for digital TV, the typical 6 MHz channel used for analog TV broadcast supports about 19 Mbps of throughput for terrestrial broadcast. Since audiovisual signals with standard resolution can be compressed using MPEG-2 to sustainable rates of around 6 Mbps, then around 3 or 4 digital TV channels can be safely supported in a single physical channel without congestion. Moreover, enough bandwidth remains within the same Transport Stream to provide several additional low-bandwidth non-conventional services such as: weather reports, stock indices, headline news, software download (for games or enhanced applications), image-driven classified ads, home shopping, pay-per-view information, and others.

It is therefore practical to anticipate that in the future, the list of services (virtual channels) carried in a physical transmission channel (6 MHz of bandwidth for the U.S.) may easily reach ten or more. What is even more important is that the number and type of services may also change continuously, thus becoming a more dynamic medium than what we have today.

An important feature of terrestrial broadcasting is that sources follow a distributed information model rather than a centralized one. Unlike cable or satellite, service providers are geographically distributed and have no interaction with respect to data unification or even synchronization. It is therefore necessary to develop a protocol for describing system information and event descriptions which is followed by every organization in charge of a physical transmission channel. System information allows navigation and access to each of the channels within the Transport Stream, whereas event descriptions give the user content information for browsing and selection.

In this document we describe the development of a transport-based implementation of the PSIP protocol using examples. Our hope is to introduce the reader to the most important concepts and components that constitute the protocol.

D2. ELEMENTS OF PSIP

PSIP is a collection of hierarchically-associated tables each of which describes particular elements of typical digital TV services. Figures D.1 and D.2 show the different components and the notation used to describe them. The packets of the base tables are all labeled with the base PID (base_PID) which has been chosen as 0x1FFB. The base tables are: the System Time Table (STT), the Rating Region Table (RRT), the Master Guide Table (MGT) and the Virtual Channel Table (VCT).

A second set of tables are the Event Information Tables (EIT) whose packet identifiers (PIDs) are defined in the MGT. A third set of tables are the Extended Text Tables (ETT), and similarly, their packet identifiers (PIDs) are defined in the MGT.

The System Time Table (STT) is a small data structure that fits in one packet and serves as a reference for time of day. Receivers can use this table as a reference for timing start times of advertised events.

Transmission syntax for the United States' voluntary program rating system is included in this standard. The Rating Region Table (RRT) has been designed to transmit the rating standard in use for each country using the standard. Provisions were made for different rating systems for different countries and multi-country regions as well..

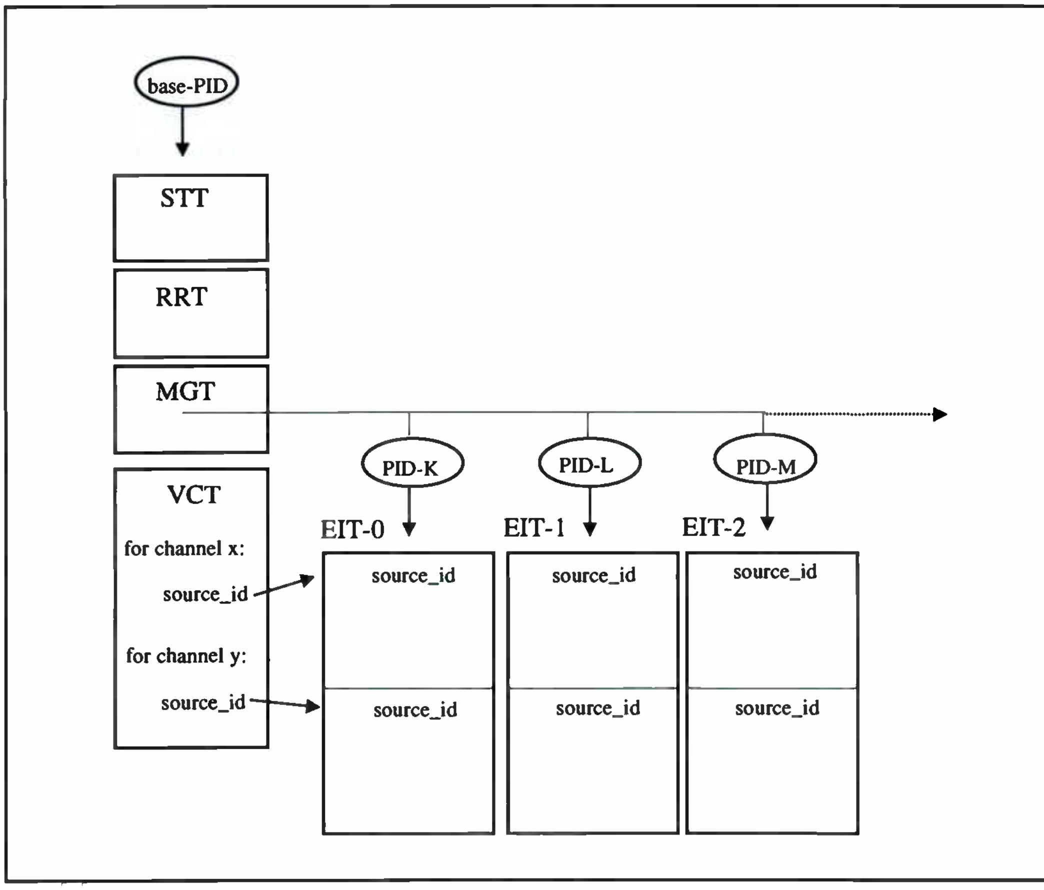

Figure D.1 Main Structure for the PSIP tables

The Master Guide Table (MGT) provides general information about all of the other tables that comprise the PSIP standard. It defines table sizes necessary for memory allocation during decoding; it defines version numbers to identify those tables that need to be updated; and it gives the packet identifiers (PIDs) that label the tables.

The Virtual Channel Table (VCT), also referred to as the Terrestrial VCT (TVCT), contains a list of all the channels that are or will be on-line plus their attributes. Among the attributes we have the channel name, navigation identifiers, stream components and types, etc.

As part of PSIP there are several Event Information Tables, each of which describes the events or TV programs associated with each of the virtual channels listed in the VCT. Each EIT is valid for a time interval of 3 hours. Since the total number of EITs is 128, up to 16 days of programming may be advertised in advance. EIT-0 always denotes the current 3 hours of programming, EIT-1 the next 3 hours, and so on. As a minimum, the first four EITs must always be present in every Transport Stream

Start times for EITs are constrained to be one of the following UTC times: 0:00 (midnight), 3:00, 6:00, 9:00, 12:00 (noon), 15:00, 18:00, and 21:00. Imposing constraints on the start times as well as the interval duration is necessary for the purpose of re-multiplexing. During re-multiplexing, EIT tables coming from several distinct Transport Streams may end up grouped together or *vice versa*. If no constraints were imposed, re-multiplexing equipment would have to parse EITs by content in real time, which is a difficult task.

For example, consider a broadcast corporation operating in the Eastern time zone of the U.S. This corporation decides to carry 6 EITs (18 hours of TV program information). If at present, the Eastern time is 15:30 EDT (19:30 UTC), then the coverage times for the EIT tables are:

Table D.1 An Example of EIT Coverage Times

EIT number	Version Num.	Assigned PID	Coverage (UTC)	Coverage (EDT)
0	6	123	18:00 - 21:00	14:00 - 17:00
1	4	190	21:00 - 24:00	17:00 - 20:00
2	2	237	0:00 - 3:00	20:00 - 23:00
3	7	177	3:00 - 6:00	23:00 - 2:00 (nd)
4	8	295	6:00 - 9:00	2:00 (nd) - 5:00 (nd)
5	15	221	9:00 - 12:00	5:00 (nd) - 8:00 (nd)

The abbreviation "nd" denotes next day. Before 17:00 EDT, the MGT will list the currently valid PIDs as: 123, 190, 237, 177, 295, and 221. At 17:00 EDT, table EIT-0 will become obsolete while the other ones will remain valid. At that time, the PID list can be changed to 190, 237, 177, 295, 221, maintaining the version number list as 4, 2, 7, 8, 15. Therefore, by simply shifting the listed PID values in the MGT, table EIT-1 can become EIT-0, table EIT-2 can become EIT-1, and so on.

However, it is also possible to regenerate one or several EITs at any time for correcting and/or updating the content (e.g. in cases where "to be assigned" events become known). Regeneration of EITs is flagged by updating version fields in the MGT. For example, if table EIT-2 needs to be updated at 16:17 EDT, then the new table must be transmitted with a version number equal to 3. Whenever the decoder monitoring the MGT detects a change in the version number of a table, it assumes that the table has changed and needs to be reloaded.

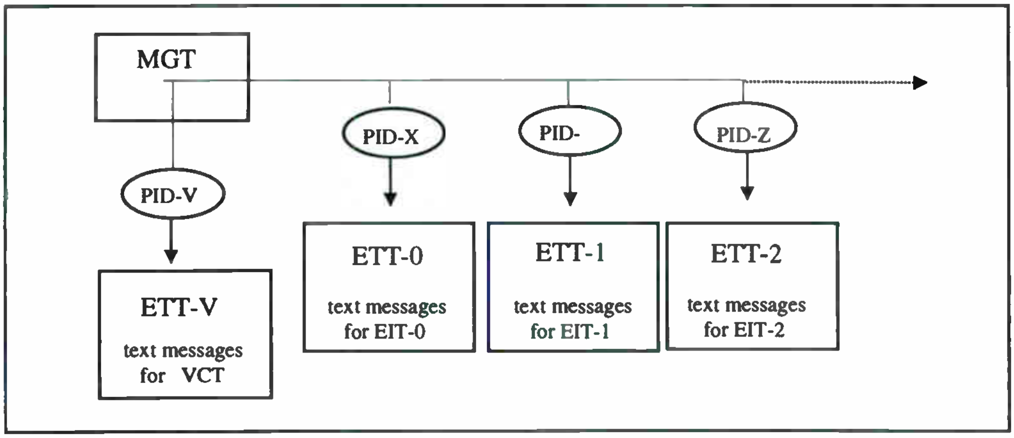

Figure D.2 Extended Text Tables in the PSIP hierarchy.

As illustrated in Fig. D.2, there can be several Extended Text Tables (ETTs), each of them having its PID defined in the MGT. Each Event Information Table (EIT) can have one ETT. Similarly, the Virtual Channel Table can have one ETT. As its name indicates, the purpose of an Extended Text Table (ETT) is to carry text messages. For example, for channels in the VCT, the messages can describe channel information, cost, coming attractions, etc. Similarly, for an event such as a movie listed in the EIT, the typical message is a short paragraph that describes the movie itself. Extended Text Tables are optional.

In this final section paragraph we review once more the requirement list. The minimum amount of information required in an ATSC terrestrial digital Transport Stream is the VCT, the MGT, the RRT, the STT, and the first four EITs. All of the other elements are optional.

D3. APPLICATION EXAMPLE

For the purpose of this example, we assume that a broadcast group, here denominated NBZ, manages the frequency bands for RF channels 12 and 39. The first one is its analog channel whereas the second one will be used for digital broadcast. According to the premises established in this document, NBZ must carry the PSIP tables in the digital Transport Stream of RF channel 39. The tables must describe TV programs and other services provided on RF channel 39 but can also describe information for the analog RF channel 12.

Assume that NBZ operates in the Eastern time zone of the U.S., and that the current time is 15:30 EDT (19:30 UTC). NBZ decides to operate in minimal configuration, therefore only the first four EITs need to be transmitted. As explained previously, EIT-0 must carry event information for the time window between 14:00 and 17:00 EDT, whereas EIT-1 to EIT-3 will cover the subsequent 9 hours. For the first 6 hours, the following scenario applies:

Table D.2 The first 3-hour segment to be described in VCT and EIT-0

		14:00-14:30	14:30 -15:00	15:00 - 15:30	15:30 - 16:00	16:00 - 16:30	16:30-17:00
PTC 12	NBZ	City Life	City Life	Travel Show	Travel Show	News	News
PTC 39 VC #1	NBZ	City Life	City Life	Travel Show	Travel Show	News	News
PTC 39 VC #2	NBZ	Soccer	Golf Report	Golf Report	Car Racing	Car Racing	Car Racing
PTC 39 VC #3	NBZ	Secret Agent	Secret Agent	Lost Worlds	Lost Worlds	Lost Worlds	Lost Worlds
PTC 39 VC #4	NBZ	headlines	headlines	headlines	headlines	headlines	headlines

Table D.3 The second 3-hour segment to be described in VCT and EIT-1

		17:00-17:30	17:30-18:00	18:00 - 18:30	18:30 - 19:00	19:00-19:30	19:30 - 20:00
PTC 12	NBZ	Music Today	NY Comedy	World View	World View	News	News
PTC 39 VC #1	NBZ	Music Today	NY Comedy	World View	World View	News	News
PTC 39 VC #2	NBZ	Car Racing	Car Racing	Sports News	Tennis Playoffs	Tennis Playoffs	Tennis Playoffs
PTC 39 VC #3	NBZ	Preview	The Bandit	The Bandit	The Bandit	The Bandit	Preview
PTC 39 VC #4	NBZ	headlines	headlines	headlines	headlines	headlines	headlines

Similar tables can be built for the next 6 hours (for EIT-2 and EIT-3). According to this scenario, NBZ broadcasts four regular digital channels (also called virtual channels and denoted as VC), one matching the analog transmission (simulcast), another for sports, and a third one for movies. The fourth one supports a service displaying headlines with text and images.

D3.1 The Master Guide Table (MGT)

The purpose of the MGT is to describe everything about the other tables, listing features such as version numbers, table sizes, and packet identifiers (PIDs). Fig. D.3 shows a typical Master Guide Table indicating, in this case, the existence in the Transport Stream of a Virtual Channel Table, the Rating Region Table, four EITs, one Extended Text Table for channels, and two Extended Text Tables for events.

The first entry of the MGT describes the version number and size of the Virtual Channel Table. The second entry corresponds to an instance of the Rating Region Table. If some region's policy makers decided to use more than one instance of an RRT, the MGT would list each PID, version number, and size. Notice that the base PID (0x1FFB) must be used for the VCT and the RRT instances as specified in PSIP.

The next entries in the MGT correspond to the first four EITs that must be supplied in the Transport Stream. The user is free to choose their PIDs as long as they are unique in the MGT list of PIDs. After the EITs, the MGT indicates the existence of an Extended Text Table for channels carried using PID 0x1AA0. Similarly, the last two entries in the MGT signal the existence of two Extended Text Tables, one for EIT-0 and the other for EIT-1.

MGT

table_type	PID	version num.	table size
VCT	0x1FFB (base PID)	4	485 bytes
RRT - USA	0x1FFB (base PID)	1	560 bytes
EIT-0	0x1FD0	6	2730 bytes
EIT-1	0x1FD1	4	1342 bytes
EIT-2	0x1DD1	2	1224 bytes
EIT-3	0x1DB3	7	1382 bytes
ETT for VCT	0x1AA0	21	4232 bytes
ETT-0	0x1BA0	10	32420 bytes
ETT-1	0x1BA1	2	42734 bytes

Figure D.3 Content of the Master Guide Table

Descriptors can be added for each entry as well as for the entire MGT. By using descriptors, future improvements can be incorporated without modifying the basic structure of the MGT. The MGT is like a flag table that continuously informs the decoder about the status of all the other tables (except the STT which has an independent function). The MGT is continuously monitored at the receiver to prepare and anticipate changes in the channel/event structure. When tables are changed at the broadcast side, their version numbers are incremented and the new numbers are listed in the MGT. Based on the version updates and on the memory requirements, the decoder can reload the newly defined tables for proper operation.

D3.2 The Virtual Channel Table (VCT)

Figure D.4 shows the structure of the VCT which essentially contains the list of channels available in the Transport Stream. For convenience, it is possible to include analog channels and even other digital channels found in different Transport Streams.

The field number_of_channels_in_section indicates the number of channels described in one section of the VCT. In normal applications, as in the example being considered here, all channel information will fit into one section. However, there may be rare times when most of the physical channel is used to convey dozens of low-bandwidth services such as audio-only and data channels in addition to one video program. In those cases, the channel information may be larger than the VCT section limit of 1 Kbyte and therefore VCT segmentation will be required.

For example, assuming that a physical channel conveys 20 low-bandwidth services in addition to a TV program, and assuming that their VCT information exceeds 1 Kbyte, then two or more sections may be defined. The first section may describe 12 virtual channels and the second 9 if such a partition leads to VCT sections with less than 1 Kbyte.

A new VCT containing updated information can be transmitted at any time with the version_number increased by one. However, since a VCT describes only those channels from a particular Transport Stream, virtual channels added to the VCT at arbitrary times will not be detected by the receiver until it is tuned to that particular Transport Stream. For this reason, it is highly recommended that channel addition be made in advance to give the receivers the opportunity to scan the frequencies and detect the channel presence.

The fields major_channel_number and minor_channel_number are used for identification. The first one, the major channel number, is used to group all channels that are to be identified as belonging to a particular broadcast corporation (or particular identifying number such as 12 in this case). The minor channel number specifies a particular channel within the group.

The field short_name is a seven-character name for the channel and may allow text-based access and navigation. The fields transport_stream_id and program_number are included to link the VCT with the PAT and PMT. A sequence of flags follows these fields. The flags indicate: (1) if the channel is hidden (e.g. for NVOD applications), (2) if the channel has a long text message in the VCT-ETT, and (3) if the channel is visible in general or has some conditional access constraints.

After the flags, a description of the type of service offered is included, followed by the source_id. The source_id is simply an internal index for representing the particular logical channel. Event Information Tables and Extended Text Tables use this number to provide a list of associated events or text messages respectively.

VCT

current_next_indicator = 1
number_channels_in_section = 5

major num.	minor num	short name	carrier freq. (MHz)	channel TS ID	progr. num	flags	service type	source id	descriptors
12	0	NBZ	205.25	0x0AA0	0x0AA0	--	analog	20	ch_name
12	1	NBZD	620.31	0x0AA1	0x00F1	--	digital	21	ch_name serv_locat.
12	5	NBZ-S	620.31	0x0AA1	0x00F2	--	digital	2	ch_name serv_locat.
12	12	NBZ-M	620.31	0x0AA1	0x00F3	--	digital	23	ch_name serv_locat.
12	31	NBZ-H	620.31	0x0AA1	0x00F8	--	digital	24	ch_name serv_locat.

Figure D.4 Content of the Virtual Channel Table

Two descriptors are associated with the logical channels in the example. The first one is extended_channel_name and, as its name indicates, it gives the full name of the channel. An example for channel NBZ-S could be: "NBZ Sports and Fitness". The other one, the service_location descriptor, is used to list the available bit streams and their PIDs necessary to decode packets at the receiver. Assuming that NBZ-M offers bilingual transmission, then the following attributes are tabulated within its service_location descriptor:

PID_audio_1	AC-3 audio	English
PID_audio_2	AC-3 audio	Spanish
PID_video	MPEG-2 video	No lang.

Two VCTs may exist simultaneously in a Transport Stream: the current and the next VCT. The current VCT is recognized by having the flag current_next_indicator set to 1, while the next one has this flag set to 0. Although carrying the next VCT is optional, its use is recommended to give receivers advance notification of the new parameters that become operational during a VCT update.

Assume for example that a Transport Stream contains a VCT with a version number of 6 which has been operational for 20 hours. At 10:00 p.m., a football game using much more bandwidth will be broadcast, and for this reason, the number of available channels and PIDs will be redefined. Around 9:30 p.m., simultaneous transmission of the next VCT can start with a version number of 7. By continuously monitoring the MGT, a receiver can be informed that a next VCT is available. The receiver may want to cache the new VCT for future use. The receiver continues monitoring the MGT and when this table signals a version change for the current VCT (from 6 to 7), then the cached information can be used.

When the VCT refers to an analog service type, the channel_TSID cannot refer to the identifier of a "Transport Stream" in the MPEG-2 sense. Analog NTSC broadcast signals can, however, carry a 16-bit unique identifier called a "Transmission Signal Identifier."[8] For the example VCT in Figure D.4, the Transmission Signal Identifier for channel 12.0 is 0x0AA0. A receiver can use the Transmission Signal ID given in the analog channel's channel_TSID field to verify that the NTSC signal received at the frequency given in the VCT is actually the desired signal. In the case that the Transmission Signal ID is not known or not available, the channel_TSID field may contain 0xFFFF to indicate "unknown."

D3.3 The Event Information Tables (EITs)

The purpose of an EIT is to list all events for those channels that appear in the VCT for a given time window. As mentioned before, EIT-0 describes the events for the first 3 hours, EIT-1 for the next 3 hours, and so on. EIT-i and EIT-j have different PIDs as defined in the MGT. In PSIP, tables can have a multitude of instances. The different instances of a table share the same table_id value and PID but use different table_id_extension values.

In PSIP, an instance of EIT-k contains the list of events for a single virtual channel with a unique source_id. For this reason, the table_id_extension has been renamed as source_id in the EIT syntax. Figure D.5 shows, for example, the NBZ-S instance for EIT-0. Following similar procedures, the NBZD, NBZ-M, and NBZ-H instances of EIT-0 can be constructed. The process can be extended and repeated to obtain all of the instances for the other tables in the time sequence: EIT-1, EIT-2, etc.

The three events programmed for the 3-hour period for NBZ-S are listed in Figure D.5. The field event_id is a number used to identify each event. If an event time period extends over more than one EIT, the same event_id has to be used. The event_id is used to link events with their messages defined in the ETT, and therefore it has to be unique only within a virtual channel and a 3-hour interval defined by EITs. The event_id is followed by the start_time and then the length_in_seconds. Notice that events can have start times before the activation time (14:00 EST in this example) of the table. The ETM_location specifies the existence and the location of an Extended Text Message (ETM) for this event. ETMs are simply long textual descriptions. The collection of ETMs constitutes an Extended Text Table (ETT).

[8] A method to include such a unique 16-bit "Transmission Signal ID" in the NTSC VBI is specified in the EIA-752 specification.

EIT-0
source id = 22 (NBZ-S instance)

num_events_in_section = 3

event ID	local start time	length (seconds)	ETM location	title	descriptors
51	12:30	7200	01	Soccer Live	content_advisory
52	14:30	3600	00	Golf Report	closed_caption
53	15:30	9000	01	Car Racing	content_advisory

Figure D.5 Content of EIT-0 for NBZ-S

An example of an ETM for the Car Racing event may be:

> "Live coverage from Indianapolis. This car race has become the largest single-day sporting event in the world. Two hundred laps of full action and speed."

Several descriptors can be associated with each event. The most important is the content advisory descriptor which assigns a rating value according to one or more systems. Recall that the actual rating system definitions are tabulated within the RRT. When a closed caption descriptor is included, it signals the existence of closed captioning and lists the necessary parameters for decoding.

D3.4 The Rating Region Table (RRT)

The Rating Region Table is a fixed data structure in the sense that its content remains mostly unchanged. It defines the rating standard that is applicable for each region and/or country. The concept of table instance introduced in the previous Section is also used for the RRT. Several instances of the RRT can be constructed and carried in the Transport Stream simultaneously. Each instance is identified by a different table_id_extension value (which becomes the rating_region in the RRT syntax) and corresponds to one and only one particular region. Each instance has a different version number which is also carried in the MGT. This feature allows updating each instance separately.

Figure D.6 shows an example of one instance of an RRT, defined as the first rating region and carrying the MPAA standard rating system. Changes in the content of the RRT must be defined and approved by the ATSC. Each event listed in any of the EITs may carry a content advisory descriptor. This descriptor is an index or pointer to one or more instances of the RRT.

RRT
rating_region = 1 (first instance)

rating_region_name_text= "US (50 states+possesions)"
dimensions = 1

dimension_name = "MPAA Theatrical Rating System"
values_defined = 7

value	abbrev	rating_value
0	" "	" "
1	"G"	"Suitable for All Ages"
2	"PG"	"Parental Guidance Suggested"
3	"PG-14"	"Parents Strongly Cautioned"
4	"R"	"Restricted, under 17 must be accompanied by adult"
5	"NC-17"	"No One 17 And Under Admitted to Theater"
6	"NR"	"Not Rated by MPAA"

Figure D.6 An instance of a Rating Region Table (RRT).

D4. PACKETIZATION AND TRANSPORT

In the previous sections, we have described how to construct the MGT, VCT, RRT, and EITs based on the typical scenario described in Tables D.1 and D.2. The number of virtual channels described in the VCT is 5 and therefore, each EIT will have 5 instances.

For the example, the size of the MGT is less than a hundred bytes and the VCT ranges between 300 to around 1500 bytes depending on the length of the text strings. Similarly, each EIT instance can have from 1 to about 3 Kbytes depending again on the text length.

Typically, the MGT, STT, VCT, and each instance of the RRT and EIT will have one or at most a few sections. For each table, the sections are appended one after the other, and then segmented into 184-byte packets. After adding the 4-byte MPEG-2 TS header, the packets are multiplexed with the others carrying audio, video, data, and any other components of the service. Figure D.7 illustrates this process.

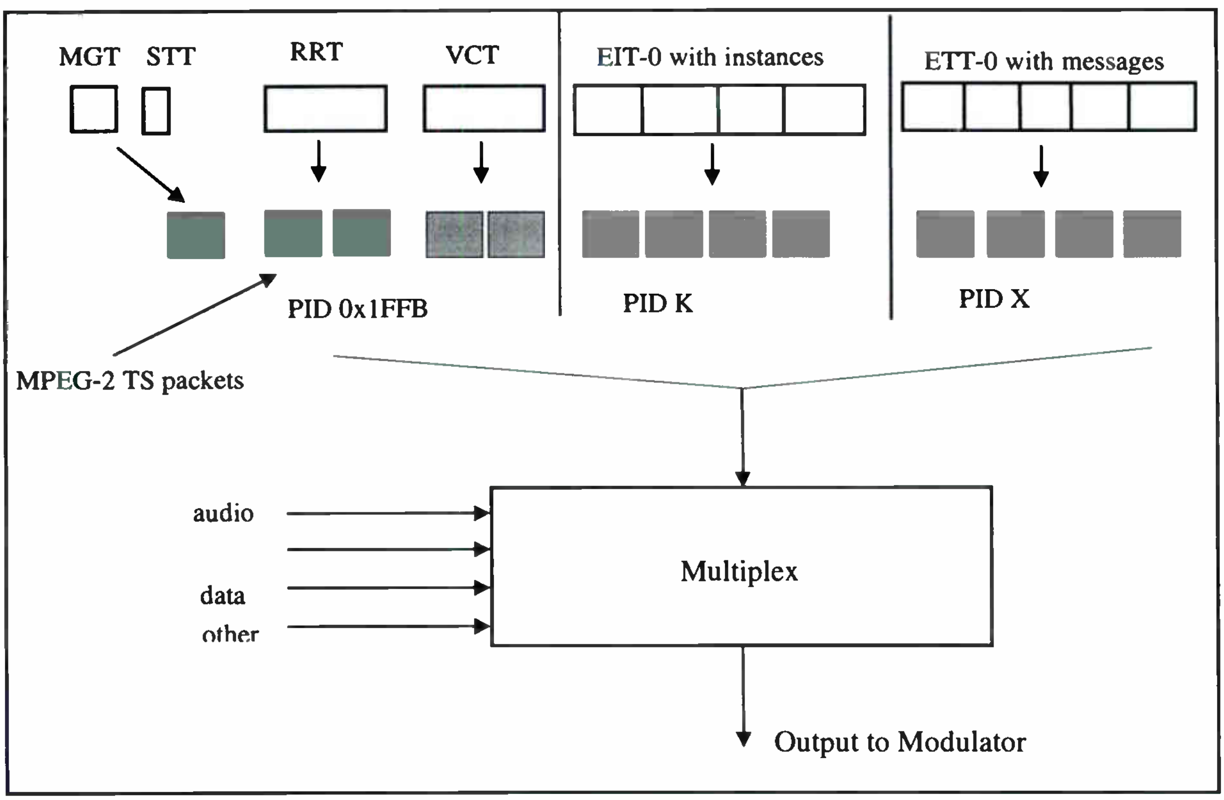

Figure D.7 Packetization and transport of the PSIP tables

D.5 TUNING OPERATIONS AND TABLE ACCESS

As described by the PSIP protocol, each Transport Stream will carry a set of tables describing system information and event description. For channel tuning, the first step is to collect the VCT from the Transport Stream which contains the current list of services available. Figure D.8 shows this process.

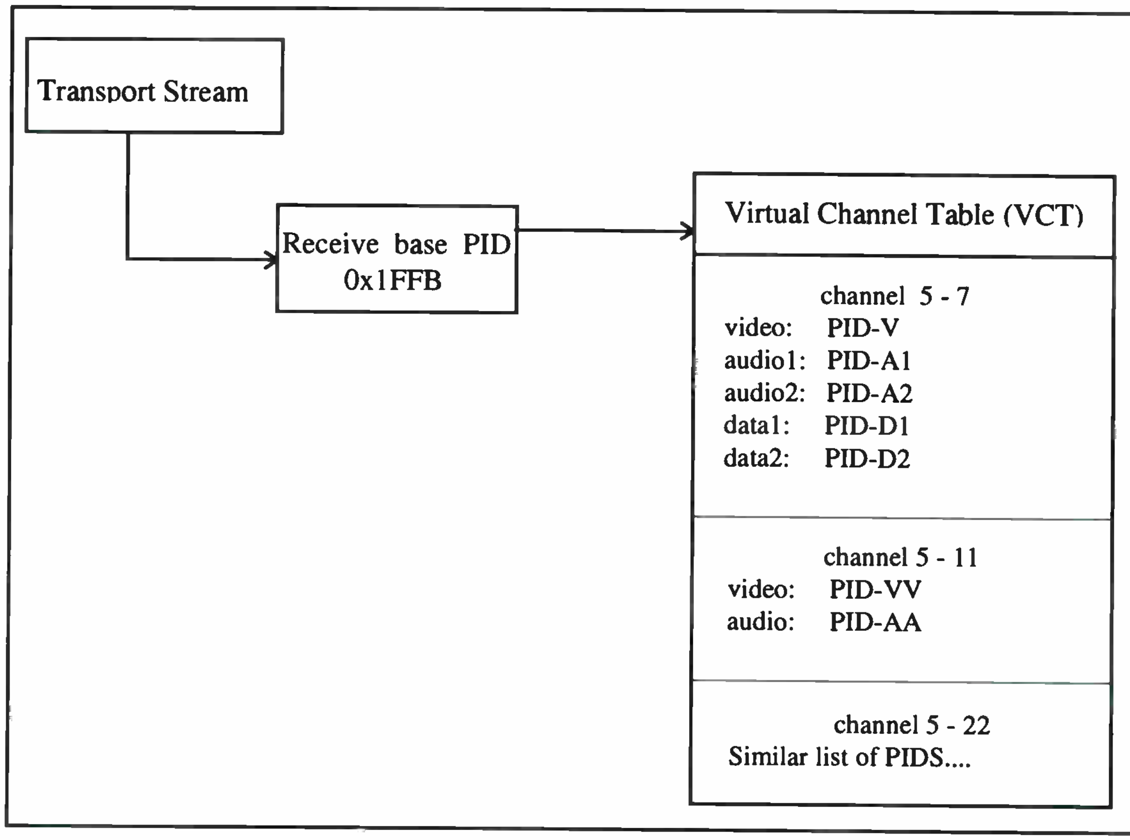

Figure D.8 Extraction of the VCT from the Transport Stream

Once the VCT has been collected, a user can tune to any virtual channel present in the Transport Stream by referring to the major and minor channel numbers. Assuming that in this case, the user selects channel 5 - 11, then the process for decoding the audio and video components is shown in Fig. D.9. For terrestrial broadcast, the existence of a service location descriptor in the VCT is mandatory and therefore there is no need to access the PAT or PMT for tuning. This feature has been included in PSIP to minimize the time required for changing and tuning to channels. However, PAT and PMT information must be present in the Transport Stream to support the general MPEG-2 compliance

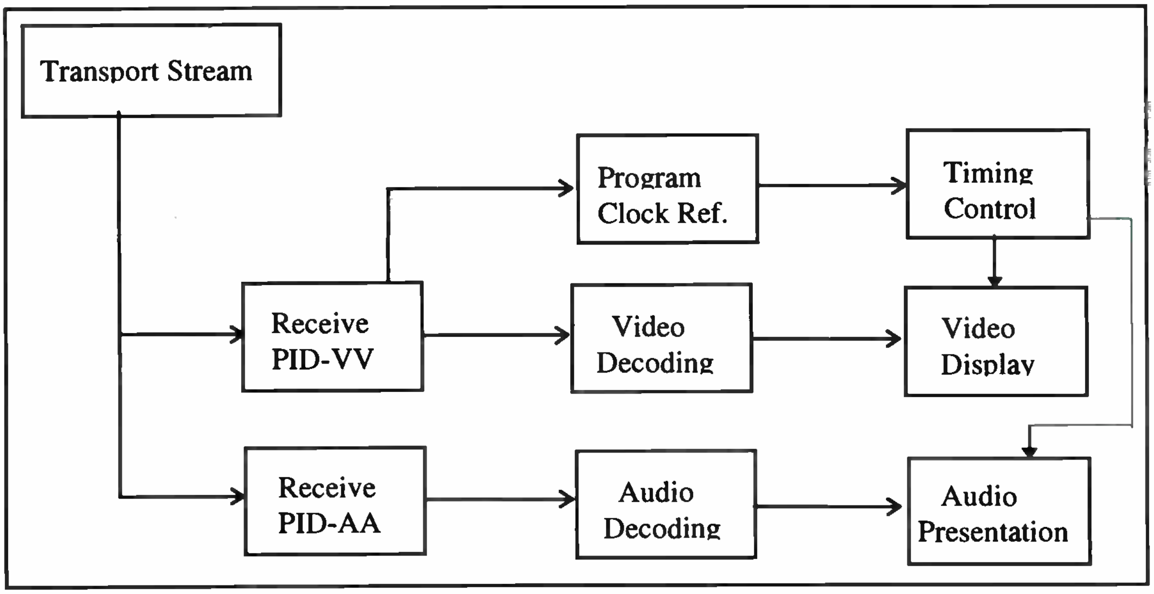

Figure D.9 Acquisition of audiovisual components

ANNEX E

(Informative)

TYPICAL SIZE OF PSIP TABLES

The typical sizes for the PSIP tables (STT, MGT, VCT, RRT, EIT and ETT) are calculated in this Section. The notation used here for the different equations is listed in the Table E.1.

Table E.1 Symbols

Symbol	Description
P	number of EITs (4 to 128)
C	number of virtual channels (analog and digital) per EIT
Cd	number of digital channels per EIT
E	number of events per virtual channel
R	number of rating regions
D	average number of rating dimensions per rating region
L	average number of rating values per rating dimension

E1. SYSTEM TIME TABLE (STT)

The typical size for the STT is 20 bytes, with the assumption of having no descriptors.

E2. MASTER GUIDE TABLE (MGT)

The typical size for the MGT (in bytes), based on the assumptions listed in the column "Assumption", is shown in Table E.2

Table E.2 Typical size (bytes) of MGT

Part	Size (bytes)	Assumption
PSI header and trailer	12	
message body	38+22*P	1. With one Terrestrial VCT, one channel ETT, one RRT instance, P EITs and P event ETTs 2. No descriptors
Total	50+22*P	

E3. TERRESTRIAL VIRTUAL CHANNEL TABLE (TVCT)

The typical size of the TVCT (bytes), based on the assumptions listed in the column labeled "Assumption" is shown in Table E.3.

Table E.3 Typical TVCT size (bytes)

Part	Size (bytes)	Assumption
PSI header and trailer	12	1. All TVCT messages are carried in one section.
message body	4+32*C	
extended channel name descriptor	20*C	2. One string and one segment per string for long channel name text. 3. Long channel name text is compressed by Huffman coding with a standard table, and the text length after compression is 10 bytes
service location descriptor	23*Cd	4. Three elementary streams per virtual channel for digital channels.
Total	16+52*C+23*Cd	

E4. RATING REGION TABLE (RRT)

The typical size (in bytes per rating region) of the RRT, based on the assumptions listed in the column "Assumption", is shown in Table E.4.

Table E.4 Typical size (in bytes per rating region) of RRT

Part	Size (bytes per rating region)	Assumption
PSI header and trailer	12	1. One section only.
message body	25+D*(14+ 26*L)	2. One string and one segment per string for all text. 3. Rating region name text is compressed by Huffman coding with a standard table, and the size after compression is 12 bytes. 4. Dimension name text is compressed by Huffman coding with a standard table, and the size after compression is 4 bytes. 5. Abbreviated rating value text is compressed by Huffman coding with a standard table, and the size after compression is 2 bytes. 6. Rating value text is compressed by Huffman coding with a standard table, and the size after compression is 6 bytes. 7. No descriptors.
Total	37+D*(14+26*L)	

E5. EVENT INFORMATION TABLE (EIT)

The typical size of the EIT (in bytes per virtual channel per EIT), based on the assumptions listed in the column "Assumption", is shown in Table E.5.

Table E.5 Typical size (bytes per virtual channel per EIT) of EIT

Part	Size (bytes per virtual channel per EIT)	Assumption
PSI header and trailer	12	1. One section only
message body	2+30*E	2. One string and one segment per string for title text. 3. Title text is compressed by Huffman coding with a standard table, and the size after compression is 10 bytes. 4. No AC-3 and service location descriptors.
closed captioning service descriptor	9*E	5. number_of_services = 1.
content advisory descriptor	(3+R*(3+2*D))*E	6. No rating_description_text.
Total	14+(42+R*(3+2*D))*E	

E6. EXTENDED TEXT TABLE (ETT)

The typical size for the ETT (in bytes per virtual channel per EIT, or bytes per event per EIT), based on the assumptions listed in the column labeled "Assumptions", is shown in Table E.6.

Table E.6 Typical size (bytes per virtual channel or bytes per event) of ETT

Part	Size (bytes per virtual channel per EIT, or bytes per event per EIT)	Assumptions
PSI header and trailer	12	
message body	508	1. A virtual channel or an event can have one text string and one segment per string for the extended text message. 2. Extended text message is compressed by Huffman coding with a standard table, and the size after compression is 500 bytes.
Total	520	

E7. AN EXAMPLE FOR TERRESTRIAL BROADCAST

Suppose that a TV provider is in charge of two physical transmission channels, one for analog and the other for digital services. Assume that the digital Transport Stream carries five virtual channels, each with 6 events in EIT-0, EIT-1, EIT-2 and EIT-3. For each virtual channel and each event an extended text message is available.

Regarding the Rating Region Table, suppose that a single rating region is defined with six dimensions and five values per dimension. Based on these assumptions, typical sizes for every PSIP table can be calculated. The results are listed in Table E.7 and Table E.8.

Table E.7 Typical sizes of PSIP tables (except ETT) for the example

Part	Size in bytes (excluding Transport Stream packet header)	Size in Transport Stream packets
STT	20	1
MGT	138	1
TVCT	443	3
RRT	901	5
Subtotal for tables identified by the base_PID	**1502**	**10**
EIT-0	2136	12
EIT-1	2136	12
EIT-2	2136	12
EIT-3	2136	12
Total	**10046**	**58**

Table E.8 Typical sizes of ETTs for the example

Part	Size in bytes (excluding Transport Stream packet header)	Size in Transport Stream packets
Channel ETT	3120	17
Event ETT-0	18720	102
Event ETT-1	18720	102
Event ETT-2	18720	102
Event ETT-3	18720	102
Total	**78000**	**425**

ANNEX F

(Informative)

AN OVERVIEW OF HUFFMAN-BASED TEXT COMPRESSION

This section describes the Huffman-based text compression and coding methods supported in the Program and System Information Protocol. In particular, this section:

- Describes the partial first-order Huffman coding used to compress PSIP text data.
- Provides background description of finite-context Huffman coding. The mechanisms for generating and parsing Huffman codes are described.
- Describes the decode tree data structure.
- Defines the character set supported by this Standard.

F1. DATA COMPRESSION OVERVIEW

Program and System Information data may use partial first-order Huffman encoding to compress English-language text. The Huffman-table based approach has the following features:

- A typical firmware-resident Huffman decode table requires less than 2K of storage.
- The encode and decode algorithms are relatively simple and fast.
- Since first-order Huffman codes are significantly influenced by language phonetics, codes produced from a sample of current program titles produce reasonable compression ratios for future program titles, even though the future program titles may be significantly different from current titles. Therefore, hard-coded tables stored in receiver non-volatile memory are helpful.

  The data compression approach has the following implementation characteristics:

- Program descriptions and program titles may use different Huffman codes. Titles and descriptions have significantly different text characteristics; for example, program titles usually have an upper-case character following a space character, whereas program descriptions usually have a lower-case character following a space-character.
- Hard-coded decode tables, one optimized for titles and one for descriptions, must reside in the receiver's non-volatile memory.

F2. OVERVIEW OF CONTEXT-SENSITIVE HUFFMAN CODING

F2.1 Overview

Each and every character does not occur with the same frequency in program titles and program descriptions. For example, the character "e" occurs more often than the character "x." With Huffman coding, the number of bits used to represent a character is inversely proportional to the character's usage frequency.

The Huffman coding compression ratio depends upon the statistical distribution of the characters being compressed. When character usage is uniformly distributed, no compression is achieved with Huffman coding. To achieve satisfactory compression, the Huffman codes are generated using statistics that match the data being compressed. For example, Huffman codes generated from Pascal computer programs would be less than ideal for compressing C programs. For text strings in the PSIP, program descriptions and program titles may be compressed with different sets of Huffman codes

Context-sensitive Huffman coding recognizes that a character's usage statistics are context dependent. For example, the character "u" has a high probability of occurrence after the character "q". The "order" of the Huffman code defines the "look-back" context by which a character is coded. With order-0, each character is coded independently of the previous character. With order-1, the Huffman code used to represent a given character depends upon the previous character. In zero-order Huffman compression, the occurrence probability of the alphabet elements is used to develop an optimal encoding tree. In first-order Huffman, the conditional probability of a character, given that the previous character is known, is used as the basis of a decoding tree. For this reason, while zero-order Huffman has typically a single tree, first-order Huffman has many, one for each character.

Huffman compression involves the following steps:

- Determine the statistical distribution of the characters or symbols in the source data.
- Create Huffman codes from this statistical information.
- Encode the source data: Translate each character into its corresponding Huffman code.

To decompress the coded data, the data string is parsed bit-by-bit and translated to the original characters. To do this, the decompressor must have the correct decode table, which maps the Huffman codes to their corresponding characters. The following example illustrates the generation and decoding of Huffman codes.

F2.2 Example

Huffman codes are mapped to their corresponding characters using a binary tree structure. The leaves of this tree are the alphabet elements to be coded. The tree is produced by recursively summing the two nodes in the tree with the lowest usage frequency. For the following example, assume that an alphabet contains the following twelve characters which occur a certain number of times in the sample database:

Table F.1 Example Character Set and Frequency of Character Occurrence

Character	Occurrence Number
'a'	144
'b'	66
'c'	30
'd'	30
'e'	18
'f'	12
'g'	6
'h'	1
'i'	1
'j'	1
ESC	arbitrary

The "escape" character is inserted into the table to handle input characters which rarely occur, and have no corresponding Huffman codes. In this example, no Huffman codes will be generated for the characters 'h', 'i', and 'j'. Instead, their frequencies will be summed into the ESC character. Whenever one of these characters occur in the input stream, the encoder inserts the ESC Huffman code, then inserts the original ASCII value for that character.

Figure F.1 shows the construction of the Huffman tree from the character frequencies. The two nodes with the lowest frequencies, ('ESC' and 'g'), are joined together, with a resulting node weight of (9). The next two lowest nodes, ('f' and the intermediate node), are then joined together, with the combined weight of (21). This process continues until the tree's root node is formed. Once the tree is completed, the bit (1) is assigned to all right-hand branches, and the bit (0) is assigned to all left-hand branches.

Decoding a Huffman string is straight-forward. Starting at the Huffman tree root, the decoder parses the string, bit by bit, until it reaches a leaf node. The leaf node is the decoded character. The decoder then moves back to the root of the Huffman tree to continue decoding the bit string. For example, the input string 10111011100010 would be decoded into 'beeaab'.

This example uses order-0 Huffman codes. With order-1, each character in the alphabet has an associated tree of Huffman codes for possible succeeding characters. The ESC character would be inserted into each of these order-1 tables to handle statistically unlikely character pairs.

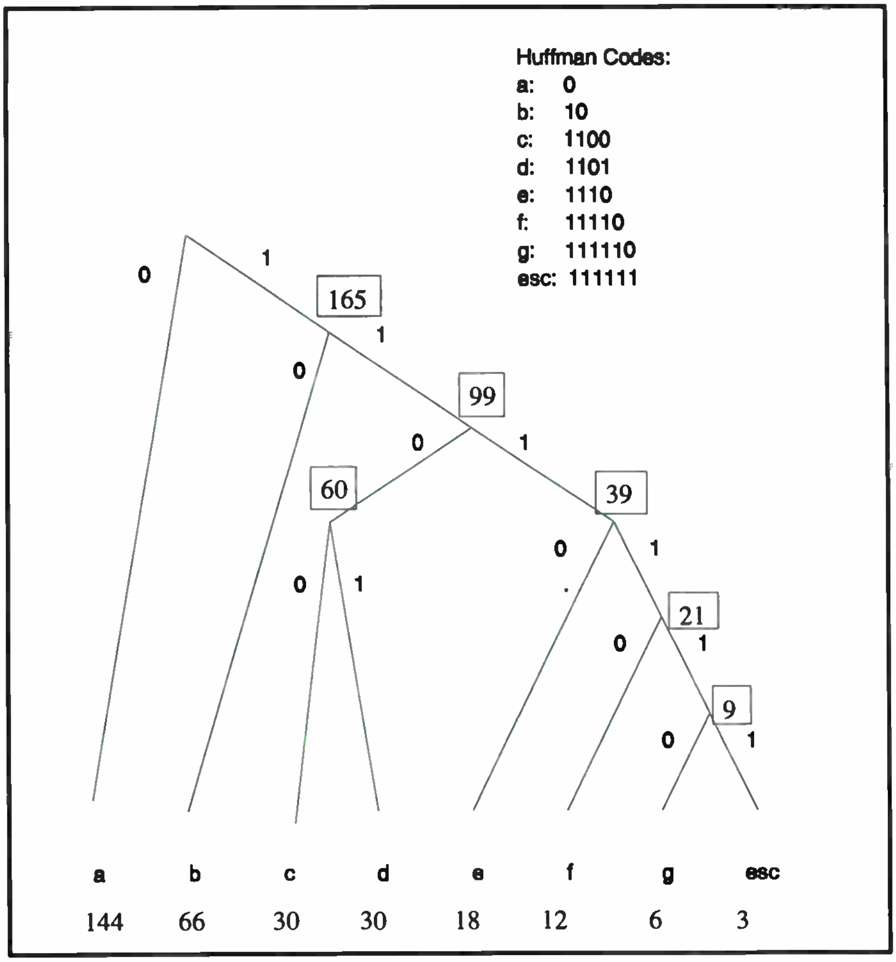

Figure F.1 Example Huffman Tree

F2.3 Decode Tree Example

Actual implementations of Huffman decoders need to map the trees into a suitable data structure that can be used by a computer or processor to traverse the tree top-down. In Annex C, a possible method for representing the trees was described and explicitly defined. Such a method is used here to build the decoding tree data for the example given in Figure F.1. Although an order-0 tree, this table is representative of order-1 decode trees, except that the bytes of each order-1 tree start at a byte location specified by the corresponding tree root offset (rather than starting at location 0).

Table F.2 Decode Tree Example

Byte #	Left/Right Child Word Offset or Character Leaf
0 (tree root)	225 (ASCII "a" + 128)
1	1 (word offset of right child)
2 (tree node)	226 (ASCII "b" + 128)
3	2 (word offset of right child)
4 (tree node)	3 (word offset of left child)
5	4 (word offset of right child)
6 (tree node)	227 (ASCII "c" + 128)
7	228 (ASCII "d" + 128)
8 (tree node)	229 (ASCII "e" + 128)
9	5 (word offset of right child)
10 (tree node)	230 (ASCII "f" + 128)
11	6 (word offset of right child)
12 (tree node)	231 (ASCII "g" + 128)
13	155 (ASCII "ESC" + 128)

F2.4 Encoding/Character Decoding Examples with 1st-order Huffman tables

As an example of using the Huffman table defined in Table C.4 in Annex C, here we show the procedure to encode and decode the string "The next" using the tables optimized for titles. The coding sequence that generates the bit stream for "The next" is described in Figure F.2.

<term>	T	→	010
T	h	→	0
h	e	→	0
e	' '	→	01
'	n	→ ' ' <esc> →	10010100
<esc>	n	→ ASCII code for "n" →	01101110
n	e	→	010
e	x	→	00010011
x	t	→	01
t	<term>	→	010

Figure F.2 Coding Example for the string "The next"

The first character 'T' is encoded assuming that the previous one was a *terminate* character. The second letter 'h' is encoded based on the Huffman tree corresponding to the prior symbol 'T.' The sequence proceeds as shown in the Figure. The combination blank-space followed by an 'n' is not listed in the tree, thus the escape character is used to switch the coding process to uncompressed mode. Once in this mode, the 'n' is encoded using its standard 8-bit ISO Latin-1 value. After the 'n', an 'e' is encoded using the appropriate n-tree and the algorithm continues until reaching the final letter followed by a string-terminate character. Uncompressed transmission of this string requires 9 bytes, while after compression, only 39 bits, equivalent to 5 bytes, are needed.

Decoding requires traversing the different trees top-down. As an example, Figure F.3 shows the tree when the prior character is 'x'. From our example, after decoding the letter 'x', the remaining bit sequence is '01010'. Traversing the x-tree top-down using this sequence shows that '01' corresponds to 't', a newly decoded character. The process now jumps to the t-tree and so on, to decode the remaining bits until the terminate code results. Notice that the trees can be obtained by examining the encoding tables or by following the semantics of the provided decoding tables.

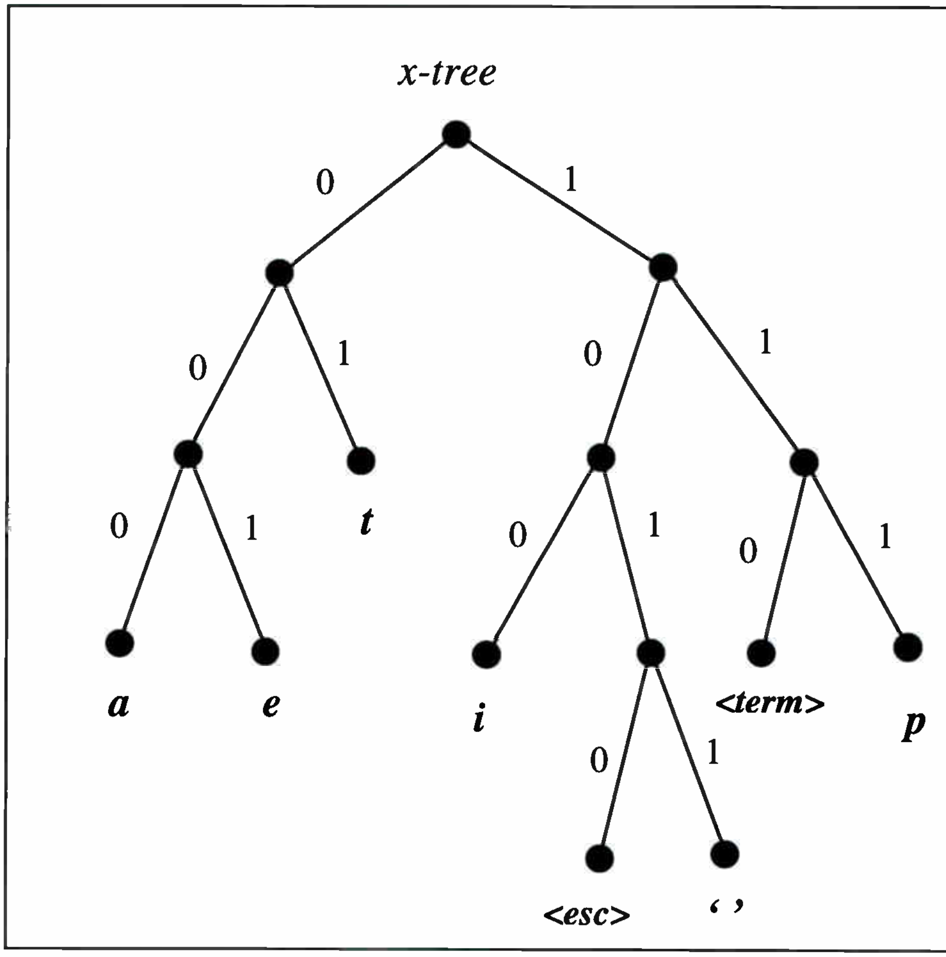

Figure F.3 Huffman tree for prior symbol "x"

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1771 N Street, NW
Washington, DC 20036-2891