PROCEEDINGS

29TH ANNUAL BROADCAST ENGINEERING CONFERENCE

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NATIONAL ASSOCIATION OF BROADCASTERS

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Mr. Dorschug

Mr. Flemming

Mr. Hill

Mr. Learned

Mr. Parker

Mr. Pointer

Mr. Smith

Mr. Spragg

Mr. Wulliman

Welcoming Remarks

Engineering Award

Vincent T. Wasilewski

President
National Association of Broadcasters

Welcoming Remarks

Good afternoon and welcome to the 29th Engineering Conference. There's a lot planned for you over the next few days and I know you'll leave Las Vegas richer in your knowledge of your profession and poorer in your wallets from the gaming tables. All except George Bartlett, of course. George is known to all the gambling houses here as the terror of the nickel slot machines.

In a nine-page story in a recent issue of *Television/Radio Age,* the author noted that "There is no question that the NAB Engineering Conference is the single most important meeting of the year for broadcast engineers."

Through the papers presented here you'll have an even keener understanding of the state-of-the-art. This conference also gives you an excellent opportunity to compare notes on the operation of your respective stations and pick up valuable information. And, of course, spread out for your inspection is the largest assemblage of broadcast equipment in the world.

NAB has been active back in Washington trying to prevent regulatory restrictions that would make your jobs more difficult and promoting those that will make your jobs a little easier.

To mention a couple—NAB has asked the FCC to reject land mobile's delaying tactics and adopt new streamlined rules for remote broadcasts. We told the Commission that its proposal for simpler rules for remote station pickup was a first step that should be extended to inter-city relays, auxiliary TV and other remotes. We also commended its action ending the operator requirement.

This was long overdue. Broadcasters have long endured attacks from the land mobile service for the use or reallocation of existing broadcasting or broadcast-related frequencies. Let land mobile look toward the efficient use of frequencies presently allocated to that service instead of eroding spectrum space assigned to other services.

Another goal we have been working toward is FCC approval of automatic transmission systems.

Chairman Wiley told a meeting of NAB's Re-Regulation Committee last month that the Commission considers this an area of great importance. Committee members pointed out that automatic transmission is possible now, but further development by manufacturers will come only after Commission action.

The Chairman said that John Taft, a member of the FCC Re-Regulation Task Force, is working full-time to coordinate action. The Commission is expected to act in about two months.

Also discussed at that meeting was the continuing problem of more convenient examination sites for the Third Class licenses. A test is now being conducted using ninety-nine Civil Service examination centers around the country. This will be completed in October and the results will be evaluated by the FCC

Field Bureau. Hopefully, both the Commission and broadcasters will be satisfied.

Mandatory next January 15 will be installation of the new monitor signalling requirements for EBS. As you probably know, monitoring systems are on the market, but some cost more than $500. The FCC's Emergency Communications Division now has a project underway to develop a circuit diagram so you can construct your own system for fifty or sixty dollars.

As you know, NAB is in the process of revising the Engineering Handbook and the sixth edition now is in the final stages of production. There are sample binders and page proofs on display during the convention. You are welcome to stop by and take a look and order now at a discount rate. The booth is number thirteen in the South Exhibit Hall.

Last, I would likel to congratulate John Silva for his being chosen to receive this year's Engineering Award. He is a man dedicated to his profession and I was pleased to see that *Broadcasting Magazine* honored him last week in its Profile section.

You are attending the 1975 Engineering Conference because you are seeking information and are receptive to new ideas. That is the kind of interest and dedication that has made American broadcast engineering the best in the world. Keep it up. After all—all the rest of us—executives, programmers, salespeople—don't have anything to do until the engineer makes it possible.

Have an excellent conference.

Engineer of the Year . . . .

John D. Silva
Director
Research and Development
Golden West Broadcasters
Los Angeles, California

Recipient of . . . .

1975 Engineering Achievement Award

NAB's 1975 Engineering Award was presented to Mr. Silva (right) by George W. Bartlett, NAB vice president for engineering, at the luncheon on Tuesday, April 8.

Presentation by Mr. Bartlett

Ladies, gentlemen, honored guests and friends.

Each year about this time we pause for a few moments to pay our respects to an individual who, in the eyes of his fellow peers, had made a significant contribution to the art of broadcasting. Needless to say, selection of such an individual is not an easy task and comes only after much discussion and soul-searching.

Today, we honor a man who has gained this respect. Without fear of contradiction, one could say that few twelve-year-old youngsters in the year of 1932 were looking forward to a career as a television engineer.

John D. Silva was born in San Diego, California; after graduating from high school he entered Massachusetts Institute of Technology where he majored in Communications Engineering. Two years later, he transferred back to his native California to attend Stanford University where he received his Bachelor of Science degree in Engineering. Following graduation, John went directly into the Navy where he served as a Radio Officer.

During his Navy career, John was in charge of setting up radar installations on the Pacific. He was later attached to the U.S.S. Shea, Destroyer Command Ship, which patrolled the Okinawa area eight days before D-Day. After D-Day, the ship was assigned radar-picket duty and fought Kamakazes downing seven of the suicide planes during one particular ten-minute period of action. He was awarded the Purple Heart for wounds received when a Japanese Bomber crashed into his ship.

John's final tour of duty was with the Chief of Naval Operations in Washington, D.C. where he was in charge of the Command Information Center Desk. He was responsible for initiating all research design and development of electronic equipment for the Bureau of Ships and the Bureau of Air.

Three months after separation from the Navy, John was hired by television pioneer, Klaus Landsberg, as an engineer for the experimental station W6XYZ, now KTLA. Evidently, John had hitched his wagon to the proper star, for as the station grew, so did he, and in six short years was named Chief Engineer.

In the early days of KTLA, John was the technical man on hand for such stories as the Kathy Fiscus tragedy in 1947 (a 5-year-old child was trapped and died in an abandoned well; the story was given national media attention); the 1952 A-bomb test in Nevada; troop embarkations during the Korean war; and, 19 consecutive Tournaments of Roses, several of such he produced and directed himself.

John developed and designed television's first—and possibly only—Telecopter, in effect a remote studio mounted in and on a helicopter. John dreamed up and built his contraption not in the recent years of portable electronic cameras, but back in the late 1950's.

The Telecopter's maiden flight was on July 14, 1958. The Telecopter was in almost constant use from then on, covering major events such as the 1961 Malibu Hills fire, the 1963 Baldwin Hills dam break, the 1965 Watts riots and the 1971 earthquake.

That's what he is being honored for by his fellow engineers; although it must be noted quickly that this is not the first honor to John for the Telecopter. He received an Emmy last year from the National Academy of Television Arts and Sciences; and in 1971, an Emmy from the TV academy's Hollywood chapter.

John resides in Los Angeles, California with his lovely wife Brooksie and their three daughters. Somehow he finds time for a hobby—golf. John had at one time a respectable handicap of eight stroker, it has now crept up to 14.

So, without further ado . . .

"In recognition for his distinguished professional career . . .

For his many contributions to our nation's knowledge in the field of communication technology . . .

For his untiring efforts to foster advances in the art of broadcasting . . .

And for his pioneering spirit which has so richly enhanced the forward progress of broadcast engineering. . . ."

It is my pleasure, on behalf of the National Association of Broadcasters, its membership and staff, to bestow upon our good friend . . . John D. Silva . . . the 1975 NAB Engineering Achievement Award.

Mr. Silva's acceptance:

I can't begin to tell you how overwhelmed, pleased and proud I am to receive the NAB Engineering Achievement Award for 1975.

It is particularly exciting because it comes from and represents the industry I love.

You know, to be honored this way from your peers suddenly makes your life's work seem so much more meaningful—and for that I am truly grateful.

I would like to thank and pay special tribute to my lovely wife Brooksie. I could never repay her for the countless hours she's spent waiting for me, or the many dinners she's had to warm over—that had long before been cooked— and I must say, she's never failed to constantly support and encourage me.

And finally, to those who helped make all this possible, thank you.

Committee Reports

Panel Discussion

Technical Papers

Workshop

Robert W. Flanders

Vice President for Engineering
McGraw-Hill Broadcasting Company, Inc.
Indianapolis, Indiana

NAB Engineering Advisory Committee Report

As chairman of the NAB Engineering Advisory Committee, I would like to submit the following report on this committee's activities for 1974-75.

1. Tower Icing:

Tower Icing continues to be of concern to broadcasters. At the Committee's recommendation, NAB has provided funding towards the effort to reach a satisfactory solution to this problem. Despite a warm winter and minimal icing conditions which hampered research, investigation will continue.

2. TV Subtitling, (*RM #2108*)

The Joint Committee for Inter-Society Coordination (JCIC) on Ancillary signals and the Public Broadcasting Service studied and field-tested the National Bureau of Standards (NBS) and Hazeltine Corporation systems of captioning. The test results are now being evaluated by the respective committees and a report is due momentarily.

The NAB Captioning Subcommittee and the JCIC Ancillary Signal group will also evaluate the captioning experiments being conducted by the British Broadcasting Corporation (BBC) and the Independent Television Network in London.

3. Radio Re-Regulation, Part #73:

Documents have been submitted to the Commission by the Radio Re-Regulation Subcommittee on:

a. Revising the operator licensing procedures so that operator tests may be taken at the various Civil Service Test Centers.
b. Changes in the Mechanical Reproduction Announcement.
c. Lighting inspection responsibility for multiple antennae on a common structure.
d. Remote control of broadcast facilities. (To make AM rules equivalent to TV, which permits one hour delay before the necessity to terminate broadcasting in the event of Remote Control line failure.)
e. Methods of measuring FM and TV aural modulation.
f. Automatic transmitters.

4. OSHA Radiation Levels and Measurement.

The Committee resolved that serious consideration be continued on the subject of electromagnetic radiation as it pertains to the communications industry and to seek out a feasible radiation measuring instrument which would be available to the broadcasters to monitor radiation hazard levels to insure that compliance with the current OSHA rules, for frequencies between 10Mhz and 100 Ghz, would be possible.

The Committee recommended and the NAB Board approved funding the initial research project.

Field measurements of the existing antenna radiation field on the Hancock Building have now been made. A computer-derived theoretical analysis of the antenna radiation field on the Sears Building has also been made. The Hancock *measured* results and the

Sears *theoretical* results will be evaluated and compared to provide cross-substantiation.

5. Auxiliary Frequencies for Broadcast Use.

A number of filings are pending before the Commission for use of auxiliary frequencies for specialized non-broadcast services.

The Committee agreed that the following actions should be taken:

a. File a pleading to restore to broadcasting service the 5 Mhz of spectrum between 942-947 Mhz once allocated to aural studio-to-transmitter links but now assigned to Land Mobile Service on a "reserve" basis.
b. File a pleading with the Commission requesting the sharing of 6425-6525 Mhz presently assigned to the Television Remote Pickup Service but exclusive to the Common Carriers.
c. Continue to oppose the Petition for Rulemaking filed by Teleprompter Corporation (RM #2208) requesting expansion on a first-priority basis of the Community Antenna Relay Service band to include all the frequencies between 12,700-13,250 Mhz, which are now assigned to Television Remote Pickup Service.

6. Automatic Transmission Systems:

The Automatic Transmission Systems Subcommittee has concluded its initial work and its conclusions were filed with the Commission on August 21, 1974. We were supported in our position by a document filed by Electronic Industries Association. (It is anticipated that a Notice of Proposed Rulemaking will soon be issued by the Commission.)

7. Revision of Part #74 of the Rules (*Docket #20189–RM #1735*):

A considerable amount of time was devoted by the Committee in reviewing the draft of the proposed comments in the matter of amending Part #74, Subpart D of the Rules and Regulations pertaining to Remote Pickup Broadcast Stations.

The NAB filing refutes the Land Mobile Service statements that existing remote pickup channels used by broadcasters are lightly loaded; and further maintains that any additional channels obtained by the channel splitting process must be used by the broadcasters to satisfy their greatly expanding needs generated by the newly emerging Electronic News Gathering technique. NAB rejects the idea that the Land Mobile Services is in desperate need of additional space.

The NAB comments were filed with the Commission on February 21, 1975.

8. Operator Licensing:

The Committee reviewed the operator licensing situation, noting the Commission's interest in correcting examination deficiencies. Also reviewed was the certification program for Broadcast Engineers and Technicians as proposed by the Society of Broadcast Engineers. (A full discussion on the certification program will be given in an Early Bird Workshop tomorrow morning at 8:00 AM.)

Comments were filed March 5, 1975, requesting that the Provisional Third Class License be made valid for one three-year period.

9. World Administration Radio Conference (*Docket #20271*):

The Committee noted that a World Administrative Radio Conference on Satellite Broadcasting will be held in 1977 and a World Administrative Radio Conference is scheduled for 1979. Since the decisions reached at such international meetings result in binding actions for at least the next decade, the Committee passed a resolution stating that the Association should devote whatever time, energy and resources necessary to assist in developing the United States' position.

The resolution was presented to the NAB Board. The Board concurred and the NAB comments were filed with the Commission on February 14, 1975, requesting that NAB be represented on any group or committee that might be established. As of the time of this writing, the Commission has called one such meeting. NAB President Wasilewski was present.

10. OTP VHF-TV Drop-in Proposal:

At this moment, although the OTP drop-in proposal is somewhat dormant, it would have a significant effect on television broadcasting. The Committee is directed to remain up-to-date and is prepared to actively participate at the appropriate time.

11. Amendment Of AM Allocation Rules (*Docket#20265*):

The Committee reviewed the Commission's proposed rulemaking and agreed that the far-reaching ramifications of the proposal as it relates to various classes of stations and the interlocking effect the proposal has on station assignments is very complex. The Committee did recommend that a pleading for improved Class IV facilities be submitted to the Commission in the near future.

12. Technical Seminars:

The Engineering Management Seminar and the Directional Antenna Seminar continue successfully. The Committee recommended that a new, two-day seminar on Digital Techniques be developed. A firm date of May 21-22 has been set for Cleveland, Ohio.

Others:

Other subjects under consideration by the Committee include, but are not limited to: The Commission's recent action in establishing its Environmental Policy Requirements for stations constructing towers in excess of 300 feet and any all-directional systems; visual and/or aural coding of commercials; the Engineering

Handbook (expected to be completed in mid-1975); CATV; cartridge recording and reproducing standards; quadraphonic sound; TV sound; AM stereo; EBS; and satellite broadcasting.

This concludes the NAB Engineering Advisory Committee Report. I wish to thank the following committee members who have given so willingly and diligently of their time to these complex problems:

Charles F. Abel, Manager of Engineering, Station KFMB, San Diego, California

Ernest L. Adams, Vice President for Engineering, Cox Broadcasting Corporation, Atlanta, Georgia

Ralph F. Batt, Vice President & Manager of Engineering, WGN Continental Broadcasting Co., Chicago, Illinois

Albin R. Hillstrom, Vice President—Engineering, KOOL Radio & Television, Phoenix, Arizona

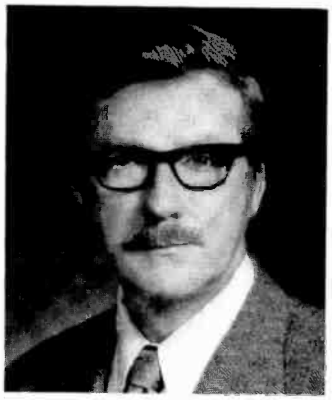
John R. Kennedy, Vice President—Operations & Engineering, National Broadcasting Corporation, New York, New York

Leslie S. Learned, Consulting Engineer, Mutual Broadcasting System, New York, New York

James D. Parker, Staff Consultant—Telecommunications, Columbia Broadcasting System, New York, New York

R. LaVerne Pointer, Director—Broadcast Engineering, American Broadcasting Company, New York, New York

Benjamin Wolfe, Vice President for Engineering, The Post-Newsweek Stations, Washington, D.C.



Robert A. O'Connor

Director, Transmission Engineering
CBS Television Network
New York, N.Y.

Chairman,
JCIC Ad Hoc Committee on Television Broadcast Ancillary Signals

Status Report on the JCIC Ad Hoc Committee on Television Broadcast Ancillary Signals

By way of review, the JCIC Ad Hoc Committee on Television Broadcast Ancillary Signals was set up to study the overall question of special signals intended for inclusion in the television program signal. The charge given the Committee included a determination of possible uses for such signals, the priority to be assigned such uses, identification of possible locations within the program signal which conceivably could be used for the signals, and the establishment of an overall technical framework for the accommodation of these signals, while maintaining the complete integrity of the program signal.

With respect to locations within the program signal which conceivably could be employed, the Committee in the interest of covering all possibilities, designated four categories for detailed study:

1. The horizontal blanking interval
2. The vertical blanking interval
3. The program audio signal, using time or frequency multiplexing techniques
4. The program video signal, using time or frequency multiplexing techniques

During the past year, theoretical studies and test programs have been conducted in all four areas with the following results to date:

The Horizontal Blanking Interval

The possibility of the use of this "time gap" in the program signal was suggested by the various time division multiplexing schemes that are in use, or have been proposed, in the U.S. and throughout the world, for the long distance transmission of the associated program audio signal. Such schemes involve pulse-amplitude, pulse-code, and pulse-width modulation techniques, using portions of the front porch, tip of sync and back porch including even encoding of the color reference burst.

With respect to the possibility of the use of such techniques for the *radiated* signal, however, grave concern has developed over the potential for improper television receiver operation that might be caused by signals in the sync or back porch area, or the potential for visibility of positive going signals on the front porch.

Some tests are scheduled for a pulse-amplitude front porch signal and a minimal 4-bit pulse code signal in the tip-of-sync which could have a bearing on the final conclusions. But as of now, the horizontal blanking interval appears to be the least desirable location for ancillary signals.

The Vertical Blanking Interval

The successful use of the vertical blanking interval for ancillary signals has been demonstrated over the years with test and reference signals, and tests are currently underway of a vertical interval signal in digital format.

Signals in the vertical blanking interval meet the basic, number 1 requirement, that they are not visible in the picture, (or can be made "invisible" by a slight adjustment of an external receiver control). Additionally with the use of conventional processing equipment, vertical interval signals can readily be removed, with new signals subsequently inserted. Information relating to the *reliability* of systems using vertical interval data signals will be determined in the course of tests currently underway.

As a result of the FCC's Report and Order which concluded the rulemaking procedure relating to the reservation of line 19 for the VIR signal, *all* of line 17 was made available for radiated signals. As part of this action the Commission adopted a few corollary proposals of the Committee which should further enhance the usefulness of the vertical interval.

Figure 1 shows the situation with respect to vertical interval signals that are currently being transmitted. On the line currently authorized for radiated signals, are depicted:

Line 17, both fields—The standard network transmission test signals—which are inserted at the key network origination points. These same signals are also transmitted on outgoing satellite feeds to other countries in accord with international agreement.

Line 18 both fields—A second appearance of these same signals which may be inserted at a regional network distribution point.

Line 19, both fields—The vertical interval reference signal.

Line 20, Field one—Under a Special Temporary Authority granted by the FCC, the commercial networks are transmitting the "source identification signal".

Additionally, on line 21, an area not presently authorized for regular transmission, the Public Broadcasting Service has authority to transmit the signal depicted which is used to provide a program captioning service for the hearing impaired. (All of the networks have been active participants in the work of the Committee and the data derived in the course of these tests will be made available to the Committee).

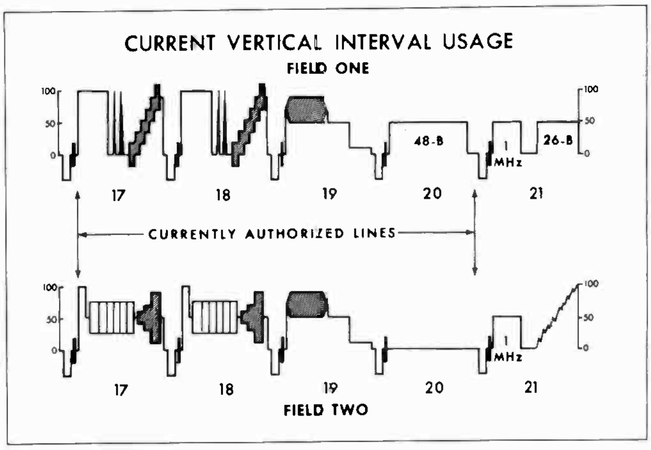


Figure 1.

Since the subject of a vertical interval signal in digital format is a very timely one, it may be of interest to point out a few key features which are common to all such signals.

As an example, Figure 2 is an expanded version of the source identification signal. It consists of a 48-bit "word", consisting of one-microsecond-wide pulses at half-amplitude in a non-return-to zero mode. The location of a "one" bit, or a pulse at half-amplitude, results in the addition of the value assigned to that space, in accord with the conventional pulse-code system.

Looking at the month interval, for example, four spaces are provided which allow for 24 or 16 combinations, which is, of course, more than adequate to cover all possible combinations.

In the illustration shown there is no pulse in the "8" position, pulses in the "4" and "2" position and no pulse in the "1" position which result in the sum of 6, indicating the sixth month. Prior to the information bits which start with the "source identification" (network and location) bits, are the so called "housekeeping" bits, the phase reference and start of message bits, which are used to time and synchronize the decoder with the encoder at the transmitting end.

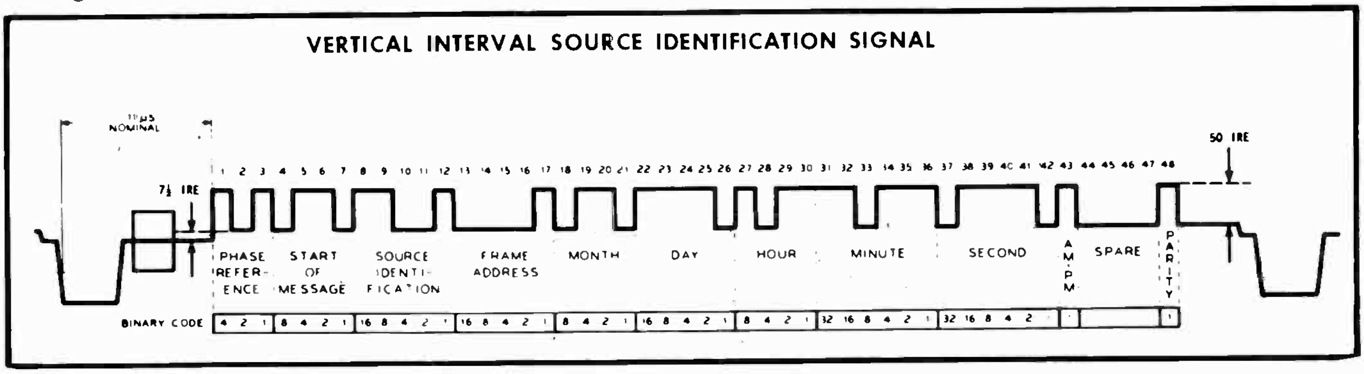


Figure 2.

Based on the prototype equipment being used for the field trials, this information as detected is printed out on a conventional teletype machine. In a future application of such a system, the information might also be stored in a solid-state memory device which could be polled, perhaps on a daily basis over the switched telephone network from a central processing location.

As indicated before although this signal was designed for a specific function that of network source identification, the general principles are applicable to vertical interval data signals in general, which the Committee believes hold a great potential for meeting many of the other proposed ancillary signal functions.

The Program Audio Signal

The concept of the use of the program audio signal for ancillary signals is an intriguing one since the technique would be applicable to all media: tape, film, and records, and also applicable to the aural broadcast services: AM and FM.

Two specific proposals have been made to provide a program or commercial identification system, both specifying time "windows" or "notches" in the audio band spectrum into which low level tones would be inserted, and modulated to provide the necessary information. These two systems, however, proposed "notches" at different ends of the audio spectrum and different methods of modulation.

To assist the Committee in its study in this area, the SMPTE established a Working Group on Aural Program Identification Systems to investigate the entire subject of frequency notches with inserted tones, and the effect of system imperfections on the reliable detection of such tones.

The subjective effect on the listener of notches does not follow any regular pattern across the audio band. There are discrete, optimum values where the likelihood of detection by the human ear is a minimum, and this phenomenon is currently under study.

With respect to imperfections in the system included are: velocity variations in origination equipment, distortion introduced in network transmission facilities, and frequency changes in the recorded tones when multiple copies of films and tapes are made. As indicated above, one source of error is the change in the frequency of the tone due to velocity errors at the origination point, that is in the telecine projector or video tape recorder.

As a first step in its study, the Working Group designed a test system for measuring start up and steady-state velocity variations, as well as a computer program for analyzing results. Measurements were made on the effect of frequency-shift keyed (FSK) signals recorded in the audio track of a test film and a test tape.

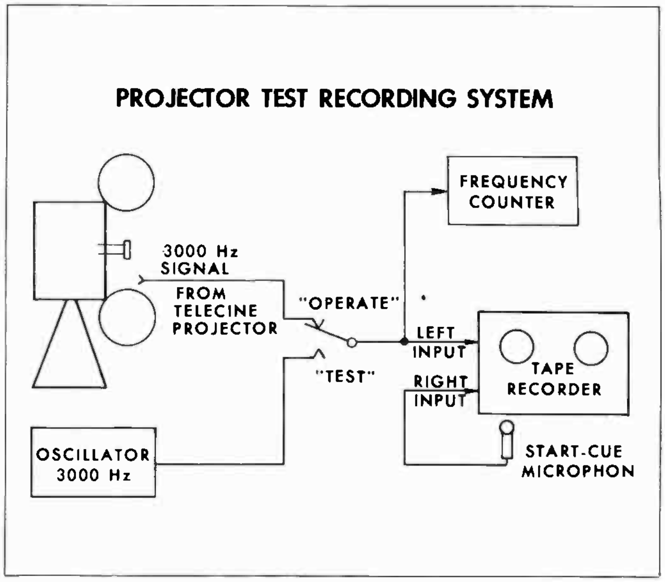


Figure 3.

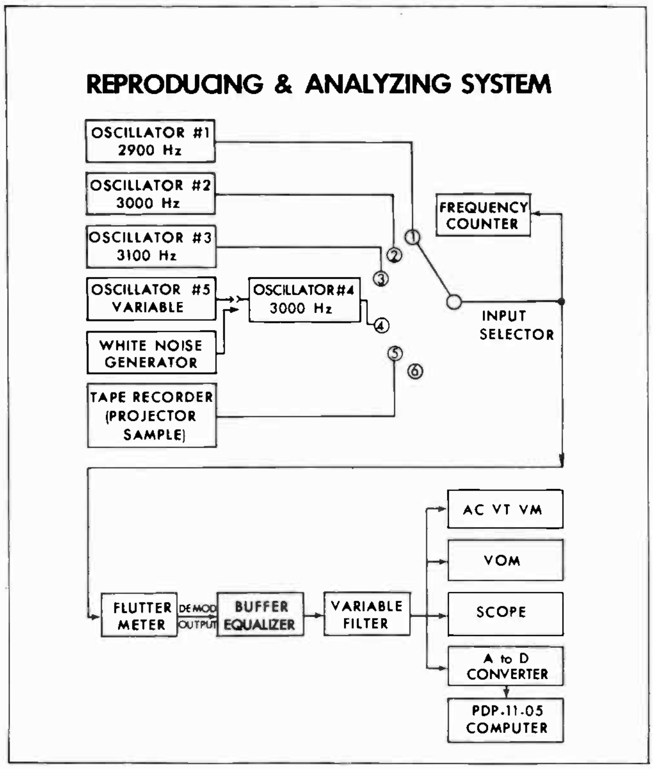


Figure 4.

The test recording system is shown in Figure 3—, and the reproduction and analysis system in Figure 4. A total of 46 telecine chains and 10 videotape recorders were tested.

It turns out that velocity errors in video tape machines are much less than in telecine machines with the result that the telecine projector represents the greater problem with respect to the source of the tones.

Figure 5 shows the average percentage error that would result from a given frequency shift, and given low pass filter cutoff frequency, the post detection low pass filter being used to eliminate as much of the noise caused by the velocity errors as possible.

For example, for a frequency shift of 50 Hz—where a "one" bit wold be 50 Hz above the center frequency, and a "0" bit 50 Hz below the center frequency,—and a filter cutoff at 200 Hz the *maximum* error rate would be 0.039%. (This error rate could, of course, be reduced by message redundancy and error coding techniques). The Committee believes that this type of analysis can provide the technical framework against which specific proposals can be analyzed. The other factors that effect reliability will also be studied by the Working Group.

The Program Video Signal

Under this category are included two basic techniques, one involving a *time* division multiplexing addition of an ancillary signal, whereby the added information replaces some video information, and a *frequency* division multiplexing addition, similar to the technique whereby color was added to the monochrome signal, using a subcarrier "interleaved" between two harmonics of the line frequency.

With respect to the time division technique the Committee has concluded that this technique, by definition, holds the potential for visibility in the picture and hence does not meet the basic, sine-qua-non, requirement of ancillary signals that no degradation to the picture should result from the addition of the ancillary signal. Also, the Committee believes that with the variety of techniques made available by current technology there is no *need* to employ such a system.

With respect to the frequency multiplexing technique, the Committee conducted a test of a system developed by Hazeltine Research Labs using such a technique.

This system adds information in alphanumeric format to the video signal, excluding the horizontal and vertical blanking intervals, that can be decoded and displayed on a home receiver. A low amplitude subcarrier around 2.5 MHz is injected at an odd multiple of one half line scan frequency, and bi-phase modulated in such a manner that the presence of the subcarrier is not perceptible in the picture.

Figure 6 shows results of tests conducted by Hazeltine on the perceptibility of the added signal on a color receiver viewed at a distance of four times picture height as a function of the level of the added ancillary signal. Note that at a level of about 43 dB below picture signal level, no observers were able to detect the presence of the added signal.

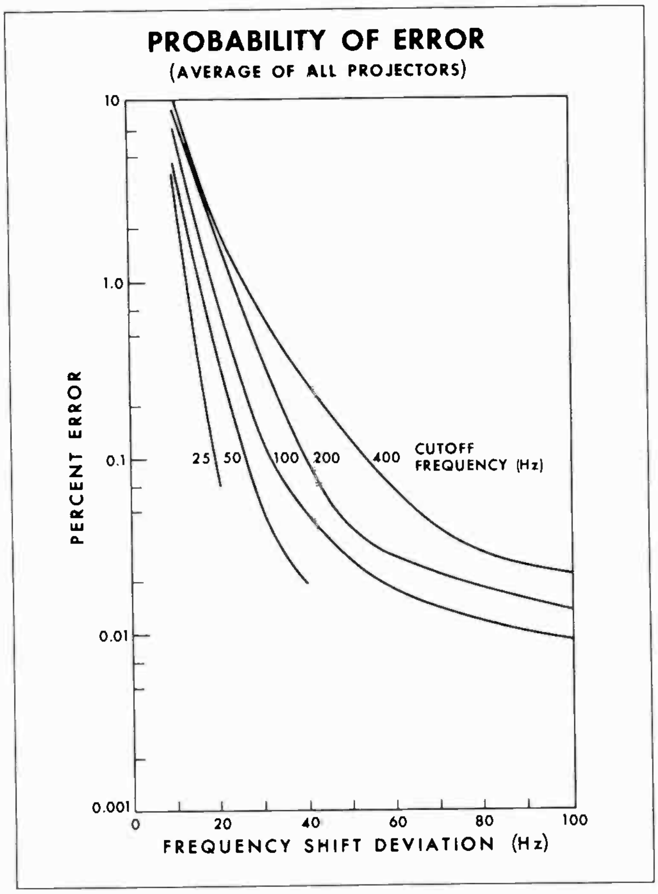


Figure 5.

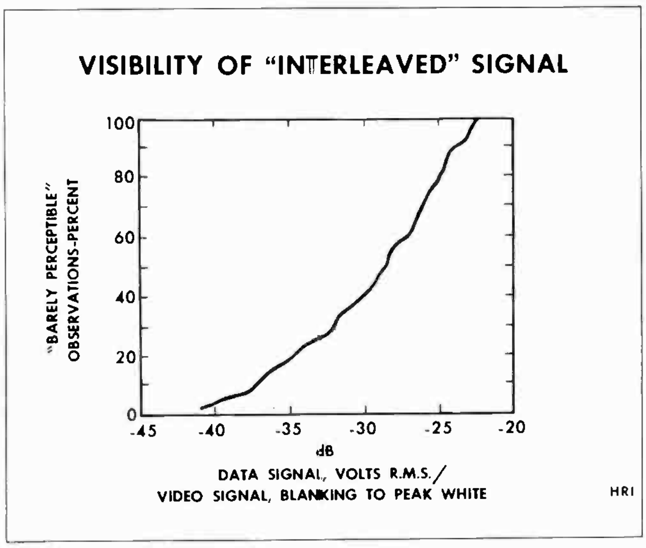


Figure 6.

The over-the-air tests conducted by the Committee employed a level of about this value, and it was determined that such a level was capable of providing a generally reliable service. (The *degree* of reliability is, of course, determined by other factors including data transmission rate and extent of error correction).

The Committee outlined certain operational problems relating to the system and the possibility of increased visibility on both monochrome and on color television receivers using wider color demodulation filters. It concluded that the technique has the *potential* for meeting the basic requirements of non-degradation and reliability. Although this test involved a specific system, the Committee believes that the results can be applied to the general technique of the use of interleaved "slots".

Conclusion

In accordance with the charge given it by the parent JCIC, the Committee submitted to the FCC its Interim Report No. 1 dated September 1, 1973.

The Committee plans to submit Interim Report No. 2 on June 2 of this year. This latter report, which will be also available to the industry, will cover in considerable detail the work previously described, and based on these findings will suggest general guidelines and a technical framework against which any future proposals for ancillary signals may be analyzed and evaluated.

Daniel R. Wells

Director of Engineering
Public Broadcasting Service
Washington, D.C.

Chairman
JCIC Ad Hoc Committee for Study of Television Sound

Status Report of the JCIC Ad Hoc Committee for the Study of Television Sound

The purpose of the JCIC Ad Hoc Committee for the Study of Television Sound is to examine the entire television sound system from the studio origination to the sound as heard in the home, to identify problems and opportunities for improvement, and to assign tasks to the appropriate organizations to accomplish these improvements.

The Committee consists of the five constituent organizations of JCIC (namely, EIA, IEEE, NAB, NCTA, and SMPTE, with SMPTE acting as the administrative organization). In addition, there are representatives from The Audio Engineering Society, the Network Transmission Committee and the PBS Engineering Committee. There are observers from the Canadian Broadcasting Corporation and the FCC.

With this broad base of representation, the Committee has the capability of looking at the entire television sound system, and this system approach is very necessary.

The work of the Committee has been divided into six panels:

Production Process—Chairman, Pierce Evans representing NAB.

The scope of this panel includes the original sound pick up and all processing in the broadcasting plant up to the interface with the common carrier.

Panel on Distribution—Chairman, Hank Ahnemann, representing NTC.

"Distribution" is defined, for purposes of this study, as transmission by the common carrier, both long lines and local loops.

Broadcasting—Chairman, Dick Burden, representing SMPTE.

Areas covered are the television transmitter and the over-the-air process.

Reception—Chairman, Mike Palodino, representing EIA.

This panel covers the home television receiver.

Cable and Master Antennas—Chairman, Blair Benson, representing NCTA.

State of the Art Techniques—Tom Keller, representing PBS/EC

This panel looks at new developments and their application to improved sound in television.

With regard to the overall system, the Committee is preparing a document on the assignment of tolerances to the various segments of the television system; that is, from the microphone through the home receiver. It will be arranged to include two parts. The first would identify the current level of engineering practices and the second part would tabulate areas where improvements could be expected, and the level of the improvements possible within the current state-of-the-art. The document would not be intended to be a firm specification but rather a set of design targets and guidelines.

Motion Picture Film is part of the study of the Panel on the Production Process. Of course, a large percentage of the program material in television originates on film, so this is one of the main areas of concern. At present, the Academy Curve equalization is being reevaluated. Also, the methods by which syndication prints are created is being reviewed. At present, it appears that syndication prints are created with little regard for the audio record contained therein.

The Committee is dealing with the question of the suitability of standard volume indicators (the so-called VU meter) as compared to peak reading meters. The American National Standards Institute has recently proposed withdrawal of the Practice for Volume Measurements of Electrical Speech and Program Waves, which includes specifications for the standard volume indicator. While there is much merit to utilizing peak reading meters as a supplement to, or in some cases instead of, the standard volume indicator, we should be aware that the VI is a standard recognized throughout the world, and that before giving up one standard, we should thoroughly understand why it was adopted in the first place and agree on a substitute standard.

The Network Transmission Committee has been preparing Report No. 6 which deals with Reporting Limits for audio transmissions carried by AT&T. In the first edition, the limits would be based on the current 5KHz service. NTC is also preparing a tutorial paper on the historical origins and current recommended practices of audio transmission. This would be a supplement to Report No. 6. Agreement has been reached within NTC on all aspects of Report No. 6 except for the transmission level at which tests are to be made, especially harmonic distortion tests. Broadcasters, of course, want to test at +18 dBm which is the instantaneous peak program level that the transmission system must be capable of handling within distortion limits. AT&T acknowledges this requirement but feels that steady state tones used for testing would cause interference with other services in the telephony super groups. Two solutions are being considered. One has been proposed by the ABC Television Network which involves the use of a chirp tone with momentary peaks of +18 dBm, but with an average level such that it would register reference (that is, zero at +8 dBm) on a standard volume indicator. The second approach being considered is to test at a time when AT&T traffic is low to minimize the loading effect.

It should be noted that at the present time there are *no* reporting limits published or agreed to by AT&T, so, the completion of NTC Report No. 6 is very important.

AT&T is also looking to the future and investigating diplexed audio and video as a means of improving quality and reducing costs. Tests have been completed on four diplexed systems, all of which would have 15 KHz response, and a +18 dBm transmission level capability. The results are currently under evaluation with regard to cost and tariff questions as well as technical performance.

There is a new topic which the Committee will begin studying which is domestic satellite audio transmission standards. The presently authorized satellite common carriers namely, Intelsat, Telesat Canada, Western Union and RCA, all have different methods of transmitting audio. Intelsat uses a separate R.F. carrier. Telesat uses a subcarrier at 6.8 MHz onto which three audio channels are modulated. Western Union intends to use a subcarrier at 7.5 MHz, and RCA is still studying alternatives.

The significance of standardization, at least among U.S. domestic satellite carriers, is that if a broadcast station owned its own satellite ground terminal, the equipment should be able to receive the sound portion regardless of the carrier from whom a satellite transponder is leased.

The AHCSTS asked the EIA/BTS Committee to review the question of multiple sound channels for television. BTS formed a Subcommittee for this purpose with Carl Eilers of Zenith as Chairman. The Subcommittee prepared a bibliography and chart depicting the essential features of all previously proposed multi-channel systems. It conducted a survey to determine if there was a need for multichannel sound in television. And it suggested an organizational structure suitable for addressing the questions of standards for over-the-air broadcasting of multichannel sound for television.

The BTS committee accepted this report without a recommendation as to whether the work should proceed. Instead, it forwarded the report to the AHCSTS for its consideration.

The AHCSTS Committee recently received the report and has not as yet acted on it.

The Panel on Broadcasting, that is, the over-the-air process, has been conducting tests on the pre-emphasis curve and whether that curve should stay at 75 microseconds. As part of this investigation, the panel is also looking into the effect of the standard volume indicator and peak reading meters as they relate to pre-emphasis. The panel is also looking into the question of limiting at the transmitter, whether the limiting should be before or after pre-emphasis, and what type of limiting should be used.

The panel plans over-the-air tests soon which broadcasters and various industry representatives would be invited to attend.

The State-of-the-Art Panel has been conducting tests on incidental phase modulation, that is, the carrier shift that occurs during AM modulation. In the intercarrier detection system of most television home receivers incidental phase modulation results in the audio being modulated with a 60 Hz hum. The panel is studying what corrective action can be taken.

These are some of the activities underway by the JCIC/Ad Hoc Committee for the Study of Television Sound. Many others remain to be addressed.

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Chairman
JCIC Ad Hoc Color Television Study Committee

Status Report of the JCIC Ad Hoc Committee for Color Television Study

Since many of you may not be acquainted with the work of the Ad Hoc Color Television Study Committee, a brief review of its organization and the charge from the parent Joint Committee on Intersociety Coordination (JCIC).

The Ad Hoc Committee was organized in 1968 by the SMPTE under the authorization of the JCIC, the latter representing the EIA, IEEE, NAB, NCTA and SMPTE. The charge to the group was to pinpoint the causes of the variations in color television pictures as viewed in the home, particularly in regard to variability in hue, saturation and color quality, and to initiate appropriate corrective action by the industry. The continuing investigations have been concerned with every element of the television system from the staging and lighting for electronic and film production, through studio and network transmission, and lastly through the home receiver.

The work has not been limited to paper or theoretical studies, but has involved extensive field tests in several cities in the U.S., laboratory measurements, and detailed surveys of operating practices in over 250 television stations. From the onset the findings have emphasized the complexity of the problem. In fact, the causes of color variability have been found to be so numerous, and to originate from so many elements in the overall system, that in only relatively few instances can any significant improvement be realized by the introduction of fixed corrections at intermediate points. As we all know, color balance can vary widely from program-to-program, and in fact from scene-to-scene in any single program. Furthermore, differences can be found among stations, cameras, and receivers and monitors.

In the initial exploratory investigations, the activities of the Committee centered on some of the more obvious parameters related to signal transmission. This effort resulted in the assignment to the EIA Broadcast Television Systems (BTS) Committee to come up with recommendations for tighter signal tolerances and to develop special test signals for transmission control. One of the most successful and noteworthy of these BTS activities has been Vertical Interval Reference (VIR) Signal, now a recommended practice for the broadcasting industry.

However, as was pointed out in our report to the NAB last year, since that time the work of the Committee has been directed more toward the problem of colorimetric uniformity in film and live origination, and with the color fidelity of display devices used in both home receivers and studio monitors. Although the original charge to the Committee by the Joint Committee on Intersociety Coordination (JCIC) was to point out the sources of color variability and to refer the problems to cognizant committees for action, since this Committee is the only group meeting regularly with representation from all of the JCIC organizations, it has served appropriately as a forum to discuss the direction and progress of the working groups.

Film Camera Colorimetry

The SMPTE Telecine Subcommittee, now the Telecine Working Group under the new engineering committee organization of the SMPTE, has held meetings in both New York and Los Angeles in order to facilitate involvement of cognizant engineers on both coasts. In addition, tests of camera characteristics in several television stations using various test slides have been completed. Because of the wide variations found, these tests have reaffirmed the need for a reference telecine camera characteristic and recommended operational adjustment procedures.

One of the slides used in the station tests was an experimental color bar slide developed by Eastman Kodak. The Working Group Members and station engineers expressed an interest in the availability of such a slide. However, specification of the test slide characteristics for industry use will have to await a definition of telecine colorimetry. Nevertheless, for the present the experimental slide has been found to be a valuable assist in appraisal of telecine equipment if attention is paid only to the relative reproduction of colors, not the absolute value.

The Working Group also has agreed that there is need for a new, short telecine performance verification film. Accordingly, the SMPTE Reference Film Working Group is conducting an examination of the current use of the presently available SMPTE Color Reference Film. Out of this investigation may be a proposal for a new and different film, specifically directed toward telecine performance verification.

In summary the Telecine Working Group has completed investigations of:

1) Degree of variation in gray-scale reproduction.
2) Usefulness and need for a color bar slide.
3) Evaluation of the degree of image enhancement or sharpness improvement suitable for telecine.

and in addition is continuing work on:

1) A reference telecine characteristic, working with full cognizance of a parallel activity being conducted by the EBU.
2) Consideration of specification for a new reference, or verification color test film.

Studio/Field Camera Colorimetry

The specification of live camera colorimetry, as well as that of telecine, has been held up pending agreement on specifications of the color display system characteristic. At last year's NAB Convention reports were given on this subject by Mr. DeMarsh, Chairman of the Camera Colorimetry Working Group, and Mr. Davidoff, Chairman of the Broadcasters' Task Force.

In order to demonstrate the differences among phosphors and the effect of matrices to compensate for phosphor difference, a demonstration was presented last August at Eastman Kodak for the Ad Hoc Committee and other industry representatives as an Interim Report of the Camera Colorimetry Working Group. Five different phosphors were used for the demonstration. These were:

Sony Consumer Products
Sony EBU-Specified
Sylvania Consumer Products
Conrac Controlled
Sylvania NTSC

Suitable compensating matrices were available to show the effect with and without correction for the Sony and Conrac controlled tubes. Video tape recordings made by two networks were used as the source material.

The consensus of opinion generally was that the NTSC phosphors and the Controlled phosphor with one particular matrix produced the best overall picture of the various scenes presented. However, it further was the consensus that the statement could not be made for all broadcast subjects until the field tests planned by the Broadcasters Task Force have been completed and evaluated. The charge to the Task Force covers three areas of investigation as follows:

1) Establish broadcasters typical operating practices with respect to live studio camera as information for the Colorimetry Subcommittee.
2) Study the technical problems involved in using a monitor with a correction matrix in conjunction with broadcasters operating practices.
3) Conduct a field test to investigate the use of a monitor with a correction matrix under actual broadcasting conditions.

In the meantime, inasmuch as the FCC Rules and Regulations are based upon NTSC phosphors, and the demonstration reflected no reason to deviate from that position, it was urged by the SMPTE Engineering Vice President that the Camera Colorimetry Working Group continue its standardization efforts based upon such a display and make recommendations for the best possible approach to achieve a standard display, simulating NTSC phosphors. Accordingly, the Working Group will proceed in this manner, and until subsequent evidence dictates otherwise, will assume that so-called 85% matrix as the most acceptable compromise with controlled phosphors.

Conclusion

Thus, it appears that substantial progress is being made on these most difficult and complex system colorimetric problems. Through the diligent and highly cooperative contribution of the committees and Working Groups, we may expect specifications and techniques to be developed in the very near future which ultimately will contribute substantially to more uniform and faithful color reproduction for the home receiver.

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What the New Environmental Impact Statement Requirements Mean to Broadcasters

The Federal Communications Commission, on August 4, 1972, issued a Notice of Proposed Rule Making regarding the Implementation of the National Environmental Policy Act of 1969. This act requires each governmental agency to consider the effects on the quality of the environment of any construction activity under its jurisdiction. The Act specifically requires that the agency consult with and obtain comments of expert Federal agencies before taking any major action significantly affecting the quality of the human environment and, following such consultation, to prepare a detailed environmental impact statement.

It is important to note that the responsibility for evaluating the environmental issues rests with the agency—the FCC in this case—and it must take responsibility for the scope and content of draft and final environmental statements. Implicit with this responsibility is the requirement for the agency to identify the types of actions requiring environmental statements, identify the typical environmental consequences of those actions, and set forth the basis to be used in evaluation of environmental impact.

The Commission received industry comments on its proposed rule making and on December 16, 1974 released its Report and Order (Docket #19555) promulgating the Rules and Regulations regarding environmental issues to become effective on January 20, 1975. The consequences of these new Rules may be very significant to broadcasters proposing new facilities or modifications to existing facilities.

The National Environmental Policy Act was intended to protect the public from the adverse effects of major private and public construction activities; such as dams, airports, refineries, industrial complexes, and the like. The typical environmental assessment for the construction of an airport would, for example, consider the following areas:

- Noise Pollution (Aircraft)
- Noise Pollution (Construction)
- Air Pollution (Aircraft)
- Air Pollution (Airport Facilities)
- Water Quality/Waste Treatment
- Alteration of Surfact Features (Drainage)
- Probability and Effects of Fuel Spills
- Social Effects
  - Relocation of People
  - Impact of Traffic
- Economic Effects
- Impact on Recreation Areas
- Impact on Wildlife
- Aesthetic and Visual Effects

Obviously, an environmental assessment of such a project requires the involvement of many areas of expertise and the preparation of a voluminous report discussing the various aspects.

Fortunately for broadcasters, the Commission has determined that broadcasting is generally a "clean industry"—in so far as its facilities are concerned, and with no reference intended to some of its programming—and, therefore, the major area of concern is with the last of the areas previously listed, namely the visual or aesthetic impact of proposed facilities. This is not to say that this will be the *only* concern as many of the areas listed in the case of the airport could be factors in certain broadcast situations. More on this later.

Having thus identified the major concern as the visual impact, the Commission defined the types of actions which it considers to be major—i.e., major in terms of the provisions of the National Environmental Policy Act. Broadcasters should take note of the distinction between major actions under the NEPA—which involve environmental considerations—as opposed to major actions under FCC guidelines—which involve service area and population considerations.

The FCC will require the submission of environmental information with all applications having major environmental consequences and has defined the major areas involving broadcast activities as

- AM/FM/TV towers over 300 feet.
- All AM directional arrays without regard to height.
- Other antenna structures over 300 feet (STL, microwave, translators, etc.).
- Satellite earth station antenna over 30 feet in diameter.
- Any facility located in
  - Designated wilderness area
  - Designated wildlife preserve
  - Listed historical districts
  - Areas recognized for scenic value

This listing is tailored to broadcast interests for this presentation and has omitted specialized areas of concern to other communication entities, such as common carriers where long microwave routes or underground cables are involved.

For further clarification, the Commission identified specific actions considered to have minor environmental impact and thus not requiring the submission of environmental data. These are as follows:

- Construction proposed in an existing "antenna farm" area.
- Modification of existing facilities not involving a substantial increase in tower height.
- Temporary structures.
- Replacement of an existing structure on the same site.

It should be noted that an "antenna farm" area is defined as a location in which similar towers are clustered and does not have to be officially designated as such under FAA terminology and definition. Further, a substantial change in tower height is defined as exceeding the greater of 20 feet or 10% of existing tower height.

It must be emphasized that these criteria are really guidelines for "normal" situations. The ultimate environmental responsibility rests with the Commission and on its own motion, or the motion of any interested party, may determine that the environmental consequences of a particular action are such as to warrant the preparation of an environmental impact statement.

In situations where broadcast applications fall into the "major" category, from an environmental standpoint, the applicant must file with his FCC application a narrative statement giving the following information:

- A description of the facility and its intended purpose, or service area.
- A discussion of the reasons leading to the selection of the site and the alternatives available.
- A description of the towers, access roads, utility lines required and any environmental considerations perceived.
- A statement of the local zoning situation and whether local approvals have been applied for and/or granted.
- A statement regarding any local controversy or objections to the proposed construction on environmental grounds.
- If adverse environmental effects are perceived, a discussion of the applicant's efforts to eliminate or minimize such effects by various means, including selection of alternative sites.
- Supporting statements by experts in the specific environmental disciplines involved in the event of unavoidable adverse environmental effects.

The applicant's discussion of reasons leading to the selection of the site will usually involve non-environmental considerations, such as:

- FCC requirements on station location relating to coverage and separations.
- Airspace and FAA approvals.
- Economic factors, including land values and station profitability.
- Availability of suitable parcels of land, particularly for large guyed towers or AM directional arrays.

If the applicant receives local approvals in the form of zoning variances or building permits, he will generally receive Commission approval and, in fact, the Commission will probably rely heavily on local approvals. However, in the case where the applicant receives approvals through the exercise of legal actions because his use of the land for broadcast facilities is technically within the permitted land uses, but this approval is received over the strong objection of local citizens on environmental grounds, he can expect a further review by the Commission.

The processing of these statements will be handled by Broadcast Bureau personnel in a parallel or simultaneous fashion with the standard portions of the particular application. In the usual case where no environmental considerations are involved, the processing procedure will not be lengthened by the additional processing required for the environmental statement. However, in cases where environmental considerations are involved, the processing of an application may be considerably delayed.

If the Commission determines that environmental considerations are involved which have not been perceived or adequately explained by the applicant, it will attempt to resolve the issues on an informal basis with the applicant. This procedure will also apply to parties objecting to the proposed construction. The Commission may, for example, seek to determine if the objectors have pursued the matter at a local level and, if not, suggest that local authorities be contacted, if appropriate. Applicants will be allowed to amend their statements and/or applicants will be allowed to amend their statements and/or applications to remove any environmental problems.

If the informal procedures fail to eliminate the environmental problem, the Commission will prepare a so-called Draft Environmental Impact Statement. This statement will cover the same points outlined earlier and, in general, it will enumerate the environmental consequences and, to the extent possible, evaluate their magnitude and significance. In addition, it will discuss alternatives, including those proposed by the applicant, advocated by opponents, and developed by the Commission. The Commission also may set forth action which could be taken to eliminate objections.

When the Draft Statement has been completed, the Commission will publish a notice in the Federal Register and will send copies to concerned Federal and other public agencies, as well as the Council on Environmental Quality. The notice will specify that comments will be accepted by the Commission for a 45 day period after publication. The applicant will have an additional 21 days to respond to the comments.

Subsequently, the Commission will prepare a Final Environmental Impact Statement. This statement will be distributed to the applicant and other parties of interest. The Commission will then consider the Final Statement in determining whether to grant the application or designate for hearing on the environmental issue. Broadcasters should note that a new area has been added to the list of reasons for which an application can be designated for hearing.

A typical timetable for processing an application on which an environmental issue has been raised may be increased by approximately the following number of weeks:

• Informal Review with Applicant	4
• Commission Prepare Draft Statement	4
• Public Notice Period	7
• Reply Period	3
• Prepare Final EIS	4
• Review and Issue	2
	24

Thus, the processing of an otherwise uncomplicated application could be delayed for up to six months if this procedure had to be followed. Note, however, that this does not include the time involved if the Commission should designate the application for hearing on environmental grounds. As we all know, this could involve an indeterminate period well in excess of a year.

At this point, it would be well to mention some additional considerations which could be factors in making environmental assessments, even though their occurrence will be rare:

- Radiation Hazards
- Hazards to Migratory Birds
- Falling Ice Hazards
- Energy Consumption
- Activated Related to Construction of the Facility
- Noise from Standby Generators

A facility is deemed to constitute a radiation hazard if persons in the area surrounding the antenna would be exposed to radiation in excess of 10 milliwatts/CM^2 for over six minutes. This standard is derived from OSHA requirements. Broadcasters should note that this is a preliminary value under current study. It does not take into account the operating frequency. As an example, this standard would require that a person should not be able to get within approximately 500 feet of the antenna of a full power UHF TV station. Because of the vertical radiation characteristics of high gain antennas, such as those normally used in UHF broadcasting, it is unlikely that anyone would be exposed to the major lobe within this distance. An exception to this rule of thumb might arise in the case of an antenna mounted on top of a tall building in a downtown area where persons in the upper floors of nearby buildings might be exposed to high levels of radiation. Consideration must also be given to personnel on the roof of the buildings on which such antennas would be mounted. Stations operating weather radar equipment should also be concerned about the very high Effective Isotopic Radiated Power (EIRP) levels achieved by that equipment, and the fact that the antennas are usually installed at relatively low elevations. The Commission has also advised all existing licensees to review their existing installations to determine whether a radiation hazard exists. If it is so determined by the licensee, he is required to notify the Commission and initiate corrective or protective measures.

The matter of migratory bird problems is particularly significant in terms of tall towers. The Commission has indicated that it expects applicants to seek out, consider and report information concerning local migratory bird patterns. The main hazard appears to be to smaller birds who fly at low levels at night and are attracted by tower lights. There is some indication that strobe lights may minimize this hazard. Applicants faced with such considerations will find that they will probably have to consult with local ornithologists at state agencies or universities.

Ice hazards, noise, and construction activity problems are self-explanatory. These would obviously be of most concern in facilities located in or near residential or other nonindustrial type areas.

The Energy Consumption aspect is not presently a problem, but could be of long range concern if power shortages continue to develop. Transmitters for full power UHF facilities might be open for criticism, particularly if circular polarization becomes a reality.

In summary, the applicant for an FCC broadcast construction permit is faced with a new area of problems involving the impact of his proposal on the quality of the environment. In the future, he will have to "do his homework" in selecting a site for a new facility to document his efforts to achieve a design which has a minimum impact on the environment while meeting all of his objectives and FCC requirements in terms of broadcast service to the public.

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What's Been Happening at the FCC— A Review of Pending Dockets

Again this year, as was done at the NAB Engineering Conference in Chicago in 1972, the NAB Engineering Conference Committee has requested me to set the stage for the FCC Technical Panel which follows, by outlining the FCC Rule Making procedures and by reviewing some of the more recent actions that have occurred, as well as some of the pending actions awaiting final disposition. I have two disclaimers: first, this presentation will be concerned only with matters involving technical elements; and, second, I am speaking not as a CBS employee but as a member of the NAB Engineering Advisory Committee, which acts on behalf of the NAB membership in these matters.

The Rule Making process is basically a two-step process involving, first, a petition, and second, the proceedings; except, of course, for proceedings initiated on the commission's own motion. For the initial step any interested party may petition for the issuance, amendment, or repeal of a rule or regulation. If the petition meets certain basic statutory requirements, it will be given an "RM" file number, and promptly thereafter public announcement will be made through an FCC "Public Notice". Interested parties may file within 30 days of the date of the Public Notice statements in support of or in opposition to the petition, and replies thereto may be submitted within 15 days thereafter. Past experience seems to indicate that broadcasters pay little attention to this initial step. This should not be. It is important to make your views known at this stage because it could have a bearing upon the Commission's disposition of the petition. The Commission has three alternatives; first, it may institute formal rule making proceedings; second, if notice and public procedures are not required, it may issue a final order amending the rules; or, third, it may deny the petition.

Turning now to the second step, the proceedings, notice of formal proceedings is ordinarily given by publication of a "Notice of Proposed Rule Making" or "Notice of Inquiry" in the Federal Register. Public notice is also given through FCC releases issued daily. The full texts of such "Notices" are available from the Commission, and they contain among other things, the Docket number assigned to the proceedings, the substance of the proposed rule, or amendment, a description of the issues involved, and the date for filing comments and replies thereto.

Any interested party may file comments in rule making proceedings. The NAB actively participates in many rule making proceedings, and where technical matters are involved, seeks the advice of its Engineering Advisory Committee. It is most important that individual broadcasters make their views known to the Commission, but if you do not find it possible to submit your views formally you are urged to pass on your suggestions to Mr. Bartlett, NAB Vice President for Engineering, or to any member of the NAB Engineering Advisory Committee so that the NAB may be able to present the composite views of the broadcasting industry as fully and as convincingly as possible.

After all comments have been placed into the record, including oral argument when deemed warranted by the Commission, the Commission, upon consideration of the full record in the proceedings and other relevant facts will issue its decision, usually in the form of a "Report and Order", incorporating therein its findings and a statement of the reasons therefor. Normally, any rule issued by the Commission will become effective not less than 30 days from the date it is published in the Federal Register.

The purpose of reviewing the basic procedural steps involved in FCC rule making proceedings is to encourage you as individual broadcasters to participate and to make your views known. The steps involved are set forth in Part 1, Subpart C—Rule Making Procedures of the FCC Rules and Regulations. In addition, I should like to call to your attention a Memorandum sent to the NAB Membership by the NAB Legal Department in August, 1971 on the subject "Filing of comments in FCC rule making proceedings". This memorandum explains in some detail the form and manner of filing comments. Copies of this memorandum are available from the NAB.

Turning now to "what's been happening at the FCC". Far be it for me to have an "inside" answer. I can only report as an "outsider" from what appears in the record.

As far back as April, 1972 the FCC instituted a broadcast regulation study and established a "Part 73 Task Force" to receive informal views and suggestions in connection with the study. This effort has been labelled "re-regulation of radio and television broadcasting". Hundreds of comments and suggestions were received, covering a wide range of topics, not all of which were technical. Since then the Commission has issued several re-regulation orders relaxing the rules on a number of technical operating matters, such as frequency of meter readings, frequency of inspection of transmitting equipment, logging requirements, auxiliary transmitter testing, visibility of transmitter, operation of unattended TV auxiliary broadcast stations, posting of operator licenses, and inspection, calibration, and testing of TV remote control equipment, to mention only a few.

Of the Dockets that were "terminated" within the past year, there are perhaps five that are of general interest.

In Docket #19555, the FCC adopted rules to implement the National Environmental Policy Act, adding a new Subpart I to Part 1. Of particular interest to broadcasters was the classification as a "major" facility any AM, FM, Television and international broadcast antenna tower or supporting structure, which exceeds 300 feet in height, and any AM directional array regardless of height.

In Docket #19906, the rules were amended to permit the use of extension meters for transmitters subject to certain restrictions regarding the physical accessibility of the operator's location in relation to the transmitter, i.e. they must be in the same building no more than one floor apart, and the path length must be no greater than 100 feet.

In Docket #19907, the Rules regarding special signals in the vertical blanking interval were modified to reserve line 19 for the transmission of the Vertical Interval Reference (VIR) signal, and to move the test signals previously specified for transmission on lines 18 and 19 by stations operated by remote control to lines 17 and 18.

In Docket #20012, the Commission amended its Rules to provide the technical framework within which FM broadcast subsidiary service, operating with an SCA, may be rendered by visual as well as by aural means. The amended rules establish engineering standards applying to visual systems such as broadcast-related facsimile, slow-scan television, teletypewriter, and data transmissions.

In Docket #20215, the Commission amended its rules to clarify the requirements for calibration of remote control instruments at stations with remote control authorizations in the several broadcast services.

In another action, but not through "Docket" proceedings, the Commission announced on December 4, 1974, that it had adopted a Two-Tone Attention System for inter-station signalling in the Emergency Broadcast System (EBS). Because of an anticipated lead time of 12 months to allow for the manufacture and installation of Two-Tone generating equipment at all AM, FM and TV broadcast stations, the revised rules require the use of the existing Attention Signal through January 15, 1976.

Turning now to actions currently pending at the FCC they will be reviewed in two categories: first, those matters which have been formalized into Docket proceedings; and, second, those matters which are the subject of Petitions that have been submitted, but which are awaiting disposition. Needless to say, it is quite impossible to review all pending matters, because they are simply too numerous; many are of specialized interest, such as changes in the Table of FM Assignments, and many others are concerned with non-technical matters.

Of the really old dockets, i.e. more than 10 years old, that have not been formally terminated, there are four that might be mentioned in passing: Docket #6741, the clear channel proceedings; Docket #11279, subscription television; Docket #14185, revision of the FM rules; and Docket #14614, prohibition of duplicate STL or inter-city frequencies.

Turning now to more current activity, i.e. less than 10 years old, first there is the combined Docket #16004/18052. Docket #16004 was initiated in May, 1965 with a proposal to revise the F(50,50) field strength curves and to adopt new F(50,10) curves for TV and FM services. In Docket #18052, initiated in March, 1968, it was proposed to provide a new method of measuring the location of service contours or the establishment of fields prevailing at specific locations. In April, 1971, these dockets were combined along with a proposal to redefine the Grade B contours for TV stations.

Docket #18063 was initiated in March, 1968 seeking comments on a proposal, by Collins Radio, to eliminate the requirement for meters as indicators on AM modulation monitors, permitting in lieu thereof an instrument incorporating a series of indicating lights that would respond successively to specified levels of peak modulation.

In March, 1968, responding to a Petition submitted by Collins Radio, the Commission issued a Notice of Inquiry on the matter of automatic FM broadcast transmitters. The matter came up again in Docket #18930 relative to operator requirements for AM and FM stations. In finalizing this latter proceeding in June, 1972, the FCC stated that it did not have sufficient information at that time to initiate formal action on the matter of automatic transmitters. Since that time the NAB Engineering Advisory Committee conducted a study on the question of automatic transmission systems for AM and FM and submitted a report on February 3, 1973 to the FCC summarizing its findings. Later, this Committee prepared a follow-on report on automatic transmission systems for television, which was submitted to the FCC on August 19, 1974. In addition, the Electronics Industries Association (EIA) submitted a report to the Commission on November 7, 1974 on the subject of unattended television transmitters. It is understood that this whole matter now has high priority within the Commission.

In March, 1969, the Commission on its own initiative, issued a Notice of Inquiry, Docket #18505, in the matter of possible changes in certain TV signal specifications to facilitate international program exchange. This inquiry was also addressed to the matter of availability of lines in the vertical blanking interval for the transmission of test signals and for other purposes, but, certain aspects of this particular issue were considered in Docket #19907, noted earlier, which reserved line 19 for the VIR signal.

On the matter of identification of program material simultaneously with its transmission, Docket #18877 relates to the inclusion of coded information in the aural transmissions of radio and television stations for this purpose. On the same general matter of program identification, but specifically with respect to the inclusion of program identification patterns in the visual transmissions of television broadcast stations, it will be recalled that in terminating Docket #19314 in October, 1973 the FCC reaffirmed the amendments to the rules adopted in Docket #18605, namely, to permit coded patterns within the first and last ten microseconds of lines 21 through 23 and 260 through 262 (on a "field" basis), for a period of two years, ending November 30, 1975. During this period the FCC expects "that intensive efforts will be made to modify the existing identification system so that it will be capable of functioning satisfactorily in accordance with this rule". The reference to the "existing identification system" is to the one developed by the International Digisonics Corporation (IDC). IDC has submitted two progress reports. The matter will be reviewed by the FCC at the end of this two year period.

On the question of aural STL and other operations in the 947-952 MHz band, the Commission initiated Docket #19130 in January, 1971, proposing to permit aural STL operations for AM and FM stations in the 2150-2160 MHz band, on a shared basis with common carriers. It will be recalled that the broadcasters lost the lower half of the 942-952 MHz band to the land mobile service in Docket #18262. In April, 1972, Docket #19130 was combined with Docket #19494 in which latter docket the proposal for shared use of the 2150-2160 MHz band was withdrawn, and in lieu thereof it was proposed that AM and FM stations in ten of the larger cities could utilize frequencies in the 2110-2113 MHz band for aural STL operations.

In March, 1971, the FCC issued a Notice of Inquiry, Docket #19183, seeking to develop data and recommendations as to TV receiver design and FM broadcast assignment principles to be used to alleviate various kinds of interference to television reception.

In February, 1973, the FCC issued a Notice of Inquiry and Notice of Proposed Rule Making (Docket #19692) on the matter of establishing standards for the design and installation of sampling systems for antenna monitors in standard broadcast stations with directional antennas.

In Docket #19931, adopted in February, 1974, the FCC proposed to amend its rules to incorporate therein specifications for the use of high intensity (strobe) obstruction lighting systems on skeletal structures as described in an FAA Advisory Circular. The major concern here for broadcasters was whether such specifications would be mandatory or permissive.

In what could well be characterized as a major undertaking, the FCC adopted in September, 1974 a Notice of Proposed Rule Making in the matter of amendment of Part 74, Subpart D, relating to the operation of remote pickup broadcast sations (Docket #20189). This Notice was in part in response to a petition filed by the NAB—"in part" because the NAB petition had also requested amendments to Subparts E (aural broadcast STL and inter-city relay stations) and F (television auxiliary broadcast stations). The proposal involves a complete revision of Subpart D and features the system concept of licensing, rather than separate licensing of individual stations, the creation of additional channels in the 450 and 455 MHz bands through "channel-splitting", i.e. the reduction of channel width from 100 kHz to 50 kHz, and the creation of several 10 kHz channels for non-voice emission.

In Docket #20195, adopted in September, 1974, the FCC issued a Notice of Inquiry and a Notice of Proposed Rule Making in the matter of the use of wireless microphones on unassigned VHF television channels 7 through 13.

In Docket #20265, adopted in November, 1974, the FCC is proposing to amend its rules regarding AM station assignment standards so as to permit power increases for existing AM stations if such increases can be accomplished without any derogation of present interference rules and would otherwise comply with the rules. Comments may also be submitted, if desired, on possible rule changes governing new daytime-only stations, new unlimited-time stations, and changes in frequency or nighttime operation by existing daytime-only stations.

As a final item on the matter of dockets, there is one of major significance—Docket #20271. This relates to the United States preparations for the 1979 World Administrative Radio Conference. This Conference will be concerned with possible revisions to the international Radio Regulations, which include the international table of frequency allocations, and is the first such major Conference since 1959. Comments in response to the first Notice of Inquiry, adopted in January of this year, have already been submitted. The FCC held its first public meeting in February to describe the Commission organization that has been established to coordinate this preparatory effort, and to explain how the public may participate. The focal point will be an all-FCC Steering Committee, under the chairmanship of Mr. Fine of the Office of Chief Engineer. Reporting to the Steering Committee will be five Radio Service Working Groups, one of which will be the Broadcast Working Group, under the chairmanship of Mr. McNaughten of the Broadcast Bureau, which will be concerned with all broadcast and broadcast related services. Any interested representative of industry may serve on the Working Groups as an "observer". Time does not permit a more detailed discussion of all facets of the organizational structure, but full information is available.

Turning now to the matter of petitions, there is a significant "back log" of petitions awaiting disposition by the FCC, but in the time available only a few will be mentioned, the older ones first, working up to the more current petitions.

The question of the transmission of discrete four channel quadraphonic sound was initiated in RM-1847, filed by Pacific FM in August, 1971.

The National Bureau of Standards proposed in its Petition, RM-2108, that all of line 21 in the odd field and the first half of line 21 in the even field, in the vertical blanking interval, be allocated for the transmission of the "NBS TvTime signals" and other forms of digital information.

In early 1973 the Land Mobile Communications Council requested in RM-2114 that the Commission issue a Notice of Inquiry on the matter of the UHF television channel assignment "taboos".

In June, 1973, TelePrompTer, in its Petition RM-2208, requested that the remaining upper portion of the 13 GHz band (12.95—13.25 GHz), presently available on an exclusive basis to TV auxiliary stations, be made available for the Cable Television Relay Service (CARS), on a shared basis, as is presently authorized in the lower portion of this band (12.7-12.95 GHz).

In March, 1974, the United Church of Christ, and others, in Petition RM-2346, requested the addition of new VHF channel assignments in the top 100 markets.

The Atlantic Research Corporation, in Petition RM-2473, filed in October, 1974, has requested amendment of the rules to permit the transmission of a "dot", 12 scanning lines high, and of equivalent width, within the television picture, for the purpose of providing an ancillary information channel for receivers equipped to decode the digital transmissions.

In November, 1974, the National Association of Business and Educational Radio (NABER), in Petition RM-2475, requested that the channel bandwidth in the 450-451 and 455-456 MHz remote pickup broadcast service bands be reduced from 100 kHz to 25 kHz, with some of the additional bands thus created reallocated to the Business Radio Service, and some held in reserve. The Petition further requests that the Commission defer action on that portion of Docket #20189 which affects these bands.

Finally, ABC filed a petition, RM-2518, in February, 1975, requesting that the rules be amended to permit the use of circular polarization for television broadcasting. As you are aware, this subject will be discussed in some detail in the television session.

In conclusion, it is hoped that this brief review of accomplishments and pending actions at the FCC on significant technical issues has raised some questions in your minds. However, the real experts on "what's been happening at the FCC" will appear on the FCC Technical Panel, which follows immediately.

Moderator:

R. LaVerne Pointer

Director, Broadcast Engineering
American Broadcasting Company
New York, N.Y.

FCC Technical Panel

Mr. Pointer:

Good afternoon, ladies and gentlemen. During the past few moments, Jim Parker has given us a rundown on the status of the various dockets pending at the FCC, and now is your opportunity to ask questions. But now let me introduce the panel to you.

To my immediate left, is Neal K. McNaughten, Chief of Rules and Standards Division, Broadcast Bureau. Next is Phil Horne, Chief, Field Operations Bureau. Next is Dennis Williams, Acting Chief of the Existing Oral Facilities Branch, again of the Broadcast Bureau. Wally Johnson, Chief, Broadcast Bureau.

Let's move right ahead with the questions. . . .

Q —I was wondering if the panel would comment on the newly required EBS two-tone system. Whether stations can construct their own equipment. What can we use for equipment? Is there going to be FCC approval or Type Acceptance for certain circuits, or what?

Mr. McNaughten:

The staff is giving considerable consideration to the two-tone matter at the moment directed toward Type Acceptance of the encoder and possibly Certification of the decoder.

In order to discuss this more intelligently, would Ray Sedden come up and give us a hand. Ray is Chief of the Emergency Communications Division and could fill us in on some of the details that maybe we're not really acquainted with here. I know some of the questions we've been encountering at the FCC booth relate to percentage of modulation for each of the tones.

We have had some concern expressed by licensees who have been approached by dealers with equipment to sell. We're also faced with the problem of grandfathering.

Mr. Sedden:

To answer the specific question about Type Acceptance and Certification of the equipment associated with the two-tone EBS attention signal, there is considerable interest among the staff to develop some procedures to require Type Acceptance of the encoder and Certification of the decoder.

I think you're concern lies with what happens if you purchase some equipment now and the Commission comes out with rules and regulations later on to require these procedures. I think the answer there is that if you were to purchase equipment today and the Commission promulgated these rules at a later date, then if the manufacturer would submit a prototype of that same piece of equipment and it were approved by the Commission, I see no problem then with the manufacturer certifying that piece of equipment that he previously sold to individual broadcast stations.

Now, for those stations that wish to construct their own equipment, I know this is an expensive procedure to get Type Acceptance and Certification of equipment through the Commission, and there is precedent in the Commission rules for equipment that is built by individual broadcast stations for their own use in not more than five units and not for resale, or not for sale, to not be required to go through these Type Acceptance procedures. However, it's up to the individual stations to test equipment and to certify that it meets the Commission's specifications in 73.906 of the rules.

Q—As a manufacturer of equipment is the date for compliance with the new two-tone system January 1976?

Mr. McNaughten:

Yes, that is the present date.

Q—We have to manufacture and sell this equipment before that date. When are we going to know exactly what the Commission will require?

Mr. Sedden:

We realize this is a problem, and we are trying to pomulgate rules as soon as possible. There is a possibility that the January 15, 1976 date may be extended, but hopefully we'll be able to meet the deadline.

Mr. Pointer:

May I remind you that technology today is becoming more complex; lead times are becoming greater. And I don't see anything new of this nature taking less than six-nine months minimum.

Q—Just out of curiosity's, the two-tone type of system as proposed for EBS is nothing new. Emergency communication pagers, pocket pagers and so forth, have been using it for years. Why wasn't a system like it used. With commercial equipment already available for these particular applications why devised a new system that's a lot more stringent than the tone-type systems that are used in two-way radio communication?

Mr. Sedden:

This matter was studied and rehashed and restudied for over 10 years. I think one of the reasons to go with new two-tone frequencies was so that there would not be any confusion and chance of false activation of the EBS system. When it was first conceived, it was to be a home-warning system and receivers were to be placed in the home. But in 1971, OTP recommended to the Commission that it delete the warning function from the EBS. So then the new signal just became an inter-station broadcast signaling system. And for that reason it was felt that a brand new system would be advisable to avoid receivers going off inadvertently because of other similar systems.

Mr. McNaughten:

I think one of the things you ought to know about this system is that, it was originated by probably one of the most outstanding groups of audio engineers in the country. In the selection of the tones, we had people who had worked on the touchtone system for the Bell Labs, therefore we had no problem in selecting the frequencies. There was something in the order of a quarter of million dollars worth of receivers and generators purchased to operate on these, and they were tested over a period of months in several different parts of the country, which proved that the system worked. That was what we were after and why we're trying to stick to the present rules.

Granted you can buy a two-tone burst generator for your amateur rig for $26 that's within one percent of 1800 cycles and 2000 cycles. I don't know what decoders cost, but that's what Heathkit lists for encoder.

Mr. Sedden:

Neal was the original project officer on the development of the two-tone system before he moved on to bigger and better things.

Q—I would appreciate some comment on the contents of the Commission's newly adopted Notice of Inquiry relating to unattended transmitters.

Mr. McNaughten:

You mean in connection with the EBS signal, or is this a separate proceeding?

A: No.

Q—While the gentleman's still here, I'd like to get some explanation on the percentage of modulation associated with the transmission of the two-tones.

Mr. McNaughten:

We had no trouble during the tests. Those stations who had the generators and who ran the tests for us were asked to leave all their audio processing gear in line, at the 40% level. We had one station that nearly damaged their transmitter trying to come up to 80% when both tones were applied. But actually, the tone was quite adequate because the decoder consisted of nothing but a Schmidt trigger which provided a square wave to excite the reeds.

We are not proposing that manufacturers use this method, but in this system the 40% plus or minus 5 was quite adequate, which gives you between 70% and 90% modulation level. Now, this is off-air modulation we're talking about, as measured on your modulation monitor. You can certainly set up a 40% modulation figure on the single tone. And we never did have any problem with the level of the tones on the carrier, whether it was AM, FM or television, and it was tested on all three services.

Q—I believe it's written up as a plus or minus 5%.

Mr. McNaughten:

Well, I don't think it's that sticky, really, because what we wanted to do was to be sure. If you get up around 95 or 100 percent, then you're bringing into action a lot of audio processing equipment. I'll agree you bring in some of it below that, but this was the figure which the committee picked. I'm afraid we'd have to go back in the record, to give you all the details on it. But 90% seemed to be the maximum that was needed, not that was desirable, but was needed.

Mr. Johnson:

One more question on EBS and then we'll get to the automatic transmitter question.

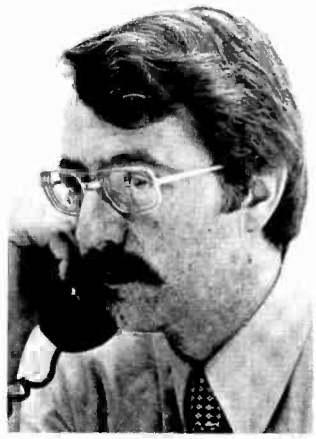
Mr. Pointer:

Do we have another question on the EBS?


Mr. Johnson


Mr. McNaughten


Mr. Horne


Mr. Williams

Q—I read the release by the FCC about three times and I'm still confused as to the need for a television station to be equipped with the generating equipment. I can well understand why we need the receiver, obviously, because it's an inter-station system. Now, why do we have to buy a generator for TV broadcast stations?

Mr. Sedden:

This same question came up earlier, and a gentleman asked why nonparticipating stations were required to install the signal generating equipment. The reason is that all broadcast stations, regardless of their participation at the national level, are eligible to participate in state and local emergency planning and situations. And in the event that there is a statewide or local emergency and the audience is either watching television or listening to their AM or FM stations in the area, and if the EBS should be activated at the state or local level, then these stations, regardless of their participation at the national level, can participate at the state and local level and transmit the attention signal. And anybody that's monitoring them would receive this information.

Without the requirement that all broadcast stations must have this equipment, then it's possible that those stations would not then be in a position to relay this information to the next station down the line.

There's another consideration also, and that is, in some areas, regardless of the participation of a television station, it may be that due to propagation conditions and local conductivity, that the only thing they can monitor is either a nonparticipating or a television station. And for this reason, it's required that all broadcast stations be capable of transmitting the attention signal.

Q—Is participation in the state system not voluntary?

Mr. Sedden:

That is correct. That's why I tried to explain that at the national level nonparticipating stations would go off the air. But at the state and local level, in the event of an emergency, nonparticipating stations do not need an emergency broadcast system authorization to participate at that level, so that they could participate to the extent that they desired to participate.

Q—But how can you impose on a station that is, in a voluntary situation, an affirmative requirement for the equipment? In other words, if a station chooses not to participate, then why do you impose the requirement for the equipment?

Mr. Sedden:

Approximately 95% of the total number of broadcast stations in the country are participants in the EBS. The reason to require those 5% to transmit the attention signal is that in the event they would participate at the state and local level, they'll have the equipment available. And also in the event of a national emergency then the stations that can only monitor those nonparticipating stations would have an alternate method of receiving the emergency action notification that there is an emergency situation.

Q—In the past year we've had an awful lot of tornadoes that have hit this country. The key EBS station will not transmit the emergency action notification of a local weather bureau request. They say, "It's voluntary. I don't have to do it."

Personally, I feel that in a weather situation it should be mandatory because we do not have direct wire service to the weather bureau that they have. They get an alarm from the weather bureau, and there's nothing to alert us at the station level.

Mr. Sedden:

I haven't heard of any situations such as that, but I have heard of one situation where the National Weather Service requested a key station to activate the EBS because of a tornado warning, and that station erroneously was under the impression that it had to be cleared by the local Civil Defense authorities before they could activate the EBS.

However, there is some thinking in government among a special warning steering committee to require broadcast stations to transmit weather warnings, either live or verbatim, and this whole matter has gotten further interest due to the hurricane or tornado that went through Xenia, Ohio recently, Congressman Brown initiated a study of the entire warning picture by his staff and by GAO. This matter is under consideration as to what degree the broadcast industry or the broadcast stations themselves should participate. The present thinking is whether the broadcasters like it or not, the public, in the event of an emergency situation, turns their radio or TV on for the full information.

The next consideration is to what extent are the broadcasters participating. The studies that we have conducted during recent emergency situations indicated that the broadcasters were participating in an excellent manner. For example, we conducted a study of the recent severe snowstorm in the Midwest area and found that the broadcast stations there participated in an excellent manner and were transmitting emergency information right up until the emergency situation no longer existed.

But I can't answer that question on whether or not the broadcast stations should be required to broadcast the weather warnings live or on a voluntary basis.

Q —I was interested in unattended transmitters.

Mr. Johnson:

The Commission actually adopted a Notice of Inquiry, on automatic transmissions last Tuesday. I would have bet anybody in the room that we would have had copies available by the time the convention started. That was one of the things we were trying desperately to do. To get action before the convention. And the Commission actually acted on this item one day ahead of the usual Wednesday meeting. So the Commission adopted this item last Tuesday, and copies are still not available. Needless to say this would have been a beautiful opportunity to have discussed in more detail what we're actually trying to do.

First of all, the item does not speak in terms of an automatic transmitter. It speaks in terms of an automatic transmission system. The way we see it conceptually is that the automatic transmission system consists of the transmitter, the antenna system, the monitoring devices, control circuitry, all of these interconnected units so that the power, frequency, modulation level, and antenna performance can be operated in an automatic manner and controlled to the point that if these various parameters go beyond certain levels, that the transmitter will actually shut down.

We are convinced that this is going to be a real developmental type effort in order to achieve a truely automatic transmission system. We're convinced that, with our present technology in broadcasting equipment and the ingeniousness of our manufacturers, that it is very possible to go into an automatic transmission system, to have the Commission recognize it and to provide ways that this system can actually operate.

We were convinced that we didn't have enough information to actually come out with proposed rules recognizing this type of a system. Our first step is a Notice of Inquiry. In this Notice of Inquiry we've indicated the problems that we have in trying to achieve this type of a system. We have posed very specific questions and problems upon which we want the expertise that's represented in this room and any expertise that we can possibly get our hands on to help us devise the system so that our Rules and Regulations will actually permit this system in a very reasonable way. We actually see this system in the future.

Let me just tell you some of the problems that we have. We've asked specific questions, for instance, about directional antennas. As you know, directional antennas are going to pose some pretty complex problems for us, which are compounded when stations change directional antennas day and night. Also just changing from non-directional to directional.

We're asking questions about power. We're trying to determine how can we actually monitor the power levels in AM, FM and TV. What should we do about simple things like the input resistance to a directional antenna system? How are we going to handle that as far as an automatic system is concerned? How are we going to provide the gear to actually cut the transmitter off within specific tolerances? What are we going to do in emergency situations when our present rules say that a station can operate with reduced power? We have to take cognizance of the fact that stations can operate in emergencies with some other type of operation. What are we going to do about modulation levels? Is automatic monitoring control and shutdown feasible in a case of video modulation? What do we do about little things like objectionable loudness? What changes can we make to compensate for modulation problems which result in objectionable loudness? We're concerned about frequency stability. How are we going to handle frequency stability when we're comparing transmitter oscillate frequencies against external frequency sources? We no longer have frequency monitors. What is a reasonable recognition in an automatic system of some calibration by some outside reference source? What do we do on an automatic system when we're in an EBS operation when emergency situations develop? How do we handle that? Little things like tower lighting safety and security precautions. The security precautions will, we think, be an added problem that we're going to have to consider. Are we going to permit some manual override capability, and under what circumstances will we permit a manual override? Our feeling is that we're going to have a black-box type of an operation that is not going to be touched by human hands and will automatically cut off within certain limits. If we do that, do we need operators? What's the role of the operator in the future? The Communications Act says that we must have operators at broadcast stations. Is it possible we can eliminate operators or change the type of operators that we need? Should we go to Congress and get a change in the Communication Act? What kind of records are we going to keep? Is it necessary to keep records? Is it necessary to keep logs? What's going to happen when our inspector comes out to the station? What's he going to inspect? How's he going to know that the station is operating properly? What are the provisions for assuring that the station is operating within the various levels that we have?

These are the type of questions. We go into great detail in the Notice itself, expressing our concerns in these areas and then asking for your assistance. But we think this is one of the top priority items as far as the Commission is concerned, so we're going to do everything possible to try to resolve this whole issue as soon as possible. And I'm assuming by the next NAB convention we'll be quite a ways down the road recognizing and identifying the actual details and design features of this type of a transmission system.

You may have some other questions. I think it's tragic that we didn't have the Notice available. But you all know that sometimes, no matter what you do, the secretary is the one that blows it. In this case it was our whole system, trying to get reproduction of this document. So you're going to have to read more about it as time goes on. You can come down to our booth and ask questions about it. We can show you at least the copy that we have which we used at the Commission.

We think this is a quantum jump, if you can imagine the Commission considering a quantum jump in its regulation, but we think it is a recognition that technology is now at the point where we can do something completely different.

Q—My question's not relevant to automatic transmitters. Are field measurements going to be accepted in the future, in relation to AM allocation problems in particular communities?

Mr. Williams:

Not to my knowledge.

Q—I have an application for a station in a community where another town which is about 50 miles away delivers a 3.16 into the community; therefore it qualifies for two services. In reality, the transmitter is about 35 or 40 miles away and there's a big mountain about 20 miles away from that transmitter. So this community I'm wanting to put another station in does not get any service at all from the station. You can't even hear it. But theoretically, it provides 3.16 nv.

Mr. Johnson:

The question relates, if I can just abbreviate it, to the problem of making measurements as far as FM is concerned, and you were relating it to measurements for FM to determine problems as far as your AM allocation is concerned in identifying where the service actually occurs.

We have Docket #16004, that involves adopting new curves for both VHF and UHF television, including FM. It also has a procedure in for making measurements on both FM and television. That docket is practically ready for adoption and I would imagine it'll be adopted within the next couple of weeks. Then we will require that the new curves be used in your predictions. It'll also recognize a method for making measurements. There's also a terrain roughness factor that is involved in it. We think as a result of adoption of this docket we'll be able to predict much more clearly and more accurately what actually happens in the real world as far as the various signal levels are concerned.

I would imagine in a couple of weeks you will be able to see what we've come out with and make your predictions with the new curves and possibly make measurements in accordance with the procedure in that Rule Making.

Q—For a station that might be looking to buy a transmitter, here at NAB, is this Notice of Inquiry going to obsolete any device that we might buy in the near future?

Mr. Johnson:

Well, I think you're being very optimistic that there'll be a transmitter available. Maybe I shouldn't be that pessimistic, but this is a pretty complex proceeding and there are a lot of unresolved questions. I would not advise you not to buy the transmitter. I think the automatic transmission system is somewhere down the pike, and I would hate to try to predict now whether you can actually come to the NAB convention and see one in two years or actually buy one in two years. Conversely all these problems may be resolved and you may be able to. But even at the time that the automatic transmission system is available, the station, as we see it will still have a choice either to continue the manual type of operation that we have now or go into the automatic system. And you're probably going to have to make some decisions that will be both economic and contain other aspects as to whether you want to go to the automatic system or whether you want to continue the present type of transmitter.

Q—On February 25th or 28th, a special PSA authorization was given stations operating on Canadian clears under an agreement with Canada during the daylight saving time period. Is any study or anything being done to see what actual ill effects this is having with Canada and possibly trying to make this situation permanent on the Canadian clears?

Mr. Johnson:

No, there's not been any evaluation. This requires agreement from Canada. It's part of our NARBA agreement. Actually it was a very difficult thing to try to get Canada to go along with us, even with this reduced power, and they did it because of the emergency situation we had as far as power was concerned and trying to reduce the consumption of gasoline. And they tied their agreement, as far as operating pre-sunrise on their clears to the Congressional Act. They were willing to accommodate us as long as Congress had the daylight saving time law in effect. And as you know, Congress has nothing before it now to extend daylight saving time beyond the last Sunday in October. So come the last Sunday in October 1975 the agreement with Canada dies. It's tied directly to the emergency situation as recognized by Congress.

Q —What I was trying to say was would an interference study, at this time, while our stations are operating, be beneficial in negotiating a permanent PSA with Canada.

Mr. Johnson:

I doubt seriously whether Canada would admit they've had no interference problems. They only have six clear channels and they're trying to use those to cover the vast areas that they have which have very little population. It was a very difficult thing for us even to get them to consider the pre-sunrise authorization we have. In order to evaluate this, I'm sure it would have to be on a theoretical basis. And on a theoretical basis, there is considerable interference being caused to Canadian 1A operation, even with the low power that we have in our country.

I don't want to give you any real hopes that this is something that's going to continue. I look at this as an accommodation by Canada because of the power problem that we were having.

Q —I'd like to delve for just a minute into the broadcast auxiliary services. About a year ago we signed a contract to replace our existing television intercity microwave system with new equipment. This entailed a change of manufacturer only—new equipment, same path, same dishes, same buildings, same power levels. It took about nine months to get the rubber-stamp approval through the Commission. And the last of the licenses, as a matter of fact, just dribbled in about a matter of six weeks ago.

I'd like to know what steps the Commission is taking to hopefully speed up such simple changes as far as authorization is concerned, and also what steps are being taken, if any, to expand or make available much more rapidly microwave facilities for electronic journalism. I'd hate to think with the rapid advances that are being made in ENG at this time that we have to wait a year or two years, three years for permission to use a microwave to get it on the air.

Mr. Johnson:

In government you have a very peculiar thing. I know you think that we've got hundreds of people working on these various things, but we can't hire people and train them to replace somebody that's leaving. And we have lost some very key people. In this particular area you're talking about, we lost the two key people that were operating that particular area. So what we've had to do is to move new people into it. We have actually three engineers that are working in this area now, and there is a little period where these things are going to fall behind. I guess that's an understatement. But we're very optimistic that we will catch up very soon on those type of applications. We've actually detailed people in from other areas to help us in processing these types of applications.

This is one of the difficult areas as far as backlog is concerned. We're very cognizant about it. And the only thing we can do is put additional people in. And up until the time they get experience it is going to be slow.

Mr. Pointer:

It's quite obvious by the attendance this morning at the nine o'clock workshop that electronic journalism is one of the paramount questions and areas that the broadcaster is concerned and is moving into very rapidly. So we've got to work together here.

Q —I just had a further comment on that same subject. It's possible now to replace an entire AM transmitter, as long as it's Type Accepted, by just notifying the Commission. And yet to replace a one-watt microwave requires a full-blown application. Is there any work being done to get some of the paperwork out of the way so your inexperienced people won't have so much of it?

Mr. Johnson:

You make it sound very simple. That's an area that we have to look at. We came out with new rules for this whole area. And I think we have to take a look at it now because of the problem we have to see if we can't speed it up.

One of the things that we're working on in the Commission is use of the computer. And if we can get this onto the computer, and I think we're quite close to doing this, then this is going to help resolve our problems. And to us, at least, it looks like the computer is the answer to these high-volume areas than actual manual processing. I think the next thing that you're going to be seeing is new forms which will lend themselves to computer operation and then having these applications actually processed in the computer.

Q —On the subject of AM sampling loops and lines, do you think the Commission will take action on this particular docket this year?

Mr. McNaughten:

Yes. We're very near to a solution on that. I should imagine within a month or six weeks we should be out with a Report and Order.

Q —On Docket #20265, about how soon should we get a report out on it?

Mr. McNaughten:

In Docket #20265 what we did was try to pull back a little bit on some of the restrictions on improving existing AM facilities. What we did was pull in a couple of the restrictions, hopefully that it would enable Class II and Class III stations to come up to their full power; that is, under our present provisions.

Where we got into trouble on it was that we did not include Class IV stations. Well, by and large, Class IV stations are already up to their limits in power, both day and night, and it was not for this purpose that the rule was being changed. In fact, it was nothing but a carryover in the old rule, which has been in there for years, which had said for exception of Class IV stations. And yet we got literally hundreds and hundreds of congressional letters as to the fact that we were discriminating against Class IV stations.

Well, really there was no discrimination against Class IV stations at all because we were just continuing a rule that had already been there.

Mr. Williams:

There's no change at all proposed in the technical requirements, or interference requirements for power increases for Class II and III stations. Right now there's a legal selection process on who can get power increases, based on the existing coverage of the city. Now it's two stations have coverage now, then you're ineligible for the power increase.

This is what the docket is aimed at. Whether or not the existing restriction should be continued. Not the technical interference standards at all.

Q —Do you see an easing on official Commission enforcement of pulse width standards until the networks and program suppliers and such can be brought into compliance?

A [Inaudible]

Q —Is there any chance that daytime stations would receive fulltime operation in this pending docket?

Mr. Johnson:

I wish you would have been in the small market panel that we had across the hall. I apoligize for coming late, but this was a panel that the chairman was the master of ceremonies and I couldn't leave. But one of the last questions that came up on the floor in there was by a daytime only station, and his suggestion was that we come up with a freeze on all changes by existing stations or grant of any new stations for a period of five years, and then the Commission use all of its personnel and all the expertise in the industry to completely revamp the AM standards and allocation plans. The end result would be to let all daytimers operate unlimited time, reduce the power of existing unlimited-timers to accommodate them. And I almost fell off the platform because I figured with a five-year period to hold up any action, by the time we ever finished that bucket of worms, we may have a different system than broadcasting completely.

But the other thing he said was, "What can we do," which intrigued me, "to eliminate daytime operations?" And those of us who have been at this game for all these years—I think daytime-only stations have probably caused more problems and work for us in trying to figure out how can they possibly get uniform hours, from six AM on ? How can they go to six PM. What can we do as far as the agreements we have with other countries? What do we do as far as interference problems are concerned? What do we do as far as coverage is concerned?

What I would suggest is that if you have a scheme, maybe you could meet this man that was across the hall. Maybe the two of you could come in with some sort of a proposal we could consider in Rule Making. But unless we make some very basic changes in our allocation standards and go into negotiations with other countries, I think we've gone about as far as we can go as far as extending the time of operation of daytime stations.

In this Rule Making the proposal was to see if we could permit existing stations to increase power. That was the basic change we were looking at. But in that proceeding we also asked questions regarding daytime stations, as to the possibility of daytime stations in certain circumstances going unlimited time. That is one of the questions that is in this Rule Making.

Harold Kassens:

Mr. Horne, do you have your inspectors in the field have a quota system?

Mr. Horne: No.

Mr. Kassens:

Would you be willing to put that in writing to all the field offices?

Mr. Horne:

They have basic instructions that do not include a quota system.

Mr. Kassens:

A lot of the industry seems to have a different impression. Instead of coming in with a violation notice with maybe one or two items, it just seems to trail off into item after item. And people get the impression that there really is a quota system.

Mr. Horne:

Well, that may be developed from certain personalities, but I think you have to realize with 300 inspectors out in the field, that they're not all identical and operate differently. In some cases it may be an implication that certain ones may act that way. But there is no policy as such. That would purely be an individual choice by an office or an inspector, and it is not coming from my office.

Mr. Kassens:

Is there any way you could set up a procedure so that we could discuss violation notices with the people in your office in Washington? As the system now works, the inspector issues a violation notice; we reply; he then judges that reply and sends it in to you; and we never hear about it again until we get a notice of violation. If there were some way that we could have an informal hearing with the authorities in Washington, rather than trying to argue with an inspector out in the field, it would help. Is there some way we could set up a procedure like that?

Mr. Horne:

Oh, I'm sure a procedure could be set up on that. However, when they are sent in, then we refer them to the Broadcast Bureau, being this is broadcast.

Mr. Kassens:

But we never find out about them and we never get to explain to somebody who knows what's going on.

Mr. Horne:

Well, I'm sure we can work that out. We can get together on that one.

Mr. Kassens:

I have been amazed in looking at licenses of directional antenna stations, some of the amazing places that the Commission puts monitoring point radials. What do we have to do to have a monitoring point radial changed or eliminated?

Mr. Williams:

Monitoring points, of course, are assigned as part of the application. When a station comes in, they submit maximum expected radiation values in all directions, and the Commission picks the directions that we think would control the pattern. If those monitoring point directions are within the maximum expected operating values, then we assume that the pattern's in adjustment.

Now, if for some reason these monitoring points are not in the dead hulls, like we do it today, but because they were assigned 15-20 years ago and they were assigned at that time on a protection basis to a certain station, and provided they were not put on the construction permit as the results of a hearing process, we can change the actual radial with a modification of a construction permit. And it doesn't entail a full engineering study, like a new pattern one, a simple modification of the construction permit.

Mr. Kassens:

What form do we file?

Mr. Williams:

FCC Form #301.

Mr. Kassens:

And what do we supply with the 301?

Mr. Williams:

Just a statement from the consulting engineer of the reasoning behind it and what radials you intend to measure in the future and which ones you intend to substitute for what and what the reason is.

Mr. Kassens:

The NAB has filed a petition with the Commission to change the life of the provisional third class operator permit with endorsement from one year to three years. Will you advise the Field Bureau how they should handle that petition? And under the Freedom of Information Act, what are you going to tell them to do with it?

Mr. McNaughten:

I understand the 30-day comment period was up last Friday on this. What we are going to instruct—we haven't instructed them yet, Harold, so actually there's nothing that could fall under the Freedom of Information Act. There's been no exchange of comments, except verbal.

Actually, I think Field Engineering Bureau will handle it. It will be in cooperation with the Broadcast Bureau, and it will be a joint item to the Commission.

Now, which way it's going, I don't know, because I haven't even read the petition yet.

Mr. Kassens:

I understand that the Commission just this past week accepted an application of a limited-time station in Los Angeles operating on a clear channel for unlimited-time operation. Does this mean that the Commission has now broken down the clear channels?

Mr. Johnson:

About the Freedom of Information Act, which is very interesting for us, because those of you familiar with it, there's a sunshine-in-government provision that is being considered on the Hill. The end result of that will be that all Commission meetings will be open to the public, if this goes through. They're already asking the Commission to experiment in making all meetings open, so then you can all come and you can follow through all the actions on your particular proposals. Who said what and what happened.

I don't think we have any secrets now, that all of our memos and so forth are open for public inspection. We have the United Church of Christ VHF drop-in item, and those of you who are familiar with that, you notice all of the various internal memorandums and critiques of the OTP plan, which we put into the docket. Pretty soon there'll be no secrets left.

But now that I've said all that, do you still want an answer to your question?

Mr. Kassens:

I'm wondering what the significance of that action was.

Mr. Johnson:

Well, the significance of that action was that it was a very specific case. It has very specific elements in it. The action that we took was only to accept the application. It did not grant the application. It involved KGBS actually supporting Roswell, New Mexico, a Class IIA station, to increase their power to 50 kilowatts. So the Commission weighed the gains of that possible action on KGBS and decided that it would be in the public interest in this specific case under these specific circumstances to at least process that application. So what the Commission has done is to accept that application.

But I think there is something inherent in that item. There again, this was acted on last Tuesday. We should have had copies available for you. But there's a lot of language in there trying to identify all of the elements in that case. It's about as purple a cow as we could possibly make it. It's a very unusual type of case.

So, on that basis, I think it would be very difficult for others to come in using that case as a precedent. We also in that order indicate that it may be necessary for us to actually open up the two remaining problems as far as our clear channels are concerned.

So, I wouldn't say that we're not going to open up the clear channels, but I don't think it's going to be on the basis of the KGBS operation.

Q—The point was brought up that sometimes stations that were assigned many years ago have monitored radials on perhaps the site of a minor lobe, which as any consulting engineer knows increase very rapidly with small changes in parameters, and in some cases they'd like to move this into a null situation. But now, these very same stations perhaps have had a very difficult time in years past proving their pattern, and over the years have gotten extended monitor point limits.

Now, if we wish to move this monitored radial into a null that is not now currently monitored what type of exhibit would be required? And would we then be starting from ground zero as far as the current 10% rule is concerned?

Mr. Williams:

What's the 10% rule?

Q—Well, we get 10% increase over the monitoring point limit.

Mr. Williams:

Well, that's not a rule. It depends on whether or not you have room. It depends on how you've proved the pattern in, of course.

There are two types of monitoring point changes under discussion. One is that you can change a specific point on an existing radial. Now, we can do that.

Q—No, we're talking about moving the bearing into a null.

Mr. Williams:

But when you actually move the bearing, we'll have to have a proof-of-performance at the same time, because it's never been measured before. We have no idea what the history of it is or what the inverse distance field. It hasn't been established on that radial.

So, if you're planning to do a proof-of-performance and you want to change the monitoring points, well, that would be the time to get it specified, before you did the proof, and we could pick it up.

Q—Possibly the question is: Could we perhaps get away with measuring that one radial completely and do a partial on the others, on the other bearings?

Mr. Williams:

I don't know.

Q—Especially if a complete proof-of-performance has been done fairly recently.

Mr. Williams:

I don't know. We'd have to look at that on your specific case. Ther's no general rule one way or the other on that.

Q—Based on the proposed curves which are coming out for FM and TV and the effect that this will undoubtedly have on UHF television, is the Commission proposing to perhaps change the basis of carriage for cable-TV systems to some sort of a signal-level rule rather than a mileage rule?

Mr. Johnson:

That's the item that's holding up adoption of that 16004—the cable involvement. What we're trying to do is to come out with something on a mileage basis as far as the carriage rules are concerned.

Q—Would it be possible to require some sort of field-strength measurements and say any signals over a certain level must be carried by the system? Is this viable?

Mr. Johnson:

The trend in cable right now is to try to come out with mileage instead of field strengths or Grade A or Grade B contour, and that's the thing that we're trying to do also in relation to 16004.

Q—Of course there would be a wide variation in the actual strength of the station itself, depending upon terrain and the power. It could be up to five megawatts for some UHFs and down arbitrarily low. But how would this dovetail in with the proposal to measure actual radials with field-strength meters? In other words, if there were measurements on file showing that the signal in the direction of a cable-TV city is very, very low, could this then be cited by the cable operator as evidence that he need not carry this station, regardless of the mileage rule itself? Or the inverse?

Mr. Johnson:

Measurements in television particularly, is awfully difficult to repeat. One of the things we're trying to do in our measuring recognition in that Rule Making is at least to come out with a procedure recognizing the type of measurements we'll accept.

But as far as cable is concerned, we're trying awfully hard to avoid the use of measurements and trying to come up with some distance figures which we think will be reasonable as far as the problems between cable and television. So we're trying to avoid using measurements as far as cable problems are concerned.

Q—One quick question with regard to the proposed curves. As I understand it, these would not primarily affect 3.16 contours, like things we have in the works right now that come close or are marginal. We don't want to file these and then have them come bouncing back when the new curves come out.

Mr. Johnson:

When the order comes out on 16004, the changeover from the contours based on the present method against the contours based on the new method is one of the items we're considering, and the chances are you will not be required to make any change until your next renewal. One of the points is that when the next renewal comes up, we'll probably require contours based on the new curves. Or if you run into a problem or you're challenged in some way, where the location of the contour becomes critical, then you'll probably be required to file under the new curves.

But I doubt very seriously that we'll come out with an order establishing the new curves and then require all stations to, in 30, 60, 90 days to file new contours. It'll probably be when the renewal is filed.

Q—When an applicant comes in and gets a construction permit for a standard broadcast station based on certain maximum expected operating values or a standard radiation pattern that antenna system is then adjusted with some particular values in the null directions where there are monitoring restrictions.

My question is this: What is the justification for setting forth some arbitrary percentage, in terms of 5 or 10 percent above the measured inverse field, and ignoring completely the actual design maximum specified value as a criterion?

Mr. Horne:
Well, the maximum expected value indicated on the construction permit, I would say just roughly now, 50% of the time is used. It's when the pattern has proved in to only one-half of that value, then you get into the situation where the monitoring point value could vary over 200, sometimes 300, percent. And in those cases, we just have to start somewhere, so we arbitrarily start off with a value of around 10%.

Q—Isn't that quite unrealistic, though, for a deep null, when the design engineer has particularly taken that variation possibility into account?

Mr. Horne:
Well, after the value is assigned, there's generally a letter sent out to the licensee or permitee requesting subsequent data covering a month's period—usually it's weekly. And at that time, if he needs a further relaxation, he can request it at that time.

Q—Unfortunately, it doesn't take into account the seasonal variations which actually do occur. That one-month period.

Mr. Horne:
Well, you can submit a further extension of that, if you like, over a longer period of time.

Q—Well, it just makes a lot of work for everybody when it's really unnecessary.

Mr. Pointer:
We are going to have to bring the questioning to a close. We're beyond our time due to runover of the earlier papers. I would like to thank the members of the FCC Panel and you the audience for your kind participation.

E. M. Tingley

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A Review of the Quadraphonic (Four-Channel) FM Field Tests

Introduction

As a representative of the organization sponsoring the NQRC, I appreciate this opportunity to present a progress report to you who will ultimately be responsible for bringing quadraphonic FM broadcasts to the public. Four-channel needs broadcasting to be viable.

The NQRC was formed three years ago to report to the FCC the Committee's evaluation and final technical conclusions regarding four-channel FM broadcasting transmission. Participation has been non-exclusive and democratic in keeping with our tradition for this type of effort. The report is based on studies that:

1. Determine the basic channel requirements from the sound reproduction standpoint.
2. Clarify the technical issues between proposed systems that meet these requirements through appropriate field tests.
3. Determine FM signal specifications based on these field test data and the best scientific data available.

The report must show that this is practical and in the public interest. When the report is filed with the FCC in the near future, you will have the opportunity to study it and determine for yourselves if indeed this is the case. Specifics must by and large be confined to the report; however, I will endeavor to give you background on the proposed systems and some of the general findings.

Test Program

The three-part test program reflects these considerations. Subjective listening tests were conducted in 1973 to assess the listening characteristics for basic systems having two, three, or four transmission channels. Closed circuit tests of FM broadcast systems proposed by five companies were made last summer and the over-the-air transmissions that followed completed the series.

Before describing either proposed systems or specific tests, it is appropriate to examine the philosophy of the tests and the test signals which were designed to:

1. Yield meaningful measure of:
   (a) Compatibility with existing services. This includes the present FCC station assignment plan, the provision for Subsidiary Communications Authorization (SCA) and the ability of a listener to receive substantially all the program that is broadcast

regardless of whether he has a monophonic or stereophonic receiver. To fully evaluate compatibility of the proposed vs. the present service, two categories of FM stereo receivers were tested along with the proponents' receivers: a mass-produced type and a high-fidelity type. Reception in an auto was evaluated for multipath effects. 67kHz SCA receivers were also employed.

(b) Quality of quadraphonic service. This includes an assessment of audio frequence response, separation between audio outputs, distortion, signal/noise ratios, and interference from the SCA service.

(c) Susceptibility to transmission disturbances. This includes measurement of multipath distortion and susceptibility to impulse noise interference. The studio-transmitter link was also evaluated, as well as the transmitter-antenna bandwith.

2. Be fair to all proponents. This refers primarily to the test signals and the musical selections. The test signals were chosen to fully exercise the baseband transmission channels with statistically independent signals. Only sine wave test signals were used.
3. Be practical for field testing. Tests did not depend on special equipment or unfamiliar techniques. They were repeatable. They were not expensive in terms of time, money, or facilities.
4. Be clearly and completely defined.

Proposals

Seven systems, plus one two-channel receiver option, have been tested. The seven systems themselves have further options, involving the presence or absence of pilots, and/or SCA. The authorized two-channel pilot tone stereophonic system with SCA served as a reference, the proposed systems all being extensions of this system. The third transmission channel (second subchannel) is in quadrature with the stereophonic subchannel. They basically differ in the handling of the fourth transmission channel and the provision for SCA. The fourth channel is a double sideband, suppressed carrier, amplitude modulated subcarrier for three proposals. The fourth channel of two systems uses lower vestigial sideband transmission. SCA may be at 67kHz, 95 kHz, or omitted. The information on the third and fourth transmission channels completes the description of the quadraphonic sound field with:

1. The difference between the front and back signals.
2. The diagonal difference signals.

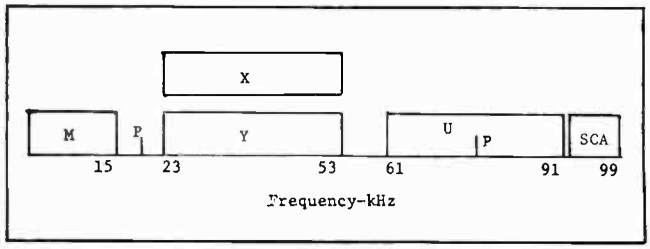


Figure 1. Quadracast System

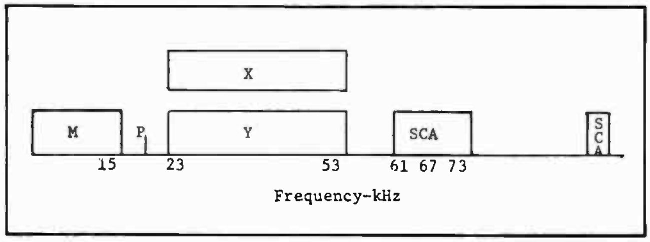


Figure 2. RCA 3-Channel System

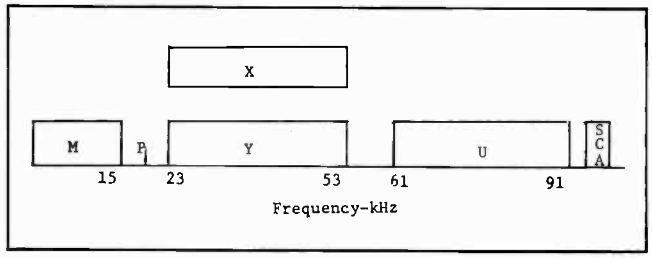


Figure 3. RCA 4-Channel System

The four baseband transmission channels are written as:

M = LF + LB + RF + RB	Monophonic
Y = (LF + LB) − (RF + RB)	Left — Right
X = (LF + RF) − (LB + RB)	Front — Back
U = (LF + RB) − (RF + LB)	Criss — Cross

The systems may now be briefly described[1] without getting into the relative advantages or disadvantages of the various proposals.

Quadracast Systems, Inc., (QSI). The system is described in the May 1971 filing with the FCC covering the laboratory and field tests. As indicated in Figure 1, the U channel, the third subcarrier located at 76kHz, is double-sideband suppressed carrier AM. A 76kHz pilot, in quadrature with this signal, may be added as an additional means of four-channel recognition. SCA at 95kHz is an option.

The RCA proposal, illustrated in Figure 2, provides for three transmission channels with no change in the current 67kHz SCA usage. For four-channel transmission, a double-sideband suppressed carrier (U

[1] For more complete descriptions, see *IEEE Transactions On Broadcast And Television Receivers,* November 1973, Volume BTR-19, Number 4.

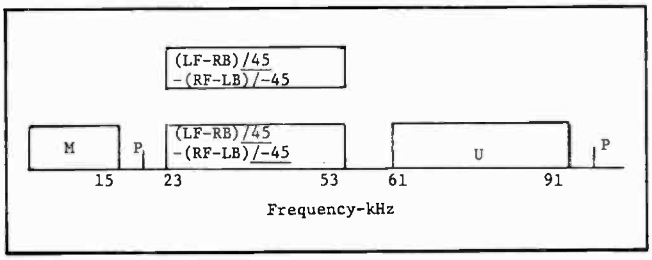


Figure 4. Nippon-Columbia 4-Channel

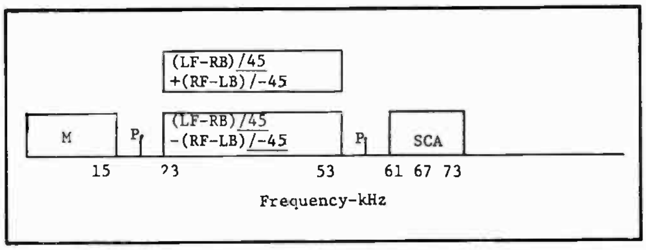


Figure 5. Nippon-Columbia 3-Channel

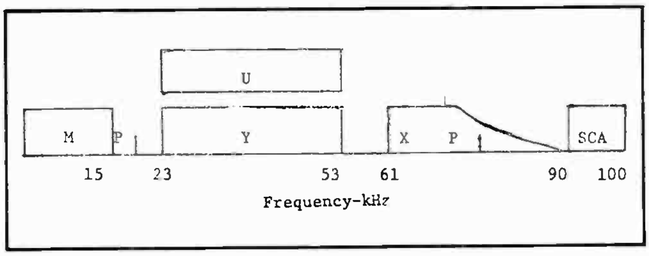


Figure 6. GE System

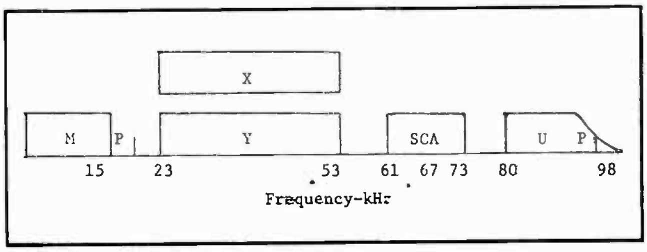


Figure 7. Zenith System

channel) AM subcarrier at 76kHz is added, as shown in Figure 3. A 76kHz pilot indicates the four-channel mode. A 95kHz narrow-band SCA suitable for telemetering may be used with either mode.

Nippon Columbia. The Nippon Columbia proposal is similar to that of both QSI and RCA; however, it differs in that the X channel (shifted 90 degrees) and Y channel are encoded with reference to 45 degrees. This allows reproduction of quadraphonic programs with a two-channel matrix decoder receiver. Figures 4 and 5 depict the four or three transmission channel systems. A pilot at 57kHz indicates three-channel transmission and a pilot at 95kHz indicates four-channel transmission. SCA at 67kHz may be used with three-channel transmission.

General Electric Company (GE). The GE system is described in a Field Test Report filed with the FCC covering radiated tests conducted in October of 1971. As indicated in Figure 6, the fourth transmission channel, a 76kHz suppressed subcarrier, vestigial sideband subchannel, carries the X information. A 76kHz pilot provides for switching the receiver to quadraphonic. During the NQRC tests the additional feature of variable compression was applied to the vestigial sideband (X) channel to maintain spectrum allocation. SCA operation at 95kHz is proposed.

Zenith Radio Corporation, (Zenith). The Zenith proposal places the fourth transmission channel at 95kHz, as shown in Figure 7, with provision for SCA at 67kHz. A 95 kHz pilot provides for receiver switching to quadraphonic. Optional 57kHz subcarriers are designed to provide a signal/noise ratio improvement by employing companding coupled with variable or dynamic preemphasis and deemphasis.

The control information is contained in double sideband, suppressed carrier, amplitude modulated, subcarriers in quadrature. Both the frequency independent and frequency dependent amplitude controls are applied equally to all quadraphonic audio inputs (LF, LB, RF, RB) at the transmitter and output channels at the receiver so that they track.

Subjective Listening Tests

An extensive series of listening tests were conducted at FM radio station K-101 in San Francisco and General Electric in Syracuse to compare sound reproduction of quadraphonic encode/decode systems having four, three, and two transmission channels.

The ability of almost 200 people to indicate the location of sound sources, both real and phantom, was tested under listening room environment using both simple and complex sounds. Localization is generally recognized as the most definitive characteristic of a quadraphonic reproduction system. That is, the precision with which a listener can locate an isolated sound source, or a sound source in concert with other sources.

The simple localization tests were conducted using bursts of pink noise. Complex localization tests, simulating musical material, were run using bursts of a chirp signal while at the same time applying pink noise background to all four speakers.

People's preferences with carefully selected music were also made. There was close correlation between the rankings for the two series of tests. That is, four transmission channels ranked first, two transmission channels ranked last. In no case was electronic enhancement of the systems employed.

The compatibility of a quadraphonic program when listened to on stereophonic or monophonic receivers was also considered. The proposed four, three and two transmission channel systems all have identical fold-down from quad to mono, with the monophonic signal in each case being simply the sum of the four quadraphonic signals. Consequently there was no need to compare system performance in this case. The fold-down for the Nippon Columbia two-channel stereo is different from the others, so a stereo compatibility test was conducted to compare the stereophonic sound reproduced using the same musical material.

A report on this work will appear in the January/February and March issues of the Journal of the Audio Engineering Society with sufficient data and detail to enable independent analysis of the results or duplication of the tests under the same conditions.

Closed Circuit Tests

Tests run closed circuit were those requiring freedom from interference. They were conducted using the facilities and some equipment of the College of San Mateo using FM exciters and station monitors. Receivers were isolated in a screen room.

1. Protection ratios. This term is used to designate the rejection of co-channel, first adjacent (±200kHz) channel, and second (±400kHz) channel interference. They serve as a measure of the proposed service to meet the present FM channel allocation requirements. When setting output level on the desired signal and for modulation of the interfering carrier, the carrier was fully modulated with a test signal consisting of one to four sine wave(s) that independently fully exercise the baseband transmission channel(s). Measurement of mono-stereo compatibility receivers for both monophonic and stereophonic transmission served as a reference for the existing receiver population.
   Test results indicate the proposed systems are compatible.
2. Signal to random noise. Receiver output was set using the above sine waves. Measurements were also made with a strong RF signal input in the presence of a fixed amount of random noise that served as a practical check on a theoretical analysis of the signal/noise ratios made of the proposed systems.
3. Intermodulation distortion (IM). The same test signal of from two to four sine waves, as above, was used to measure IM distortion by eliminating the fundamental tones at the receiver output.
4. Multipath distortion. This was assessed by measuring IM distortion, as in 3, by combining a second RF path, having about a 5us delay and reduced somewhat in amplitude, with the straight path. The phase of the delayed signal was adjusted to give maximum distortion. Recordings of musical material served as a subjective reference for the measured values.
5. Radiated spectrum. Pictures were taken of the spectrum analyzer display for both composite baseband and RF output under worst case conditions.

   The radiated spectrum for quadraphonic systems compared favorably with those of monophonic and stereo.

Over the Air Tests

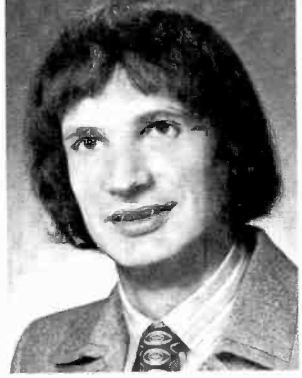
These tests are being conducted between San Francisco FM station K-101 and a receiving site some 30 miles south. The tests:

1. Amplitude, separation, and single tone distortion vs. frequency. These tests were run using a swept sine wave.
2. SCA interference. SCA interference into monophonic, stereophonic, or quadraphonic service is measured.
3. SCA service. Measurements were made to determine interference from the monophonic, stereophonic, or quadraphonic services together with assessment of distortion, signal/noise, and frequency response. Both 67kHz 95kHz were checked.
4. Subjective quality. Recordings of musical program material were made for later evaluation. Selections included those used for the Subjective Listening Tests. Compression was not employed, except as called for by the one system. The compatibility of monophonic and stereophonic reception in an automobile were checked by recording this material while the vehicle was driven over a prescribed course at a given speed.

Additional Studies

Information on the compatibility of matrix systems, now permissible under FCC Rules, will be included. Requirements for interconnecting facilities and broadcast transmitters are also being examined.

The five system proponents can be expected to make separate FCC filings in support of their own proposals, which will include information gathered through optional tests made during the closed-circuit or over-the-air series.

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A New Technique for Maintaining High FM Modulation Without Degrading Audio Quality

It is clear from practical experience that all current devices purporting to prevent FM overmodulation actually don't. One result of this is that it is impossible to set up FM modulation using tone; instead, it must be done with program material—usually one of the station's "difficult" records which inexplicably causes the new type-approved stereo monitor to go wild. Most broadcast engineers tend to somehow blame their limiters. Yet observing the output of any of these devices reveals peak levels very tightly controlled. Plainly, something is happening between the output of the limiter and the RF modulator to cause this unpredictable overmodulation.

A brief review of the peculiarities of the stereo FM system is in order. The FM stereo "pilot tone" system used in the U.S. transmits the sum of the two stereo channels (L+R) on the main channel, which modulates the transmitter from 50 – 15,000 Hz. It transmits the difference (L−R) on a double-sideband supressed-carrier AM subchannel centered at 38 kHz. A 19 kHz pilot tone at 9% modulation is utilized to provide a phase reference to regenerate the 38 kHz carrier for purposes of demodulation. In addition, both left and right channels are pre-emphasized at 75 microseconds for noise reduction.

The result is that the left and right channels must be lowpass filtered at 15 kHz. Otherwise, high-frequency components could interfere with the pilot, and the main channel could leak into the subchannel, and viceversa (a phenomonon known in communication theory as aliasing). The filters required are rather high-performance types, as they must be essentially flat to 15 kHz and down 50 – 60 dB at 19 kHz. In order to achieve this performance with reasonable economy, the sophisticated "Elliptic Function" filter is used in most transmitters.

Unfortunately, the elliptic function filter has an extremely non-linear time delay close to its cutoff frequency. The result is that it introduces phase shift and ringing. By changing the phase relationship between the various harmonics in a complex wave, the filter also changes the peak level, thus introducing the possibility of overmodulation. Further phase shift and ringing can be caused by the input transformer, and mismatches between the transmitter preemphasis network and the deemphasis network in the limiter.

Figure 1 shows a photograph of the response of a highly regarded modern stereo generator/exciter to a 1 kHz square wave. The ringing introduced by the lowpass filter is equivalent to instantaneous overmodulation of roughly 144%. Figure 2 shows the frequency response of the filter. It should be noted that it is extremely steep, with cutoff occurring at roughly 17 kHz.

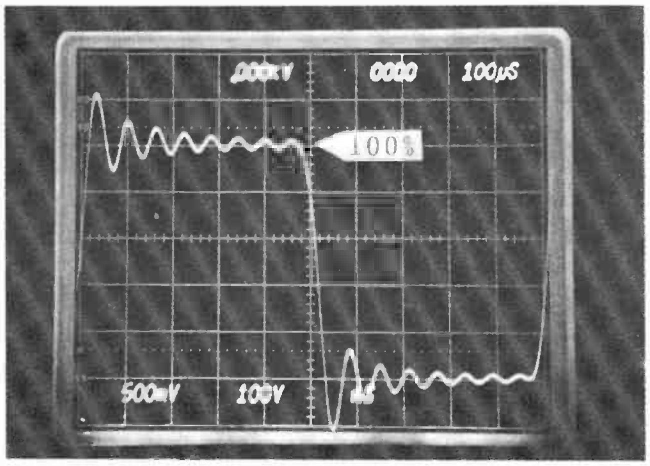

Figure 1.

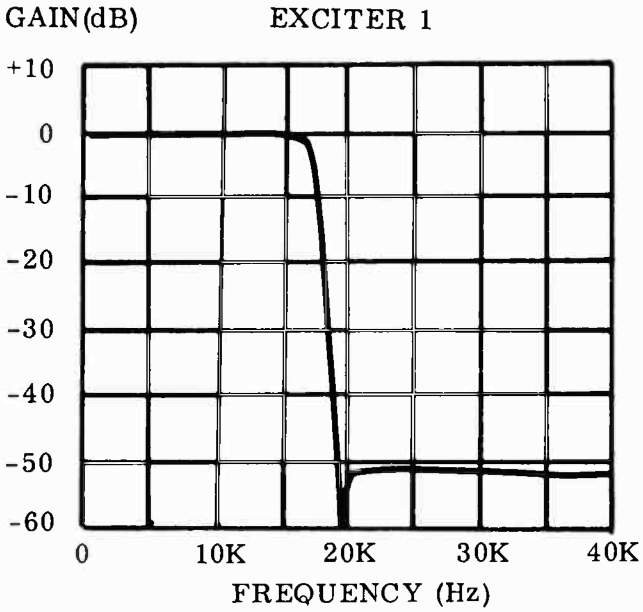

Figure 2.

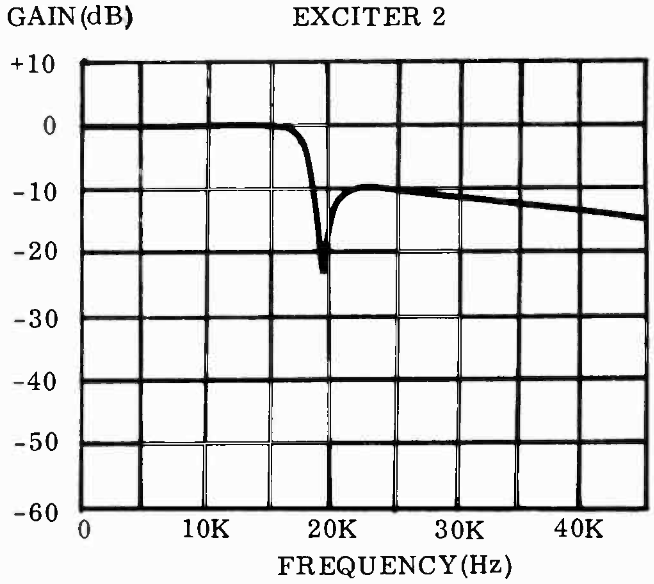

Figure 3.

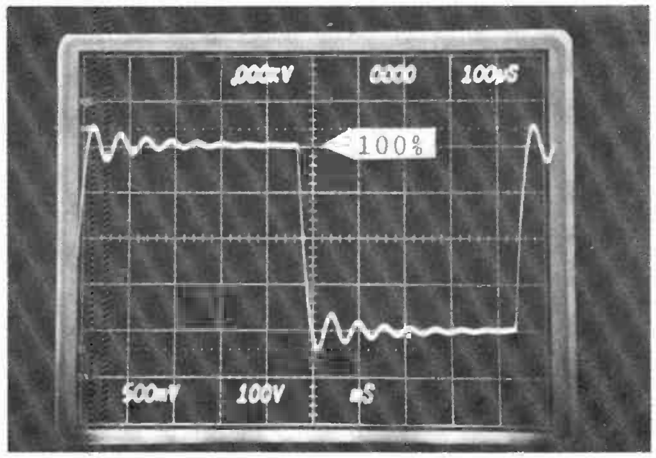

Figure 4.

Figure 3 shows the overall frequency response of another modern exciter. It has a much gentler frequency cutoff, and its ringing Figure 4 although much less, still corresponds to 122% modulation.

Another potential cause of overmodulation is much more subtle, but nevertheless significant. In order to be effective, all currently manufactured FM limiters must use some form of clipping or instantaneous limiting at the output. In some cases, this is used as the sole mechanism to control overmodulation due to preemphasis; in other cases, its purpose is to control overshoots due to the relatively slow attack time of the broadband and/or high-frequency limiting sections.

The result in either case is the same. For some period of time, the clipped waveform resembles a square wave. This means that the clipping has introduced high-frequency components which were not present in the original program material. Often, these components extend far above 15 kHz, and are therefore removed by the lowpass filter in the stereo generator.

Figures 5 and 6 show the respective responses of exciters 1 and 2 to 5.7 kHz square waves of the same amplitude as the 1 kHz square waves shown in Figures 2 and 4. The sharp cutoff of exciter 1's lowpass filter results in almost all of the harmonics being removed. However, the remaining harmonics plus the fundamental reach 132% modulation.

The gentler cutoff of exciter 2's lowpass filter permits substantial amounts of third harmonic to remain. In addition, the phase shift in the region of 15 kHz (where the third harmonic is) has changed its phase relationship to the fundamental and increased the peak level to 143% modulation.

When we apply Fourier analysis to a square wave, we find that the peak level of the fundamental is 2.1 dB above the peak level of the square wave as a whole. This means that even if we could filter out all of the

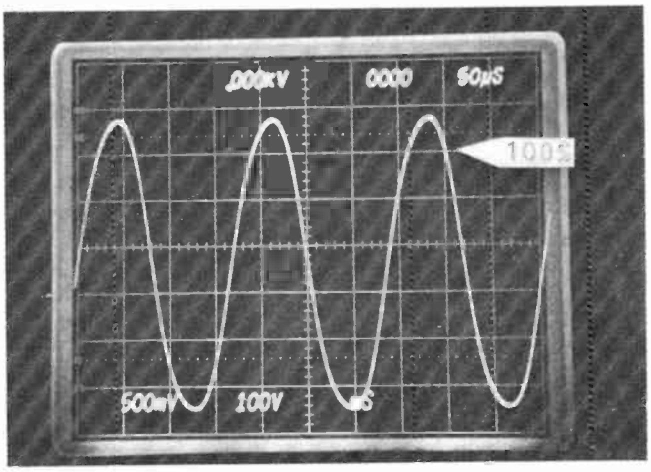

Figure 5.

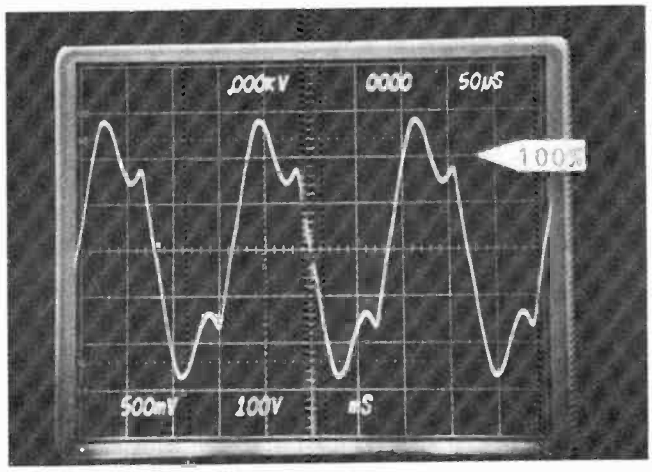

Figure 6.

harmonics introduced by clipping with an ideal filter having no phase shift or ringing, the level of the output of the filter would still exceed the level of the input by 2.1 dB—equivalent to 127% modulation. When we deal with a real filter, the situation becomes much worse because of the phase shift/ringing problems described above.

In view of all of this, it is a smaller wonder that one's peak light seems to flash at unreasonable levels and at unpredictable times!

Fortunately, there is a solution to this dilemma. It involves a radical departure from the old way of doing things. However, its justification is its dramatic effectiveness in bringing the peak overmodulation problem under control.

A solution is this: The broadband limiter, preemphasis, high-frequency limiter, instantaneous limiter, lowpass filter, and stereo generator may be engineered as a single system, in a single package, so that the interface between the various processing stages can be consistently defined, and sophisticated corrective techniques can be employed in a precisely controlled manner. This is what we have done.

Figure 7 shows a block diagram of the audio processing section of the system. The basic system logic can be seen to be similar to conventional systems. Following RF interference filtering and the input transformer, the signal is applied to a voltage-controlled amplifier, realized with a junction FET functioning as a voltage-controlled resistor. The output of the VCA is applied to a dual comparator. If the positive or negative peak value of the output attempts to rise above the level of the reference voltage, the comparator turns on and charges the integrator until the overload disappears.

The integrator determines the basic "sound" of the unit, and great effort has been expended in its development. It incorporates four distinct time constants, which have been carefully arranged so that "pumping" is totally absent. In essence, the unit establishes an average level, and permits approximately 2 dB of very fast limiting above that level, with

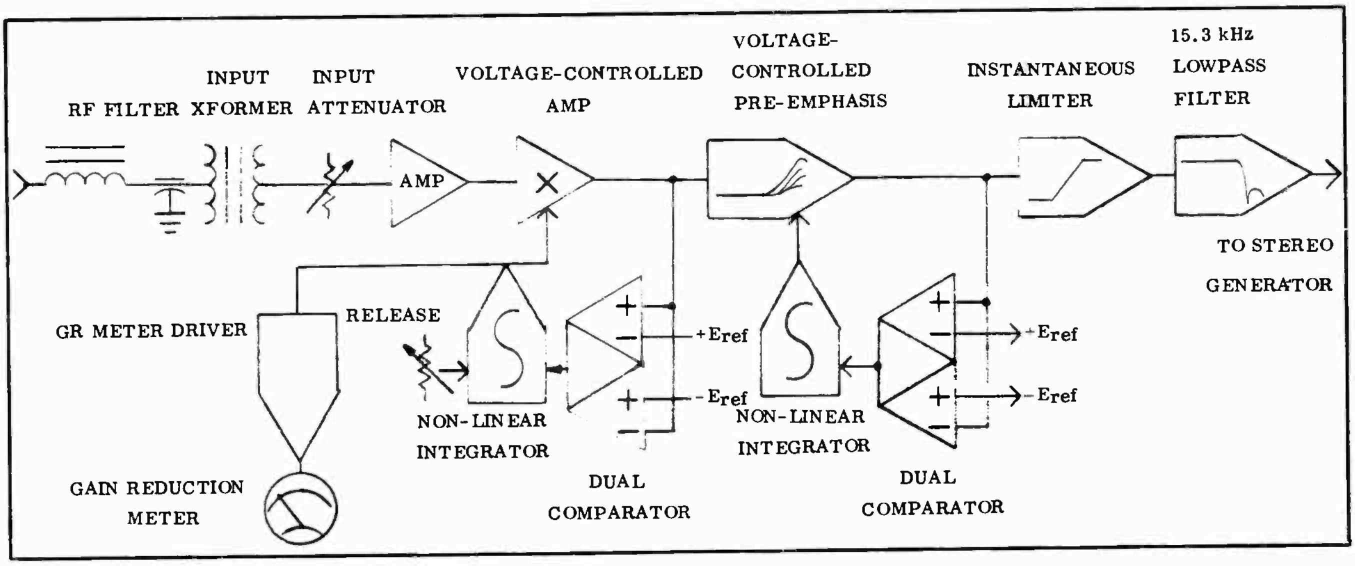

Figure 7.

a release time in the order of 70 ms. If the average level of the input changes, the integrator will release very slowly, on the order of 2 dB/second. A large, sudden increase in input level will cause the release time of the circuit to decrease for a short time, and then recover to its steady-state value. In addition, a time delay occurs between the onset of an input peak and the onset of release. In this way, low-frequency distortion is eliminated.

The range of gain reduction has been purposely limited to 15 dB. Extensive listening tests with all sorts of modern recordings, ranging from classical to hard rock, have revealed that the ideal amount of compression of low-level passages in the music is 8 – 10 dB. This increases the perceived signal-to-noise ratio at the receiver, but simultaneously does not damage musical values because of the nature of the release time constants. In particular, the medium-speed release time which causes pumping and listener fatigue has been totally eliminated.

Thus, if the average amount of compression is 8 – 10 dB, some 6 dB is left over for "headroom" for fast limiting, operator error, etc. Increasing the compression ratio to greater than 15 dB will result in severe audible distortion.

An additional advantage of using 8 – 10 dB compression is that gating is not required for musical programs. A station with a talk format would probably wish to use an "intelligent compressor" ahead of the limiter, and use the limiter for peak limiting only. However, for music formats, further compression will only degrade quality.

While the basic "sound" of a limiter is determined by the broadband section, it is largely the design of the high-frequency limiter that determines whether or not that sound will be preserved or whether it will be grossly degraded by the need to control potential overmodulation due to high-frequency preemphasis. Our high-frequency limiter uses a new principle, whereby the pre emphasis filter itself is voltage-controlled by the feedback signal from a comparator. This maximizes the accuracy of the operation, and also permits a tailoring of the filter curves under high-frequency reduction to match the typical spectral distribution of music, thus assuring that small overloads in one part of the spectrum will not force gross filtering of another part of the spectrum, with the attendent coloration familiar to all contemporary FM listeners.

The integrator controlling the variable preemphasis filter utilizes delayed release for low distortion, and releases in approximately 15 ms.

The high-frequency limiter is followed by an instantaneous limiter in order to control overshoots due to the moderate attack times of both the broadband and high-frequency limiters. The spurious harmonics produced by this process are dealt with in the lowpass filter, so they do not cause overshoots.

The typical harmonic distortion through the broadband and high-frequency limiters at any frequency and any degree of gain reduction does not exceed 0.2%. Deemphasized noise in the 50 – 15,000 Hz band is better than 80 dB below 100% modulation. Stereo coupling of the two limiters is achieved by using one comparator per limiter per channel, wire "or'ing" Corresponding comparators, and using the integrator output to drive matched FET's. In this way, gain reduction in both channels tracks whichever channel requires the most gain reduction at any instant.

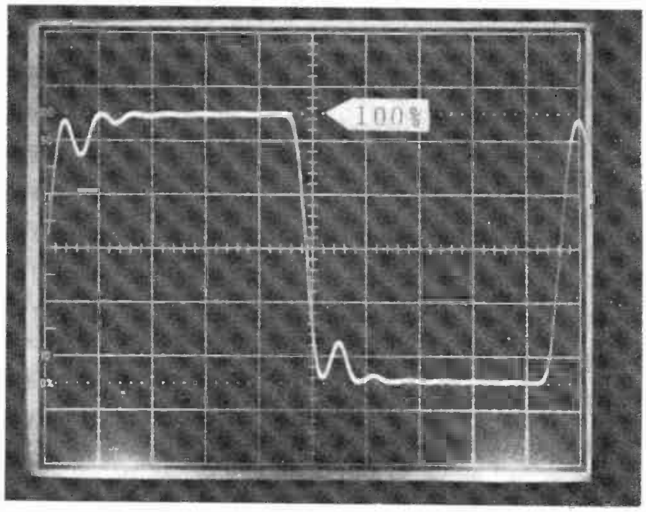

Figure 8.

So far, the system merely consists of refinements of current techniques. However, the lowpass filter deviates substantially from current design procedure in order to control the overshoot and ringing exhibited by typical exciter lowpass filters. Actual design details of the filter are extremely mathematical and beyond the scope of this paper. However, its performance can be described generally.

The filter is non-linear and non-minimum phase. Its frequency response is flat within +0, −0.6 dB from DC to 15.2 kHz. It is down approximately 60 dB at 19 kHz, and approximately 45 dB from 23 to 53 kHz, reaching 70 dB in the SCA region. It is realized with resistors, capacitors, and operational amplifiers, thus eliminating spurious responses caused by stray capacitance associated with inductors. Theoretical overshoot is 3% maximum. Harmonic distortion is under 0.1%.

Figure 8 shows the 1 kHz square wave response of this filter, which should be compared with figures 1 and 4. Note that the ringing is in the form of undershoot, and that overshoot is less than 103% of the steady-state value of the square wave.

Figure 9 shows the 5.7 kHz square wave response of the filter, and Figure 10 shows its frequency response. Note that the sharp cutoff at 15.3 kHz has removed essentially all of the harmonics from the wave, but the remaining fundamental reaches precisely 100% modulation.

The practical result of this filter, directly coupled to the stereo generator, is an almost complete absence of fast peak overmodulation. Therefore, the average modulation can be raised accordingly, and greater

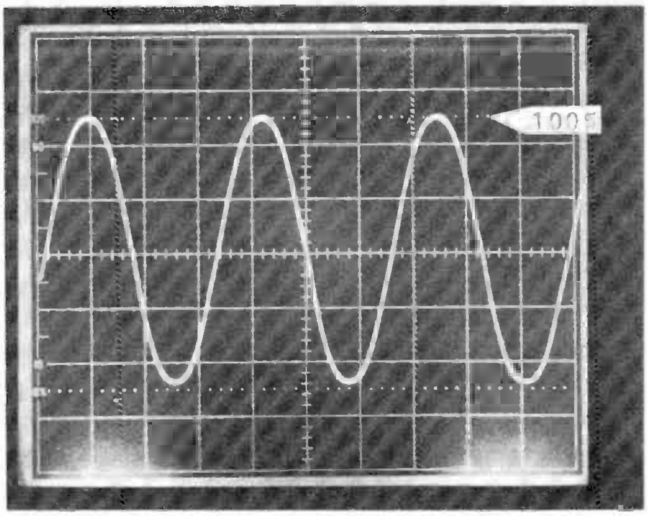

Figure 9.

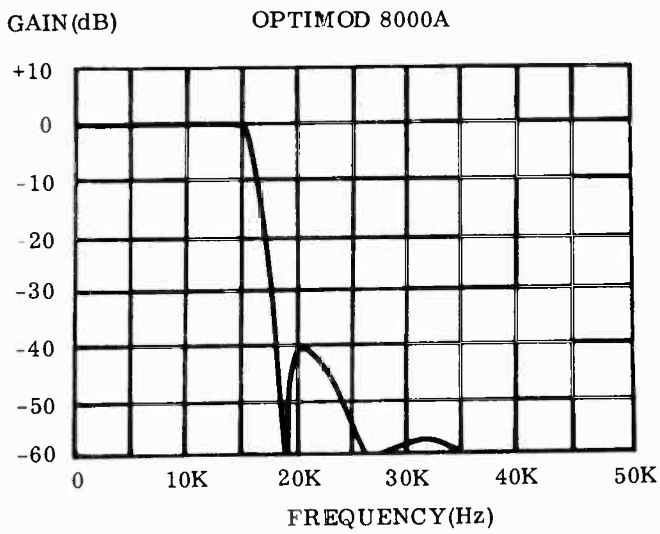

Figure 10.

loudness is obtained with no degradation in audio quality whatever.

The stereo generator contains a number of novel design features worthy of comment. Historically, stereo generators which used the "time division" or switching technique to generate the stereo baseband signal have been popular. In the days of vacuum tubes or discrete semiconductors, the simplicity of this technique was extremely appealing. However, the fly in the ointment was the necessity of filtering the baseband signal above 53 kHz to remove the spectra around the odd-order harmonics of the 38 kHz square-wave switching signal, such as 114, and 170 kHz. If permitted to modulate the transmitter, these signals would cause severe interference to adjacent channels, due to these harmonics' high frequency.

Unfortunately, the design requirements of the 53 kHz lowpass filter are very stringent because of the necessity for low phase distortion. Such filters are very expensive to manufacture, and have in addition caused mediocre high-frequency separation (40 dB or so).

The direct technique of generating the baseband signal, using a balanced modulator for the L−R component, is compromised by the relatively low performance of transformer/diode balanced modulators. In addition, these balanced modulators were sufficiently slow to require phase correction in the L+R channel to compensate for the intrinsic phase delay of the modulator.

Since the invention of the Gilbert principle of transconductance multiplication in 1968, the situation has reversed. The Gilbert multiplier permits construction of a fast, quiet, low-cost balanced modulator that requires no filtering of the baseband output, and no L+R phase correction. With certain circuit refinements, the Gilbert multiplier can be made outstandingly linear, yielding modulation linearities in the order of 0.01 − 0.02% harmonic distortion.

Figure 11 shows a block diagram of the stereo generator. In order to maintain the high performance capabilities of the basic modulator with time and temperature, the 19 kHz oscillator level, the 38 kHz level (and therefore the L−R gain) and the 19 kHz/38 kHz phase relationship are all stabilized by feedback loops, using the very precisely regulated positive power supply as a reference.

The 19 kHz crystal oscillator is stabilized by a feedback loop using a comparator and integrator. The output of the oscillator contains about 0.1% harmonic distortion, therefore requires no additional filtering.

The 19 kHz output is fed through a voltage-controlled phase shifter to a doubler, which produces a 38 kHz square wave. This square wave passes through a voltage-controlled amplifier, and is then lowpass filtered to approximately 0.04% distortion. The output of the lowpass filter feeds a comparator/integrator which controls the VCA so as to stabilize the 38 kHz level at the output of the filter. 38 kHz amplitude is adjusted by varying the reference voltage on the comparator. The filtered 38 kHz output is fed, along with the 19 kHz, into a phase detector. The output of the phase detector is integrated, and controls the voltage-controlled phase shifter, thus forming a phase-locked loop. Because of the integrator, steady-state phase error is theoretically zero.

The left and right outputs from the limiter lowpass filters are applied to a summing amplifier and differential amplifier to roduce L+R and L−R respectively. The L−R signal is multiplied by the 38 kHz lowpass filtered signal to form the double-sideband amplitude-modulated supressed-carrier subcarrier. The subcarrier is summed with the L+R and the pilot in the output amplifier, which passes through an attenuator to the RF modulator of the station's exciter.

The stereo generator is capable of 60 dB separation at midfrequencies, reducing to approximately 47 dB at 15 kHz. Sub-main and main-sub crosstalk due to

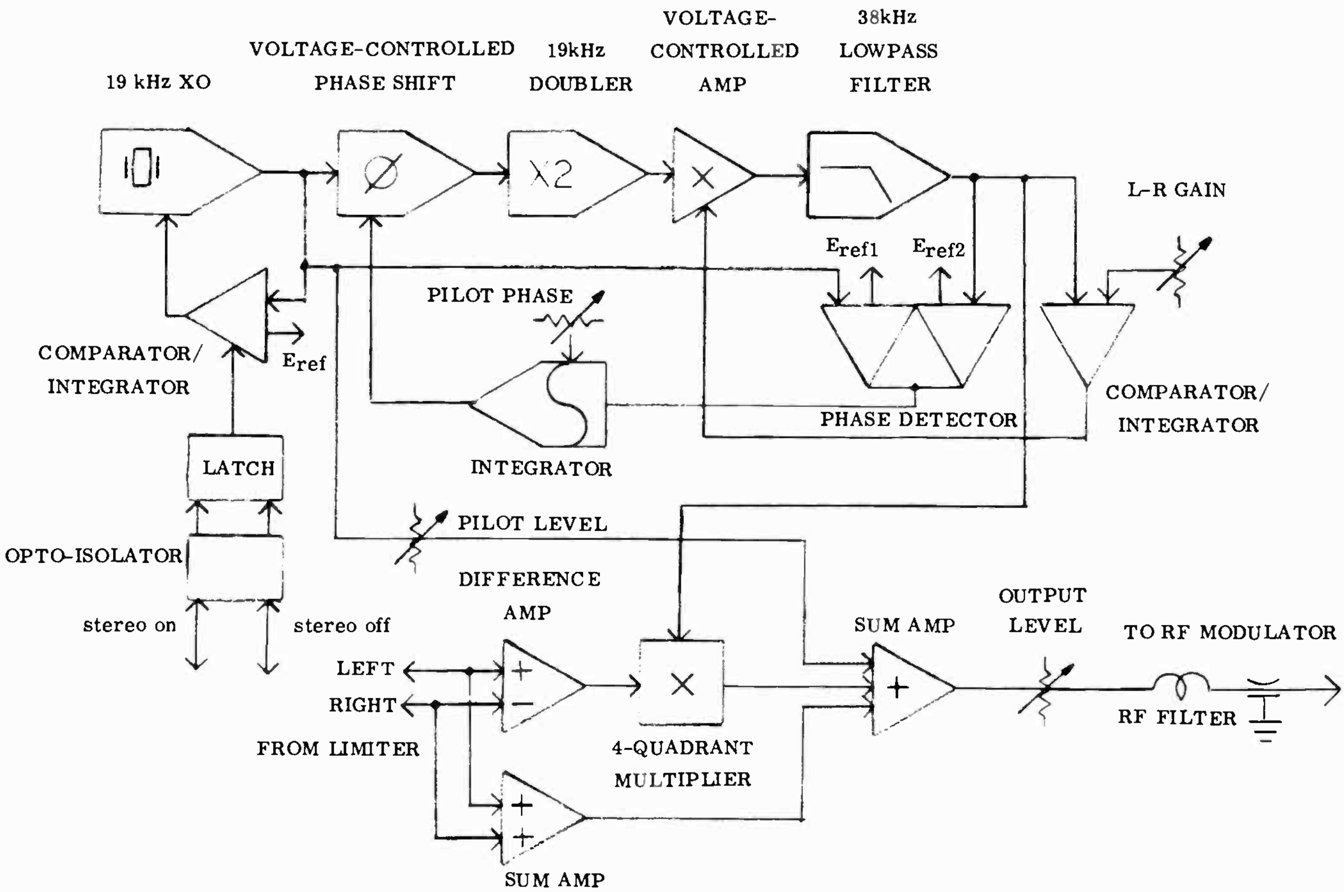

Figure 11.

modulator non-linearities are below −70 dB. 38 kHz suppression is typically better than −55 dB. 76 kHz suppression exceeds −70dB. The only major spurious output is 114 kHz at −57 dB, which is sufficiently low to be negligible. Noise and distortion are limited by the performance of the earlier audio processing chain.

Stereo/mono switching is accomplished by killing the 19 kHz oscillator, which in turn kills the 38 kHz and eliminates the subcarrier. The stereo/mono control is interfaced to any remote control by means of opto-isolators for RF and ground loop supression. A simple set-reset flipflop provides latch for the stereo or mono mode, and is arranged so that the system always comes up stereo in the event of an AC power interruption.

The systems approach makes setup of the unit almost foolproof. The outputs of the left and right input amplifiers are metered, as is the output of the L-R amplifier in the stereo generator. Setting input levels is merely a matter of sending a mono audio signal from the studio, advancing the left input level control until the gain reduction meter reads the desired amount of compression, and then advancing the right input level control until the L-R metering nulls. Then the output level is advanced until 100% modulation is obtained. In a new installation, it is sometimes desirable to tweak the pilot phase and L-R gain controls to compensate for RF modulator quirks, particularly with older exciters.

Interfacing with RF modulators is most readily accomplished by means of the interface modules which all major manufacturers of exciters can provide to interface to composite STL receivers. The output of the stereo generator has been designed to look like the output of such a composite STL receiver electrically. However, we have had considerable success in interfacing the system to certain exciters directly, provided that the baseband output lead length is limited to less than twelve inches.

A prototype of the system under discussion has been tested on the air at several stations. In general, the system has performed according to expectation. Subjectively, there is extremely little sense of limiting or compression, and no audible difference in high-frequency response between "air" and "program" for approximately 95% of all current recordings. Any high-frequency loss that does occur sounds perfectly natural, without timbre modulation or distortion common to other systems, and is detectable only in a direct A/B comparison with the original source.

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Gates Broadcast Equipment Division
Harris, Corporation, Quincy, Illinois

Design Criteria for a 1 kW Solid State AM Broadcast Transmitter

This paper will describe the design criteria of the world's first FCC Type Accepted all solid state AM Broadcast Transmitter, the Harris one-kilowatt MW-1. This advanced new transmitter was developed to provide the 1 kW broadcaster with a powerful, clean sounding signal, an established characteristic of other Harris MW transmitters, the 5 kW MW-5, and the 50 kW MW-50.

This transmitter is all solid state because transistors have now become cost competitive with the tubes presently used in 1 kW designs. The technology necessary for this transmitter has been available at Harris since 1967, and in fact, the power amplifier portion of the MW-1 has been field proven by using 4 power amplifier modules as a 500 watt RF Driver in our MW-5 transmitter.

After establishing the need for a new 1 kW transmitter, and deciding it should be solid state, the following criteria were used to provide an optimum design for the 1 kW broadcaster:

1. Reliability: Obviously, reliability is of prime importance. Transistors are inherently reliable; however, if too many devices are employed, overall reliability could suffer unless proper sensing and protective techniques are employed. For example, when many devices are used in parallel, the circuitry must be such that the failure of one device does not cause failure of the entire unit.
2. Transient Response: The transmitter must provide superior transient response equal to or better than that discussed last year in our NAB paper entitled "Enhancing AM Signal Coverage through Improved Modulation Techniques." This will increase loudness or coverage.
3. Distortion, Response, and Intermodulation: These must be better than that previously available. For instance, intermod in older transmitters could run as high as 10%.
4. Cost to the User: The initial or acquisition cost to the user must be reasonable, but the long term cost should be less than tube type transmitters through lower operating and maintenance cost coupled with increased revenue because of improved signal quality and high modulation capabilities.
5. Redundancy: Since transistors are limited in power output per device, we know that many transistors will be necessary to provide 1100 watts carrier and 6000 watts peak power required for a solid 125% peak modulation capability. Therefore, the power amplifier and modulator transistors must be paralleled in such a fashion as to provide redundancy.
6. Maintainability: In the average 1 kW station, the Chief Engineer has more duties than that of maintenance and repair. He will have less time to understand the intricate workings of any new device. Since the transmitter must work for the station to be on the air, troubleshooting devices and indicators must be provided to help the engineer locate faults and repair them. This also means making the transmitter as modular as practical.
7. The transmitter must protect itself against brownouts and lightning.

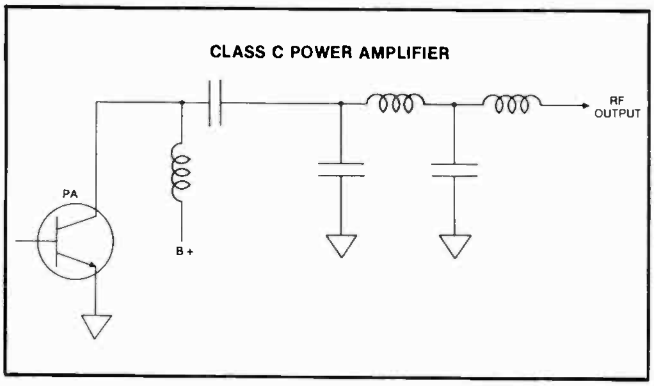

Figure 1.

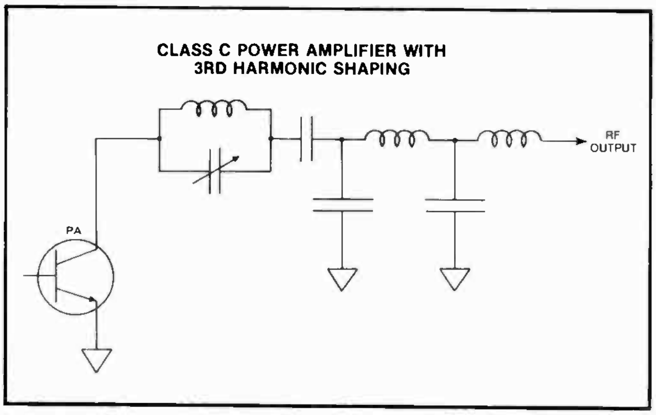

Figure 2.

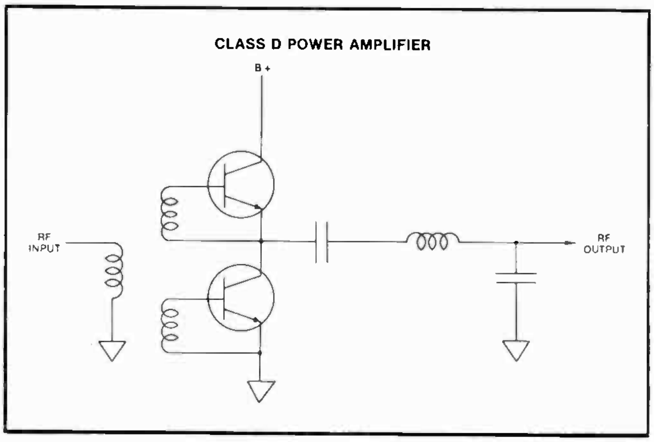

Figure 3.

8. Accessibility: Accessibility and size are related. We decided the transmitter size should be the same as our present BC-1H to help make all components as accessible as practical.

9. Efficiency: The new transmitter must be more efficient. However, in the 1 kW range where the tower lights can easily take more power than the transmitter, it was decided that efficiency should come second to simplicity if a trade off was required.

Let's keep this criteria in mind as we look at individual parts of our MW-1 transmitter.

First the power amplifier design. Let's look now at some of the alternatives considered before our final design was chosen. First, the conventional single ended Class C power amplifier. **Figure 1** This circuit will provide about 60% of the power per device that could be supplied by other PA designs— and it has a second serious drawback, which is poor lightning resistance. The single ended Class C amplifier provides no way of limiting the collector voltage while the transistor is in the off condition (which is most of the time). Thus, a lightning strike could cause the collector voltage to rise high enough to exceed the transistor voltage rating and destroy the device. Transistors have almost no overvoltage tolerance. To summarize, the conventional single ended Class C amplifier would require more devices than other methods and provide poor, if any, lightning protection.

Next is the single ended Class C power amplifier using third harmonic wave shaping. **Figure 2** We get good power output per device, but we have added more tuning controls and still have poor lightning protection.

Let's look at the Class D push-pull power amplifier used in the MW-1. **Figure 3** First, by operating the transistors in a push pull circuit, we will get some automatic lightning protection. The collector voltage of the power amplifier cannot exceed the power supply voltage even during a lightning transient inpulse coming back through the output network, due to the collector-to-base diode in the transistor. Second, the push-pull design allows us to operate the transistors at twice the voltage possible in the single ended design. This allows us to drop the supply current in half and reduce the power supply losses and increase the overall efficiency of the transmitter. Finally, we operate the transistors Class D, which is a square wave switching mode to attain an efficiency higher than that of the Class C amplifier using wave shaping, and we add no new tuning controls.

After deciding on the circuit for the Power amplifier, we know that 12 pairs of transistors are required to provide the 1100 watts carrier and 125% positive peak capability. The combining technique used to combine these 12 pairs of transistors becomes critical. We chose to combine using 90 degree networks, the same as used in our MW-5 RF Driver. **Figure 4** This 90 degree network does not require adjustment. It is a result of the design Q of the tank circuit. The single-knob tuning shown on this slide is that of the conventional PA tuning. These coils are all ganged together.

The 90 degree network isolates each of the 12 modules from one another. At first glance, it could be assumed that the failure of one module would cause the remaining modules to be more heavily loaded, causing more stress on the remaining transistors. This is not the case. By using 90 degree networks, the loading is actually decreased on the operating modules, causing less current to flow.

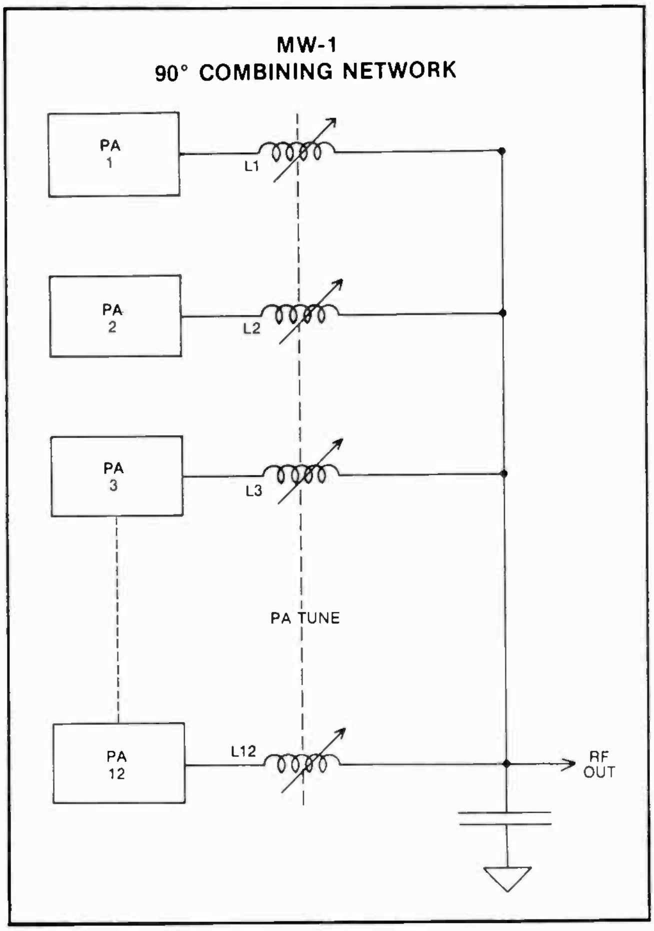

Figure 4.

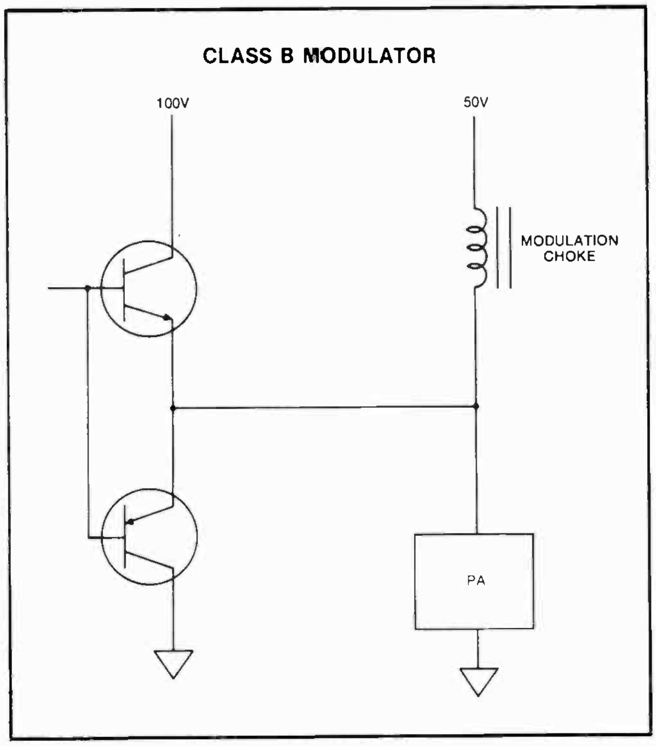

Figure 5.

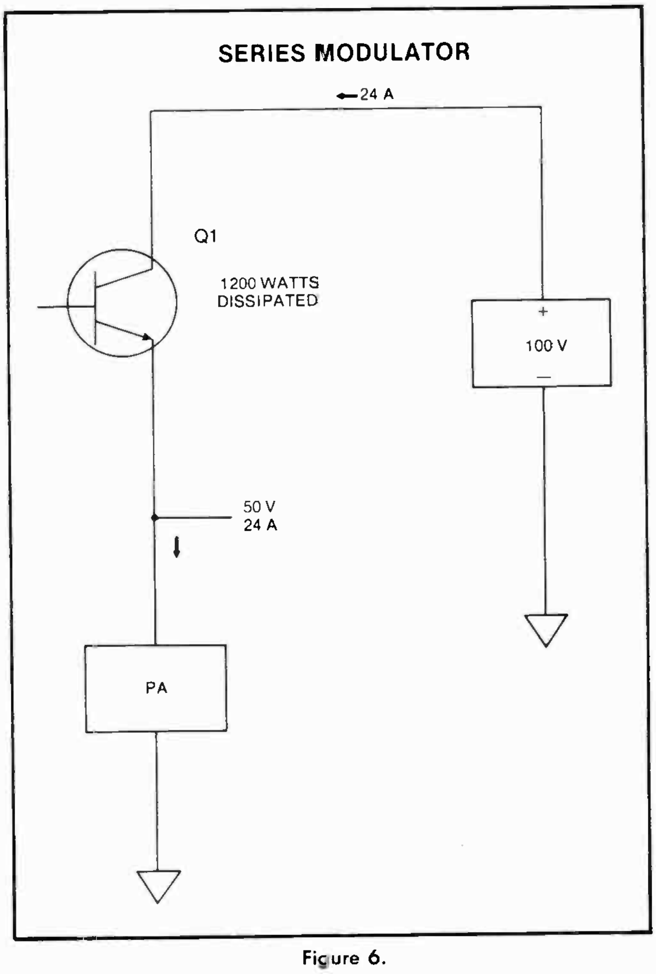

Figure 6.

Now it's time to consider the possible modulator technique that could be used. First, the Class B push-pull modulator, using complimentary transistors. **Figure 5** This would certainly be the most well known from an operator standpoint. However, a modulation choke would be required. There should be nothing embarrassing about using a choke, except that we will have difficulty meeting our transient response characteristics, and lose the ability of using overall DC feedback for automatic power control and carrier shift correction, as used in our MW-50 and MW-5. Also, this modulator will not split up easily if we would choose to put a modulator with each power amplifier.

This leads us to the series type of modulator. **Figure 6** The simplest type, shown in **Figure 6**, consists of one set of pass elements to control the power amplifier voltage, similar to the pass element in a conventional regulated power supply. Its efficiency is too poor even for the 1 kW broadcaster. Let's briefly look at this. Assume carrier conditions where the PA requires 50 volts or 24 amps. The power supply voltage must be twice this to provide 100% positive peak. This means that 50 volts are across the modulator during carrier and the 24 amps required for the PA are flowing through it. The

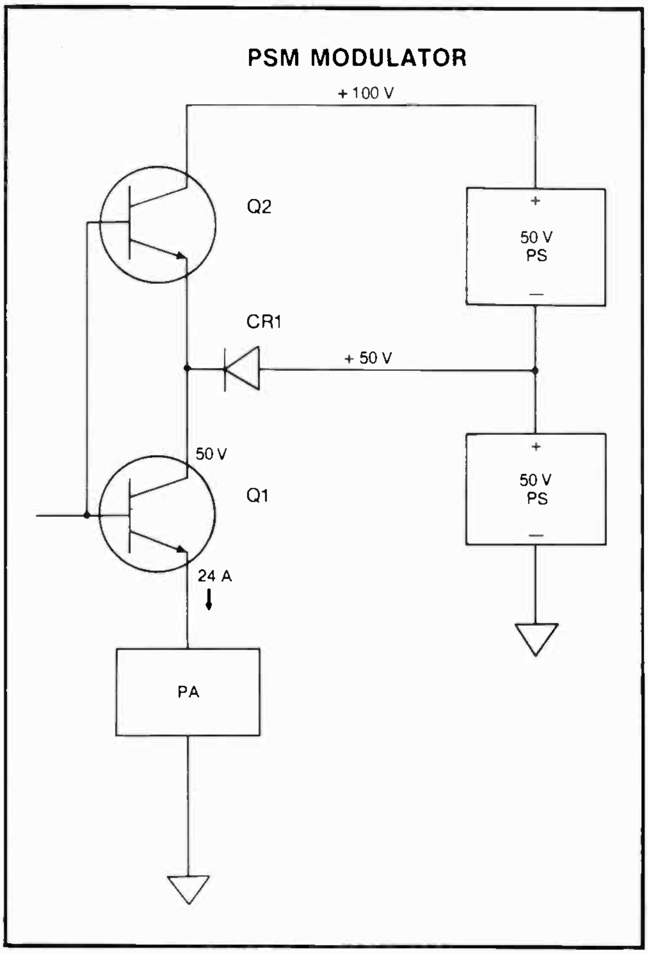

Figure 7.

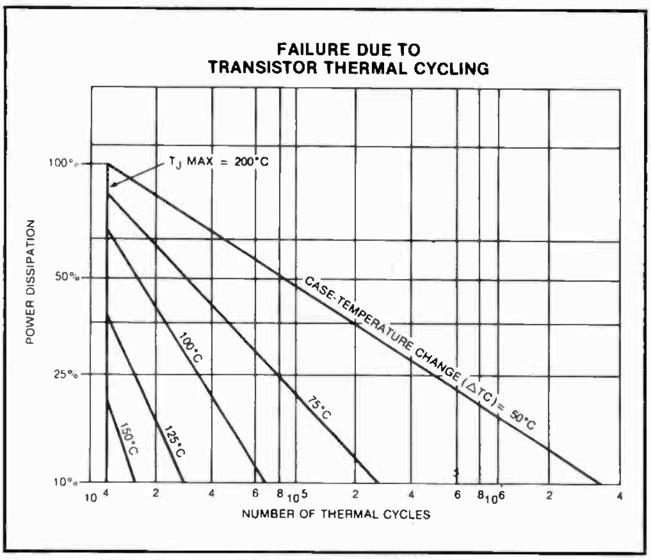

Figure 8.

power lost in the modulator pass transistors would then be 1200 watts which is just too much. This modulator can be DC coupled, would have excellent transient response, and could be broken into to 12 separate elements if desired—its only drawback being poor efficiency.

We really need a new invention. The modulator must be simple, efficient, DC coupled, and have the capability of being split into 12 parallel elements. We call this invention PSM or the Progressive Series Modulator. **Figure 7** The PSM series modulator consists of 2 emitter followers in series rather than one as in the simplest series circuit. It requires no modulation transformer or reactor, no filter inductor, and no 70 kHz filter, and it uses standard audio transistors.

Let's look at the efficiency of this circuit at carrier conditions. Again assume that 50 volts and 24 amps are required for the PA. During carrier, this 50 volts will come from the lower supply thru CR1 and Q1, which is near saturation. This slide shows no loss in the modulator; however, in real life, these voltages are offset a little to provide the 125% modulation capability and the losses will be about 100 watts or so. During modulation, this circuit acts similar to the Class B modulator with the lower transistor, Q1, modulating in the negative direction and the upper transistor, Q2, modulating in the positive direction—Q1 being saturated. Diode CR1 automatically disconnects the lower 50 volt supply when Q2 begins to conduct. In summary, by using the PSM method, we get the simplicity of a conventional Class B modulator, the performance of a simple series modulator and good efficiency. Our old transmitter was less than 38% efficient whereas out new MW-1 is 50%.

We now have the power amplifier, output network and modulator. PSM allows us to put a modulator with each power amplifier to form a module.

To aid in troubleshooting, we place an RF indicating sensor at the output of each PA/modulator module. Now, should a module fail, for any reason— the respective indicator will light on this panel telling of the failed module—the same method used in the MW-5 RF Driver. This makes troubleshooting to the PA and Modulator very straight-forward. The transmitter will operate normally with a failed module, or with a module removed from the transmitter. If a module fails, it may be removed at a convenient time, replaced with a spare, if desired, and repaired on site with a VOM, or returned to Harris for repair. If more than one module should fail, the transmitter will reduce its power, but will still fully modulate.

We must package the transmitter with reliability and accessibility as the main criteria. The first problem is the heat sink for the PA/Modulator module. This appears to be a simple task. However, when the utmost in reliability is required, the cooling of the transistor is critical to prevent thermal cycling fail-

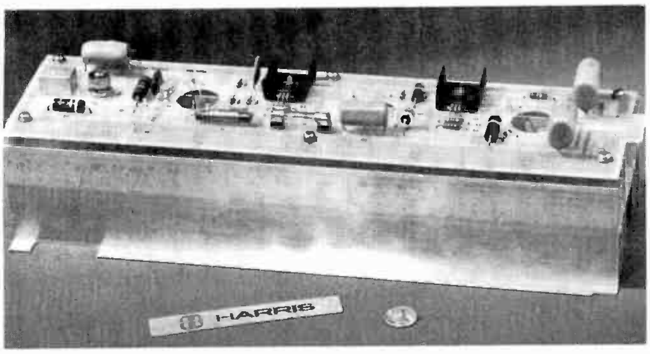

Figure 9.

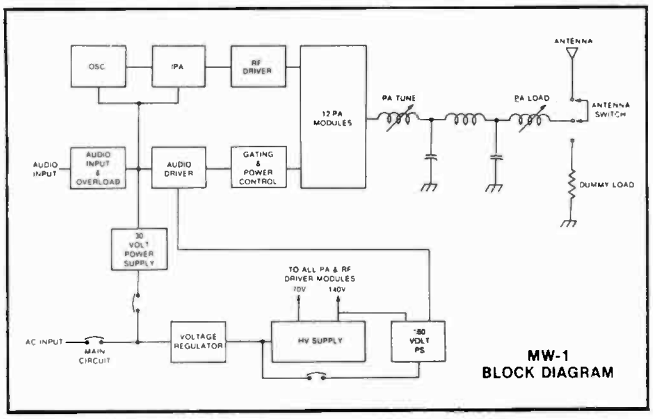

Figure 10.

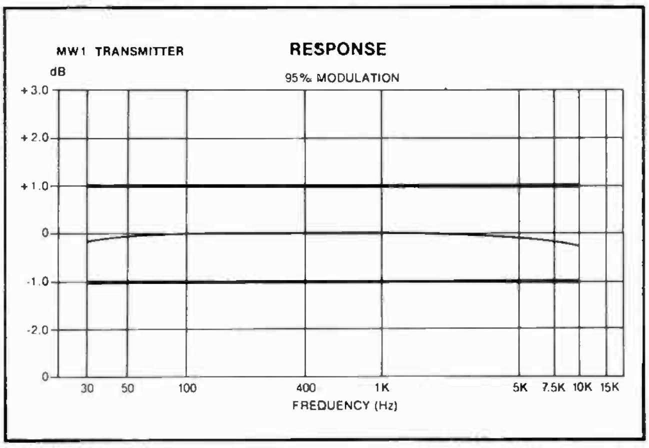

Figure 11.

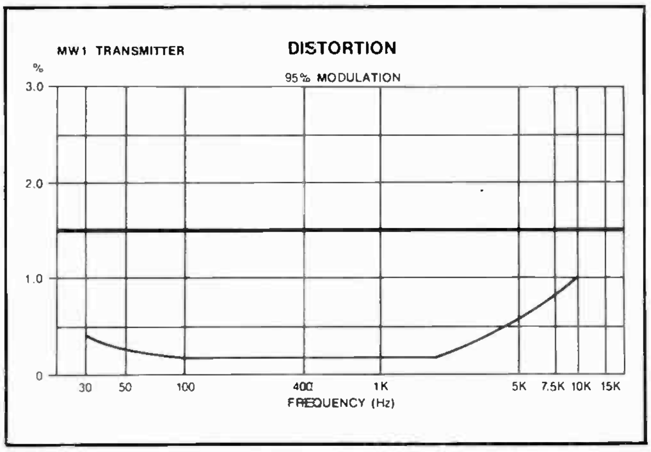

Figure 12.

ures. Let's see how thermal cycling affects transistor life. **Figure 8** In a transmitter, the total power output can change by 50% between a rock record and a voice commercial. This also means the transistors are changing in temperature by about 50%. **Figure 8** shows how temperature cycling affects transistor life. We can begin to see failures in 10 to 20 thousand cycles. This is only 2000 hours of operation, if 10 records were played per hour. Therefore, we package the power amplifier/modulator modules on very large heat sinks and cool them with a fan which keeps the total temperature change from turn-on to max modulation with a highly processed record, to less than 8 degrees C, or off this chart, and near zero. Our final module looks like the module in **Figure 9.** Each PA/Modulator module provides about 100 watts carrier and 500 watts peak. Because of the large heat sink and low dissipation, the transmitter may be operated with no fan at reduced power.

The Block Diagram of the transmitter is shown in **Figure 10.** We have a built in voltage regulator for brownout protection and ease of installation.

Next, there are 12 PA/Modulator modules operating as the PA and Modulator. One more identical PA/Modulator module operates as an RF Driver. This allows the operator to use a PA in the RF Driver position and operate one short in the PA should the RF Driver fail.

We decided to stop the redundancy at this point as high power stages are always the most failure prone. Also, there is really no practical way to have total redundancy without actually having two transmitters.

The oscillator has only 3 transistors and one IC and the Audio input and overload has only 8 transistors (all overload and tally functions). The other stages of the transmitter are simple plug in modules. The IPA has one transistor, the audio driver has 4 transistors.

Now, for the results of this venture—performance:

1. Frequency Response—**Figure 11**
   Easily within our plus or minus 1 dB spec.
2. Positive Peak—**Figure 12**
   The distortion is very low, especially in the mid range where most of the modulation really is.

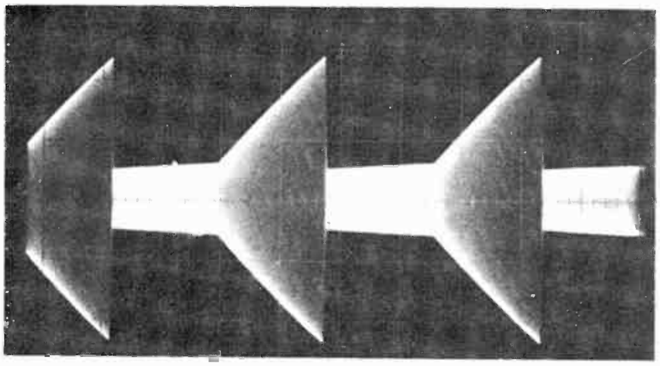

Figure 13.

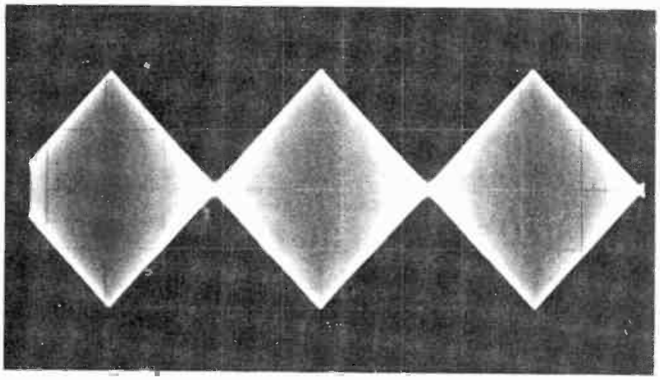

Figure 14.

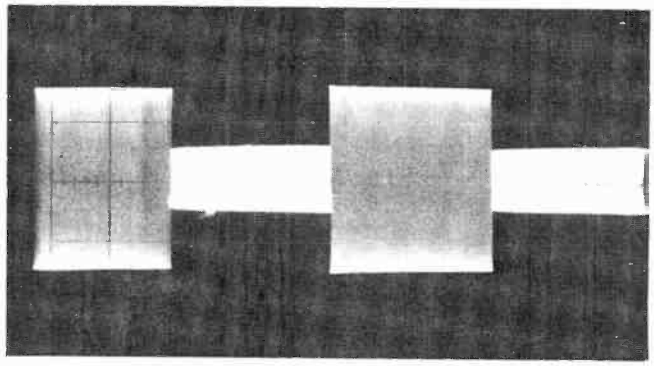

Figure 15.

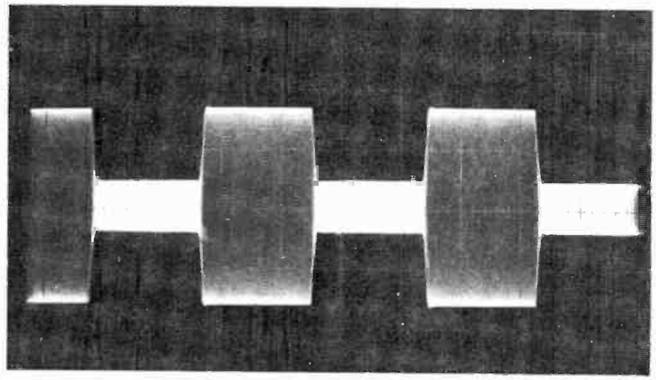

Figure 16.

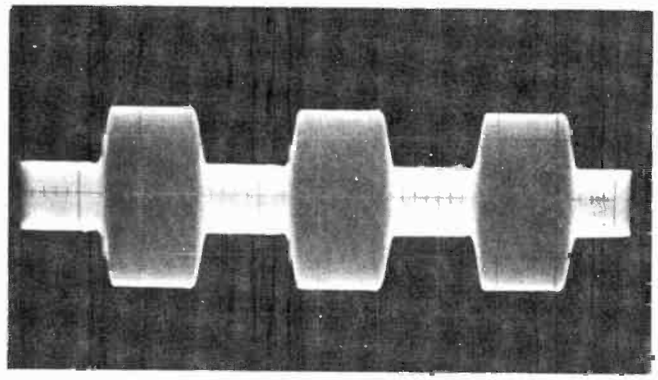

Figure 17.

3. Positive Peak—**Figure 13**
   Using an asymmetrical ramp we can see that the linearity is nearly perfect to over 125% positive modulation. This means our distortion remains low even at the highest levels of modulation.
4. Linearity—**Figure 14**
   Using a triangular wave we can see that the linearity is excellent from 100% negative to 100% positive peak modulation. This also shows why we can spec this transmitter at less than 2% IM, as no crossover or nonlinearities can be seen. By the way, IM was measured at a 4/1 ratio of 60 and 2000 and 60 and 7000 Hz at 95% modulation.
5. Square Wave Response
   First 30 Hz. **Figure 15** No tilt. Next 1000 Hz—**Figure 16** again, no tilt and hardly any detectable overshoot. Finally, **Figure 17** 3000 Hz and again, no phase shift-gain linearity or overshoot problems.

These tests are great, but what does it really do. There is only one real way of finding out and that is by using the actual program audio signal itself. Let's play music through this transmitter and connect a DC coupled dual trace scope to the transmitter. One input of the scope is connected to the audio input line to the transmitter and the other scope input to the actual RF output of the transmitter. Now we can see if the input and the output are the same, meaning a perfect reproduction. Let's backtrack one year to last year's presentation. This shows an old transmitter being modulated. **Figure 18** Bright line is input—underneath is the transmitter carrier—the input should not pull away from the output if the transmitter is to be loud— this figure is showing "bounce." The result of a clipped input is shown in **Figure 19**. The input stops at about 80% negative, but the transmitter goes uncontrollably to 100% due to bounce, tilt, and overshoot. Now, the MW-1! It can be seen that the transmitter exhibits no bounce, tilt, or overshoot even with processed audio. **Figure 20**

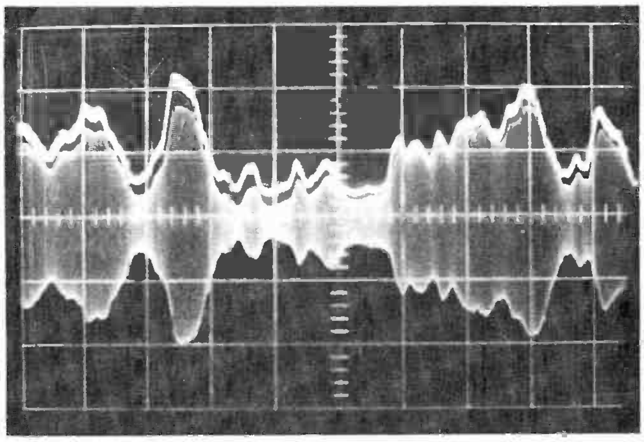

Figure 18.

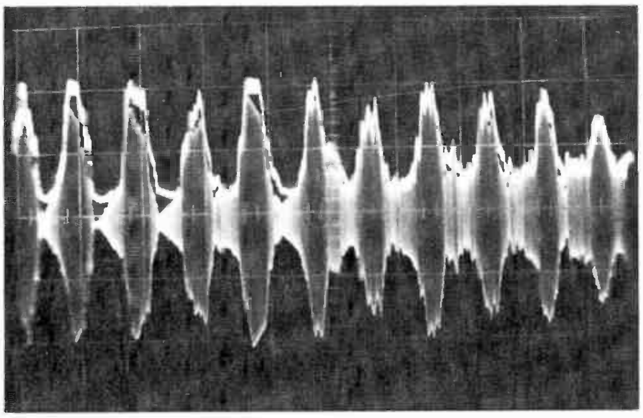

Figure 19.

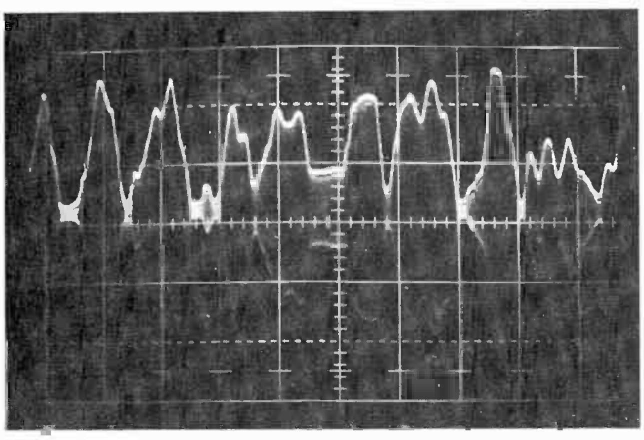

Figure 20.

The last item I'd like to bring up is lightning protection and testing. This must be the biggest concern of anyone here as it seems questionable to connect a voltage sensitive device to a 400 foot lightning rod. We have protected the transmitter against lightning in at least 4 distinct was:

1. We use a push pull power amplifier design which is inherently lightning resistant.
2. The output network is a bandpass type to limit the energy getting to the power amplifier itself.
3. We place a nanosecond Spark gap in the output network which shorts the output network to ground at the first inpulse of a lightning strike.
4. We use a fast VSWR trigger circuit to remove the PA voltage in a few micro-seconds after the lightning strike to prevent damage.

To test the operation of these methods, we did the usual things such as open and short the transmitter hundreds of times. And for the actual lightning test itself, we discharged a 1 uF capacitor charged up to about 10,000 volts into the output network.

During these tests we had no failures of the transistors.

To sum up, with our MW-1 we have provided the following features for tte one kilowatt AM broadcaster:

1. A signal louder and cleaner than that of other one-kilowatt AM broadcast transmitters.
2. Redundancy in the power amplifier and modulator stages.
3. Great reliability, proven in the field and through extensive testing. We have had an MW-1 on the air at a broadcast station since early December, 1974.
4. An efficient, uncomplicated design.
5. All solid state.

Robert H. Fasulkey

Fellow Engineer
Communications Programs
Field Engineering and Support
Westinghouse Defense and Electronic Systems Center
Baltimore, Md.

Space Age Transmitter Today

This Westinghouse transmitter has been under development for several years and in operation for the past few months, both in actual station programming and laboratory test.

Numerous solid state transmitters have been constructed which utilize linear amplifiers of low efficiency and limited power output capability.

The Westinghouse development sequence progressed through the low voltage linear amp and SCR circuitry to the classic Class D or S (switching) amplifier. None of the three methods, linear, SCR or Switching are without limitations. The reason for choosing to develop this circuit was because the SCR and the linear amp contained massive limitations, such as speed, or frequency response for the SCR, and power handling capacity (efficiency) for the linear amp.

The determination to explore and develop switching amplifiers was made in 1963 and that development is continuing at this time.

In the case of the switching amplifier, the ability of the solid state device to change state with rapidity is the limiting factor. This limit manifests itself as an upper limit in operating frequency. In much the same manner as vacuum tubes in the early days of radio, each new development extended the operating range. Since 1963 there has been a continual advancement of solid state devices accompanied by advancements of circuitry by Westinghouse.

The combination of these advances has made it possible to produce the unit that is on display today.

The use of an all solid state design in a broadcast transmitter is most effective in producing high reliability. Power transistors do not have the wear out problems associated with vacuum tubes and reliability is further enhanced by modularizing the transistors into trays of transistors.

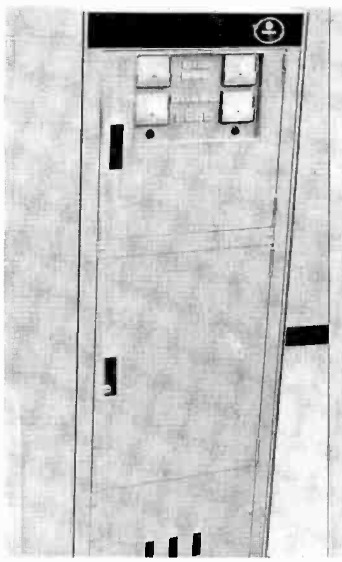

Figure 1

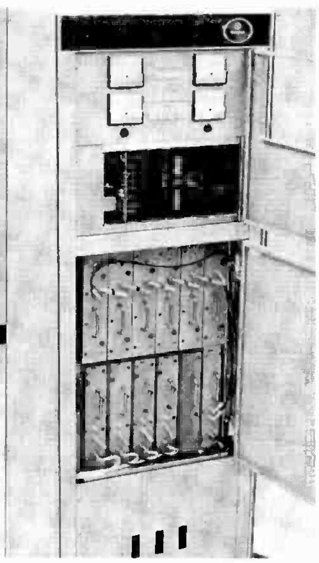

Figure 2

The modular configuration of the 5 KW transmitter is shown in **Figures 1 and 2**. The transmitter contains a Radio Frequency Driver, 6 Power Amplifier trays, an Audio Driver, and 4 Modulator trays, containing a total of 330 (TRW SVT 350-5) transistors. This provides a 20 percent built-in redundancy of transistors. **Figures 3 and 4.** Each power amplifier and modulator tray contains 32 transistors and is capable of generating 5 KW of Radio Frequency peak power under full modulation.

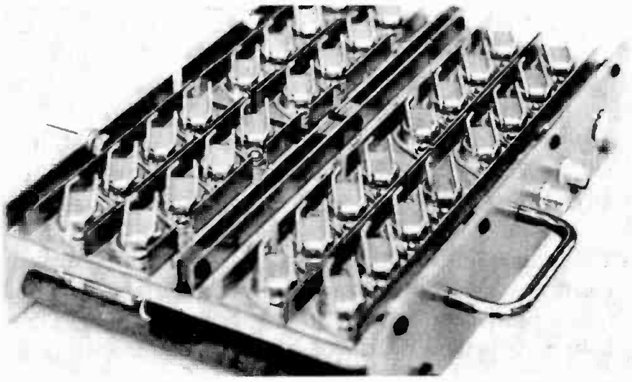

Figure 3

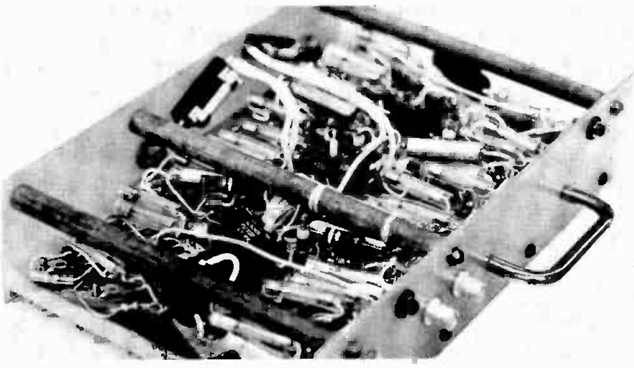

Figure 4

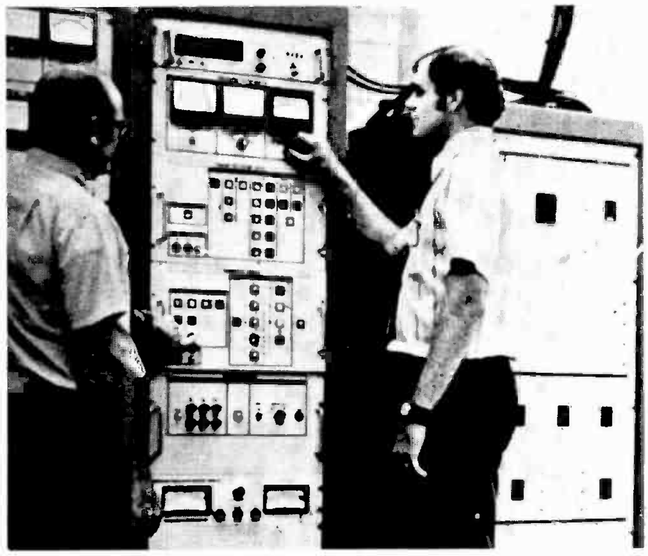

Figure 5

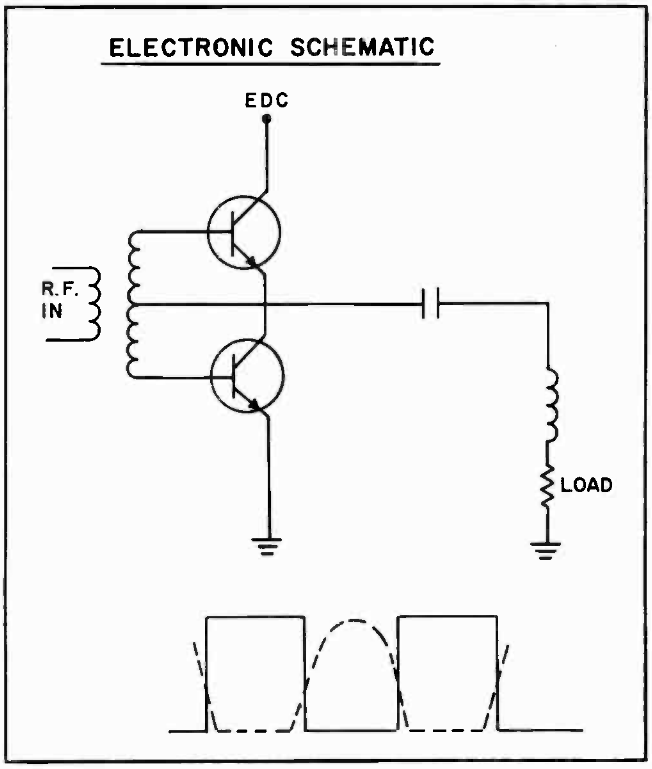

Figure 6

The use of multiple transistors, configured to be fail safe where the failure of a transistor or transistors will not cause the transmitter to discontinue operation is accomplisshed through the use of an evaporative element in both the base and emitter leads of each transistor, allowing the unit to continue to provide full power output. Degradation of the output will not begin to occur until such time as at least 25 percent of the transistors have failed.

The ability to combine multiple transistors and transistor trays into transmitters of large power output has been available for some time within the design used in these equipments. Several 20,000, 50,000 and 200,000 watt transmitters are now in operation; see **Figure 5**. Design preparations are in progress to provide units of megawatt capability.

The power transistors (SVT 350-5) are operated in a switching (Class D) mode, providing very high efficiency **Figure 6**. The choice of utilizing a Class D circuit was reached when all of the limitations of this circuit were compared to the limitations of either the low voltage of a linear or the high current capabaility of an SCR. In the case of a linear amplifier both low efficiency (10%–25%) and low voltage, combine to cause very high currents which have to be handled when high power is required. In the case of the linear amplifier the ability to operate over the usable spectrum is a decided advantage.

When analyzing the SCR (Silicom Controlled Rectifier) the ability of this device to operate at high voltage and conduct a huge amount of current, at first look, makes it seem a natural for the high power amplifiers. However, the inability of this device to cease conduction, limits its application to frequencies below 20 kilocycles. Numerous methods have been attempted to accelerate this turn-off time, but without success.

The choice of the Class D amplifier is not without limitations, but this circuit had a larger window of probable success when the choice was made in the early 1960's.

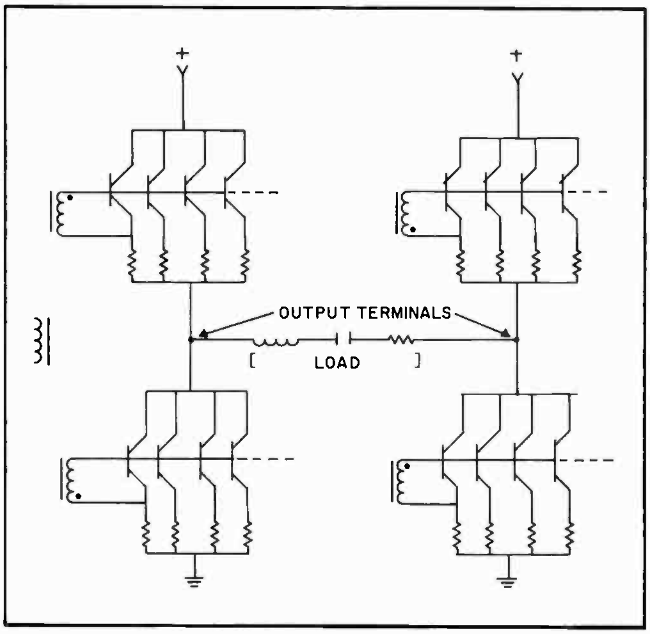

Figure 7

The basic circuit shown in **Figure 6** operates in the following manner, power supply voltage is connected to the collector (c). The emitter (e) of the topmost device is then connected to the collector of the second device and its emitter is connected to ground. The bases (b) are supplied with digital signals which are 180 degrees apart. At the junction of the upper device emitter and the lower device collector is connected a resonant load whose resonance is equal to the frequency of the digital base signal. When the base of the upper device goes in a positive direction the transistor is driven into saturation and the load is allowed to accept energy from the power supply. As the first half cycle is completed, the base is returned to cut-off and the transistor stops conducting. During this change, the lower device is taken into saturation by its base going positive. This allows the load's stored energy to flow in the reverse direction.

The condition of saturation of a device allows current to flow through the device while the voltage drop across the device is near zero. Thus, if a transistor could be turned on and off in zero time, no power would be disippated in the transistor, which would result in near 100% efficiency. Secondly if this zero time on and off could be reached, the frequency handling ability would be infinite. Unfortunately zero time cannot be realized. As is indicated on the waveforms, a small amount of overlap occurs between the voltage waveform (solid line) and the current waveform (broken line) which result in two limiting factors. First is a departure from 100% efficiency to one of 95–98% under laboratory conditions and 90–95% in practical applications, and secondly, the steepness of the vertical portions of the voltage waveform establishes the highest operating frequency. An extensive amount of design effort has been expended to push these limits upward.

Just the ability to cause a transistor to switch on and off at a rapid and efficient rate is not enough to produce a satisfactory unit of equipment. Various other factors must be considered and improved, such as total power, size, weight, and maintainability. Once the technique of switching was developed, the ability of generating more power than two transistors were capable of had to be accomplished. The obvious step is to add devices in parallel until enough current could be developed to produce the desired output. The tray shown in the photo, **Figure 3**, can deliver several kilowatts of power on a continuous basis.

Once the parameters of switching were managed, the next immediate problem was to convince the several transistors in series/parallel to change state at the same instant, **Figure 7** or with enough time between opposite transistor states to prevent a short from appearing across the supply voltage.

It can be readily seen in this circuit depiction that should each device be on at the same time, each would be destroyed as they would appear as a short across the power supply. The synchronization of the numerous transistors was accomplished with the development of the tray configuration. The tray layout is a complete bridge circuit, with 4 legs of 8 transistors each in parallel for a total of 32 transistors, 16 of which are in a conducting condition for each half cycle of the carrier frequency. The conduction cycle of each transistor is held to a mirror image of each other transistor, thus preventing an imbalanced load to any of the devices, assuring a precise current division. The technique that permits precise load sharing is part of the several patent applications on solid state switching amplifiers by the Westinghouse Electric Corp. The usual current sharing technique of using a small amount of emitter impedance was augmented and improved to near perfection by the application of strip line construction, nominally used in the gigacycle region, and through automatic drive level control, applied on a cycle basis. These same techniques not only permit the precise operation of transistors at high efficiency, but realize an added benefit of allowing operation at much higher frequencies than had ever been accomplished before. Once the switching technique had been mastered, the several parameters associated with a complete system could be optimized.

The development of switching amplifiers was begun in the early 1960's, see **Figure 8**, and at that time the maximum capability of any solid state amplifier was 900 watts per cubic foot. Accomplished values for 1973 are well over 3000 watts per cubic foot, with a trend line continuing to beyond 4000 watts per cubic foot. This line continues at a most optimistic rate for the foreseeable future.

In the category of system weight **Figure 9** the trend line rate of the preceding graph is evident.

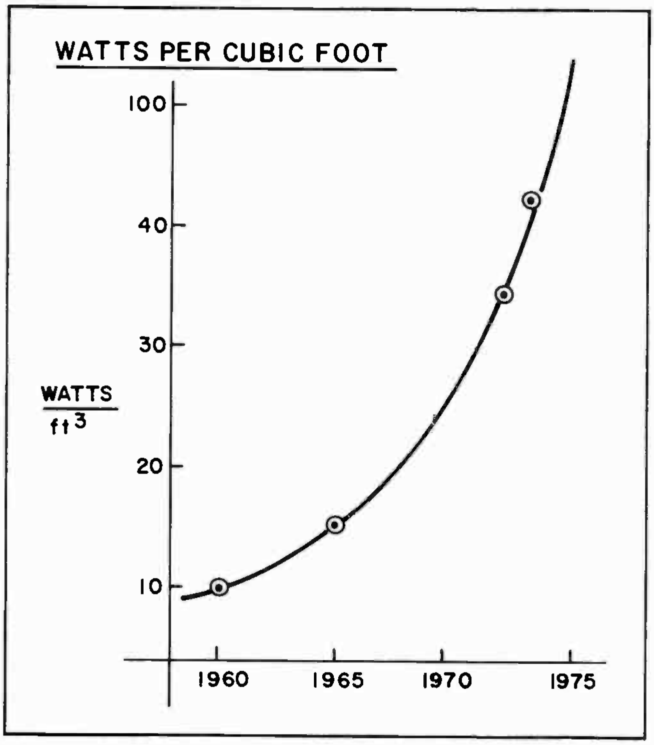

Figure 8

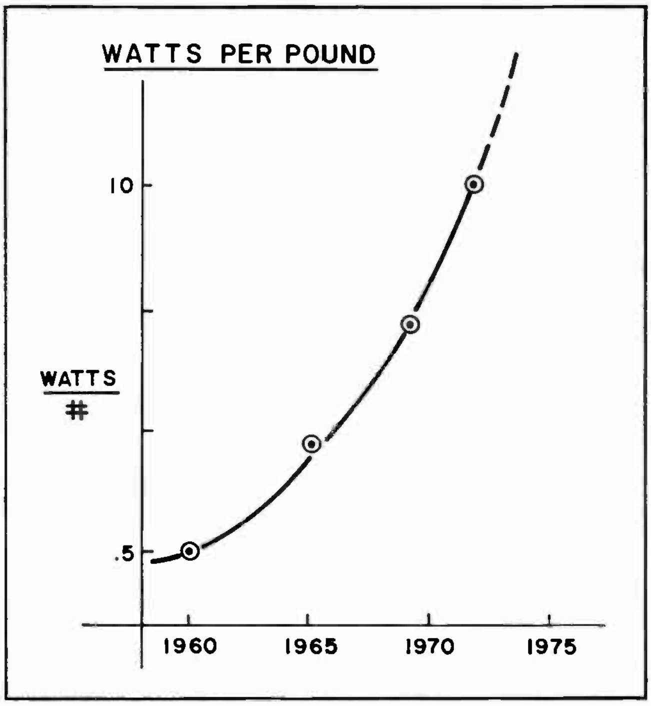

Figure 9

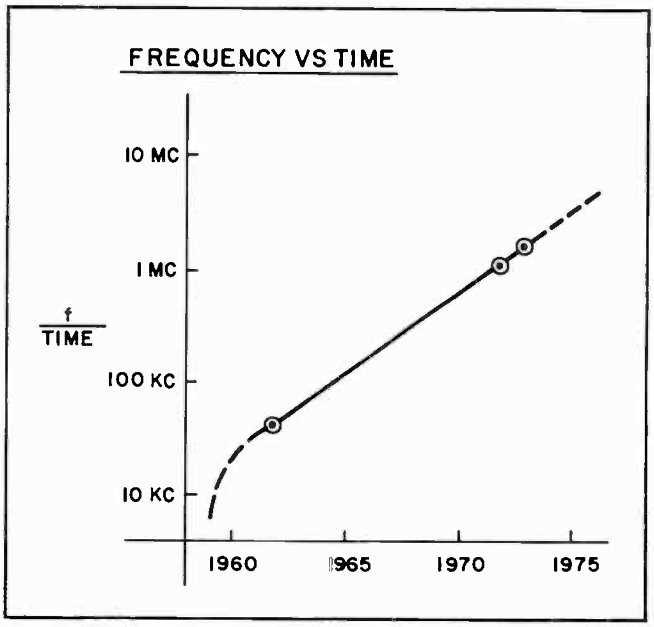

Figure 10

Again in the early 1960's, the possible amount of power per pound was very limited, with only about one half watt of capacity per pound of system weight. This factor has improved during the past 10 years by at least twenty times. Ten watts per pound for an entire system is an easily achievable quantity. On an individual transistor basis this power could approach 500 watts per pound.

Although significant, the power per volume and weight is not the factors which permitted solid state Class D amplifiers to progress into the broadcast frequencies. The opening remarks of this paper discussed the limitations of each method of achieving high power operation from various solid state devices. In the case of switching amplifiers the highest possible operating frequency corresponds to the length of time required for the device to get into saturation, i.e., full conduction, and then return to a non-conducting state **Figure 10**. During the past decade this capability has been extended from 60 kilohertz to nearly 3 megacycles, with practical operating systems to 1.5 megacycles. Contrary to some published information, the present practical limit for high power operation is possibly 2 megacycles, however, efficiencies of 90% and more with output power into the hundreds of kilowatts is quite possible.

In constructing a tray **Figure 3**, the technique of harnessing multiple transistors as a team was achieved, but the physical limitations of paralleling a hundred or more in such a manner presented the problem of current management; that is, being able to effectively utilize the power being generated. When the tray becomes too large, the magnitude of current flowing in its various legs becomes excessive for the conducting medium. This limitation sets the final size of a tray. The electrical consideration is not the lone factor. It is at this point that maintainability enters the design equation. If a multiple number of trays are used in the design, then each is easier to handle, allowing switchable replacement for automatic repair should one tray fall below its rated contribution. Assuming a design criterion of 5 kilowatts per tray, any number of trays can be added or stacked to produce a desired level of power output, **Figure 11**. However, the simplicity of such a state-

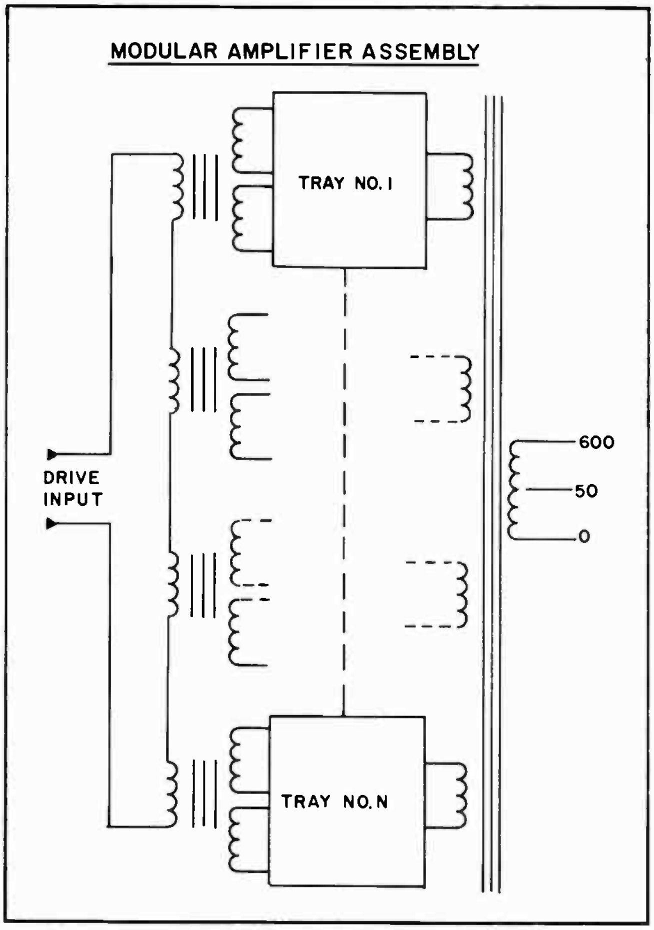

Figure 11

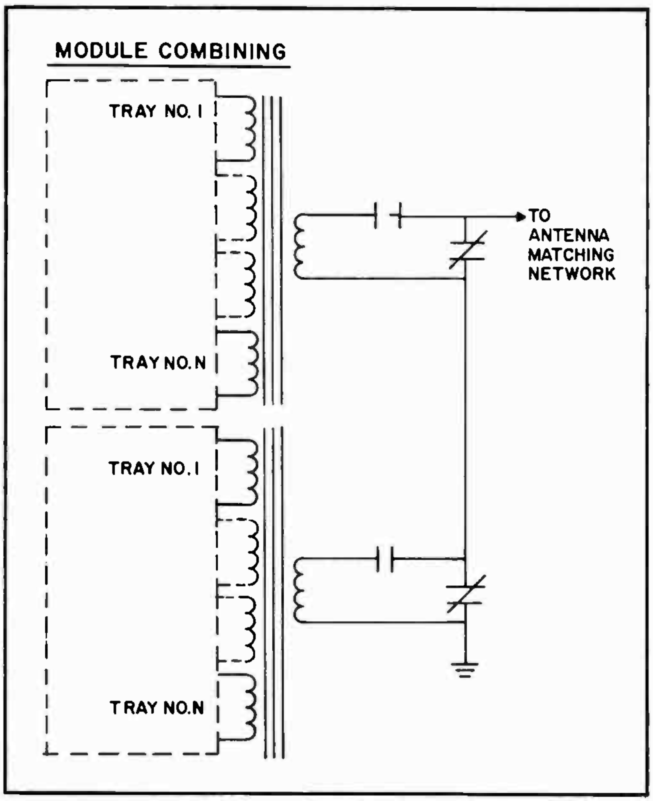

Figure 12

ment would not be realized in actual practice. Much the same as in the case of individual transistors, each tray must contribute its share precisely in phase with each other tray. An idealize method is shown in this figure. Transformer combining and coupling provides two advantages. First, isolation is obtained between trays, and secondly, the required impedance transformation can be obtained between the low level (6 – 12Ω) at the terminals of the tray and the line operating level (50–600 Ω).

When a power level of 5 KW is desired (25 KW PEP), a system determination must be made for a desired reliability factor and initial cost/repair cycle, but for the purposes of this discussion we will establish that 5 trays satisfy all the system requirements. We now have a combination of basic modules which in turn furnish the building block for higher power units. **Figure 12** The basic combining transformer is retained and taps added to the secondary so that when the number of modules is changed, they can be added in a series configuration with an appropriate tap change. In this manner several basic 5 KW modules can be combined to produce any given amount of power output. In the case of a 50 KW unit a nominal 10 modules would be employed, however, an eleventh module could be provided, so that switchable redundancy might extend the reliability and facilitate maintenance.

For example, the addition of an extra unit would extend the purchase price by 6 – 10% while extending the probable failure interval by many thousands of hours and in the case of an inadvertant failure, the ability to remain at full rated power **Figure 5** by inserting the standby module. This substitution technique is now being employed in a fully automated 50 KW station.

We have now examined several steps of power generation, from the mode of operation of the initial transistor, to combination on a tray, to the complete addition of numerous power modules. In each step the same identical piece parts are employed, which will eliminate the cost of a wide variety of spare parts, and practically eliminate the requirement for "on hand" spares. Moreover, the necessity of replacing a costly tube would be gone due to the minute segmentation of the active elements and their negligable wear out factor.

Thus far we have addressed the radio frequency generation portion of a transmitter. Of equal importance is the intelligence section or modulator. Just as was done with the radio frequency transistors, the circuitry surrounding the audio frequencies was developed to utilize the identical device (SVT 350-5) in a high efficiency switching modulator **Figures 13 and 14**. No significant difference exists between the construction of the modulator tray and an RF tray. Again as in the case of the RF amplifier, several modulator trays are used for redundancy and power handling capability and the basic building block has

Figure 13

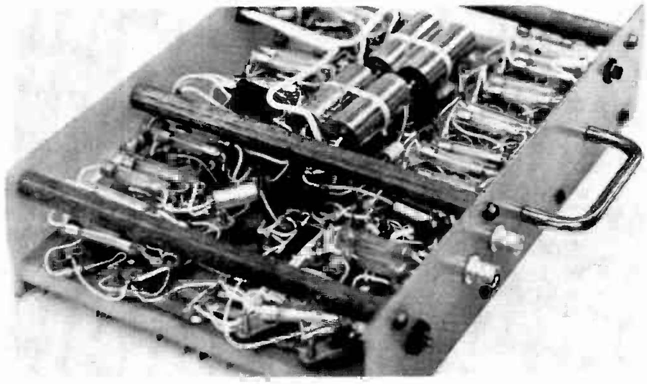

Figure 14

been retained allowing trays to be paralleled as modulation requirements increase.

In concert with the modular concept for a final amplifier, as an example, a 5 KW amplifier would be married to a modulator of equivalent capability to form a basic building block, providing a 5 KW AM transmitter, having several radio frequency generating trays and several modulator trays, **Figure 1**, several of these modular blocks have been combined to provide very high power. In all cases, should an unexpected failure occur a complete loss of power will not be experienced, rather minimal reduction in power will occur if no active spare were available. Thus if ten 5 KW modules are used to provide a 50 KW carrier the defective unit may be automatically switched off the line and the balance of the unit operated at 45 KW. With the modulator being an integral part of the amplifier segment, no other change would be required and proper operation would be retained at a slightly reduced power level. In the case of a 5 KW transmitter which contained 5 tray modules, each having 1 KW capability, should the transistor complement of a single tray be reduced by more than 25%, the tray might be switched out of service and the remaining 4 allowed to operate at some value between 4 KW and 5 KW, in this manner a station is assured of at least 90% of his coverage at all times.

Design For Reliability

There is an ever increasing interest in the automatic, unmanned facility for communications, primarily because of the significant savings which can be realized in the life cycle costs. Life cycle costs can be defined as the summation of incurred housing and real-estate costs plus equipment procurement, installation, training, operation, and maintenance costs for the life of the facility. With constantly increasing cost to train, house, pay, protect, and retire manpower, automation of equipment so that it can be operated unmanned becomes increasingly attractive. This is particularly true with the high power radio transmitter terminal where 24 hour, seven day a week shifts of several skilled personnel has historically been a strict requirement. Although the initial cost of procuring an automatic facility is, as might be expected, higher than a conventional design, the reduction in life cycle cost makes the approach a profitable one.

The benefits to be gained from automation have been apparent for some time; however, several difficulties have impeded its feasibility until recent technology provided practical solutions to the problems. First, unattended operation of a major transmitter facility essentially requires evaluation, decision, and enabling functions to direct the complex control, monitoring, test and administrative operations needed for safe and efficient operation. The recent availability of the mini-components at a reasonable cost has solved this problem and, combined with appropriate hard wire logic, can provide rapid, accurate, yet flexible automatic station management.

Another basic requirement for successful unattended operation, and in particular for the high power installation, is equipment reliability. This demand is met in the solid state transmitter by employing inherently reliable components, and utilizing modular redundancy with automatic standby switching. The fault detection circuitry is used to evaluate status and implement switching to select the optimum modular configuration for the given set of operating (or failure) conditions. This approach not only achieves the best operating capability but provides almost instantaneous selection, not subject to panic errors, which sometimes occur with manual operation. This rapid and accurate reconfiguration of redundant modules is a major contributor to realizing an impressive degree of reliability mandatory to profitable station operation.

In regard to component reliability, the use of all solid state design in the transmitter is particularly effective in producing high reliability. The power transistors **Figure 3** do not have the fragility and wearout problems associated with vacuum tubes. The use of multiple transistors in parallel, arranged to be failsafe, essentially eliminates the transistors as a cause of transmitter failure.

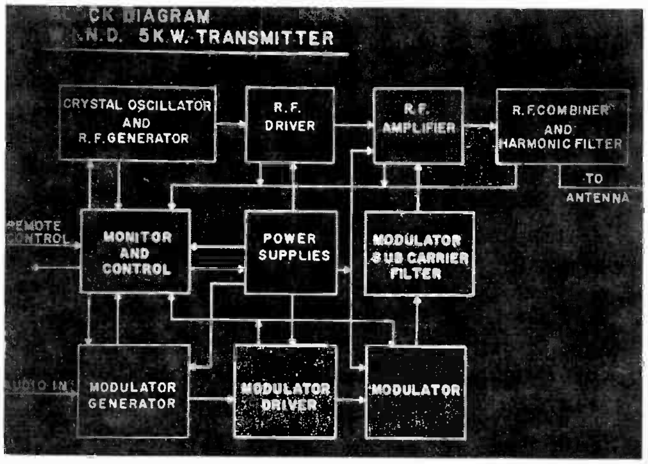

Figure 15

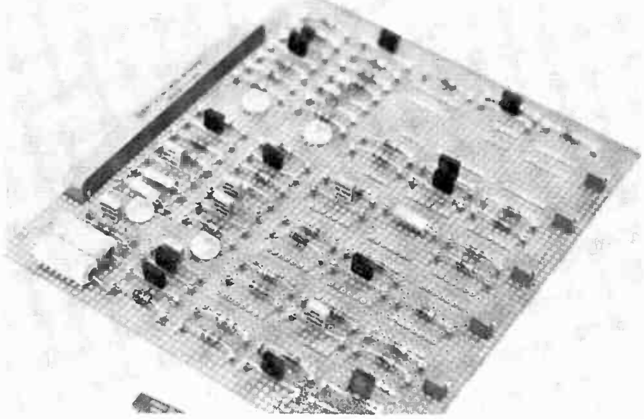

Figure 16

Design For Automation

Although there are no formal provisions for the automatic operation of a broadcaster transmitter, speculation about the basic parameters which would need to be controlled can easily be made. The need for the carrier power and frequency to remain within specified limits is quite apparent. The recent developments in frequency generating devices provide stability of the order of 1 part in a million for years at a time, so that with a choice of three basic standards, a comparison can be provided that will insure proper frequency for an infinite period. In addition, to assure proper equipment operation a figure of merit for the magnitude of the oscillator output would be necessary. The quantity of circuitry and cost for providing the automatic monitoring of these two parameters is very small.

Likewise, carrier level, modulation in both directions and carrier shift can be monitored, corrected and reported.

Thus far, the excursions of vital parameters into restricted regions has been discussed, but this is only half of the requirement. Naturally the regulating agency will not allow more than a safe percentage of increase in carrier level as well as modulation, however, no real percentage for decrease is established as cause for shut down. In this area the owner would want to maintain as high a carrier level and modulation value as possible without exceeding the legal limits. Thus we have established two sets of parameters, one residing within the other, much the same as small panes of glass in a large window.

These limitations can be named as the "owners window," where satisfactory operation and coverage is assured while violations of regulations are prevented and a second and larger window defined by regulations where the transmitter must be shut down.

A typical Solid State transmitter, as shown in **Figure 1,** has a block diagram as shown in **Figure 16,** where there is only slight deviation from a conventional transmitter. In this case, the predominant factor is the block indicated as control circuitry. With the advent of micro circuitry at a nominal cost, it is now possible to accomplish numerous functions of data processing for automation, which in the past would have been impossible.

The values of the several parameters that need monitoring are generated in the circuitry contained in the low level portion of this transmitter. **Figure 16** Each of the several cards are individually plugged into the card tray assembly, this allows for rapid exchange should a failure occur and also provides an easy method of reconfiguration or improvement. These cards are of a low enough cost that spares could be maintained in much the same manner as components are stocked at this time. The necessity for in the cabinet repair would be eliminated. And if it were desired, a repair service by the manufacturer could be utilized, or in the case of many stations who maintained an engineering staff, the printed circuit card could be plugged into a test fixture and repairs completed in a leisurely manner. The interface points of the various cards has been chosen so that the noun name of the card describes its functions.

Such names as Modulation Generator, Power Control and Fault Detection are typical of the names selected. The modulation generator accepts the coherent subcarrier and audio input as well as the necessary feedback and audio shaping signals, combining these with the carrier level controls to provide an output to the modulator driver whose square wave duration corresponds to the frequency and level of the input program. The first of the unique happenings occurs in the power control by the use of an optical coupler to control the power level of the transmitter. A light emitting diode presents an output which is proportional to the voltage across the modulator. The averaged light intensity of the L.E.D. is sensed by a light amplitude sensitive transistor whose output, in turn, is used to control the carrier level of the transmitter. In this manner changes in carrier level due to input line voltage variations is virtually eliminated over an extensive range of conditions. This capability of carrier control is much like the action of a governor never allowing excessive carrier nor allowing a reduction to go unnoticed. Power control is the form of manual interface is carried out through this board. This capability augments the automatic optical coupler and also provides sixteen (16) steps of manual carrier level control.

The Fault Detection board is much like a traffic signal, in that, numerous nerve endings are monitored in this sensor and appropriate commands are dispatched to the mundane items, such as temperature detection, cooling air, and safety interlocks, Tray Balance, Reflected Power and over current. Tray Balance is a measure of load sharing between the various amplifier trays should an inbalance condition occur a warning signal advises the station operator. The first is an early warning to the station operator that maintenance is necessary and a second or shutdown response is initiated should be imbalance indicate a malfunction. Additionally all supply voltage are compared against a reference and should any of these approach a predetermined threshold, a warning is provided, and if any of the voltages exceed a desired level or fail the transmitter would be shut down. In all cases, status indication is presented on the control panel so that rapid analysis can be made and appropriate action taken. Eight discrete conditions are sensed and acted upon in the fault detection card. Should a momentary fault exist, the fault name is illuminated on the fault panel as well as the re-set button. This status is non-evaporative even though the transmitter is again functioning properly. Such storage of data will allow a tabulation of momentary outages for use in maintenance analysis. Accumulation of outages is also sensed in the transmitter fault detection unit and if a rapid sequence of momentary faults are sensed the transmitter will shut down. Operation can be restored by initiating the normal start up sequence if the causative fault has been cleared.

The frequency control board provides the time base for the transmitter and the audio processing sub-carrier. The primary standard is two Bulova oscillator assemblies whose output is reduced to the desired carrier frequency and to the subcarrier frequency. In this manner the operating frequency is held to a very tight tolerance as well as holding the sub-carrier precisely in the center of its pass band. Whenever eight of two parameters might drift out of prescribed limits appropriate action is taken to prevent a violation of regulations. Initial operation begins on one of the clocks and if either amplitude or frequency limits are exceeded, a switching circuit will select the alternate clock and a fault indication presented on the control panel. Should a failure occur in the second clock, either in amplitude or frequency no attempt will be made to return to the first source but the transmitter will shut down, with an appropriate alarm to the operating facility. Manual selection or day to day rotation of the oscillators could also be available.

The remaining boards are straight forward circuits, consisting of power amplifiers for both audio and radio frequency. Couple, not, the modular combination of trays, switchable replacements and graceful degradation with an enhanced reliability factor and low level components which are of low cost an approach automatic/unattended operation becomes practical.

Power Sub Division For D.A.'s

During the initial part of this discussion a great deal of time was spent explaining the process of combining various power components so that higher power might be possible; however, in respect to a DA installation, the next immediate step would be to divide this energy into suitable increments to be fed separately to each tower element.

Consider, at this time, each Solid State power amplifier tray as a separate and distinct transmitter, whose operation is held to precise tolerance. That is, since it is a switching amplifier, each cycle of operation must precisely match at each tray. This is a true statement since this type of operation is an essential part of a solid state Class "D" amplifier.

In **Figure 11** each tray, 1 through N, is a discrete transmitter whose output is added to each of the others in the output combining transformer. Any combination of trays is possible, until such time as the size of the combining transformer becomes unwieldy. If the output transformer was a separate item for each tray; that is, several transformers having a single primary and secondary, each of the secondaries could be attached to an individual load. Therefore, it is possible to do the power division within the amplifier rather than in subsequent passive radio frequency circuits. Considerable power savings will be achieved with the power loss greatly reduced. In the conventional method, once the various power levels have been established, the correct phase angle must be added at that power level; that is, all phasor components must be able to handle a number of kilowatts.

The operation of a class D amplifier is essentially coherent. The driving signal is not a sine wave but a square wave, whose edges are nearly vertical and capable of causing the amplifier to respond in a few nanoseconds. Because of this ability, little or no phase distortion is generated within the amplifier. Therefore, whatever phase delay might be required could be introduced at the drive. This could reduce the cost of a DA by a significant amount. Those high voltage, high current components could become a suitable length of RG-59 cable or hardly more than a handfull of small components. A review of what has been changed in this future design shows that the transmitter could be divided into as many power modules as is required to drive separately each of the elements that might exist in any given DA. Earlier the capability of holding precise power levels was attributed to automation and the optical control over the modulator. Each of these amplifier-modulator combinations becomes a discrete transmitter driving its related antenna element and sharing a common driving signal, with the phase being delayed to produce a desired pattern. The high power divider the high power phasing networks have become unnecessary. If no further change is made there still remains the task of getting the energy from each transmitter to its related antenna by means of appropriate coaxial line. Generally these lines are relatively large and subjected to high voltages and pressurization leaks, as well as being costly to install and maintain. If some method could be found to eliminate these lines, another significant cost reduction could be achieved. Obviously the preceding discussion would not be necessary if a solution were not possible. In the preceding paragraphs, the power was subdivided. And the phasing accomplished at the 1 volt level.

The next step will be somewhat shocking, but within the realm of possibility, place each transmitter at its respective tower. The combination of several features, small size, ease of power control, reliability, and stability could permit such an installation. An installation were the large transmitter building and its attendant services would disappear to be replaced by a small box on the side on each tower. Possible—Maybe.

With the break through in the state-of-the-art using solid state circuits, it now allows for the use of modularity and totally automatic transmitter systems. With this innovation one can start thinking out into the future. It is envisioned that today's concept of AM Broadcast transmitter sites change entirely.

Considerable changes could occur with respect to non-directional systems with a reduction in the building facility and the complete elimination of large transmission lines. In the case of the non-directional, it appears that it will be possible to mount the transmitter in a small building at the base of the tower. With the packaging techniques that can be employed with these new transmitters, there will be no necessity for either heat or air-conditioning. Any requirement for either heat, air-conditioning, or additional space would be for convenience of the personnel performing maintenance, etc., and would not be required for proper operation of the transmitter equipment.

It would be possible to mount the transmitter on a platform in the tower if so desired, because these transmitters can be packaged for exposure to most environments. If this type of installation were adopted, then, all that would be required would be an external power source, and a small building for housing misc. equipments that would be used at the site. Coupling and matching networks could be mounted in an enclosure attached under the platform.

As can be seen even with a non-directional system there could be considerable initial ownership cost saving in the construction of a new transmitter site. These savings would be in the areas of possible reduced building requirements, reduced transmission line, reduced power equipment requirements and reduced requirement for aux equipments. Considering the above comments with respect to non-directional systems should enthuse those with requirements for directional systems and the many applications that potentially be employed using the new Westinghouse Solid State Transmitter System.

In turn, an added feature can be incorporated into this system beyond the pre-set control/compensation. This system can be designed so as to contain the features for total unattended operation the same as those for over/under power, plus or minus modulation, and carrier shift. The system would be designed using the two window approach. As the pattern shifted or varied approaching the limits of the inner window, the (program) would attempt to correct current into the antenna and phase angle to hold the pattern within those limits. If it was impossible to hold the pattern within the limits of the inner window and pattern shift or variation approached the limits of the outer window the transmitter would automatically shut down. The limits of the inner window would be those established by the Broadcaster, which should be more restricting than FCC requirements, and the limits of the outer window would be those as specified by the FCC regulations.

All of the above functions can be incorporated into the overall system at the transmitter site including the pre-set control system. Any or all of these functions can be monitored and remotely transmitted back to the studio. It is also possible to incorporate over-ride remote control for some of the operational parameters. However, it is strongly recommended that **"NO"** over-ride control be provided for any condition that demands transmitter shut down. If this is allowed then the sole purpose of the total unattended transmitter site has been defeated.

As you can see, it appears that the broadcast industry is on the threshold of a revolutionary new mode of operation which should be a tremendous boon. Not only will initial ownership costs be reduced but with increased reliability and the automatic/unattended features coupled with reduced power consumption and spares requirements, operating costs will be greatly reduced. This should allow for a much more profitable industry.

The closing comments are offered as food for thought and a goal for the future. They will be achieved, however, getting from here to there will require a lot of hard work and cooperation from all elements associated with this industry.

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Is AM Stereo In Your Future?

Introduction

We are experiencing the beginning of a new era in multichannel sound entertainment. In the recording industry, stereo long ago replaced the monophonic product, and today, quadraphonic recordings are fast becoming a new force in the marketplace. In broadcasting, FM two-channel stereo is well established as an independent medium, and means are now being explored for expanding this service into three- and four-channel transmission. More than 400 FM broadcasters are currently providing their audiences with excellent quadraphonic programming in the 4-2-4 matrix format, while the National Quadraphonic Radio Committee (NQRC) is engaged in a study of more complex methods of multichannel broadcasting.

Despite all this activity in the recording and FM-radio industries, AM broadcasting has remained essentially unchanged since the earliest days of its inception. Some interest in AM stereo was in evidence in the early 1960's when the National Stereophonic Radio Committee (NSRC) was examining potential candidate systems for both AM and FM stereo broadcasting. However, there seemed to be some feeling that the fledgling FM service should not at that time have to suffer additional competition from AM stereo, and the idea was dropped. Now, with FM and FM stereo broadcasting firmly established, AM broadcasters are beginning to ask again about the possibilities of expanding the scope of their own service.

In 1970, AM stereo transmissions were begun at station XETRA in Tijuana, Mexico. This station primarily serves the southern California area, and its efforts have attracted considerable attention. More recently, WFBR in Baltimore, Maryland, received an experimental authorization and is currently conducting broadcast tests with a system similar to that at XETRA. With this renewed interest in evidence, it is appropriate to review the historical and current proposals for AM stereo and also to consider what the future may offer.

Early History

It is typical of many of our modern day advances in communications and entertainment systems that their origins can be traced to the earliest days of the electroacoustic era. In 1881, in Paris, for example, one of the first demonstrations of binaural transmission and reproduction was held. That demonstration at the Paris Opera used two widely-spaced microphones connected to a multiplicity of binaural handsets. In 1925, a New Haven, Connecticut, radio station, WPAJ, made the first wireless stereo transmission by broadcasting on two separate AM transmitters at different frequencies. The signals for this broadcast were derived by two standard studio microphones placed seven feet apart.

In 1926, R. K. Potter of AT&T was awarded a patent[1] for a system for broadcasting binaural sound by quadrature modulation of a single radio fre-

1. *R. K. Potter, "System for Binaural Transmission of Sound," U. S. Patent No. 1,608,566, November 30, 1926.*

quency carrier. In 1933, A. D. Blumlein of EMI, Ltd., England, was awarded a patent[2] for a complete two-channel system which seemed to anticipate nearly all the elements of present day technology. Blumlein's patent contained 70 claims, including the specifications of a single-groove binaural disc employing 45°/45° modulation. His suggestion of sum and difference audio matrixing appears to be one of the earliest mentions of this technique. Although his patent does not give specific details on how to achieve stereo broadcasting, he did suggest that binaural transmission by duplex modulation of a single carrier could be accomplished by separate amplitude and frequency modulation or amplitude modulation of two quadrature carriers.

During the same period of time as Blumlein's research, Harvey Fletcher of Bell Laboratories was conducting a series of studies in auditory perspective. These tests culminated in a demonstration in 1933 of long distance transmission of stereo sound from the Academy of Music in Philadelphia to Constitution Hall in Washington, D. C. For this demonstration, Leopold Stokowsky worked the control board while his assistant conducted the Philadelphia Orchestra. To provide a sensation of dimensional sound, three microphones were placed in front of the orchestra, and the output of each was transmitted over a separate circuit. At Constitution Hall, the loudspeaker associated with each microphone was placed on the stage in a corresponding position. The event created quite an excitement among those who heard the reproduced concert. Some felt that the new system promised even greater emotional appeal than live music, but cooler heads attributed their reaction to the "enhanced volume range of the reproduced sound."[3]

It was not until two decades later that stereo broadcasting received renewed interest when, in the early 1950's a number of commercial stations began experimenting with AM/FM transmission. This method of using two stations for a single broadcast was to remain in vogue until 1961 when the Federal Communications Commission established the present rules and regulations for FM stereo. The FCC action closely followed the completion of a study begun in 1959 by the NSRC under the sponsorship of the Electronics Industries Association. The NSRC originally studied proposals for stereophonic techniques to be applied in AM, FM and Television broadcasting. In view of the existence of FCC Docket 12517 which was aimed at consideration of FM multiplex techniques, the NSRC placed first emphasis on FM systems with the intention of postponing action on the AM and Television proposals until after the FM case had been resolved. As the situation developed, according to the NSRC report,[4] "the President of the Electronic Industries Association, following a refusal by the FCC to institute a procedure which would satisfy certain requirements of the anti-trust section of the United States Department of Justice, placed the NSRC on a "stand-by basis" on February 12, 1960." This committee never did reconvene to complete its work on AM and TV systems. However, some of the original members are presently participating in the work of the NQRC.

Audio Matrixes

Although we have come to accept the use of sum and difference audio matrixes without question, the choice was not always so unanimous. In fact, around 1960, there were more than a half dozen stereo matrixes under active consideration. It is obvious, of course, that with L + R and L − R matrixes, it is possible to entirely recover the original Left and Right signals through simple additive and subtractive operations. The same can be said for other matrixes such as L + R and L, or L + R and R, although if the noise of the broadcast subchannel is equivalent to or higher than that of the main channel, a lack of noise symmetry will be apparent in the Left and Right reproduction.

Since compatibility with existing two-station AM/FM stereo broadcasts was an issue in 1960, several means were suggested to allow new FM multiplex broadcasters to continue service to their AM/FM audiences. These approaches all required compromises, either of maximum separation or compatibility of monophonic reception. One example of such a matrix employed L − R/2 on the main channel and R − L/2 on the sub-channel. More complex schemes used similar approaches, but with frequency band splitting.

A number of delay matrixes were developed by Becker and Schroeder[5] of Bell Laboratories. These matrixes took advantage of the psycho-acoustic principle that listeners determine the direction from initial sources of sound, and that delayed sounds can be presented at various locations without substantially changing the apparent location of the intended source. A simple delay matrix is shown in **Figure 1**. The left and right channels are cross-coupled through two delay lines. A delay of approximately ten-milliseconds is used—enough to permit localization of the undelayed sound, but not so long as to

2. *A. D. Blumlein, "Improvements In and Relating to Sound-Transmission, Sound-Recording and Sound-Reproducing Systems," British Patent No. 394,325, June 14, 1933.*

3. *F. K. Becker, "A Compatible Stereophonic Sound System," Bell Laboratories Record, November 1959.*

4. *Report of Panel I, "Systems Specifications," National Stereophonic Radio Committee, Vol. 1, February 29, 1960.*

5. *F. K. Becker, "Criteria for Compatible AM-FM Stereo as an Interim Method for FM Multiplex Stereo," J. Audio Eng. Soc., Vol. 10, No. 1, January 1962.*

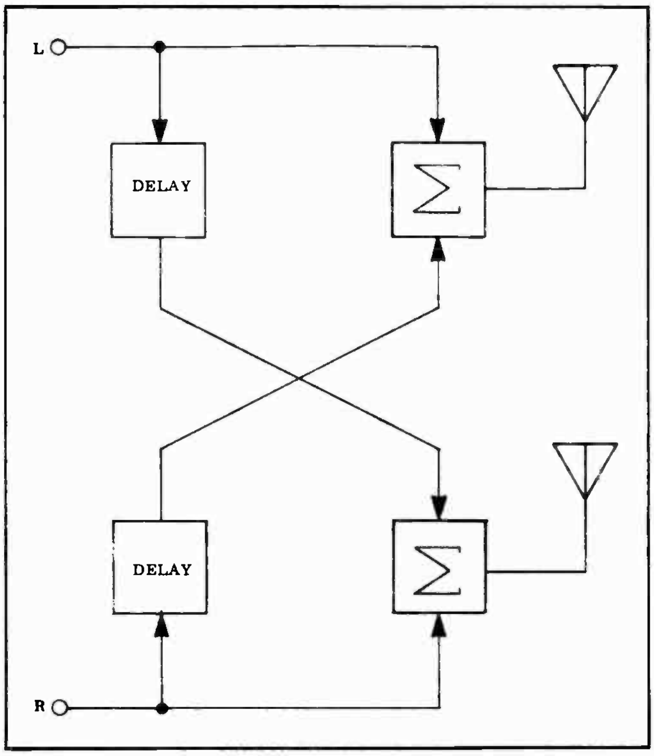

Figure 1. Delay Matrix

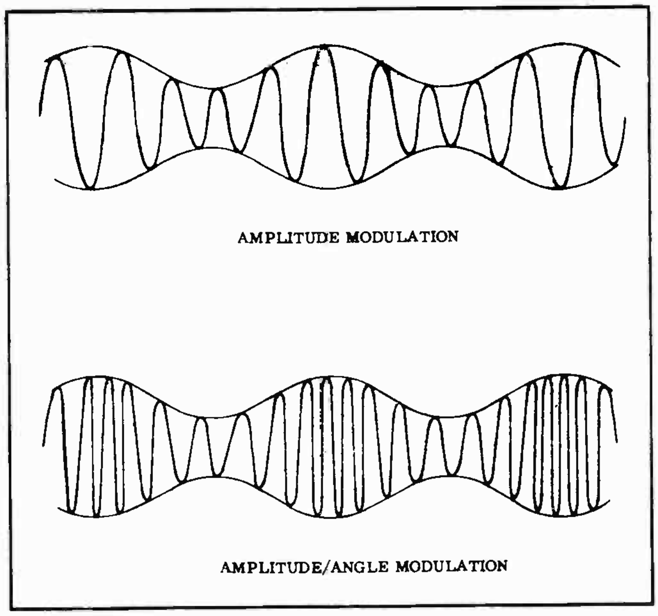

Figure 2. Modulation Waveforms

give the sensation of an echo. With this arrangement, left program signals are reproduced in one loudspeaker directly and delayed in the other. Conversely, right signals are reproduced directly in the right loudspeaker and delayed in the left. The stereo listener localizes on the direct sound only, and the listener with monophonic equipment receives all signals—or nearly so. One of the difficulties with this approach is that coherent (center) signals suffer from comb filtering effects. Other variations of delay matrixes reduced the comb filtering of coherent signals, but at the expense of adding some similar effects to the independent signals.

Another approach which received considerable attention at the time was developed by Percival[6] of EMI. Percival also took advantage of the precedence effect provided by initial transient sounds. His system used an L + R matrix on the main channel and a 100 Hz band directional signal on the subchannel. This latter "steering" signal was derived from an $\frac{L}{L + R}$ matrix and was used to dynamically control the gain of the two reproducing channels. The Percival system provided a good stereo effect because of the psychoacoustic principle that permits panning a signal to a different location after the initial transient, without changing its apparent source.

6. *W. S. Percival, "A Compressed Bandwidth Stereophonic System for Radio Transmission," Inst. of Elec. Eng. Paper No. 3152E, November 1959.*

Modulation Techniques

Except for the work of Percival, most technical approaches for AM stereo broadcasting utilize two full-band channels and involve the combined use of amplitude modulation and angle (frequency) modulation. Due to the relatively narrow band spectrum allocations for AM service, the sidebands for the amplitude and angle modulation processes must necessarily overlap. Adequate spectrum space does not exist for development of a composite signal such as is used in FM multiplex broadcasting. The various AM systems which have been proposed over the years differ primarily in the way the signals are generated—frequency modulation, phase modulation, quadrature modulation, independent sideband modulation—and in the ways signals are modified to provide greater compatibility or ease of detection.

The waveforms in **Figure 2** illustrate the nature of a basic AM stereo signal. In the upper waveform, there is only amplitude modulation. Ideally, this should be a function of the sum of left and right information, to provide maximum compatibility with the existing monophonic AM service. In the lower waveform, the amplitude modulation is maintained, with added angle modulation of the carrier by a signal which is a function of the stereo difference. In an ideal system where the transmission channel has sufficient bandwidth to accommodate all significant sidebands arising from these combined modulations, there will be no interaction between the two forms of modulation, and the original left and right signals can be perfectly reconstructed. If any part of the sidebands is lost or subjected to amplitude and phase perturbations, then the two modulations may interact with one another and produce distortion in the demodulated signals.

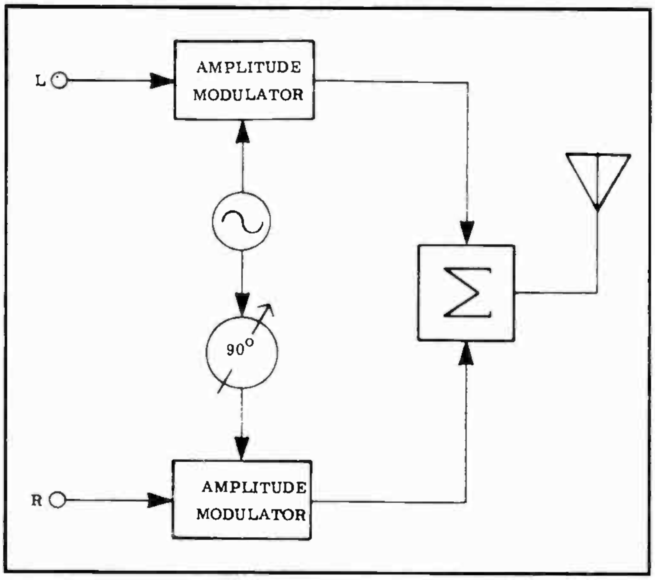

Figure 3. Quadrature Modulation

The first known AM stereo system, proposed by Potter[1] in 1926, is illustrated in **Figure 3**. A single carrier is distributed to two modulators. In one path, the original phase is maintained, in the other, the carrier is shifted by 90°. Left channel audio program modulates one carrier, and right channel program modulates the other carrier. The two carriers are then linearly combined. The resultant waveform is a single modulated carrier that is varying both in amplitude and phase, in accordance with sum and difference modulations, respectively. All sidebands fall naturally within the usual limits, but synchronous detections must be employed for stereo reception.

A more familiar format, proposed independently by Philco[7,8] and CBS Laboratories[9] in 1961, is shown in **Figure 4**. A conventional sum and difference matrix first converts the audio program to L + R and L − R. The sum signal amplitude-modulates the carrier in a conventional manner. The difference signal feeds a balanced modulator to provide a quadrature double-sideband suppressed carrier AM signal. The two resultant signals are then linearly combined, as in Potter's system.

The more direct FM-AM approach shown in **Figure 5** was originally proposed in various configurations by G.E.[4], Westinghouse[10], and RCA[11]. Shown here in simplified format, the technique first uses the audio difference signal to frequency-modulate the carrier, which is then amplitude-modulated by the sum signal.

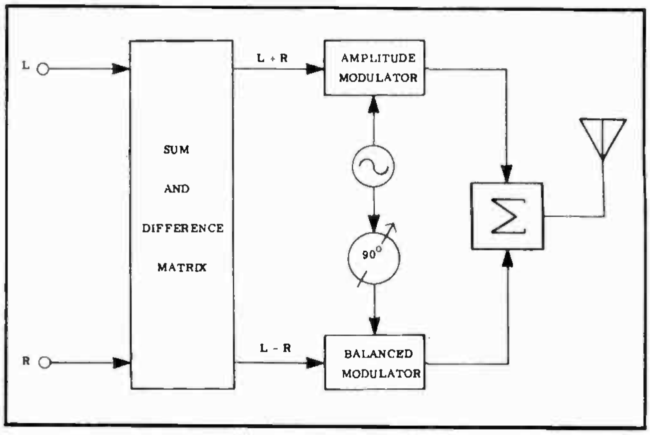

Figure 4. Supressed Carrier Quadrature Modulation

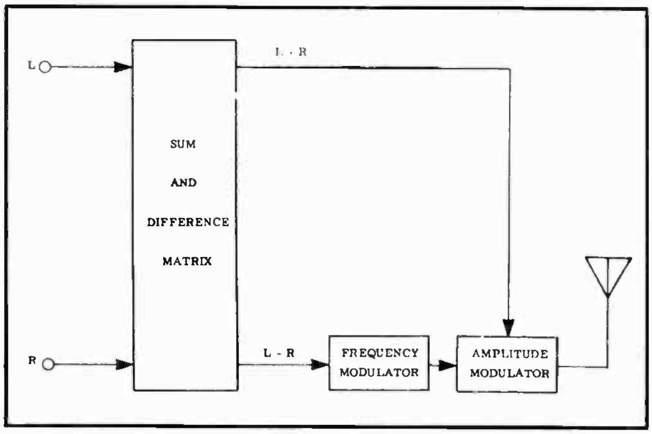

Figure 5. Amplitude/Frequency Modulation

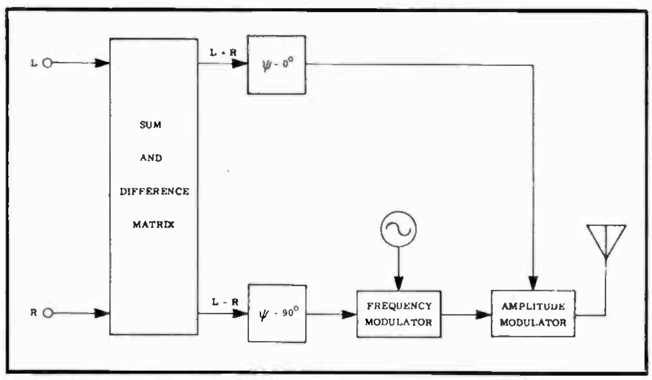

Figure 6. Independent Sideband Modulation

7. *H. B. Collins, Jr., and D. T. Webb, "Optimized Compatible AM Stereo Broadcast System," IEEE Trans. Broadcasting, Vol. PGBC-14, November 1959.*

8. *R. C. Moore, "Single Channel Stereophonic Broadcasting System," U. S. Patent No. 3,067,293, December 4, 1962.*

9. *J. M. Hollywood and M. Kronenberg, "A Stereophonic Transmission System for AM Broadcasting," J. Audio Eng. Soc., Vol. 9, No. 2, April 1961.*

10. *H. E. Sweeney and C. W. Baugh, Jr., "Multiplex Communications Systems," U. S. Patent No. 3,069,679, December 18, 1962.*

11. *J. Avins, et al, "A Compatible Stereophonic System for the AM Broadcast Band," RCA Review, Vol. 21, No. 1, March 1960.*

More recently, an approach employed by Kahn Laboratories[12] **Figure 6** achieves resultant left-channel modulation on one of the AM sidebands and right-channel modulation on the other. The modulation technique is similar to that of **Figure 5**, except that the sum and difference signals are shifted 90° with respect to each other before modulation. The independent sideband approach was first suggested by C. A. Lovell[13] at Bell Laboratories in 1941, except that the filter method of single-sideband generation was used at that time. Independent sideband signals have the interesting property of permitting simple stereo reception by slight opposite detuning of two separate receivers.

All of the above systems may be considered similar in performance at low modulation levels (up to phase deviation of approximately one radian). They differ primarily in how high level modulations are handled and, consequently, in the specifications of appropriate receivers. Angle modulation by high level difference signals introduce even-harmonic distortion into the main channel. This distortion is detectable by ordinary radios which employ envelope detection. In the past, various approaches have been used for reducing the distortion—a reduction of the difference signal amplitude, a reduction of the low-frequency response of the difference signal, or the addition of a correction signal. In the first case, reduction of the difference signal amplitude, the desired improvement in compatibility is achieved, but at the expense of a degraded signal-to-noise ratio on the difference signal or reduced left-right separation after audio dematrixing. In the second, more common, approach, reduction of the low frequency response of the difference signal, the low cutoff frequency is limited to about 300 Hz. Although the electrical separation for such a system will be poor at low frequencies, it is generally accepted that such a mixed-lows approach gives an adequate stereo effect to the listener. The removal of low-frequency information from the difference signal also removes the corresponding distortion components which would have been located at mid-frequencies. The third, and more complex, approach adds a special correction signal to reduce the effect of distortion in receivers. Since not all receivers are alike in IF bandwidth and other characteristics, such a correction signal can be ideal for only one case. For all others it is a compromise. The use of a correction signal also tends to reduce the modulation index of the main carrier and for complete correction, additional spectrum space may be required.

12. *L. Kahn, "A Stereophonic System for Amplitude-Modulated Broadcast Stations," IEEE Trans. Broadcasting, Vol. BC-17, No. 2, June 1971.*

13. *C. A. Lovell, "Stereophonic Reproduction by Carrier Wave Transmission," U. S. Patent No. 2,261,628, November 4, 1941.*

While some of the above techniques for improving capability may be more applicable to one approach than another, it is not meant to be implied that AM stereo broadcasting cannot be compatible with the existing service. Quite the contrary, the distortions cited are only second-order effects. By being able to maintain the main channel modulation at high levels, by containing sideband energy within the present spectrum allocations, AM stereo can be quite compatible, indeed.

Looking Ahead

Although there is not presently an open FCC docket on AM stereo, the recent broadcasts at XETRA have been watched with considerable interest by government and industry alike. The current tests at WFBR in Baltimore should provide an additional opportunity to study the effects of such broadcasts.

An interesting possibility exists for immediately expanding the scope of AM stereo into AM quadraphonic broadcasting. The recording industry has dramatically demonstrated that only two transmission channels are required for quadraphonic performance. The 4-2-4 matrix format, most successfully represented by the CBS SQ system[14], has been adopted by major record labels and by approximately 120 brands of playback equipment manufacturers, and SQ-encoded program is presently being broadcast by nearly 400 FM stereo broadcasters. The SQ matrix has the unique advantage of maintaining full front-stage separation in the stereophonic mode, and at the same time of providing for equal-power compatible reception of all four corner signals in the stereophonic and the monophonic mode. It is obvious that due to spectrum limitations, AM transmissions must be limited to two channels for the foreseeable future. Accordingly, the SQ matrix offers a compatible, ideal approach for AM quadraphony.

Although at first they may seem unrelated, the present studies of methods for FM quadraphonic broadcasting should be considered in the context of future AM quadraphonic broadcasting as well. The public interest will be best served by the adoption of a universal standard for all quadraphonic service. In the future a single quadraphonic decoder can be employed for all forms of home entertainment systems—AM, FM and phonograph reproduction. The public will be the major benefactor of such standardization, but broadcasters, too, will enjoy the benefits of a large, ready audience, eager to participate in this new exciting audio experience.

14. *E. L. Torick, "The SQ Broadcast System," IEEE Trans. Broadcast and Television Receivers, Vol. BTR-19, No. 4, November 1973.*

Carroll Cunningham

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Grounding, Bonding, Shielding, Isolation & Decoupling—A Primer

After consulting with some of our colleagues over on the show biz side of this industry, I began to realize how dry this subject might be. A leisurely cruise through the joke books didn't reveal anything that could be hammered into a wise crack that would even remotely relate to this subject.

I know I will be telling many of you things that you already know, that's why we call it a primer. I think the problems related to this subject matter are ubiquitous enough that some percentage of you—no doubt those currently fighting the problem—will find some of the things I have to say of interest.

So what are we talking about? We're talking about getting signals where we want them, that are contaminated as little as possible by noise and here we define noise as anything we don't want. Thus we include such things as cross-talk, oscillations and the ever present 60 cycle AC hum.

Grounding—Just What Is It?

Some place in the perfect world there is a magic spot: that spot is called perfect earth ground. I've never seen the place, nor do I expect to ever see it. It is an ideal that we technical people harbor, very much like the concept of linearity.

In the world of the broadcast engineer, coping daily with RF signals, perfect ground is indeed a long ways away. So we end up settling for something quite distant from that magic spot that we have to, some way, make as good as that mystic place on which we have to build our own little technical universe.

In the end, what we're attempting to do is to eliminate noise. Usually the noise is AC hum, though some of it is noise that we make in our own systems.

There are two basic modes of getting noise into an electronic system and for lack of better words, we shall call them the electrostatic mode and the induction mode. The most common model for electrostatic coupling is stray capacitance from your circuitry to the unwanted noise source. Magnetic or inductive coupling is just that, coupling of a signal by means of a magnetic field from one circuit to another by what amounts to transformer action.

The Faraday Shield—It Still Works

Back in the old days, when two coils were placed in proximity with the desire of coupling by magnetic means between the two, they were often separated by a Faraday Shield. The purpose of the Faraday Shield was to reduce the capacitive coupling between the two.

It was most commonly little fingers of metal inserted physically between the two coils connected at one end and carried to "ground."

Figure 1 simiply shows how stray capacitance between the coils terminates on the Faraday Shield and prevents capacitive coupling between the coils.

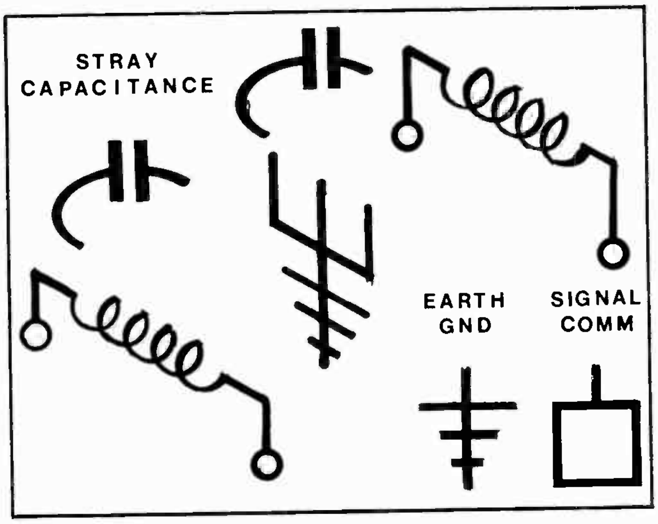

Figure 1.

The reason the Faraday Shield is built as it is—as a memory refresher—is to provide an electrostatic shield without disturbing the mutual inductance between the two coils. So the fingers at one end are open and connected at the other thus eliminating a closed loop and circulating currents and inductive effect from the Faraday Shield.

This is a good model for electrostatic shielding in an electronic system such as you would find at a broadcast station. In this case, the fingers on the Faraday Shield are composed of the shield (braid or foil) surrounding the signal conductors. They fan out through the entire system, considerably more physically complex than the simple Faraday Shield shown in Figure 1 and are brought together at a single point and carried through a single conductor to ground. This means earth ground, station ground, and power system ground.

In this paper, we will show this ground as the conventional and well established symbol for ground. Later on, we will be talking about another ground, which we will call signal ground and here we will show that ground as a simple open block.

Magnetic shielding or induction shielding is accomplished by two means. One is enclosing the circuit system or conductors in an enclosure that serves as a MAGNETIC shunt around the things enclosed. It is also accomplished by "screen-room effect" by enclosing the system in a good low impedance conductor which is also usually serving the function of an electrostatic field. The most common example of this is shielded, twisted pair. The shield on this wire serves a dual function: electrostatic shielding and induction shielding, the latter being accomplished in much the same way as a screen-room works.

You might legitimately inquire as to what it would hurt to tie both ends of the shield down, say for example in shielded cable, especially since we don't intend to be subjecting our system to heavy magnetic fields.

The answer is circulating currents induced not only by magnetic fields but induced by electrostatic means as well in a closed loop shield (and possibly a resonant one) could very well end up inducing noise into the very thing that it's attempting to shield.

Since I started out by saying that we could never get to ground, what then is the purpose of tying this shield system down to ground or the point we call ground? Actually, all we're doing is establishing an equipotential point, and then we will attempt to reference the entire system back to that point and minimize the differences in the signal circuits between their ground or signal common point and this selected point of equipotential.

Even if we had some magic point, truly 0 potential, we wouldn't have to go far to get away from it, 10 ft. at 100 mHz, that's approximately a wave length and not all that far at 1000Khz, the middle of the AM band. If you call the center of the ground screen at the base of the tower as close as you can get to true zero potential from a magic ground point, AM stations would never be able to ground there. Afterall, practically all transmitters are at least a quarter wave away from the base of their tower.

To digress a moment further, station grounds at transmitter sites should of course be single, unclosed ground busses. Even that's not totally safe, since Murphey's Law says that the length of that single ground buss or one of it's branches will be exactly resonate and then the fun really can start. This really does happen with unbelievable regularity and it is sometimes necessary to shorten or lengthen the ground buss to eliminate standing waves on it and pull it away from the resonance.

Shield Ground and Signal Common

It should be apparent now that when we speak of earth ground we speak of the point to which we bring all shields and enclosures to a common tie point, a short-circuit to the noise of the universe.

Most electronic systems operate from DC and are associated with a power supply which has a **Common** that ends up being the reference for all **Signals** in the system. This is called signal common and we'll show it as an open square block as a symbol.

Do they ever get together? And if so, why?

It might seem, since we're attempting to establish our own little universe, that we might be better off having two: one being an equipotential point called signal common, and when we use battery power supplies selfcontained in a unit, we can actually get by with this . . . especially, if we can avoid ever tying that signal common down to some other signal common which is tied back to the universe, principally that represented by the AC power system. This is done all the time of course, utilizing transformers to isolate these two references and it's commonly done also in microphones, also usually using a transformer.

Most times, however, Figure 2 applies. We have connected our system to the AC power line which in this country is required to be tied to earth ground.

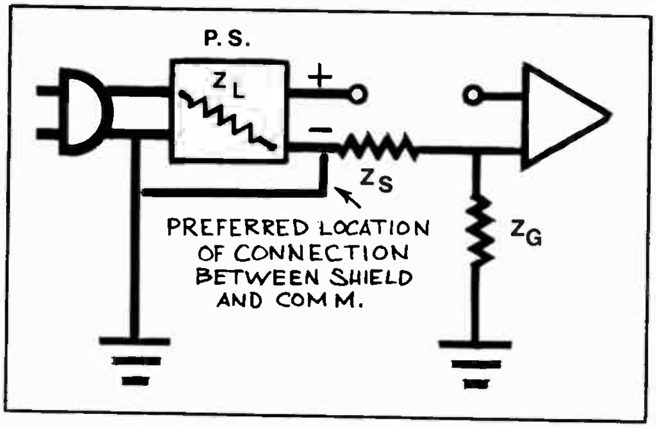

Figure 2.

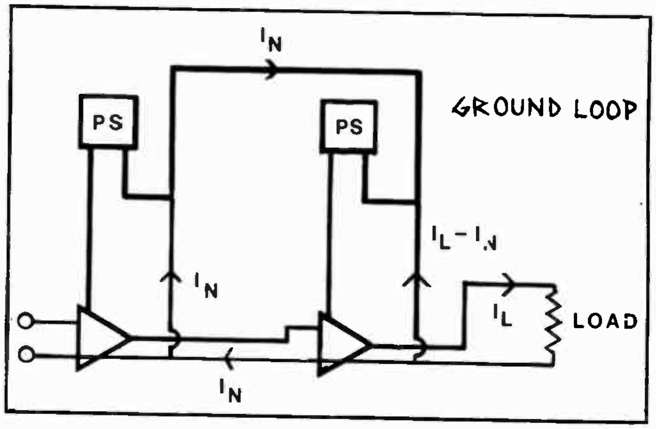

Figure 3.

The absolutely unbelievable fields, both static and magnetic existing in the average building in a broadcast station, are coupled to the common DC terminal of the output of the power supply. This is represented by the impedance shown inside the power supply of Figure 2 as Z1 which in most cases represents the coupling though the power transformer, usually capacitive, of the AC power line to the common output terminal of the power supply. (Of course it's also coupled to the plus terminal, but generally we employ a filter that essentially ties that terminal to the signal common.)

Zs in Figure 2 represents the usually very small series impedance in the signal common buss and Zg represents the usually very large impedance of the floating power supply to shield ground. We then have a voltage divider or a group of them composed of Z1, Zs, and Zg.

Even though Z1 is very large, Zg is equally large, which means that the signal common buss has substantial amounts of voltage on it. This alone introduces some common mode problems, but coupled with the additional drop through Zs (which is in the signal path) directly introduces unwanted noise into the system. Supplying the connection from the common lead, back to ground—shield ground—shorts out the voltage divider and eliminates the unwanted voltage from the signal common buss. We could have had almost the same results, particularly as far as common mode problems are concerned, by supplying the connection between shield ground and signal common at the input to the amplifier terminals. That would, however, leave us with the voltage divider of Zs and Z1, which would continue to result in an unwanted signal on the signal buss.

You say it looks like we are annihilating fleas on the rear end of an elephant with a shotgun because Zs is extremely small. That is true, it is small and it should be as small as possible. But what levels are we really looking at?

In a typical modern audio circuit, a typical output level into 600 ohms for an amplifier would, of course, be 0 dBm. And it might typically also have a gain of 60. It is entirely achievable to expect the output noise to be 70 dBm—nearly—below the output level. (That's in round numbers and staying within our pass-band of interest.) That means referred back to the input, we're looking at noise levels of 130 dBm, again in round numbers. At 600 ohms, that's approximately 1/3 of a microvolt and less than that at 150 or 75 ohms.

That isn't much! As an instructive exercise, however, assume 1 milliampere of current flowing through 1/1000 of an ohm. Guess what? That's a whole microvolt.

And a milliampere of unwanted current flowing through a milliohm of impedance the wrong way is, if we're not careful, in this day and time not altogether improbable. It's fairly common in broadcast systems to have preamplifiers near amplifiers delivering a watt or two into a low impedance of 8 or 16 ohms. We might have an ampere, or so, going into a speaker load on peaks and it just isn't that hard to have a milliampere of that get away from you and go in the wrong direction.

So we have two grounds: shield ground and signal ground or signal common, and they do, perforce, have to get together. The place for that to happen is at the power supply, and the further away the power supply is from the amplifiers it's supplying, the more important it becomes to make that connection at that place and only at that place.

Ground Loops

Most of you can see that we've been heading in the direction of discussing loops in the grounding systems. In fact, to most of us, the very words **Ground Loop** strike terror into our hearts. Nevertheless, it is amazing how few people have ever sat down and analyzed just what a ground loop is. I have seen engineers examine a printed circuit board, observing that the signal common trace is a band all along the outside edges and declare it to be a dangerous ground loop situation. It probably isn't.

Figure 3 shows a typical ground loop situation in a broadcast station. We have two pieces of equipment separately supplied by their own power supply. For example, a preamplifier driving a power amplifier.

After the current leaving the output terminal of the power amplifier has passed through the load, it

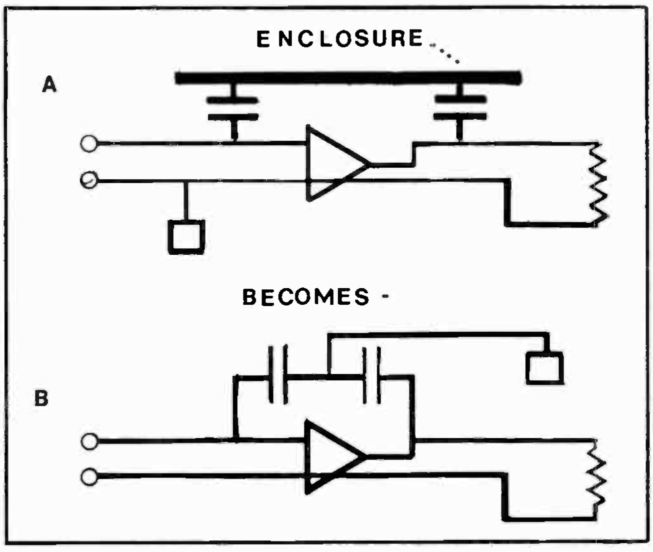

Figure 4.

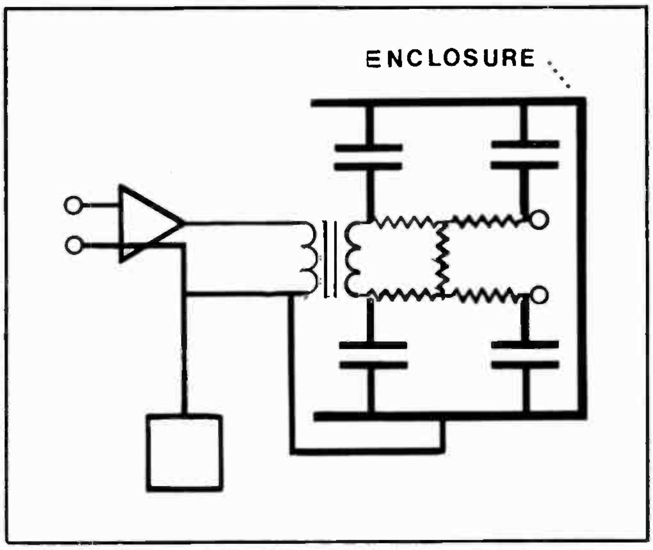

Figure 5.

returns to the low side of the power amplifier. A portion of it, I1 minus In, returns to the low side of the power supply, driving the power amplifier. Another portion, In, heads down the signal common buss to the low side of the preamplifier power supply back through the connection shown in red which is the ground loop, to join it's main component at the low side of the power amplifier power supply.

This current, In, flowing through the impedance of the signal common buss amounts to noise signal on the input to the power amplifier.

The most common means of accomplishing this in a real system is a ground loop formed by the shield connections between the two amplifiers. It would almost seem that we're damned if we do and we're damned if we don't.

It's this situation that has caused interstage transformers to become so popular around broadcast stations.

That's the most ready means of breaking up this problem. And short of rearranging the circuit, the only means. Of course, lifting the shield ground at one of the power supplies breaks up the loop, but it may do so at the expense of increasing the noise by the mechanism shown in the previous Figure 2. The one to lift, of course, would be the shield connection from the power supply of the power amplifier, noting that the common to shield connection would still be made, but partly through the signal common buss returning through the preamplifier power supply.

Enclosures and Shields

We have indicated earlier that it is necessary to enclose our system in conducting enclosures and preferably magnetic ones to eliminate the effects of magnetically induced noise. As usual, it's not as simple as it sounds.

Very often in high frequency systems the enclosure itself must bear some relationship to signal common as well. Figure 4 illustrates the problem.

Stray capacitance between the input terminals and the enclosure and the output terminals and the enclosure, provide, through that means, a path between input and output of the system, that introduces a special form of noise, oscillation. With today's active components, it is difficult to do this and still have a system that won't oscillate at some frequency. IC's have a particular fondness for the 1 to 5 mHz oscillation range.

Sooner or later you have to achieve the situation as shown in Figure 4B that breaks up the input to output coupling situation.

You may think this seems extremely obvious, but that's far from the case. One of the leading manufacturers of audio components devised a system shown in Figure 5. At first look, he did everything just right. He transformer balanced the output of his system and inserted a very large pad, also balanced. The enclosure was tied to signal common.

We had a devil of a time trying to find out why a 40 dB pad in that system had a rising frequency response as you got to 20 and 30 kHz. As little as it was, the stray capacitance between the balanced pad and it's enclosure provided another signal path that bled off a large number of the supposed dBs inserted by the pad.

If you needed the system balanced, as we did, it was rather tough to devise a solution to this problem. To this day, I don't feel there are any elegant ones.

All of this serves to illustrate another important point, particularly when dealing with shielded wire, as most of us do. Occasionally a shield can break or for some reason or another, not get tied down to the shield ground, which is as we have already learned, ultimately becomes signal common ground also. If this happens when the input shields and output shields to a

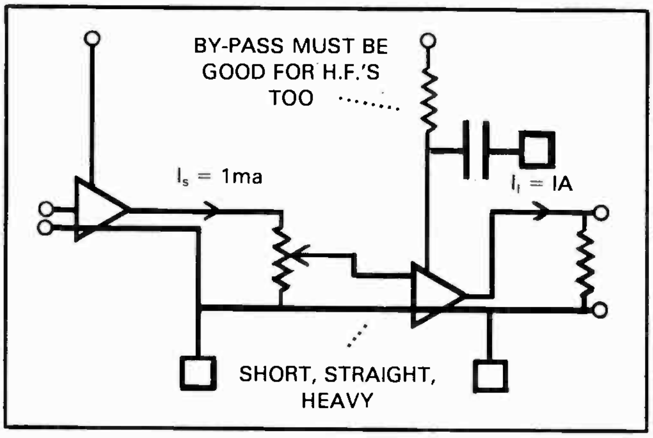

Figure 6.

high gain amplifier are tied together, you end up with a very busy and hot little oscillator.

This leads us to a good standard practice relative to shield ground. A good place to break continuity in a single signal path, insofar as shield is concerned, is at the first high gain amplifier. In other words, input shields should be carried off to the ground point independently for the output shields, continuing to bear in mind we have a very complex and involuted Faraday Shield inside of our system composed of the shields going to their respective origins.

Grounding

I feel sure that most of us are familiar with the most common decoupling problem encountered in electronics, namely "motor boating." It serves us in good stead for the neat tricks encountered with modern semi-conductor devices. Coupling between various parts of the system inputs to outputs, Channel A to Channel B, etc. through the power supply with semi-conductor electronics is far more common. Not only do they motor boat, they oscillate and they cross-talk like mad.

Power supply decoupling techniques employed in the old days, and considered standard, are still standard . . . they just aren't enough. You not only have to decouple the amplifier with massive electrolytic type filtering, but you also need to employ good high frequency decoupling by means of capacitors with low impedance at HF and VHF. Oscillations on the power supply buss, when modulated with audio, have an uncanny ability to make audio in some other channel far removed from the excited one.

The bypass capacitor shown in Figure 5 to decouple the power supply from the amplifier is, most generally these days, two capacitors: a large electrolytic and a high grade dipped mica or the equivalent.

The more difficult part of the decoupling involves the extremely fine art of controlling the paths of signal currents. In the example shown in Figure 6, once again, I have assumed a load current of 1 ampere, which is not extravagant, and a signal current of 1/1000 of that, one milliampere. The load current after it has passed the load returns to the signal common buss. The art here is encouraging that load current to go on back to the power supply and not drift off into alternate routes on the signal common buss, thus causing unwanted input signals to various other amplifiers in the system. For example, if it was a stereo system, keeping the righthand load current off of the input common of the lefthand preamplifier.

Inside of a system that you control, for example an audio console, a number of techniques can be employed. Most involve care in the physical configuration and wire routing schemes. Supply power to the system near the heaviest load. Arrange component locations so that load currents cannot pass along the common buss of preceding stages. Currents from one channel are isolated from others and from preceding stages that they may share.

In any event, every means possible to reduce the series impedance of the common buss should be employed by making it as short, heavy and straight as possible.

Where diverse pieces of equipment are interconnected, be sure to provide complete current paths for all sources and loads, thus minimizing the possibilities of extraneous currents flowing in unwanted places. In other words, don't try to scrimp on wire. Carry high sides and low sides together at all times and fully to the destination, whether it be a source to an input or an output to a load. It is also good practice to hold all impedance as low as possible and keep everything terminated. Most cross-coupling modes are high impedance and the lower the signal impedance buss the lower the noise coupled to it.

Once again, often in broadcast systems it's necessary to provide decoupling in the form of transformers, amplifiers or other isolation devices, for example, a pad, to eliminate unwanted signals on other parts of the system.

Radio Frequency Interference

I don't know of a single broadcast station that has not had radio frequency interference problems from their own RF sources at one time or another.

Semi-conductor devices are uniquely susceptible to Radio Frequency Interference. Please remember that every transistor is secretly two diodes and straight detection of unwanted RF signals is one of the most usual means for Radio Frequency Interference.

Slope detection is another.

Intermodulation mechanisms and common mode amplifier limitations are the other means of observing RF interference. It is, to me, absolutely astonishing, particularly with FM signals, how many places reasonably clear sounding audio comes out, while going in as frequency modulated RF.

I have to admit that most RFI problems are pure misery. There is so much RF and the devices are so sensitive and susceptible to it.

Of course, standard techniques and exceptional care uniformly cure the problem.

It's necessary in RF fields to install your system in conducting enclosures which, for the device or system, amounts to RF ground.

It's an extremely common failure to couple RF into the enclosure through the power line. Power lines and all signal lines input and output, should be bypassed to the case of the device with as short a lead as is possible. Again, remember that capacitors are not all the same. Some capacitors, particularly the ceramic ones have substantial impedance at RF frequencies, thus the bypass capacitor employed must be a high Q device at the frequencies of interest, be it AM band, FM band, or TV band.

It's often necessary to install small series RF chokes ahead of the capacitive bypass. Here again, and for the same reasons, keep signals terminated and at low impedance.

These techniques are usually sufficient and it leaves us with the remaining problem of common mode limitations of the device. In other words, the entire system, signal and signal common, are both bouncing around at some high frequency level relative to the universe and today's semi-conductors probably have some response at that frequency. If the differential response to the signal **High** and the signal **Low** is not the same, the resulting common mode signal amounts to interference in unwanted signal noise.

Your only defense against this is to limit the response of the system as severely and as often as you can afford to before this common mode signal gets detected or intermodulated and results in something that **can't** be controlled. This means that between every stage for every component of the system get rid of as much RF from the signal as you can afford to get rid of without affecting the signal itself, giving you, at the next stage, something of a fresh start.

Summary

Modern semi-conductor devices have opened up at least one order of magnitude, greater dynamic range and signal to noise ratio. Not only is this an opportunity that broadcast engineers should accept, thereby improving the performance of their stations and the satisfaction of their audience, but it makes previously unnoticed problems painfully apparent.

In addition, most broadcast systems have not been **Planned** to achieve minimum intrusion of unwanted signals and noise, therefore, as the system is maintained and updated, systematic procedures and techniques should be employed to take advantage of modern day devices.

An understanding of the truly simple nature of noise pick up aids in achieving this goal.

John P. Jenkins

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NAB Cartridge Tape Recording and
Reproducing Standards Committee

Review of New NAB Cartridge Tape Standards Activities

In 1958 the first cartridge machines were originated. In the early years during which time this cartridge equipment was coming into common use, there were as many as thirteen manufacturers at one time building this type of equipment. It quickly became evident that standards were required since various manufacturers were going various ways in the basic methods by which this equipment operated, both mechanically and electrically. During 1962 and 1963 the NAB organized a standards committee with the express purpose of establishing standards for cartridge tape recording. This work was completed and issued in October of 1964. Since 1964 there have been many changes, both in the cartridges and in the machines. There are various international committees who currently have these standards under study and have found that they are inadequate. The international requirements as well as domestic requirements clearly indicated that these standards needed to be revised. Therefore, in June of 1974, a new committee was formed, and work began.

The specific objectives which have been established by this committee are as follows:

1. To review the October 1964 NAB Cartridge Tape Recording and Reproducing Standards and make whatever changes are deemed necessary to reflect the present state-of-the-art.
2. Whatever revisions in the standards that are deemed to be necessary must be made in harmony with existing equipment and operational practices.
3. That a standing committee be established, charged with reviewing the Standards periodially to keep them current. It will be an ongoing committee that will meet perhaps every two years.

The committee also stated that the NAB Cartridge Tape Recording and Reproducing Standards reflect good engineering practices and that they be for the guidance of the broadcasting industry. When completed, these standards will represent the contributions of experts throughout the world.

People from various countries have been invited and are participating in this effort. Unlike the initial writing of cartridge standards, there are many broadcasters serving on the committee, as well as manufacturers of tape, tape heads, cartridges, motors, and machines. The committee represents a very high level of technical expertise and is in the process of achieving standards for the broadcast industry.

The primary areas which are currently being studied are:

1. Cue Tones and Tolerance
2. Consideration of a 4th Cue Tone
3. Need for dimensions of cartridge height, length, and shape
4. Compatible mono/stereo cartridge systems
5. Interface standard between stereo and mono
6. Flux level standard 160 nWb/m
7. Tape Speed; consideration of a second speed; high speed wind—fast forward

8. Tape Width—.246 ± .002 to .248 + 0.000—0.002
9. Tape Pulling Force
10. Method of measuring flutter plus tolerance—DIN
11. Consideration of frequency response phase differential, gain, and crosstalk
12. Mono/stereo track compability
13. Quadraphonic Requirements
14. Practical vs Scientific considerations and measurements
15. Reference Cartridges—mono/stereo format. Currently there are no reference cartridges for stereo.
16. Life, environment, and stability of tape cartridges and machines
17. International Consideration—IEC (International Electro-Technical Commission) Standards Format
18. Erase/Delay Provisions

Due to time limitations, it is not possible that I cover the new standards in their entirety. So I will address myself to major areas which have been changed.

A major change for those people from the United States serving on the committee has been that we are using metric dimensions as well as Imperial. The format of the basic document will be more in keeping with International writing. Fortunately, we have not provided this written in Arabian, French, Nigerian, or even the King's English.

It has been difficult to rework these standards, not because of trying to reach a basic standard. The basic intent of the committee has many times been clearly identified, but the difficulty arises when we want to establish a method of measurement or a unit or measurement. As an example, tape speed is very difficult to measure. One means of establishing tape speed is to measure 150 feet of tape and then play this measured length of tape in a machine. However, it is argued that the method of measurement can be inaccurate because of temperature and humidity changing the length of tape. Tape speed has not been formally adopted at this time. However, either a .2% speed accuracy or .4% is under consideration. Careful consideration was given to a second tape speed of 15 IPS. By increasing tape speed, there is an improvement in response, signal to noise, and considerable improvement in phase differential. Very limited lab and field tests have been undergone at this time, and the committee has elected to hold action on the 15 IPS tape speed until our first two-year review. It has been generally agreed that the 7½ inch tape speed of most existing cartridge machines are running .4% fast. The new standards will provide a means of measurement which will cause future machines to be built to run at an absolute 7½ IPS. The effects of this will be that if you were to play a 10-minute cartridge, it will play back 2.4 seconds slow. This minor speed variation will in all likelihood not be noticeable.

Track Dimension and Format, especially as it relates to stereo, has consumed a considerable amount of time. There have been five stereo track formats proposed, all of which offer various advantages and disadvantages. In order that the committee could best serve the broadcast industry, a questionnaire was prepared and mailed to 1,000 broadcasters. This questionnaire was primarily aimed at gaining information concerning stereo/mono capability. The five proposed systems are:

1. Use the three-track format that is currently in use.
2. Crowd to narrow tracks within the .080″ track width used in mono machines.
3. Change the mono track format so that we would have a very wide program track which would cover the two tracks currently in use in the stereo format.
4. Matrix left + right, left — right on the two existing channels with the input-output matrix circuit.
5. Use a 4-track configuration with left channel, right channel, a left + right, and a cue track.

The committee had a strong desire that a new track configuration be adopted which would provide compatibility between mono and stereo machines. The system which showed the most promise was putting two tracks within the .080″ track used on the mono machines. This provided a theoretical two-times phase differential improvement. The effects of this change if it becomes a standard in the future would be that all stereo cartridges would have to be re-recorded and all record and play heads would have to be changed on stereo machines in order to accomplish a full standardization of a new track format.

However, there are various advantages and disadvantages to each of the systems. The committee has given considerable time, test, and study to each of the above systems, and it has been decided that we will stay with the current three-track system and do more extensive laboratory and field tests with some of the other configurations. This will be brought under review at the next two year meeting. We hope to combine a new track configuration with the 15 IPS tape speed, with a possible result of a 6-times improvement in phase differential.

A major area of change which has been attempted by other domestic and international committees, but to date has not been formally adopted, relates to tape width. The NAB has formally adopted changing the old tape width of .246 + .002 to .248 + 0.000 — 0.002 inches. Tape manufacturers present on the committee indicated that they were already holding this new width tolerance. The effects of reducing the allowable tolerance in tape slitting should reduce tape skew and improve phase differential.

A change which will affect all broadcasters relates to low end equalization. Historically the NAB has used a low end time constant of 3180 micro-seconds. The committee has formally adopted new standards which will drop the 3180 micro-seconds time constant, with the low end continuing on a straight 6 dB per octave contour. The committee elected to drop the 3180 micro-seconds contour in that with most modern tapes we will approach tape saturation at the low end. The effects of this will be that cartridges which have been recorded with the old system and played in the new will have an approximate 3 dB boost at 50 cycles.

The reproduce and record frequency response has changed. The major changes relate to the fact that we no longer are referencing to 400 Hz, and that we have allowed more deviation at the low end to allow for variations caused by head shielding effect. As most of you are aware, there are humps and bumps in the response at the low end, which is caused by the shielding on the tape cartridge machine playback heads. In a cartridge machine, it is not possible to properly shield the head as it is in a reel machine since the cartridge itself occupies the space immediately in front of the head. Therefore, the following limits have been established. Reproduce limits shall be within a 2 dB window from 300 Hz to 10 kHz; 3 dB window from 150 Hz to 299 Hz; a 5 dB window from 50 Hz to 149 Hz, opening up from 2 dB to 3 dB between 10 and 15 kHz. The upper limit of the window is to be flat from 20 Hz to 20 kHz. Regarding the record limits, when reproduced through a standard cartridge equalized playback channel, the recorded tape will match an NAB test tape within a 2 dB window from 50 Hz to 10 kHz and opening up from 2 to 3 dB between 10 and 15 kHz. The upper limit of the window is to be flat from 20 Hz to 20 kHz.

You will note that the record play response is not as restrictive as it has been in the past. However, it does more nearly reflect actual capabilities of the cartridge system.

Many of you are probably familiar with the expression "nanoWeber per meter". But what does it really mean? In the past the old planet Earth has been spinning around the sun with at least three different means of measuring electro-magnetism as it relates to tape recording: the terms Weber, Maxwell, and Gauss. In order that a single unit of measurement can be standardized worldwide, various international committees in the past have adopted the unit "nanoWeber per meter". The term "Weber" is the magnetic flux which, linking a circuit of one turn, produces in it an electro-motive force of one volt as it is reduced to zero at a uniform rate. Fluxivity is the proposed name for the tape flux unit track width, which is the quantity for the magnitude of the recorded signal, and the unit is Weber's per meter of track width.

In the past, distortion standards have been established only on the overall system. The new standards have established distortion limits on the actual playback and recording amplifier as follows. Regarding the total harmonic distortion, the reproduce limit is 0.5% maximum total rms distortion with a signal generator feeding the input of the reproduce amplifier, adjusting the input level for a constant +18 dBm output level (minimum capability) of the amplifier from 50 Hz to 15 kHz. On the record limits, 0.5% maximum total rms distortion at 1 kHz with a level 18 dB above that required to record 160 nWb/m flux level on a tape of the same characteristic used for the NAB test tape.

In the system distortion, when recording 1 kHz tone with peak bias on any lubricated tape at a level at which the rms 3rd harmonic distortion is 3%, the distortion shall result from the recording process, and not from the recording or reproducing amplifiers. When recording 1 kHz tone with a 160 nWb/m level, the system distortion shall not exceed 2.0%. The effects of the distortion will be limited by tape rather than by the machine.

Note that in order for a machine to meet this standard, it must be capable of +18 dBm output.

In the old standards, the operating reference level was derived from a 3% distortion point. It was intended with the 1964 standards that all recordings would be at that operating level. In our new standards, it has been formally adopted that we will have a flux reference level of 160 nWb/m at 1 kHz, and that the actual operating level will not be specified. The reason for this is that the broadcasters may use a VU meter, PPM meter, or various types of measurements which will require different amounts of head room. Furthermore, the broadcasters may elect to record certain types of program material at a different level such as a voice only recording may be at a higher level than would be a classical music tape. This also allows broadcasters to make use of tapes which may not even yet be available and which will support a higher operating level.

A major area of addition to the standards has to do with the amplifier minimum gain requirements and interface impedance. Both the reproducer and the record amplifier have been standardized. A reproduce limit of 10 dB minimum above that required for 0 dBm output level into the rated load with a recorded tape flux level of 160 nWb/m at 1 kHz; a record limit of —32 dBm matching, or —8 dBm bridging input level for record level capability of 160 nWb/m at 1 kHz, with the gain control at or near minimum attenuation. The amplifier must be capable of accepting a bridging input level of up to +14 dBm with the gain adjusted for normal recording level, without exceeding the harmonic distortion limit.

Never before has the NAB elected to establish standards on interface impedance. However, we have formally adopted the following: Cartridge machines shall provide 600 and 150 ohm load impedance. The reproduce amplifier output impedance shall be 0.125 (maximum) times the rated load impedance over the frequency range from 50 to 15 kHz. The minimum required source impedances for cartridge machines shall be 600 and 150 ohms. The record input shall be 8 (minimum) times the rated source impedance for matching, plus 10K ohm minimum bridging over the frequency range from 50 Hz to 15 kHz. The input/output terminals shall be floating, (ungrounded) capable of being connected to sources/loads with either side or the centre-tap grounded while meeting all other specifications. All machines meeting the new standards will be capable of providing 600 and 150 ohm output at +18 dBm. All recording amplifiers will be capable of accepting 600 or 150 ohm input and 10K bridging input. The most serious effect of this standard will be the +18 output level. If, as an example, the broadcaster only requires a —20 to feed his console, it will be necessary that he insert a pad in the output of his machine or accept the loss of signal/noise that may occur if he simply turns the level control down.

Stereo phase standards have not yet been formally adopted. There have been various proposals, and there will be a standard established and made a part of the formal standards.

There have been slight changes in the standard concerning the primary, secondary, and tertiary cue tones, both as it relates to frequency tolerance and flux level standards. Basically, we will continue to use the 1 kHz, the 150, and 8 kHz tones. A fourth and fifth cue tone configuration has been considered. We have formally adopted a fourth cue tone signal and have called it a logging signal. The committee has not adopted a fifth cue tone, although considerable study was expended concerning this matter. A change from the 1964 standards is that we are not specifying duration of secondary and tertiary cue tones. Regarding the cue tone frequencies and tolerances, the following standards apply: Primary Cue — 1 kHz, ± 50 Hz; Secondary Cue — 150 Hz, ± 8 Hz; Tertiary Cue — 8 kHz, ±400 Hz; Logging Signal — 3 kHz, ± 150 Hz for single tone On/Off method, FSK logging tones to be within maximum window of 2.8 to 3.2 kHz.

Regarding the cue sensor passband and tolerances, the following standards will apply: Primary Cue —Minimum passband from 900 Hz to 1100 Hz, minimum passband from 800 Hz to 1200 Hz; Secondary Cue — Minimum passband from 135 Hz to 165 Hz, maximum passband from 120 Hz to 180 Hz; Tertiary Cue—Minimum passband from 7200 Hz to 8800 Hz, maximum passband from 6400 Hz to 9600 Hz; Logging Signal—Minimum passband from 2800 Hz to 3200 Hz, maximum passband from 2600 Hz to 3400 Hz.

The cue tone recorded flux level shall be as follows: Primary Cue tone—160 nWb/m nominal, 180 nWb/m maximum, 120 nWb/m minimum; Secondary Cue tone—360 nWb/m nominal, 400 nWb/m, maximum, 250 nWb/m minimum; Tertiary Cue tone—20 nWb/m nominal, 22 nWb/m maximum, 14 nWb/m minimum; Logging Signal—40 nWb/m nominal, 45 nWb/m maximum, 28 nWb/m minimum; Cue tone distortion—5.0% maximum total rms distortion as recorded on the tape at the frequencies and levels mentioned above.

When these tones are reproduced on a unit having the standard reproducing characteristic, the corresponding relative output levels are as follows: Primary Cue Tone 0 dB; Secondary Cue Tone —+6 dB; Tertiary Cue Tone — —10 dB; Logging Signal — —10 dB.

Regarding the cue tone burst duration: Primary Cue — 500 ± 250 milliseconds; Secondary Cue — 100 ms minimum (15 cycles at 150 Hz), no maximum duration to be specified; Tertiary Cue — 2 ms minimum (16 cycles at 8 kHz), no maximum duration to be specified; Logging Signal — 4 ms minimum (12 cycles at 3 Hz), no maximum duration to be specified. With the exception of the fourth tone or logging tone, the machines built to the old or the new standards should be totally compatible.

The 1964 standards have been totally rewritten as they relate to the cartridge and the mechanics of the machine. One example of machine change from the original standards is the distance between the playback and recording heads. The old standards were set at 1¼ inches, and very few machines were built with a 1¼ inch spacing between heads. The new standards call for 1⅛ inches between heads. This more nearly conforms to the spacing which has typically been in use.

There has been numerous changes both internally and externally in the cartridge dimensions. These changes involve several mechanical drawings and cannot be properly presented in this brief report. Cartridges built to the new standards will work in the old machines. However, if a machine manufacturer takes advantage of larger head window openings in the front of the new standard cartridges and places tape guides in this larger window, the old cartridges would not necessarily fit into a new machine.

During the past year, the committee has been aggressively pursuing the writing of new standards. An immense amount of time and study has been expended. During this time, we have never lost sight of the fact that this work is being done to improve the tape cartridge system for all broadcasters. Work is nearing completion and the new standards should be presented in the very near future.

Circular Polarization

A three-part presentation. . .

- Its history and theoretical advantages. . .
- Other considerations and special requirements for the design and testing of transmitting and receiving antenna. . .
- Results of field tests with experimental equipment. . .

And a question and answer session.

Fred L. Zellner, Jr.

Manager of Allocations
ABC
New York, N.Y.

Circular Polarization in Television Broadcasting

Since the very beginning of television, we have all been plagued with problems in getting our signal to leave the transmitter and arrive at the viewer's TV set intact. No matter how high we build our towers, we can't seem to build them high enough to overcome the tall buildings, mountains, electric toothbrushes and frost-free refrigerators that are waiting to add their "extra touch" to our signals.

The situation is full of irony for the broadcaster. Within a metropolitan area, countless viewers are unable to get a good picture from us, even though plenty of signal strength is available. The tall buildings cause severe ghosting and multipath interference.

Farther away from the city, co-channel interference can make viewing impossible. And what about the viewers located behind a hill, where all they get is reflections?

To top it all off, we have to realize that many of our viewers are using sets with rabbit ears or whip antennas, which they may or may not even bother to adjust very carefully. At best, these antennas are not really suited to the horizontal polarization we use today; the only reason they work at all is that we transmit with such high power that the signal loss they cause is tolerable.

We now have an opportunity to provide an antidote to many of our problems. Instead of radiating our signals by using *horizontal* polarization, as we do today, we can use *circular* polarization.

Circular polarization can provide substantial improvement for a number of common TV problems:

- Ghosting
- Spotty coverage
- Multipath interference
- Poor reception on whips and rabbit ears
- Misoriented antennas
- Co-channel and adjacent-channel interference

We will look at these in detail later on.

The idea of circular polarization is not new to ABC. It was proposed some 30 years ago for aircraft communication, as a means of providing more consistent transmission and reception between the ground and moving airplanes.

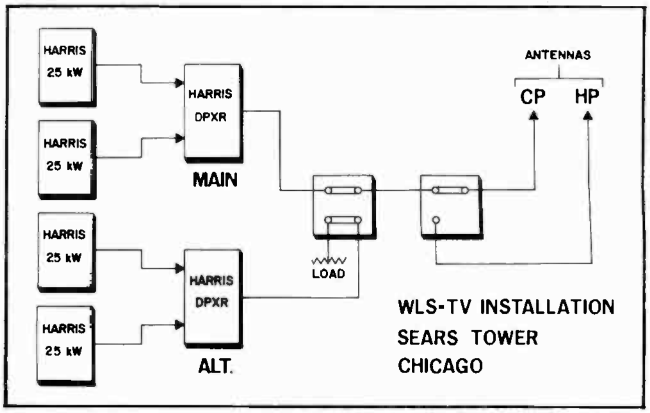
FM broadcasters also have made use of circular polarization for about 10 years to provide a usable signal for their listeners riding in automobiles or using portable sets with whip antennas.

If circular polarization works well for radio, why shouldn't it also work for television? The ABC engineering department asked that question eight years ago, and preliminary investigation of the principle led us to a series of experiments in 1968 which are known as the "Altoona tests."

A special authorization was obtained from the FCC to erect a circularly polarized antenna as an auxiliary for the Channel 10 station in Altoona, Pa., in order to measure the performance of circular signals. The terrain in Altoona was ideal for our tests, and the results were so encouraging that we decided it was time for a large-scale demonstration of the principle.

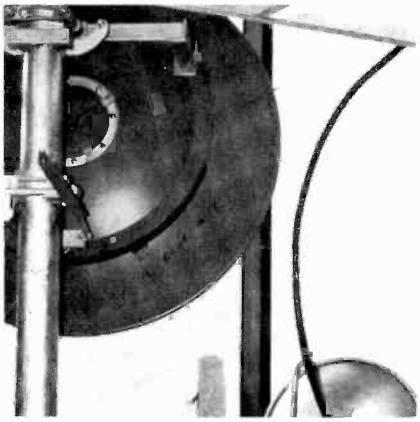
At this point, the American Broadcasting Companies approached RCA and worked out the details of the design, construction, and installation of a circularly polarized antenna. We chose our Chicago station, WLS-TV on Channel 7, as the location for the field testing, and ABC obtained special authorization from the FCC.

We have been operating for about one year using circular polarization from our new location on the Sears Tower. The installation includes both a circular antenna and a conventional horizontal antenna, which allows us to switch back and forth for comparative purposes.

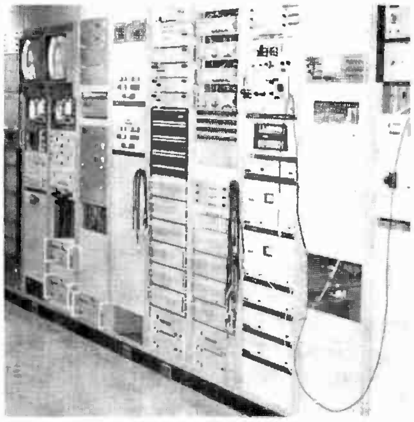


Block Diagram of WLS-TV Installation.

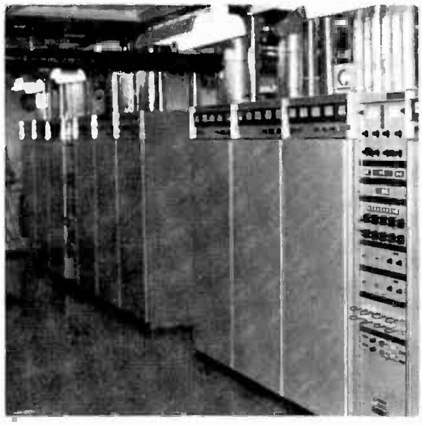
The WLS Installation in Chicago . . .



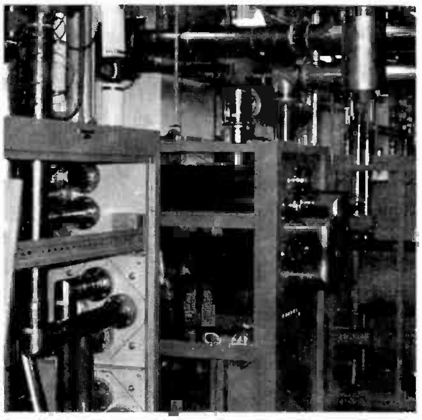
WLS-TV Microwave Dishes atop the Sears Tower.



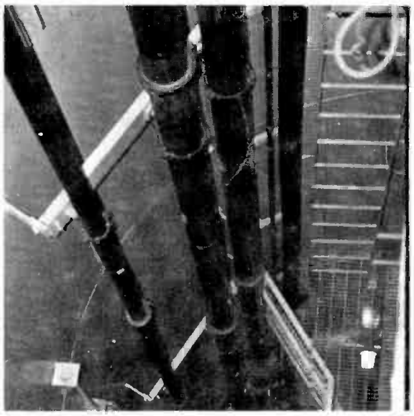
Input and Monitoring Equipment.



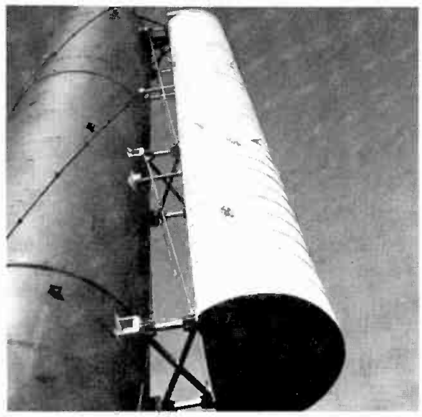
WLS-TV Transmitter.



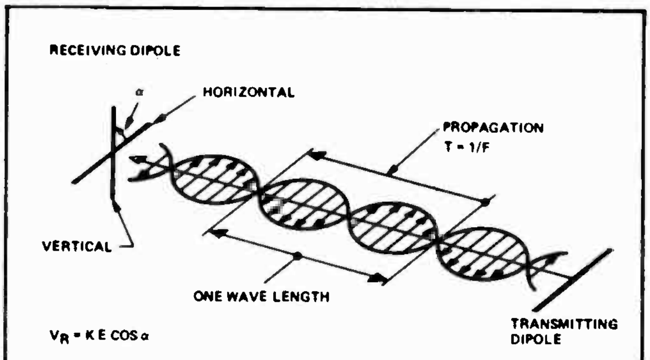
Output and Combining Equipment.



Transmission Lines in Steel Supporting Cylinder.

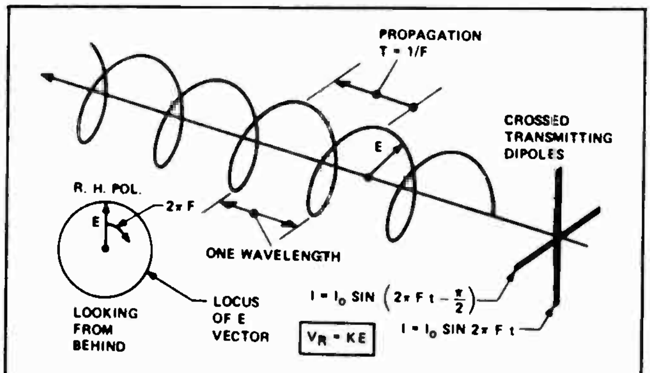


Horizontal and Circular Antennas, in Radome.



Horizontally polarized signal.

Figure 1.



Circularly polarized signal.

Figure 2.

The performance measurements which our engineers have made are very satisfying. Not only has the circular-transmission technique worked just as well as we expected, but no *unexpected* side effects have appeared.

Our tests have not progressed far enough to allow us to quantify the amount of improvement that circular polarization brings, but we can demonstrate that quite a bit of improvement is involved.

Circular polarization for television has come of age.

Let's take a look at it and compare it with horizontal polarization, so we can see where the advantages come from.

Figure 1 is a schematic representation of *horizontal* polarization used in television today. In the lower right-hand corner you see our transmitting dipole which is lying horizontally in space. If we impress an RF current on this dipole, it produces a radiated field. Because this field-voltage vector lies entirely *within* the horizontal plane, we call this horizontal polarization. Note that the field goes through zero every half-cycle.

On the left side we have another dipole which we are going to use as a receiving antenna. In order to get the maximum voltage induced across this dipole, we are required to meet two conditions: First, the receiving dipole must be facing *broadside* to the transmitting dipole; in other words, it must be perpendicular to the direction of propagation. Second, the receiving dipole must lie in the horizontal plane; in other words, it must have the correct *azimuth*, because if we depart from this plane, the voltage produced across the antenna decreases.

We can illustrate this last point mathematically by referring to the equation in the lower left corner of the Figure:

$$V_R = K E \cos\alpha$$

If alpha equals zero, which means our antenna is perfectly horizontal, the cosine is 1, so that we get a voltage maximum in this position. As we depart from the horizontal, however, and alpha increases, the cosine of alpha approaches zero, so that when the antenna is vertical, and alpha equals 90°, the cosine has gone to zero, and we get no signal whatsoever.

Figure 2 is a diagram of a *circularly* polarized arrangement. Instead of using just one transmitting dipole, we take two dipoles, mount them at right angles, and feed them in phase quadrature. Now, instead of producing a field that vibrates within a plane, we produce one that rotates around the axis of propagation.

This arrangement has a number of interesting properties. First of all, the field never goes through zero, as it does with horizontal polarization. The magnitude of the field-voltage vector, E, remains constant.

If we slice through the axis of propagation in a plane which is parallel to the plane formed by the crossed transmitting antennas, we get a picture like the one in the lower left corner of the Figure. The field-voltage vector, E, rotates around the axis with an angular velocity of 2 π F. The rotation may be clockwise or counter-clockwise, depending upon how we feed the transmitting antennas, and we refer to this as either right-hand or left-hand polarization.

We can see from this that if we place a dipole receiving antenna in this field, we no longer have to meet the *azimuth* requirement. We need only to make sure that the antenna is *broadside* to the transmitting antenna. For any angle with the horizontal, the antenna receives a constant signal.

You will notice that the equation for received voltage, which is in the box at the bottom of the Figure, no longer includes the "cosine alpha" term. Thus, we have already made it simpler for receiving sets with rabbit ears and whip antennas to receive the strongest signal from us.

The truly interesting results come when we equip the *receiver,* as well as the transmitter, with a circular antenna. We will look at those results in a moment, but for now, remember this: A circular receiving antenna is sensitive *only* to one sense of rotation. In other words, a right-hand circular receiving antenna will *reject* left-hand signals.

Let's investigate what will happen when we install a circularly polarized transmitting antenna at a television station. As we saw earlier, a circular signal will induce the same voltage in a linear receiving antenna as will a horizontal signal (for the same field strength).

Figure 3.

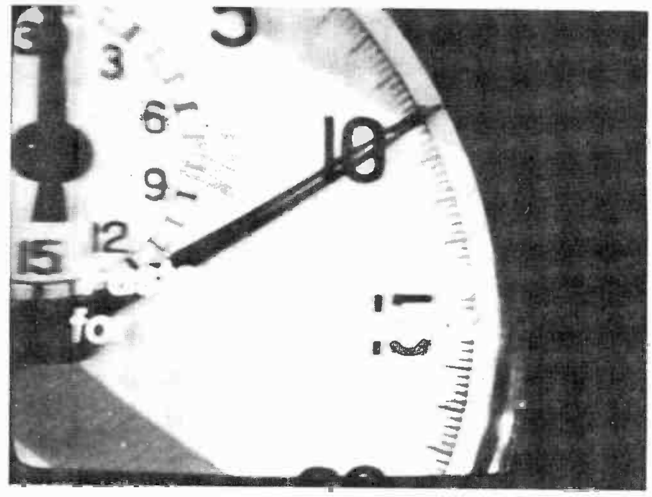


Figure 4.

Figure 5.

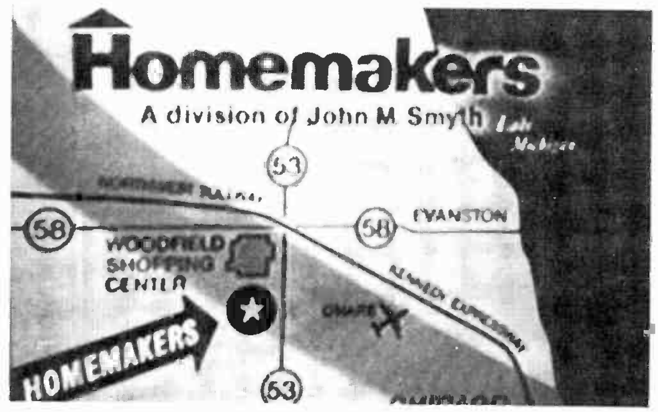


Figure 6.

Thus, those viewers who are still using horizontal receiving antennas will find their reception *absolutely compatible*.

However, as we also saw, the viewer no longer has to worry about the *Azimuth* of his antenna. Under conditions of random positioning of a whip antenna, for instance, we now have much greater assurance of full signal strength at the receiver. The viewer will find he has a much easier time adjusting his antenna.

We have therefore *upgraded* our signal for a vast number of viewers.

Suppose we now equip our viewers with *circular* receiving antennas. The first thing we observe is this: When a circular signal bounces off a reflecting surface, its sense of rotation *reverses*. A right-hand signal becomes left-hand, and vice versa.

Since, as we noted earlier, a circular receiving antenna is sensitive to only *one* of the two possible directions of rotation, the viewer will never see those ghost-producing reflections. If the transmitting station radiates a right-hand signal, and the viewer uses a right-hand antenna, the left-hand reflection signals will not appear on his set.

Figure 3 is a photograph taken from the television receiver in our test vehicle in Chicago. In this photo, the station was transmitting with horizontal polarization, and the receiving antenna was also horizontal. In other words, we used the system that is now standard for all television stations.

As you can see, *there is ghosting in this picture.*

Figure 4 is a photo from the same location, but using *circular* transmission and reception. Note the *absence* of ghosting.

Another photo is shown in Figure 5. This one was taken at a different location. Here, once again, horizontal transmission and reception were used. In this picture, *the ghosting is severe.*

The photo in Figure 6 was taken at the same location as Figure 5, but using *circular* transmission and reception. Note the *dramatic reduction* in ghosting.

In addition to ghost rejection, we can eliminate the co-channel interference problem through intelligent use of circular polarization. Our tests have shown that a circular antenna will reject a signal that has the wrong sense of rotation by an amount equivalent to a front-to-back ratio of *28 db.* If two co-channel stations radiate with opposite senses of rotation, a viewer can easily select between the two.

As with horizontal receiving antennas, circular antennas will be easier to adjust for best reception, because the only factor we have to be concerned about is facing the antenna *broadside* to the transmitting antenna.

Of course, a circular antenna will work fine with present-day horizontal signals, and with the added bonus of easier positioning.

The overall effect of this arrangement is that *wherever* the viewer is, within the coverage area, he will be able *more consistently* to get a good picture from the station. Geographical considerations will not be as important as they were before.

During our engineering tests in Chicago, we drove the test vehicle through the streets of the City to see what the reception would be like. As anyone who has attempted to watch television in a moving vehicle can attest, the results using present-day horizontal polarization are poor. Although the sound portion of the signal is generally adequate, the visual signal varies so greatly that viewing is well nigh impossible.

We confirmed the bad viewing using horizontal polarization while driving through the streets of Chicago. When we switched to *circular* transmission and reception, however, there was a drastic improvement. Using circular polarization, our engineers were able to watch an evening newscast while driving in all driections through the City streets—something they simply could not do using horizontal polarization.

Our interest is *not* in providing better television reception for automobiles, of course, but we must observe this: If we can receive a decent television signal while driving through the City of Chicago, we certainly ought to be able to obtain good reception at home.

Circular polarization, then, is evidently better than horizontal for television reception and transmission. As we have seen, the greatest benefits will come to viewers who obtain circular receiving antennas, but don't forget that the rest of the viewing audience will also benefit immediately from circular transmission, even though they are still using their existing antennas.

What does this mean for television stations that install circular polarization?

Our experiences with circular polarization have produced a number of implications. First of all, we can expect better local coverage, both from the ghost-reduction property and the greater freedom of antenna antenna-positioning. Basically, metropolitan viewers will have a much easier time tuning the station in.

At the same time, circular polarization should reach sections within the coverage area which, due to mountains or other geographic obstacles, were unable to obtain a reasonable picture before. We will have given the viewer a fighting chance to position his antenna for good reception and to reject unwanted reflections.

By the same token, we will give our distant viewers much greater ease in tuning us in for a good picture. Note that we have not extended our coverage contours one inch; they will remain exactly where they were.

What we observe, however, is this: The picture quality available to viewers *within* the coverage area will remain much more consistently *good*, no matter where they are located, instead of dropping off gradually as distance increases. Remember the example of the vehicle driving through Chicago.

What this adds up to is less viewer frustration. Not only will more viewers be able to pick up a decent signal from the station, but they will have a much easier time doing so.

Think about what it means for so many viewers to get such an improvement in reception, and you will see why we have pursued this technique so intently.

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RCA Corporation
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Circularly Polarized Antennas for Television

This paper discusses the theoretical aspects and the potential advantages demonstrated by the field tests of the circularly polarized mode of transmission. These advantages included increased local coverage, less critical receiving antenna orientation and a capability for ghost reduction.

These make the mode attractive to television broadcasters.

Further principles also are discussed and the paper includes some sketches and photographs of several receiving and transmitting antennas, particularly those used in the WLS experiments.

The last portion of the paper enumerates differences in test-range pattern measurements. The conclusion indicates that, with good test range facilities and proven test methods and precautions, adequately accurate measurements can be performed.

Three aspects of circular polarization for broadcast television will be discussed. The first section compares circular polarization with the more familiar horizontal polarization now standard in the U.S.A., and includes a discussion of the potential benefits of circular polarization as applied to TV boradcasting. The second section discusses the principles involved in the design of circularly polarized antennas and the third discusses the considerations of factory tests of C.P. antennas.

A Comparison of Circular Polarization and Horizontal Polarization

Before we get to the potential benefits of circular polarization, let's strike a few comparisons between C.P. and the horizontally polarized transmissions used in the United States. (See Figures 1 and 2 in preceding article).

In horizontally polarized (H.P.) mode of transmission, the electric field vibrates in a horizontal plane, going through a zero value every half wavelength. A dipole antenna, oriented horizontally in this field, captures energy from the field. However, this same dipole, oriented vertically in the same field, is incapable of intercepting any energy. Therefore, the intensity of the signal received by the dipole depends upon the dipole orientation. The signal magnitude is proportional to the length of the dipole as projected on the horizontal plane.

With circular polarization, the situation changes considerably. Two orthogonal dipoles, excited 90 degrees out-of-phase produces a field of constant magnitude and rotates clockwise or counterclockwise depending on the phase relationship between the two excitations. A clockwise rotating field is called a "right-hand" polarized (RH) field. A counterclockwise rotating field is called a "left-hand" polarized (LH) field.

A dipole, placed in the propagation path of this circularly polarized field captures a constant level of signal regardless of its orientation: horizontal, vertical or any angle in between because of the rotating field.

An Increase in Station Coverage

It follows then, that circularly polarized transmission is compatible with receiving antennas in common use today: the "whip" is a vertically polarized antenna; the "rooftop" antenna is a horizontally polarized device and the "rabbit ears" antenna may use both the vertical and horizontal components of the signal.

This is illustrated graphically in Figure 1. Receiving antenna orientation is expected to be less critical with C.P. transmission.

Increased Transmitter Power

The horizontal and vertical components of the circularly polarized signal are orthogonal and, therefore, independent of one another. Since we expect no appreciable twisting (or depolarization) of the fields, each signal will propagate approximately the same distance as the present horizontally polarized signal. Consequently, it is feasible to radiate both signals at authorized ERP without increasing co-channel or adjacent-channel interference. The same total ERP in each polarization could be achieved by increasing transmitter power or increasing antenna aperture or, a combination of both.

If the ERP of both components is maintained by increasing the antenna aperture, the close-in coverage will be reduced due to the narrower vertical beam width associated with larger apertures.

For an existing station, it is more desirable to maintain the close-in coverage by retaining the existing antenna aperture and increasing transmitter power. It seems reasonable to expect that this power doubling, by itself, would provide better service. This is especially true if a circularly polarized antenna is used.

Reduces Ghosting from Multipath

Circularly polarized transmissions promise considerable improvement in the phenomenon of "ghosting" from multipath signals. This expectation is based on two factors: one, the reflection behavior of C.P. fields and, two, the polarization discrimination by C.P. receiving antennas.

Figure 2 illustrates the reflection phenomenon of electric fields. The vector illustration of incident signal—a right-hand C.P. signal—shows the spatial relationships of the two component fields as radiated from an antenna. The vector illustration of the signal, after reflection, shows that the vertically polarized field experiences essentially no change in character while the horizontally polarized component reverses polarity. The total tangential electric field must be zero at the reflecting surface.

As a result, the reflected signal is a left-hand signal if the incident signal is right-handed; the sense of rotation is reversed. This reversal gives C.P. the capability to reduce picture ghosting due to multipath signals.

Polarization Discrimination by Receiving Antennas

A right-hand C.P. antenna responds only to a right-hand, circularly polarized radiation; it is essentially blind to left-hand polarization. Let me explain:

The three drawings in Figure 3 illustrate the polarization discrimination of C.P. antennas. Two receiving and one transmitting antenna are shown.

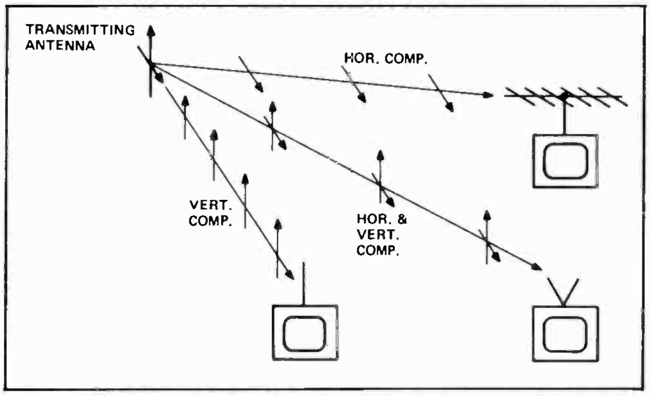


Figure 1.
More Solid Coverage

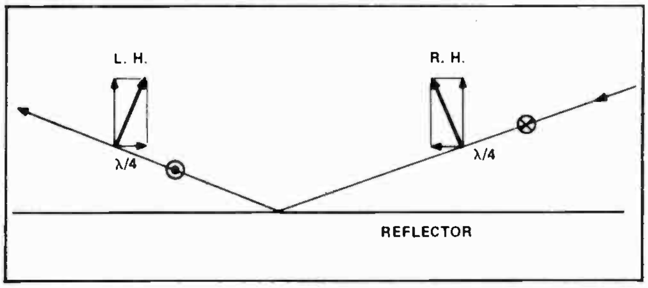


Figure 2.
Reflections of Electric Fields

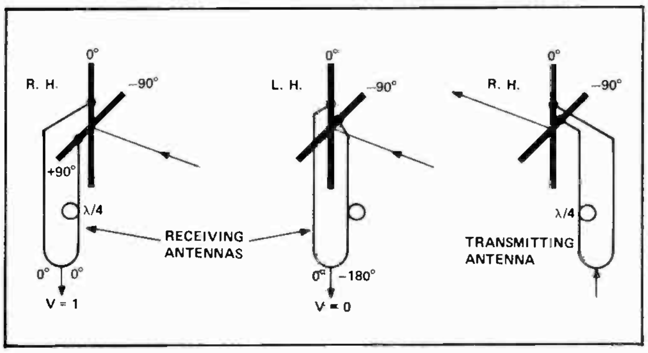


Figure 3.
Polarization Discrimination

All three use identical crossed dipoles connected in phase quadrature. The transmitting antenna radiates a right-hand (RH) signal as a result of the connections to the feedline.

The receiving antenna on the left is identical to the transmitting antenna in the drawing and it is also RH polarized. Since it's used as a receiving antenna, it faces the transmitting antenna.

The antenna in the center is wired for a left-hand (LH) polarization but it is, otherwise, identical to the transmitting antenna.

When these receiving antennas receive the signal (RH) radiated from the transmitting antenna, the RH antenna develops additive currents at the output combiner while the LH antenna develops opposing currents at the combiner. Obviously, only the RH antenna converts the field into useful current. This is

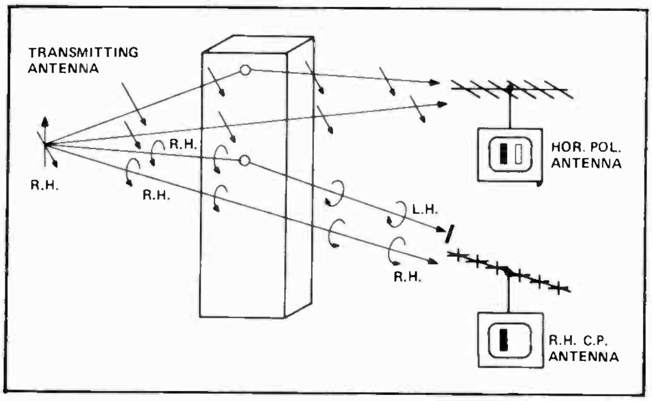


Figure 4.
Ghost Reduction Capability

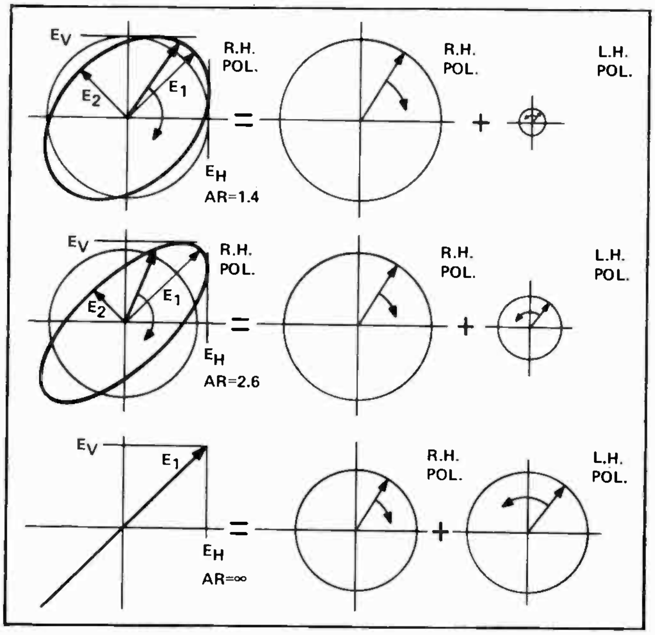


Figure 5.

the polarization discrimination so useful in the multipath ghosting problem.

The drawings in Figure 4 illustrate the combined effects involved. In this case, a tall, flat-faced building is the interfering reflector for a particular receiver location.

The upper drawing shows the situation with an H.P. signal and an H.P. receiving antenna. The receiver responds to both the direct and the reflected signals. Because the reflected signal takes a longer path, it appears on the picture screen as a ghost.

When the radiated signal is circularly polarized, the situation depicted in the lower drawing takes place. The direct signal maintains its right-hand polarization and the receiver's right-hand polarized antenna accepts it. The reflected signal, however, becomes left-handed for the reasons described and, consequently, is rejected by the RH receiving antenna. Result: the received picture contains little, if any, ghost resulting from the multipath signal.

This same mechanism may also reduce other multipath-caused distortions such as changes in picture sharpness or color saturation due to less than perfect video response of the path.

Under ideal conditions, perfect ghost rejection would result if both the transmitting and receiving antennas were prefectly circularly polarized devices. In actual practice, the polarization is always more or less elliptical rather than truly circular.

The Quality of Circular Polarization

We may describe the quality of circular polarization in two ways: For FM radio, circularly polarized antennas are specified by "polarization ratio". This is the mathematical ratio of maximum vertical component to maximum horizontal component. It's a good indicator of radiated power in each polarization but not the ellipse formed when the two signals are combined. The shape of the ellipse might range anywhere between a circle and a line without a change in polarization ratio. This appears quite adequate when the receiving antennas are either horizontally or vertically polarized.

For antennas used in television broadcast, however, we expect the term "axial ratio" to become a most important parameter of quality. Axial ratio is defined as the ratio of the major axis of the polarization ellipse to the minor axis.

The drawings in Figure 5 illustrate how an elliptically polarized signal can be represented as two circularly polarized signals that are counter-rotating. The top drawing shows an almost circular case; the middle drawing, increasingly elliptical and, the bottom drawing, an ellipse that has collapsed to just a slanted line. The circular and slant-linear are special cases of elliptical polarization.

The top drawing represents a "circularly" polarized signal consisting of a large right-hand and small left-hand polarized components. As the ellipse slims, as depicted in the middle drawing, the right-hand polarized component reduces slightly while the left-hand polarized signal increases appreciably. When the ellipse becomes a line, as in the bottom drawing, left- and right-hand components are of equal amplitude.

If we use the ghosting situation described in Figure 4, we can see that a right-hand elliptically polarized transmitting antenna radiates a large RH component and a small LH component. The receiving antenna, an RH device, accepts the RH components from both the direct and the reflected waves; a large one from the reflected signal.

The reason for the smallness of the latter is the fact that the large RH component in the incident signal is converted to a large LH component in the reflected signal while the small LH component in the direct wave is converted to a small RH component in the reflected wave. As a result, the ghost is attenuated

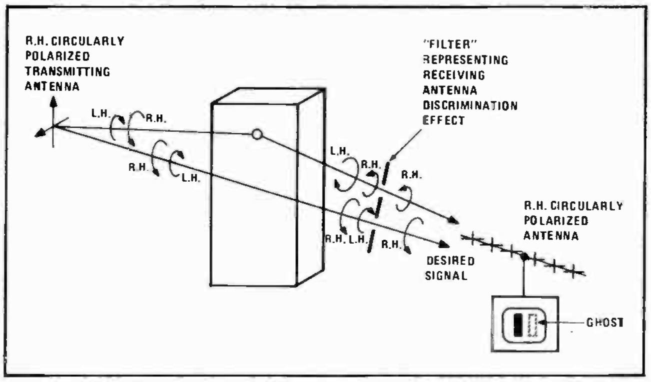


Figure 6.
Effect of Axial Ratio

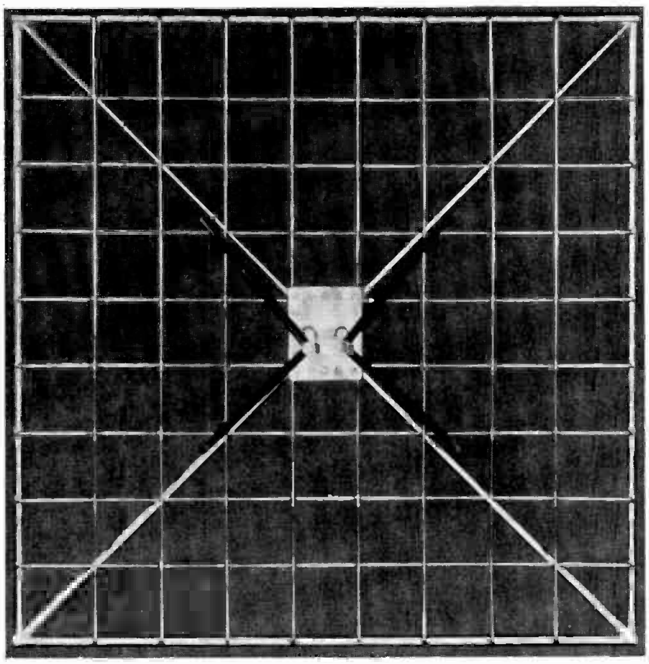


Figure 7.

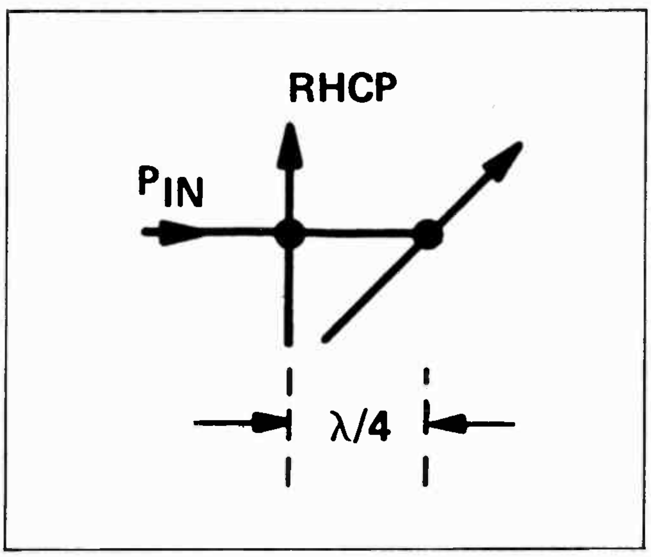


Figure 8.

considerably, possibly to a level below viewer perception. (See Fig. 6.)

Thus, axial ratio of the transmitting antenna is an important factor in determining the quality of a circularly polarized antenna.

Reduced Co- and Adjacent-Channel Interference

Another benefit of circular polarization is a possible reduction in co- and adjacent-channel interference. Advances in receiving antenna design will be a factor here.

Circularly Polarized Antennas

Antenna principles are identical for both receiving and transmitting systems. However, it is often easier, for the purposes of discussion, to consider the system as a transmitting antenna.

Generating a Signal

We need a rotating field which, for example, might be the sum of two fields of proper magnitude and relative phase and at right angles to one another.

In the earlier discussion, we used orthogonal dipoles with a 90-degree phase shift between the radiating currents. Such an antenna produces a circularly polarized field and is simple in concept. Crossed-dipole antennas are often the basis for panel-type transmitting antenna designs.

Figure 7 shows an RCA circularly polarized antenna for FM radio. It uses crossed-vee-shaped dipoles operated in phase quadrature.

A circularly polarized antenna could also be obtained by displacing the dipoles a quarter wavelength apart, as shown in Figure 8. The two dipole currents may now be in phase with the essential 90-degree phase shift provided by the quarter-wavelength displacement (*separation*) along the radiation axis.

Figure 9 shows a C.P. receiving antenna consisting of two yagi arrays in space quadrature using the displaced dipole principle of phase shift. This antenna type was built by RCA for the WLS experiments.

Figure 10 is a drawing of a crossed log-periodic antenna for C.P. In this antenna, the elements (*or steps*) in the periodic structure are interlaced to produce the 90-degree phase shift without increasing the overall length of the device appreciably.

Figure 11 is a drawing of a conical log-spiral antenna for circular polarization. The two spirals are interlaced in much the same manner as the windings of a bifilar-wound transformer. This antenna is, in principle, very simple but may not be quite as economical to manufacture as the crossed-dipole types.

The Figure 12 photo illustrates a corner-reflector type of receiving antenna for C.P. In tests, this very simple, slanted-dipole device appears to be capable of an axial ratio of better than 2dB.

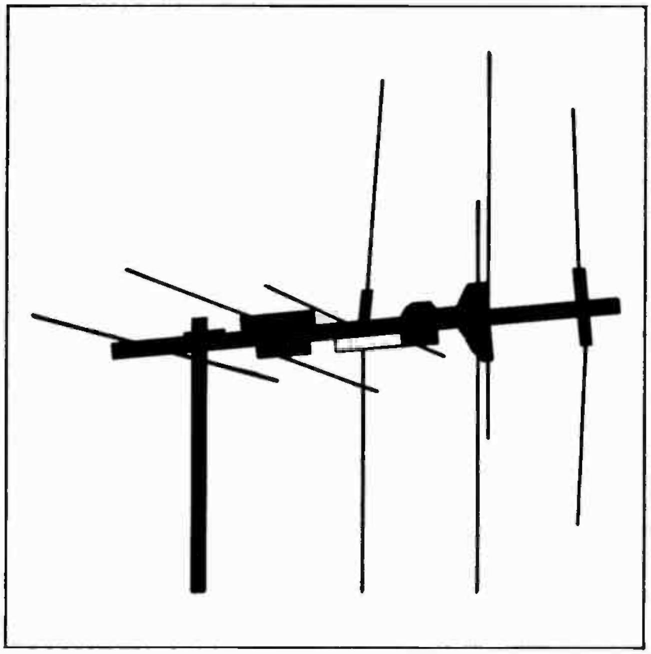


Figure 9.

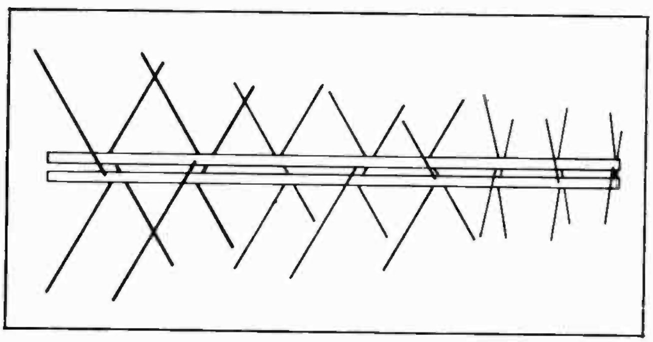


Figure 10.

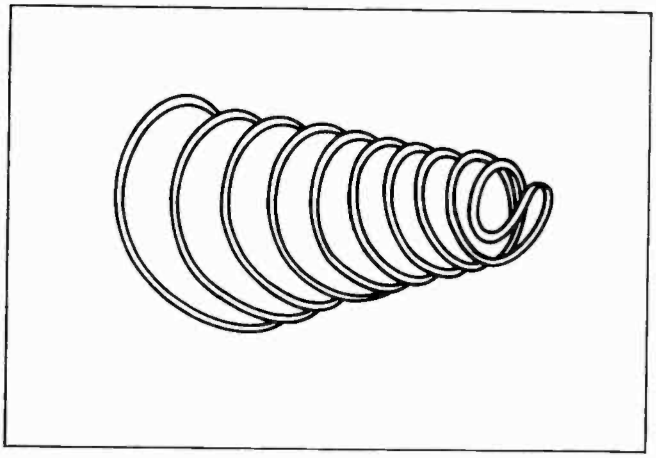


Figure 11.

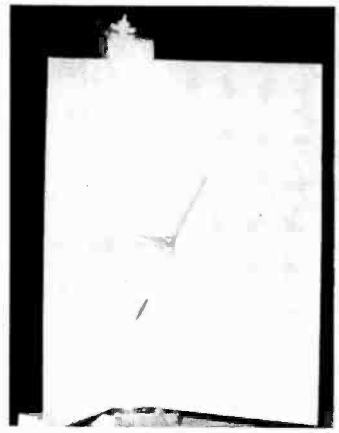


Figure 12.

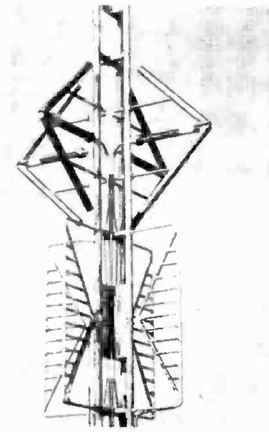


Figure 13.

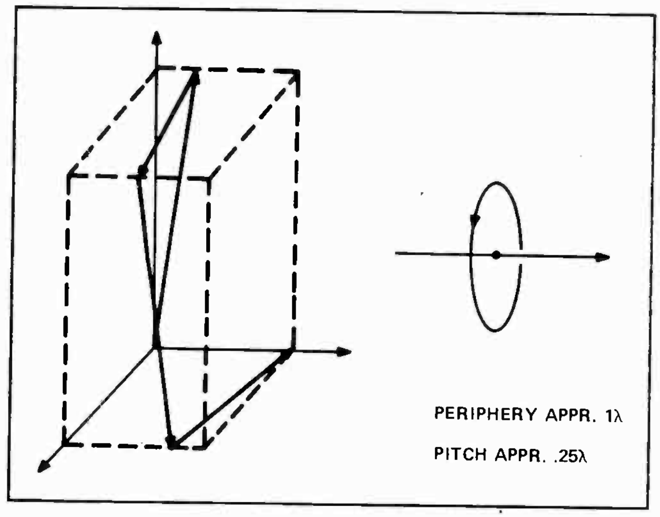


Figure 14.

Getting back to transmitting antennas, the Figure 13 photo shows one layer of an antenna we developed for channels 2-6. It uses a turnstile-type vertical dipole layer (top section) for the vertically polarized signal and a "fan-type" turnstile layer for the horizontal.

This is the antenna we dubbed the "Fan-Vee" for obvious reasons. By providing an appropriate power division and phase relationship between the "fan" and the "vee", the antenna makes a very good system with an excellent axial ratio.

If we form a helix from the straight elements as shown in Figure 14, with an approximate one-wavelength periphery and a quarter-wavelength pitch, we have an end-fire radiator. This antenna type, in cylindrical form, is popular for radar-tracking and telemetry applications. The design is also useful as a basis for broadcast antennas.

For highband VHF, we developed a circularly polarized panel antenna using a helix mounted in a reflector. Figure 15 is a photo of the antenna showing the "dishpan" reflector and the radome which protects the radiator.

The axial ratio of this end-fire-helix antenna is better than 3dB and acceptable horizontal pattern circularity can be obtained. The design is feasible for omnidirectional antennas with total combined power gains up to 16.

The drawing in Figure 16 shows the radiating elements of a high-power panel antenna for FM radio. This design uses a "backwards vee" element for the horizontal elements which series-feed vertical dipoles each—respectively—radiating the horizontal and vertical components of the C.P. signal. The stubs and discs provide matching as well as proper phase relationships between the H and V elements.

The antenna design was first used around a leg of the Mr. Sutro Tower in San Francisco. Figure 17 is a view of that array. The antenna provides a simple, single feedpoint, circularly polarized radiator for FM radio.

The WLS Transmitting Antenna

This brings us to the design used for the WLS experimental antennas. ABC requested that RCA provide suitable antennas for the experiment.

It was determined that two separate arrays were most suitable for the purpose: one, a circularly polarized, three-layer system and, the other, a two-layer, horizontally polarized system. Figure 18 shows a one-layer scale model of each antenna mounted on a triangular tower section. Note the resemblance of the circularly polarized device to the Fig. 16 drawing. The horizontally polarized layer is the RCA "Butterfly" design.

Since the antenna-test information was to be supplied to the FCC for possible use in rulemaking, the antenna was completely assembled and tested at RCA's Gibbsboro, N.J., antenna engineering facility and test range.

The antenna, in its ultimate environs, is mounted on the side of a cylinder atop the Sears Building in Chicago. Figure 19 is an air view of the building after antenna installation. Note the twin cylinders on the roof and the antenna on the cylinder are left.

The supporting cylinder affects antenna performance. Therefore, it had to be simulated on the test range. Figures 20 and 21 show the WLS antenna on the horizontal and vertical pattern turntables at the Gibbsboro facility. (Note the simulated cylinder structure.) Since the antenna design included a radome, the system was tested with the radome in place.

The horizontal patterns of the H.P. antenna and of each component of the C.P. antenna were made as equal as possible to facilitate easy correlation of the measurement data. (See Fig. 22).

After testing, the antennas were shipped to Chicago and installed atop the building late in 1973.

The system design included separate feed systems for each antenna to allow easy power transfer from one antenna system to the other.

First on-air tests took place starting in January, 1974, in the early morning hours. These tests assured that the new mode of transmission resulted in no appreciable increase in coverage area to satisfy the requirement that there'd be no increase in co- or adjacent-channel interference.

The tests that followed assured that the circular polarization caused no degradation of the received picture. Once these requirements were satisfied, the FCC granted permission, in May, to use the circularly polarized antenna on a continuous basis rather than only during off-air hours.

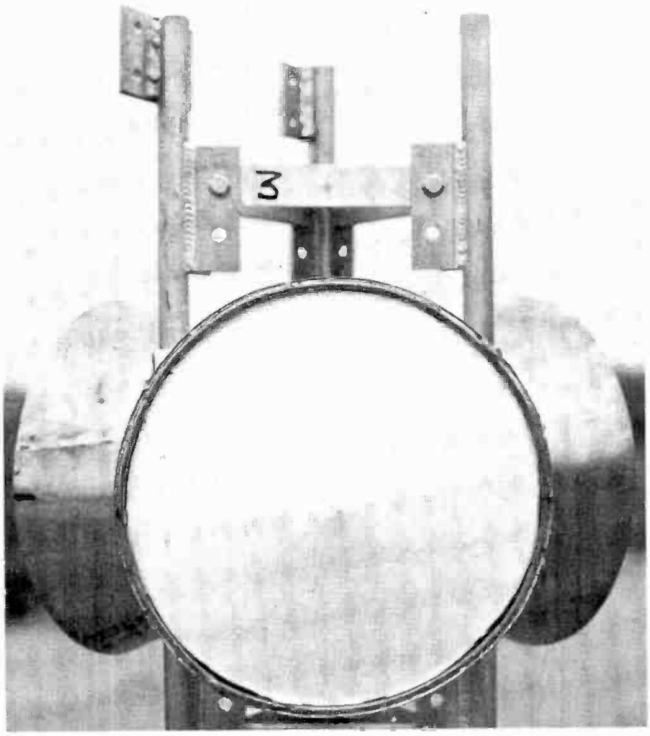


Figure 15.

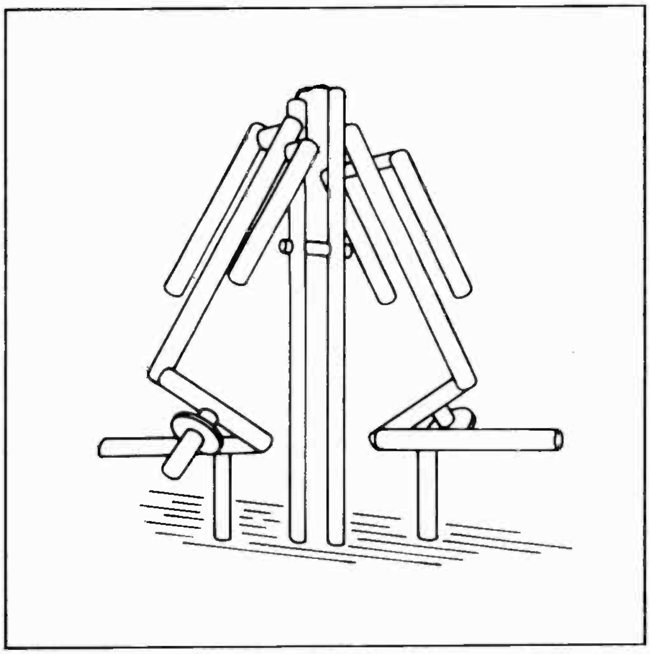


Figure 16.

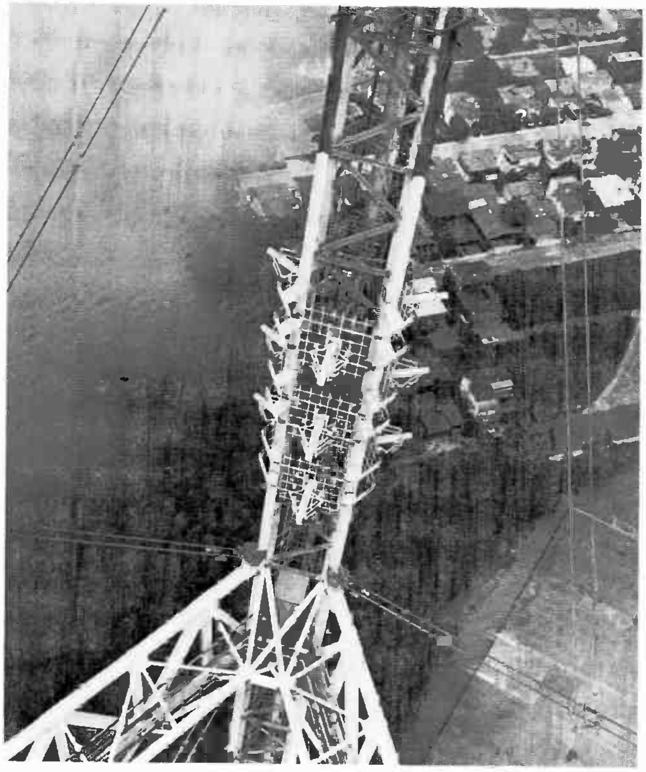


Figure 17.

Figure 19.

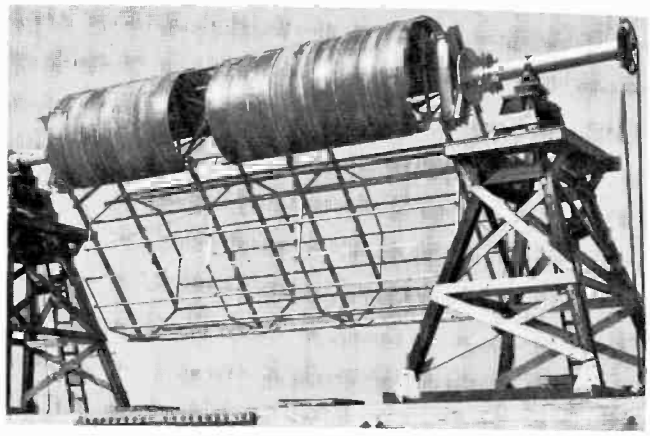


Figure 20.

Figure 18.

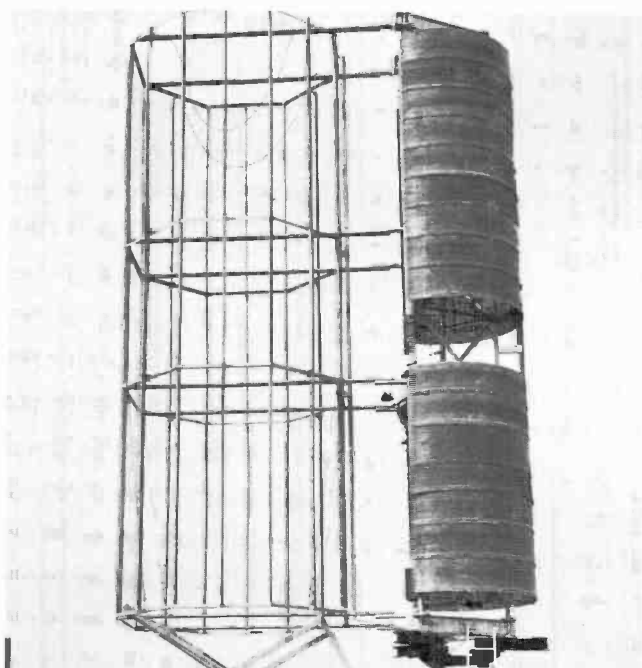


Figure 21.

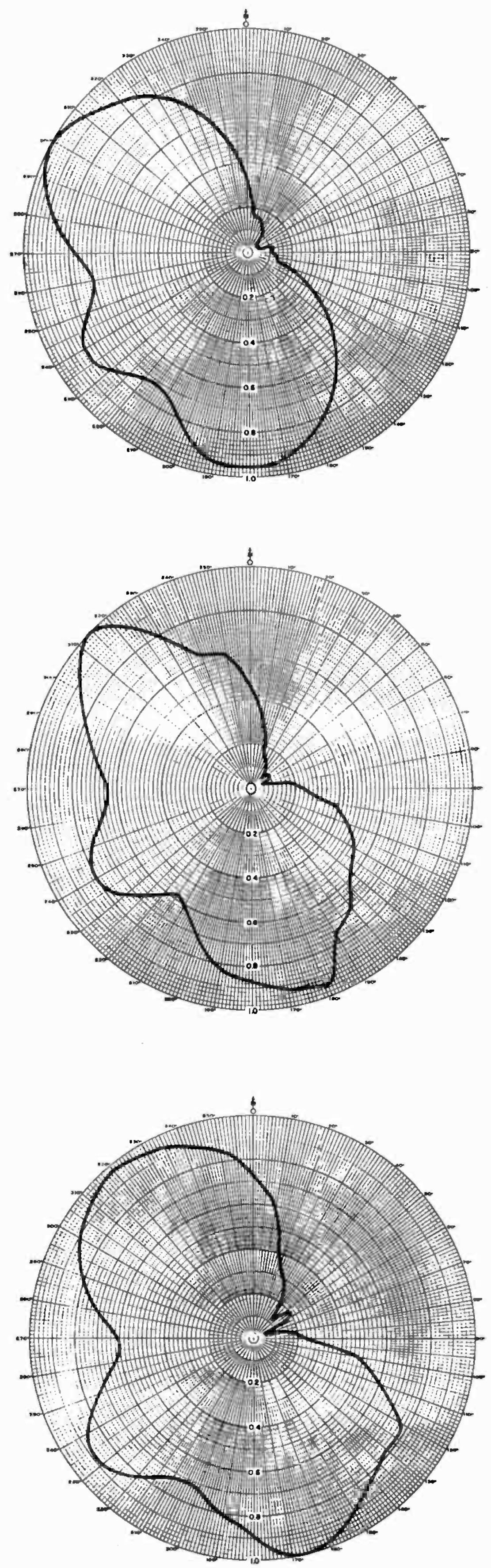


Figure 22.

Factory Measurements of Circularly Polarized Antennas

What are the characteristics required of an antenna to assure proper operation?

In a horizontally polarized system, the important antenna characteristics are its impedance across the channel, its power gain and the vertical and horizontal patterns. If vertical polarization is added, the antenna characteristics include the impedance and, in addition, the patterns and power gains for both polarizations.

In addition to the above, a circularly polarized TV antenna requires an evaluation of the axial ratio.

All these parameters are easily measured during scale model development work using a three-axis turntable, such as that in the photo of Figure 23. The antenna under test was mounted at the end of a boom, operated as a receiving antenna with the transmitting antenna mounted below, on the ground, radiating upwards. Rotating one or both antennas provides all the measurements needed. This is a technique described by Dr. Ben-Dow three years ago.

Figure 24 shows a set of recorded measurements at the horizontal plane.

The two solid lines represent the horizontal patterns of the vertical and horizontal components of radiation. The polarization ratio is derived by taking the ratio of the values of these curves at any specific azimuthal direction. If we rotate the antenna under test slowly and rotate the source antenna rapidly, we produce the serrated pattern on the chart. This is not a real pattern but a means to measure, with a linear antenna, the maximum and the minimum signals in any specific direction and thus obtain the magnitudes of the major and minor axes of the polarization ellipse to that direction. The ratio of these maximums and minimums is the axial ratio. As we have seen, all four measurements are obtained in one simple setup.

Full-Scale Antenna Measurements

Testing full-size antennas, unfortunately, isn't quite as easy. We discussed earlier how an RH signal tends to change into an LH signal at the point of reflection.

In test terms, this means that if we measure the axial ratio of even a perfect circularly polarized antenna in the presence of a reflected wave, we encounter considerable error. The curve in Figure 25 shows the extent of this variation. As the ratio of reflected wave to direct wave increases, the axial ratio of the antenna—as seen by the test antenna—increases. For example, a 10-percent (−20dB) reflection causes an axial ratio measurement error of 2dB on a perfect antenna. Obviously, this technique is unsatisfactory for full-size antennas.

Another error factor is due to propagation. With large antennas, long distances between the test antenna and the signal source are essential. Although depolarization is quite small under such conditions, we can't be sure that the propagation of the horizontally polarized and vertically polarized signals is alike.

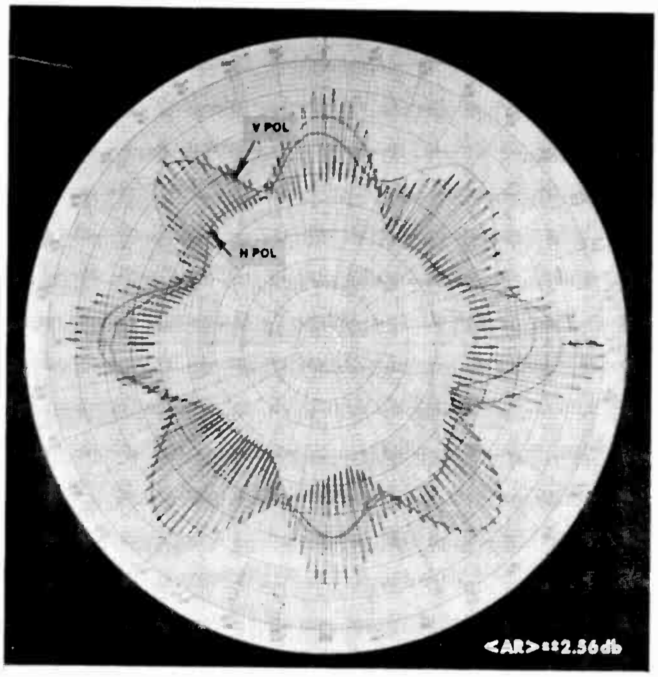


Figure 23.

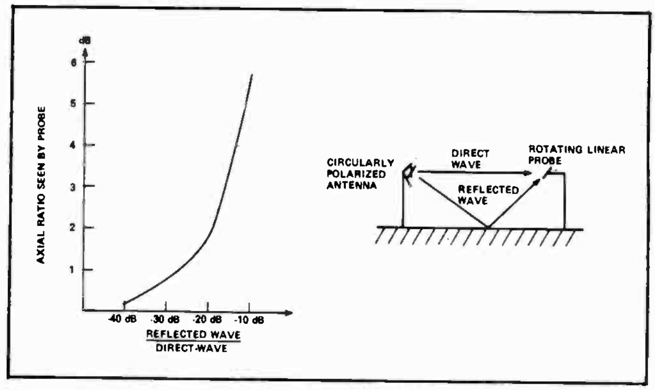


Figure 24.

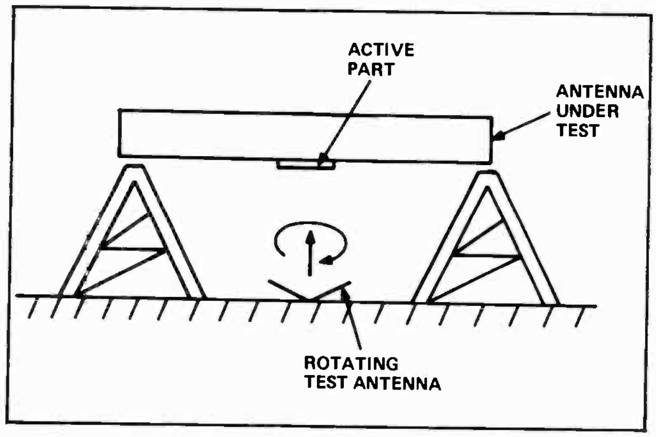


Figure 25.

This means that, if the two signals are compared, a proper comparison with a reference antenna is imperative to assure reasonable accuracy. Even then, there tends to be small errors due to depolarization.

We've found that, in practice, with a very good test range—one with a minimum of depolarization—we can make separate measurements of the vertical and horizontal components. A calibration between the two measurements can be made for gain purposes. For the measurement of axial ratio, however, other means appear necessary.

A technique used has been to mount a portion of the large antenna high above ground or energize only a portion of the large antenna and locate the signal source below it. By rotating the signal source, we can establish the axial ratio. Fortunately, axial ratios are essentially identical from one layer to another of the antenna.

The obvious conclusion is that circularly polarized antennas require more tests than horizontally polarized systems. Further, a great deal of care is required even when a good test range is available.

Conclusions

All the evidence points in the direction that the C.P. mode of transmission offers enough potential advantages to encourage broadcaster implementation.

Practical hardware principles have been developed to a point where antennas can be built to meet broadcaster's specific requirements.

Experience and methods of measurement are available to test the hardware.

Neil M. Smith

Smith and Powstenko
Washington, D.C.

Report on Field Tests of Circular Polarization in Television

Despite certain predicted advantages of circular polarization in television, theory and reality do not always go hand in hand. Logical questions have been raised with regard to such a sweeping concept, and many of these questions could be answered only through actual experimentation.

For example, does a practical, circularly polarized receiving antenna actually eliminate ghosting, and do circularly polarized receiving antennas actually exhibit the expectedly high front-to-back ratio?

Does depolarization take place, so that the vertical component of the circularly polarized wave unduly supports the horizontal component at certain locations or in certain situations, with the possible result that the service area of the station might materially increase, or that the interference potential might rise?

It is because of these uncertainties that these field tests were undertaken. The questions we sought to answer through this study fall into three basic categories:

- Does a station's service area change when it converts to circular polarization?
- Does a station's interference potential change when it converts to circular polarization?
- Does an improvement in reception actually occur?

To answer each of these questions, we designed a three-part study. The first phase consisted of TASO-style measurements along particular radials, comparing the field strength received from the horizontally polarized transmitting antenna with the horizontal and vertical components of the circularly polarized transmission, as well as with the signal received by specially designed circularly polarized receiving antennas. In addition, picture quality was observed at each measurement location.

The second part of the study was directed toward the observance of co-channel interference changes.

The third phase involved picture-quality observations at locations approximately 10 to 15 miles from the WLS-TV transmitter. In combination, these three divisions of the study covered the three basic questions.

Four different combinations of WLS-TV transmitter-mode/field-receiver-mode are discussed here. For convenience, initials are frequently used to designate these various combinations, both in the text and in the exhibit labels, and these initials (which always describe the *transmitter* mode first) are defined, as follows:

H-H	Horizontally polarized transmission, Horizontally polarized reception
CP-H	Circularly polarized transmission, Horizontally polarized reception
CP-V	Circularly polarized transmission, Vertically polarized reception
CP-CP	Circularly polarized transmission, Circularly polarized reception

We obtained our data through the use of a small panel truck. (Fig. 1). It was equipped with an AC power supply and a 30-foot, pneumatically operated telescoping mast (shown raised in Fig. 2).

For our study, RCA designed a circularly polarized receiving antenna (Fig. 3), made up of three horizontal and three vertical elements. This antenna is cut to Channel 7 and includes a quadrature combining network. For the measurement of horizontal and vertical components, we used a five-element, Channel

Figure 1.

Figure 2.

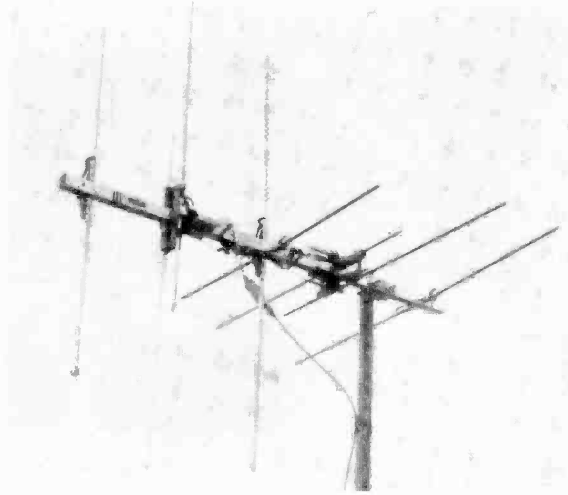


Figure 3.

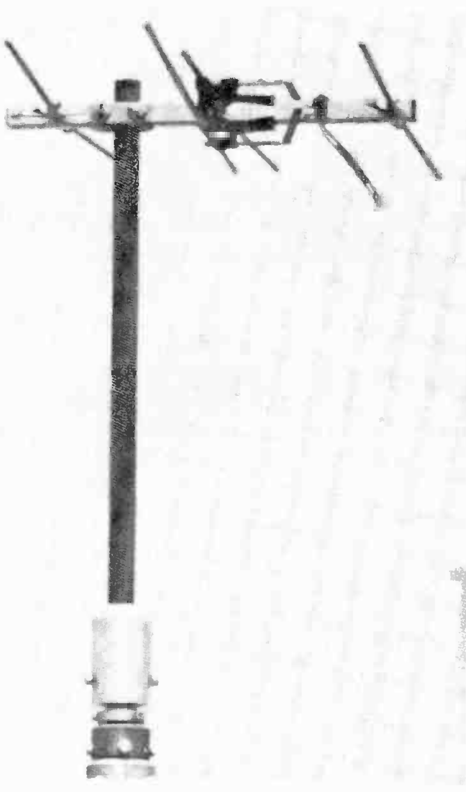


Figure 4.

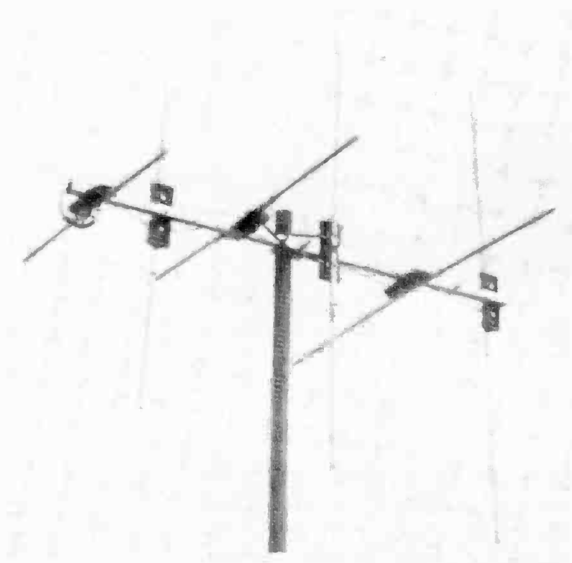


Figure 5.

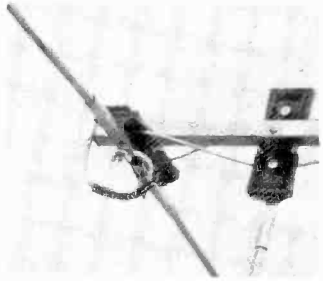


Figure 6.

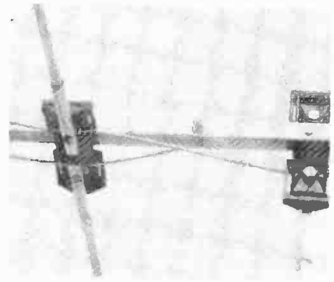


Figure 7.

7 Yagi, with its hardware modified to permit orientation in either plane.

In order to avoid electrical interaction between the mast and the vertical antenna elements, we used a plastic top section, shown in Figure 4. (The antenna is another which RCA later provided us, and it is simply half of the circular antenna.)

Recently, we received JFD's version of a CP receiving antenna (Fig. 5). This is designed for reception of Channels 7 through 13 and has an interesting feed system. From the antenna elements, a piece of line disappears into one end of the boom (Fig. 6) and comes out the other end at an "F" connector (Fig. 7). JFD hasn't yet explained what's inside.

Figure 8 shows the interior of the field truck, with the RCA BW-7A field-intensity meter on the right and a pair of Esterline-Angus chart recorders beside it.

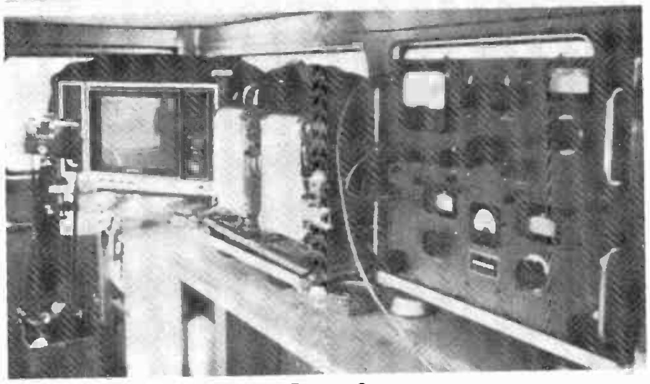


Figure 8.

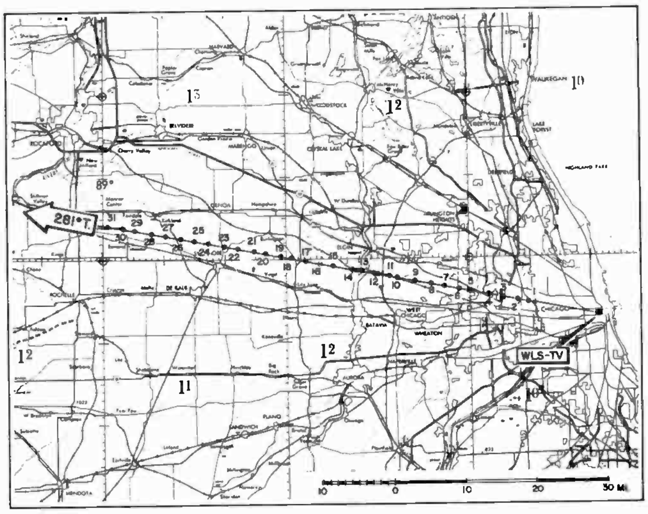


Figure 9.

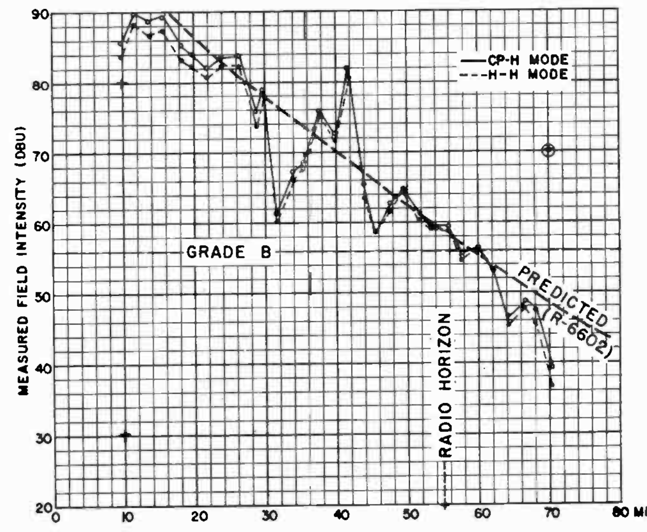


Figure 10.

The meter and its associated standard dipole have been calibrated by the National Bureau of Standards, and the antennas used in the study were, in turn, calibrated against the meter's standard dipole at one of the better measurement locations. Individual field-intensity values were measured by the usual TASO technique of recording field intensity as the vehicle was moved along a 100-to-200-foot course and extracting the median therefrom.

Picture quality observations were made through the use of a Sony Trinitron color monitor. Photographs of the television pictures were made with Nikon cameras, and picture qualities were rated by the TASO six-step standard, described as follows:

- GRADE 1 (Excellent) — The picture is of extremely high quality. It is as good as one could desire.
- GRADE 2 (Good) — The picture is of high quality and provides enjoyable viewing. Interference is perceptible.
- GRADE 3 (Passable) — The picture is of acceptable quality. Interference is not objectionable.
- GRADE 4 (Marginal) — The picture is poor in quality, and the viewer wishes he could improve it. Interference is somewhat objectionable.
- GRADE 5 (Inferior) — The picture is very poor, but one could watch it. Definitely objectionable interference is present.
- GRADE 6 (Unusable) — The picture is so bad that one cannot watch it.

Both the horizontally polarized and circularly polarized transmitting antennas are directional in the horizontal plane (and, naturally, in the vertical plane). Further, the patterns of the horizontal and vertical components of the circularly polarized antenna are not equal at all azimuths. Thus, at any given azimuth, it is unlikely (or, perhaps, impossible) that all radiation components will be equal.

All measurements discussed herein were made along an azimuth of 281° T, which is a direct line between WLS-TV and KWWL-TV, Channel 7, Waterloo-Cedar Rapids, Iowa. Along that bearing, the RCA pattern measurements indicate the horizontal transmission to be 1.0 db below the maximum allowable power at this height (19.1 dbk at 1500 feet). Similarly, RCA shows the horizontally polarized component of the circular transmission to be 2.7 db below that value and the vertical component to be 3.0 db below. In order to make proper comparisons, all data has been corrected to the 19.1 dbk equivalent.

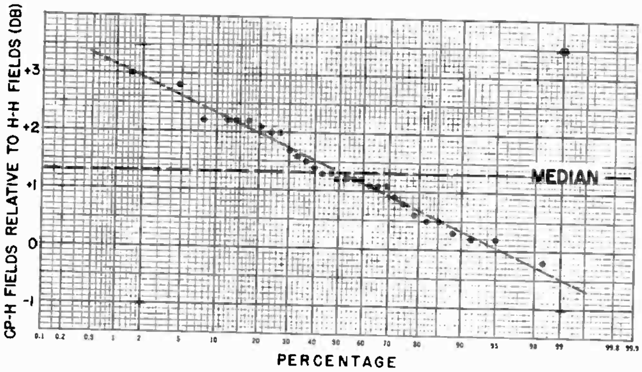


Figure 11.

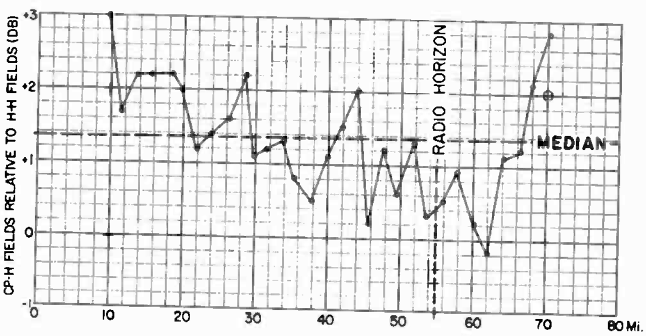


Figure 12.

Figure 9 shows the measurement locations along the 281° T radial, running out to about 70 miles.

In Figure 10 we have plotted the measured field versus distance for horizontal transmission and for the horizontal component of circular transmission. The data generally follows prediction, and the departures from prediction relate directly to terrain variations which, although not great in this part of the county, are certainly not negligible. It can be seen that the horizontal component of the circularly polarized transmission runs higher at the closer distances, and that the data is quite consistent.

A plot of the statistical distribution of the difference between the CP-H and H-H fields is shown in Figure 11. The CH-H field varied from being about equal to the H-H field to being about 3 db higher, and the median difference was 1.35 db.

When we plotted these differences as a function of distance (Fig. 12), we saw a generally decreasing trend until we got well beyond the radio horizon, where the CP-H fields seem to strengthen again. These data suggest to us that a station's Grade B contour should be essentially unaffected when changing to circular polarization.

In Figure 13 we see the horizontal and vertical components of the circularly polarized signal as a function of distance. The two sets of data are quite similar, indicating that both components propagate similarly.

The probability plot of the difference between the two components (Fig. 14) shows that the vertical fields were sometimes higher than the horizontal fields, and sometimes lower, but the median difference was only 0.5 db, which is within the overall accuracy of the

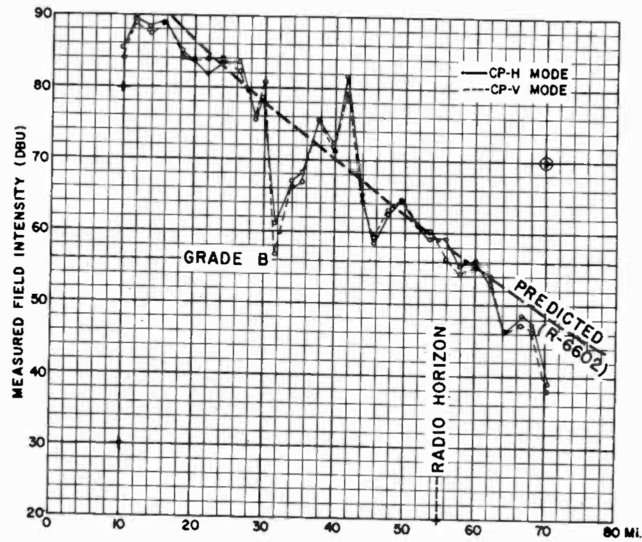


Figure 13.

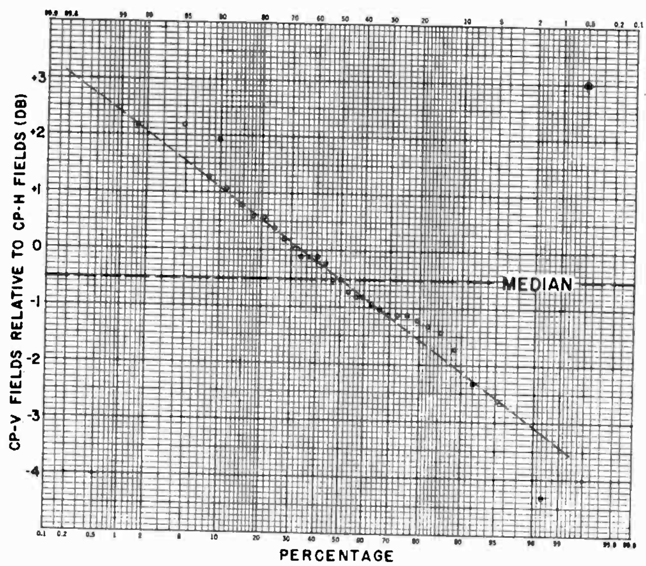


Figure 14.

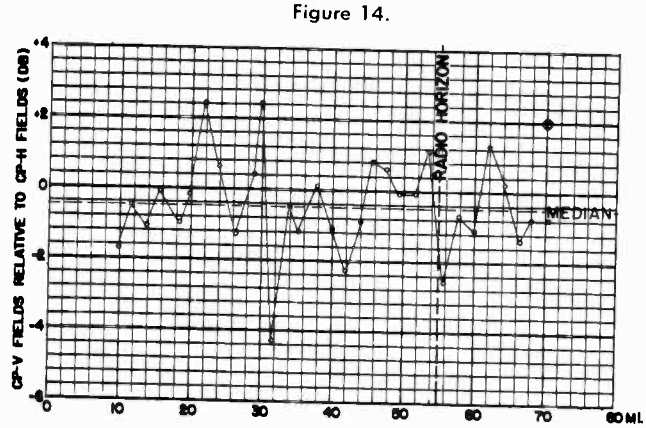


Figure 15.

measurements.

When we plotted these differences as a function of distance (Fig. 15), we saw no discernible trend at all.

It was originally believed that the use of circular polarization would tend to minimize variations in

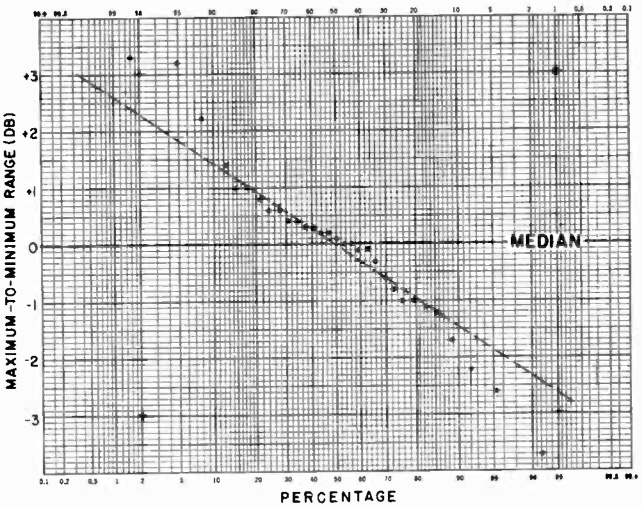


Figure 16.

signal strength. We tested this by observing the maximum-to-minimum range of signal recorded at each location for each mode of operation, and then used the variation in the H-H measurement as our standard. We compared the CP-H ranges with the H-H ranges (Figure 16) and found that the median difference was zero.

However, when we compared the CP-V ranges to the H-H ranges (Fig. 17), we found that the CP-V ranges were greater by 1 db, and very large differences were observed at about 20 percent of the locations. This suggests to us that the vertical component tends to vary more than the horizontal component, and under particularly unfavorable conditions it varies much more.

When we compared the ranges obtained with circular transmission *and* reception (Fig. 18), we found that, on a median basis, the variations were only a little greater than for the H-H mode, but the graph looks quite a bit like the one for the vertical component, with substantial variations about 10 percent of the time. Thus, it seems that the more erratic behavior of the vertical component may influence the behavior of the signal received by a CP receiving antenna.

An example of this effect is shown in Figure 19 which is a representation of the variations in signal found at one of the receiving locations. The vertical scale is receiver-input voltage, and the horizontal scale represents the path along which the recording was made. The H-H trace varies over a 6.5 db range, while the CP-H trace is virtually identical with a 7.7 db spread. However, the CP-V trace varies over a 12.5 db range, and the CP-CP trace covers a similar range. This tendency was seen quite regularly.

Another unexpected finding was that the average picture quality for the CP-H mode seemed slightly better than for the H-H mode, and the average for the CP-CP mode appeared better still. Indeed, Grade 1 pictures were observed *only* during circular transmission. For example, with 10 mv in the H-H mode, a

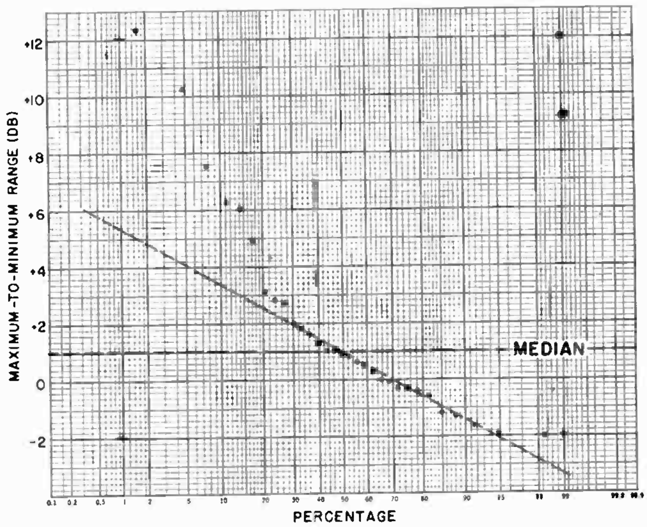


Figure 17.

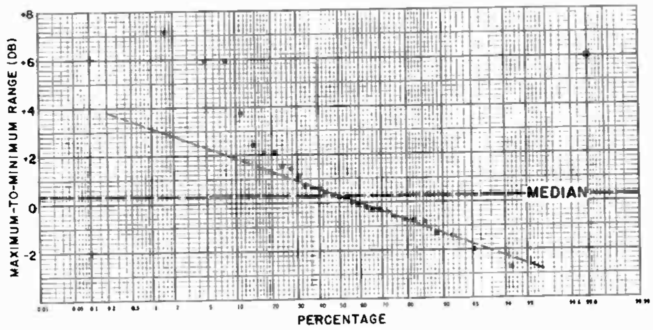


Figure 18.

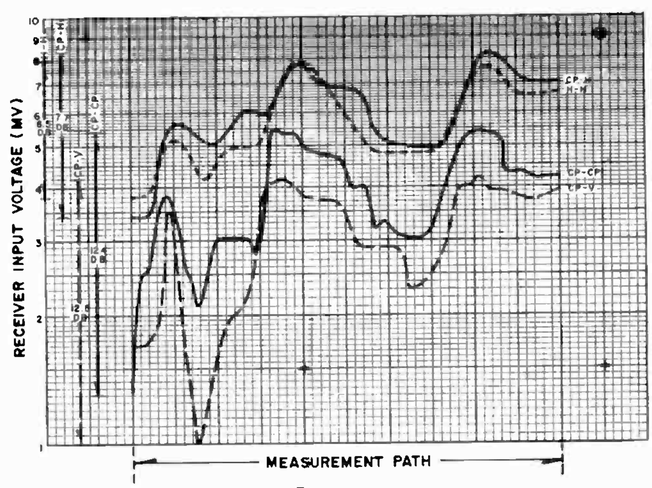


Figure 19.

Grade 2 picture was observed, while 6.2 mv in the CP-CP mode provided a Grade 1 picture.

Figure 20 shows a graph of the median receiver-input voltage which produced each picture grade. Although the graphs tend to merge for the poorer pictures (for which very little data happened to be available), there is a clear trend indicating that lower CP fields can produce higher-quality pictures.

Why this should be so is something of a mystery. In most cases, the picture degradation came simply from noise rather than ghosting; had it come from ghost-

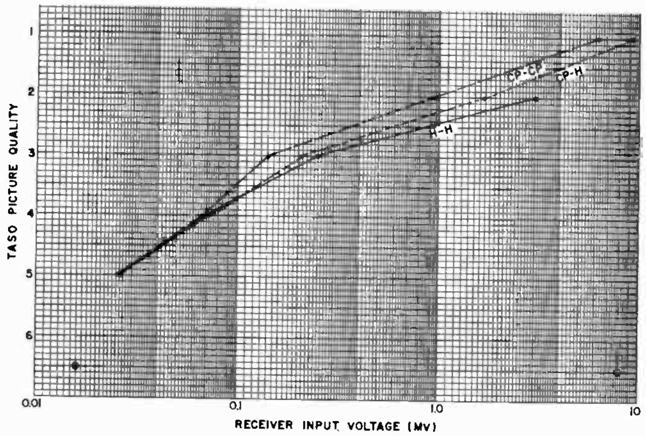


Figure 20.

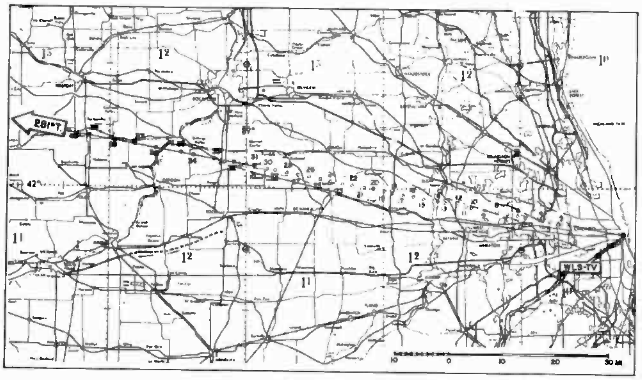


Figure 21.

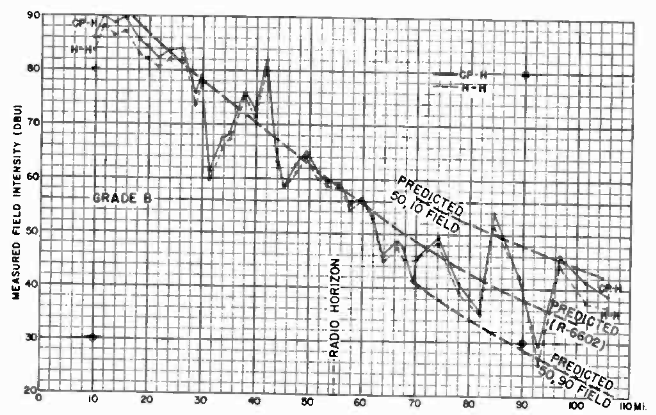


Figure 22.

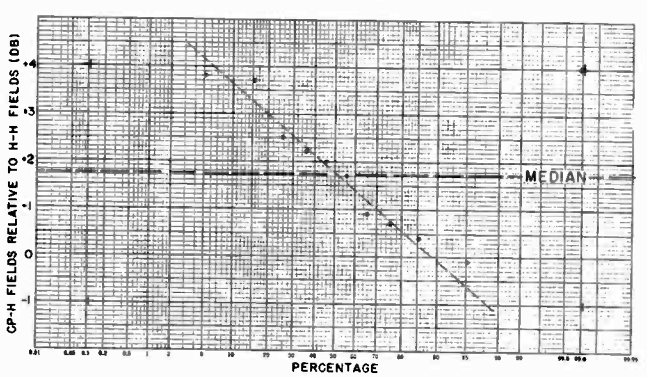


Figure 23.

ing, CP-CP operation might logically have minimized it. The CP-H and CP-CP pictures were simply cleaner and clearer and sometimes of studio-monitor quality. We do not know the reason why.

In order to investigate the interference situation, we extended our radial out to a distance of 106 miles by adding the ten locations shown in Figure 21.

To minimize the problem of fading, we changed our measurement approach. Instead of making mobile runs, we positioned ourselves in one place, and WLS-TV alternated its two transmitting antennas at regular intervals. By recording about a half-minute of signal both before and after the switch, we could observe the two signals with little influence from fading.

In Figure 22 we have added these more distant data points to our previous plot of the H-H and CP-H data. We've also shown the predicted 10-percent and 90-percent curves, to indicate the expected range of variation at these distances. One can see that the two signals continue generally to track with each other.

In Figure 23, we see that the CP-H field at these ranges appears to run above the H-H fields by about 1.75 db.

Turning to our close-in observations, let us first discuss an aspect of circular reception which had not been anticipated.

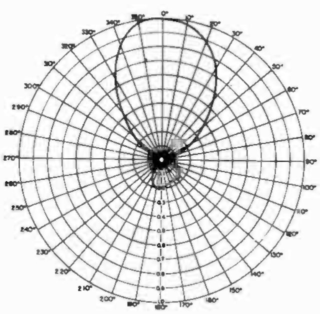


Figure 24.

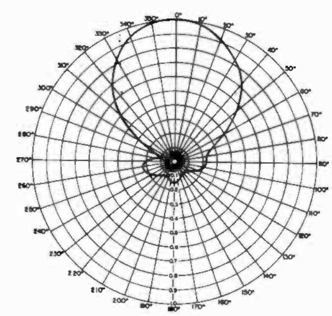


Figure 25.

Shown above are two antenna patterns, measured by us in the field (and, therefore, approximations). The pattern in Figure 24 is of the Yagi antenna oriented in the horizontal plane. It has the characteristic major and minor lobes, with nulls at approximately the 90-degree points. Figure 25 is our vertically oriented Yagi pattern. It's a little more broad, but with greater null values.

Figure 26 is RCA's pattern for its CP receiving antenna, measured in a horizontal field. Although the pattern is more broad, due to the lower gain, it looks very much like that of the Yagi, and its half-voltage beam width is about 80 degrees.

Figure 27 is RCA's pattern for the vertical reception of its CP antenna. Its beam width is only a little less.

But Figure 28 shows the pattern we measured for the CP receiving antenna in a CP field. (Actually, we should call this a plot of voltage delivered by the antenna, since there is a distinction.) The beam width

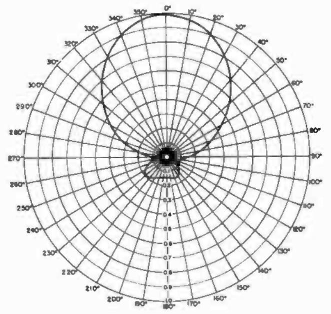


Figure 26.

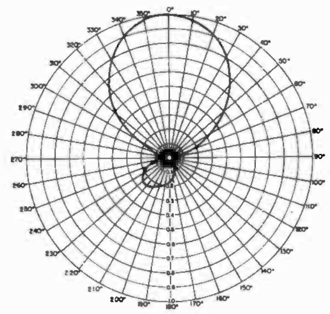


Figure 27.

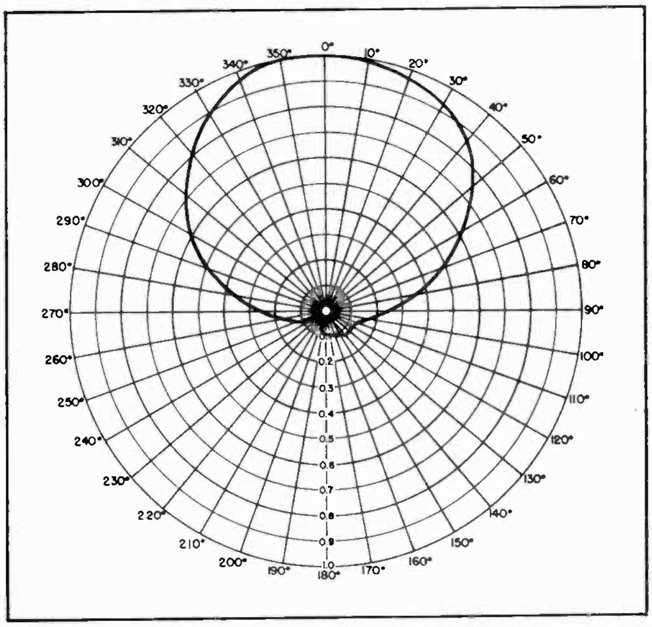


Figure 28.

is increased to 105 degrees, and the deep null is at the back rather than at the sides.

This demonstrates the unusually high front-to-back ratio which we believe will minimize co-channel interference when both stations employ circular polarization. The combination of a broad front lobe and a very deep null in the opposite direction should permit the "nulling out" of co-channel interference quite effectively. It also makes it easier to establish a compromise antenna orientation where the desired-channel transmitters are not co-located.

This seems important to us, and we include a series of pictures which demonstrates this phenomenon. At one particularly good location, we made the series in Figure 29 by observing picture quality as the receiving antenna was rotated in 20-degree increments away from the optimum orientation.

[All television photographs in this report are black-and-white reproductions of the color slides shown during the oral presentation of this paper. Unfortunately, in some cases this process has resulted in the loss of certain picture detail.]

It may be seen that an excellent picture was obtained with either combination of transmission/reception when the receiving antenna was properly oriented. Further, significant picture degradation did

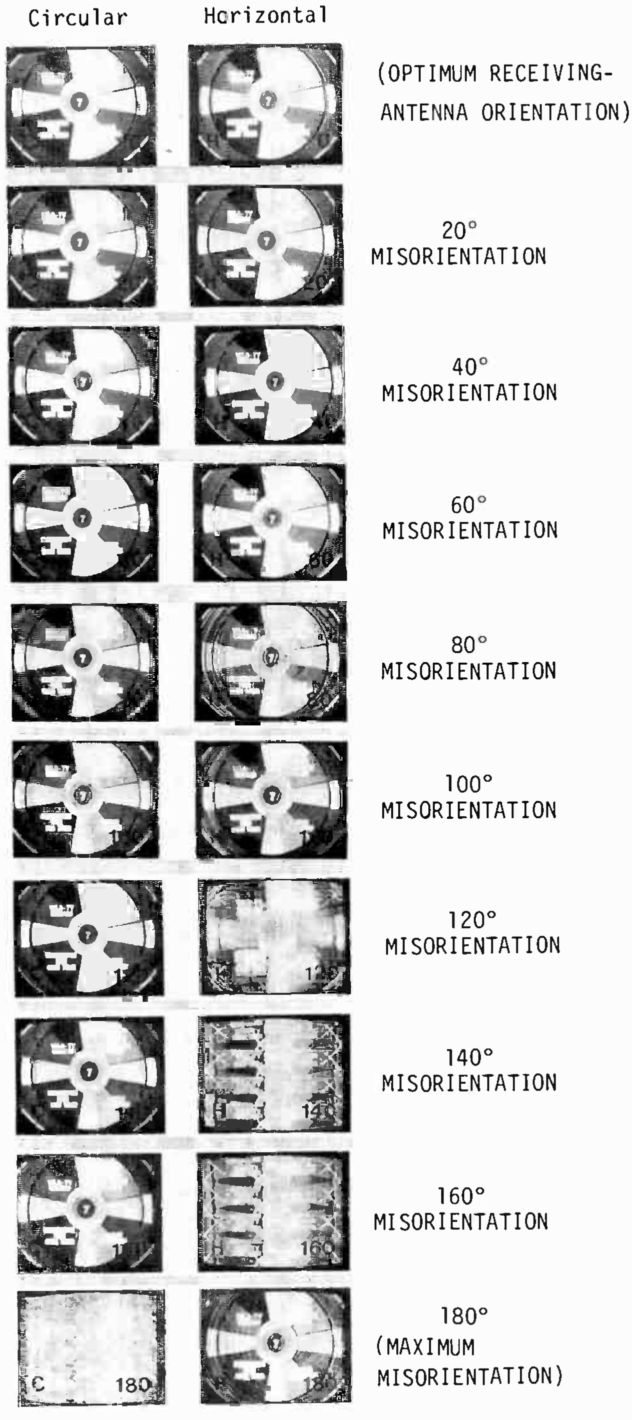


Figure 29.

not occur when either receiving antenna was misoriented by a modest amount. However, when the misorientation approached 90 degrees, the H-H reception became severely degraded, while the CP-CP reception remained acceptable although no longer perfect. As the receiving antenna was misoriented farther, the H-H picture remained unviewable, with a loss of synchronization, while the CP-CP picture held its own. It was only when the antenna was misoriented by 180 degrees that a semblance of an H-H picture was regained (due to the substantial pattern back

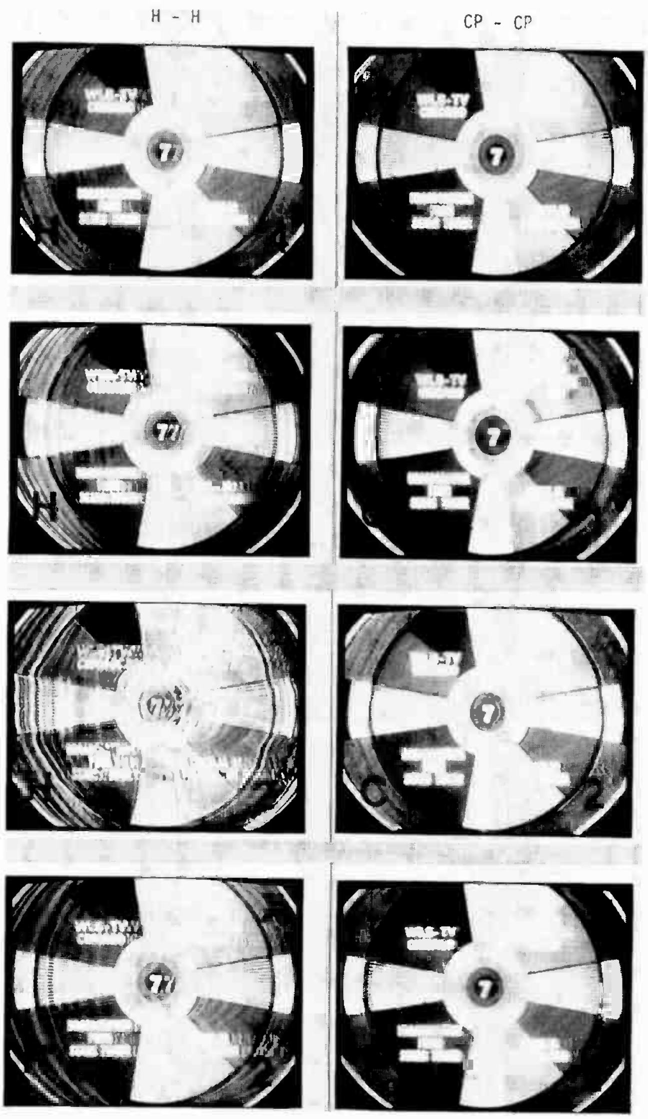
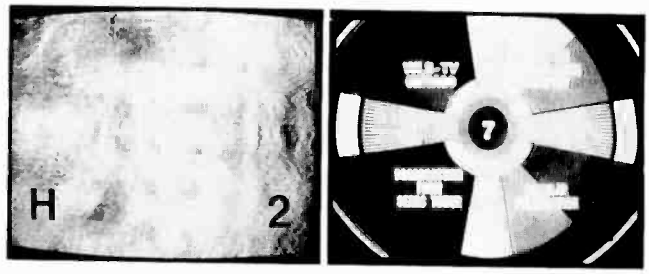


Figure 30.

lobe), while the CP-CP picture simply disappeared. Thus, with CP-CP operation, a drastic misorientation of the receiving antenna has little impact on reception where a reasonably strong signal is available. Those viewers with antenna-orientation problems may therefore benefit markedly by the institution of CP-CP reception.

The final part of this report involves the reduction in ghosting which can be obtained by circular polarization. Figure 30 is a series of pictures taken at several locations in an area where we had observed substantial ghosting during horizontal transmission.



WORST EXAMPLE OF H PICTURE OBTAINED AT ONE LOCATION

WORST EXAMPLE OF CP PICTURE OBTAINED AT SAME LOCATION

Figure 31.

Since reflections arriving from behind a receiving antenna cannot be rejected in the way that reflections from the front can, it is not to be assumed that CP-CP operation will result in consistently perfect pictures. In the area in which these particular tests were made, the streets are lined with buildings, and very strong combinations of ghosts were present. We reasoned that if CP-CP operation could cure *severe* ghosting to a reasonable extent, it could make similar *relative* improvement anywhere. It will be noted that in each case shown, an improvement in picture quality equal to at least one TASO picture grade was obtained.

Because it was only recently received, we have not been able to properly analyze the JFD antenna, but our limited tests show it to perform much like the RCA antenna, as shown in the bottom row of Figure 30.

In order to end with a flourish, we show two final pictures. These were taken while we moved the truck around, a few inches at a time, at one particular location and observed the changes in picture quality. We show (Fig. 31) the worst horizontal picture we found, as well as the worst CP-CP picture obtained at the same location.

The worst CP-CP picture is obviously not very good, but it's better than nothing, which is what the worst horizontally polarized picture provided.

Theory suggests several advantages to circular polarization in television broadcasting, and the necessary hardware is presently available. Although additional studies are to be made, the observations to date tell us that the system works.

Questions and Answers About Circular Polarization

Following the presentations, authors of the three papers answered questions from the audience. The exchanges follow:

Q: I was concerned about the effect on the predicted fields . . . in other words, the coverage you would expect between horizontal and CP. Based on this point, that in a horizontal dipole with a horizontal field, the voltage going through zero and peak twice each hertz, the effective voltage is the RMS. But with a CP receiving antenna and a CP transmitting antenna, the effective voltage is constant at the peak value.

Dr. Siukola:

Yes.

Q: It would seem to me that the effective field really is greater, so that the predicted contours would be further out. If we're to maintain the same predicted coverage with CP, the power ratio—total power would be that same ratio as the RMS to the peak on a sound wave basis.

Dr. Siukola:

I am not very sure how to answer that. It appears to me that we have a matter of how we measure what we are receiving. If we are looking for the signal with a horizontal polarized antenna, then obviously the signal for the same ERP should be the same, except for the part of the D polarization.

Now, if you look at a signal, as Neal pointed out, with a circularly polarized receiving antenna, then the terminal voltage will be higher by 3 db, but not the field.

Q: If the terminal voltage is higher, it would seem to me then that that means you receive the signal further out. . .

Dr. Siukola:

With a circular polarized antenna, I wouldn't be surprised that is true. For horizontal, probably not.

Q: Mr. Siukola spoke briefly about receiving antennas and broad-banding them to preserve the circular polarization characteristic. I note that most of these tests, or all of them, have been on a single channel or single frequency basis. And I wonder if we can get further enlightenment about how to handle our home reception where channels 2 through 13, at least in the VHF band, have to be received with a minimum amount of hardware on the roof.

Dr. Siukola:

I was trying to illustrate some of the principles which might be used. Let's take a very simple case. There are now, for instance, 2-to-13 yagi type of antennas. In principle, one could put two of these crosswise and provide a proper 90 degree phase shift, and that's where the trick is.

In other words, it may require a two-portion antenna, for instance. Hybrids have a phase shift, you know.

I don.t think these kind of things have been considered and worked out yet, but that is one possibility. Another thing is a spiral type of antennas. They are very broad. There might be another technique, the conical spirals.

As Neal used in his measurements, for instance, they are just custom-built for that specific case, and no designs have been made. That's why my pictures were also only illustrations.

Mr. Smith:

The JFD antenna, of course, is designed to be broadband over 7 through 13, but unfortunately, we only have channel 7 to look at, so we can't make any observations as to how well it works on channel 13.

Q: First a comment and then a question. In 1947 I was associated with Carl Smith in the development and test of the first circularly polarized FM antenna. We made a lot of measurements inside residences. We could transmit either vertical or horizontal fields. And we found that the wiring and plumbing does an excellent job of ellipticizing the wave. So that once you're inside a residence, it's pretty hard to find if you're getting the advantage of circular polarization or not, except possibly the power increase.

So the question is: Are you planning to go inside residences to see if rabbit ears or whips really give you the advantages?

Mr. Smith:

Yes, we're going to do that. The problem comes in how do you systematize that. One of the thoughts we had, is to get a group of people and time them as they adjusted their rabbit ears to see how long it took them to get a decent picture, and then try it the other way. It's difficult, you know, to put the scientific method into this kind of thing. But we are going to make some observations. That's on our list of things to do.

Q: I'm sure many of you have observed, as I have, that in areas of high reflection ghosting, like apartment areas, a portable set with a vertical whip antenna gives a much cleaner picture, many times, than a horizontal rabbit ears. On horizontal polarization, when you were making your picture observations, did you compare a circular receiving antenna against a horizontally polar antenna?

Mr. Smith:

Yes, we did. In some situations we would use the circular antenna for our horizontal reception. And in other situations we would use a horizontal receiving antenna.

I know what you mean about using a vertical element and very often beating the laws of physics with it. But we've done it both ways.

Q: On a horizontally-polarized transmission in a bad ghosting area, did you compare the reception between using the circularlly polarized receiving antenna and a horizontally polarized receiving antenna?

Mr. Smith:

Yes we did.

Q: I'm trying to justify why in apartment areas a single whip gives you a much cleaner picture than a horizontally polarized rabbit ears.

Mr. Smith:

We did make the observations that you're talking about and we do get a much cleaner picture, we get rid of ghosts by using circular transmission and circular reception as opposed to horizontal.

Mr. Zellner:

I think your question is basically just how easy or with what, if any, difficulty there is to adjust a set, let's say, with rabbit ears under the different modes of polarization.

We have made some of these observations, where the station was transmitting, let's say, in the horizontal mode to a receiver with rabbit ears. And we went through an adjustment procedure as to just what the position of the rabbit ears would have to be to produce a satisfactory picture. Then we, conversely, did the inverse of that and transmitted in the circular polarization mode. And we found in a number of cases, or most all cases, I should say, that the antenna still had—the rabbit ears still had to be oriented for optimum picture, but the difficulty and the number of different angular positions of the rabbit ears that you had was much easier when we were in the circular polarization—circularly polarized mode.

Does that answer your question?

Q: Not quite.

Moderator:

We're running a little behind time, and I have a question over here.

Q: I think the gentleman is making a very important point, and really his question is: If we have horizontal and circular polarized transmitting antennas and one uses a circular polarized receiving antenna, is he going to get some improvement from horizontal transmissions in this multiple-reflected environment?

Mr. Zellner:

Yes, there was a slight improvement in that situation. In other words, where you were using a circularly polarized receiving antenna and the station was transmitting in the horizontal mode, yes, there was a slight improvement. We have data to support that.

Moderator:

We are running a little behind. I do want to make one additional point which I don't think was covered. ABC has submitted a petition for a rule-making, seeking amendments to permit the use of vertical polarization for television. The announcement was made on March 4th. The statutory 30-day time limit is over, but I know in the past the FCC has accepted comments even after the 30-day limit, particularly on petitions.

What the next step will be, of course, is up to the Commission, whether it will issue a notice of inquiry or a notice of proposal making, and the timetable, of course, is up to the Commission.

Did you have another further comment?

Mr. Zellner:

Yes, just one other thing. Jampro on the West Coast is putting up a CP antenna where there'll be further measurements made and further data compiled and hopefully filed with the Commission. We at ABC are planning another series of measurements to further support our technical position, and this data will also be filed with the Commission.

Also, in regards of comments, any technical data which you can support, as broadcasters, should be filed with the Commission in support of this petition.

John L. Windle

Chief Engineer Stainless, Inc.
North Wales, Pa.

Structural Considerations When Changing or Adding Antennas on Existing Towers

In recent years there has been a substantial increase in the desire to replace or add antennas to existing broadcast towers. Since each tower has been designed for a specific arrangement of antennas and lines, it is essential that its structural adequacy to support the new loads be properly evaluated.

The purpose of this paper is to present the structural considerations that should be investigated. While many no doubt are familiar with much of the material, it is hoped that its presentation will increase the appreciation needed in the sensitivity of towers to changes in antenna loading.

Loads Imposed on Towers

In addition to the dead load, or weight, of a tower and the equipment it supports, a tower must be designed to resist the forces of nature caused by wind and ice.

The determination of the magnitudes and directions of loads from these forces is defined by various codes and standards, the most common of which is Electronic Industries Association Standard RS-222, "Structural Standard for Steel Antenna Towers and Antenna Supporting Structures."

These loads produce bending moments, transverse shears, torques, and axial forces on the tower structure. By applying the principles of structural mechanics, these may be resolved into axial and shear forces and bending and twisting moments on the individual structural elements.

When the face bracing system is continuous and there are no significant eccentricities in the connections, the only appreciable stresses in the leg and bracing members are axial.

These may be tension or compression depending upon the direction of the wind load.

Tower Types and Structural Models

There are two basic types of tower structures—self-supporting and guyed.

Self-Supporting Towers

A self-supporting tower may be described structurally as a cantilevered space frame or truss. Its principal structural elements are the vertical legs, each of which is common to two adjacent faces, the web bracing in each face, diaphragm bracing at levels of torque loading and the foundations.

The potential modes of failure are buckling of individual leg or bracing members under compressive loads and shear or tension failures of the connections.

Most self-supporting towers are designed with isolated foundations for each of the legs. These must be designed to resist both vertical uplift or download combined with horizontal shear. In addition to the internal stresses on the foundation itself it must be proportioned to prevent excessive bearing pressures on the soil which would result in unacceptable differential settlements of the individual foundations.

Guyed Towers

A guyed tower may be described structurally as a continuous beam-column on elastic supports. Its principal structural elements are the vertical legs, each of which is common to two adjacent faces, the web bracing in each face, the guy cables, the tower base foundation and the guy anchors.

The potential modes of failure are buckling of individual leg or bracing members under compressive loads, rupture of bracing members, guys or guy anchor arms under tensile loading, and shear or tension failures of the connections.

The foundation at the base of a guyed tower must be designed for vertical download combined with horizontal shear and any overturning moment. In addition to its internal stresses it must be proportioned to prevent excessive bearing pressure on the soil which would result in unacceptable settlement.

The guy anchors must be designed for the combination of vertical uplift and horizontal shear forces imposed by the guy cables. They must be proportioned to prevent excessive bearing pressures on the soil which would result in unacceptable sliding.

A guyed tower is a much more complex structure than a self-supporting tower. Whereas there is only one basic path through a self-supporting tower for the loads to be transferred to the ground, there are several for a guyed tower. The distribution of the loads among these paths is dependent upon the relative stiffnesses of guy systems and the tower shaft.

Each span of the tower has a stiffness with respect to axial and shear forces and bending and torsional moments. These stiffnesses are a function of several variables including the geometric configuration, mechanical properties and sizes of the individual members.

Each level of guys has a stiffness with respect to translation and twisting of the tower at the guy attachment level. These stiffnesses are a function of the geometric configuration, the mechanical properties and size of the individual guy cables, the amount of initial tension, the magnitude of ice load and the magnitude and direction of wind load on the cables.

By evaluating all of these it is possible to simulate all the guys at a given level as a spring having a specific stiffness. Because of the non-linearity of some of the relationships involved, the spring constant derived is only valid for a specific set of conditions and for a finite range of translation. Similarly a torsional spring constant can be derived. It is interdependent with the translation stiffness and is also valid for only a finite range of translation.

Another difference between a guyed and a self-supporting tower is the magnitude and significance of the axial load. For a self-supporting tower this is composed only of the gravity loads from the tower, its appurtenances and any ice load. It is independent of wind load and its effects on individual member loads are relatively small. The axial load for a guyed tower includes in addition to the gravity loads, the vertical components of the tensions in the various guys. Since these tensions are directly affected by the wind loads the axial load is now dependent upon wind load, and its effects on individual leg members are relatively large. It also produces an additional bending moment on the tower equal to the product of the axial load and the deflection of the tower.

Despite the complexity of the relationships involved, the availability and wide spread use of digital computers permits accurate structural analysis of guyed towers.

Effects from Top-Mounted Antennas

A top-mounted TV or FM antenna imposes a horizontal wind shear, an overturning moment and an axial download on the tower. A conclusion that would appear to be logical is that the larger the antenna and consequently the larger the loads imposed on the tower, the greater will be the stresses on the tower elements.

As a corollary, one might conclude that if a tower has been properly designed for a given antenna, replacing that antenna with a smaller one would result in lower stresses on the tower elements, and therefore no overstresses would occur as a result of such a replacement. This may not be true, particularly for a guyed tower.

As load is reduced at the tower top, the deflection pattern of the tower is changed, and the distribution of bending moments and shears and resulting loads in the tower members are altered substantially. An example of such a change in loading is shown in Figure 1.

In this case, even though the new antenna wind loads are less than half of those for which the tower was originally designed, the maximum load in a vertical leg in the top span of the tower is increased by more than thirty percent.

There are several ways of correcting this situation:

1. Add dummy loads to the antenna to obtain total shear and overturning moment approximately equal to those for the original antenna.
2. Replace the overstressed legs with new, larger members. This would require dismantling a considerable portion of the tower and would be very costly in both dollars and off-air time.
3. Relocate the top level of guys to a lower elevation. This would require field installation of guy connection fittings on the tower and resocketing of the top level of guys. Again the cost would be high.
4. Modify the stiffness of the guy system; specifically decrease the stiffness of the top level

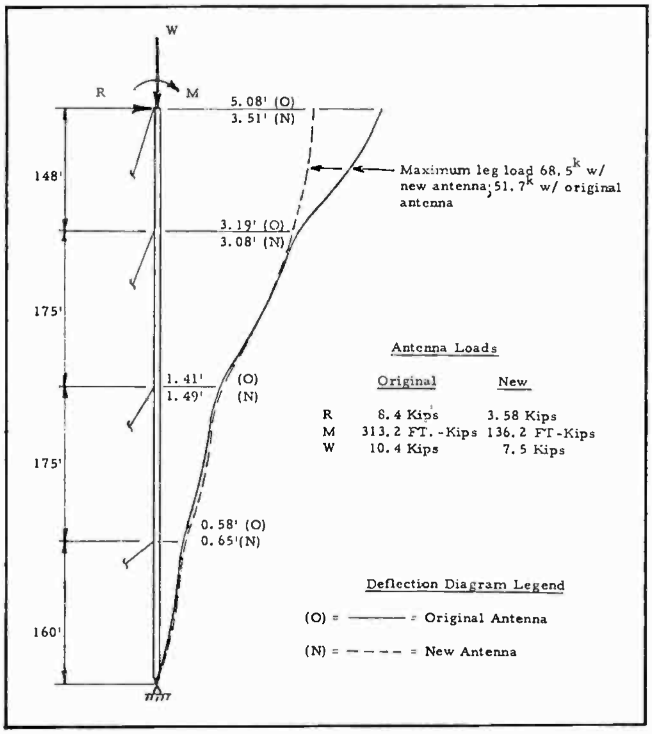


Figure 1.

and/or increase the stiffness of the next level down.

This is the most effective method and is worthy of further discussion.

The two variables affecting the stiffness of a guy set which may be changed are the initial tension and the size of the cables. Of these, the changing of initial tension is obviously the easier and less costly. However, there are limits for establishing initial tensions. Consequently, the effects on guy system stiffnesses of changes in initial tensions may not be sufficient. It is then necessary to change the size of the cables.

In the example given it is actually desirable to replace the top set of guys with smaller cables in order to prevent overstresses in the tower legs.

Effects from Side-Mounted Antennas

Many broadcast towers, in addition to the primary, top-mounted antenna, support some side mounted antennas. These may include additonal TV or FM antennas, microwave antennas or reflectors, and two-way communications antennas.

Usually when the tower is originally designed, the exact type and required location of these antennas are unknown. To provide for their future installation an anticipated arrangement of equipment is included in the specifications. However, when the time comes to install additional antennas there are frequently substantial deviations from the anticipated arrangement. The effects of these deviations must be investigated.

Each side-mounted antenna imposes a horizontal wind shear, an axial download and, depending on the nature and location of the antenna, moments in either a vertical or horizontal plane. Just as for top-mounted antennas, it is tempting to conclude that as long as these loads and moments are less than those for which the tower was originally designed, there will be no detrimental effects. A related conclusion often made is that an antenna may be mounted at any location on the tower provided it is not higher than in the original design.

For a self-supporting tower these are generally valid assumptions. However, for a guyed tower any change in magnitude or distribution of wind load will result in changes in the deflection and the distribution of stresses within the tower. Therefore, neither of these assumptions is necessarily valid.

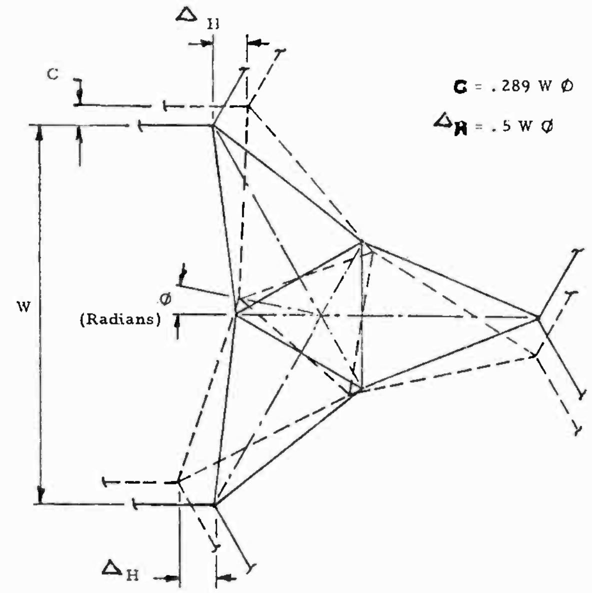
A significant additional consideration when locating side-mounted antennas is the magnitude of torsional moment. Torques cause additional loads in the bracing members and twisting about the vertical centroidal axis. Once again these effects are most significant for guyed towers.

While both the bracing members and the guy system provide resistance to torque loads, the guy system is the more effective. When these loads are relatively small no special arrangement of guys is required. The torsional resistance of such a guy set is the product of the sum of the horizontal components of guy tensions and the chord distance of the rotation of the guy attachment points on the tower. When torque loads are large, such as for microwave antennas or TV antennas offset from the tower a considerable distance, the torsional stiffness of a conventional guy set is frequently inadequate. In this case it is necessary to use two guys in each of the guy directions and attach them to frames extending from the face of the tower rather than directly to the tower legs. Now when the tower twists the horizontal span is increased for one guy of each pair and decreased for the other. This results in differences in tensions in the two guys, thereby creating an additional resisting torque.

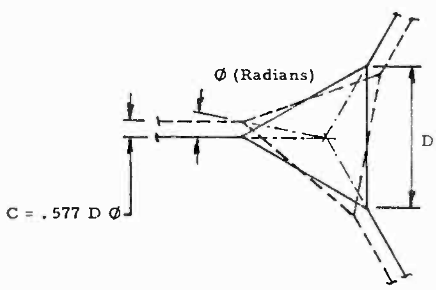
Diagrams of the two arrangements of guying are shown in Figure 2.

A basic principle to keep in mind is that the most effective way of controlling twist is to minimize torque. For microwave antennas and reflectors this can best be accomplished by designing the mount so that the line of sight path passes as near as possible to the centroidal axis of the tower.

Since it is not possible to eliminate torque loads entirely, it is essential to locate antennas imposing them near those guy levels having the greatest torsional stiffnesses.



(a) Double Guy Arrangement with Torque Stabilizing Frames



(b) Standard Guy Arrangement

Figure 2.

Effects from Lines

Each antenna mounted on a tower requires an RF transmission line and possibly an AC power circuit for deicing equipment. These lines impose additional uniform wind and dead loads along their entire length on the tower. For some antennas such as FM and two-way communications, the effects of these lines are far greater than those of the antennas themselves. This is particularly true when the antennas are mounted near the top of a tall, guyed tower.

When microwave passive reflectors are being replaced by antennas, the effects of the lines feeding these antennas must not be overlooked. Even though the total load of an antenna and line might not exceed that of the reflector it replaces, it is distributed in a different manner. On a guyed tower this change in distribution could be significant.

It is just as important to consider any changes in the quantity, size and extent of lines on a tower as it is to consider changes in antennas.

Summary

In the preceding discussion we have attempted to emphasize the sensitivity of broadcast towers to changes in antenna loading. It was not intended to infer that such changes cannot be made, but rather that they should be made only after a thorough investigation of their effects on the structural adequacy of the tower.

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A. H. Lind

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Utilization of CCDs in Color TV Cameras

The solid state image sensor of the charge coupled device (CCD) type (Fig. 1) is markedly different from the classic beam scanned pickup tube in its manner of operation. Its small size, "flat" shape, and light weight excites one's imagination when thinking of the potential impact on the design of compact, highly portable color TV cameras for broadcast service.

The CCD SID-51232, as announced by RCA, is a self-scanned solid state image sensor which contains an array of 512x320 information storage sites. These storage sites are arranged in two areas of 256x320 each. As shown in Fig. 2, one area is exposed to light and becomes the image sensing area. It is slightly larger (12.2 vs 12.01 mm diagonal) than the projected picture area in a frame of 16 mm film. The second area, which has a corresponding site for each of those in the image area, is a storage area which is screened from light.

The charge pattern developed in the image area is transferred by suitable clocking circuits to the storage area during each scanning field vertical blanking period. To obtain the real time output video signal, the charge pattern in the storage area is transferred a line at a time into the horizontal register during horizontal blanking intervals. During the active scan period, the information is clocked out at a 6.1 MHz picture element rate to provide the normal 525 line real time video.

Each storage site (or elemental cell) consists of three adjacent gate electrodes which are stacked vertically. By proper selection of the voltages applied to the three gates on alternate fields it is possible to accomplish interlace to achieve the full 512 lines of image information per frame of scan.

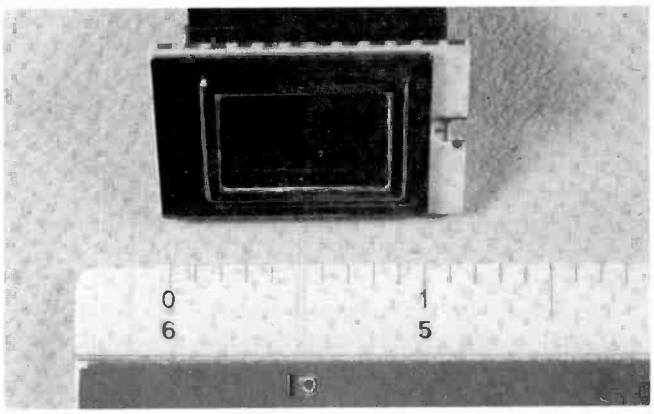

Figure 1

In order to be a good candidate for a color camera sensor, a pickup tube or CCD must satisfy a series of key requirements. These include resolution, sensitivity, spectral response, geometry, registration and lag. Others are the transfer function characteristics—highlight overload and color tracking—and signal-to-noise characteristics—dark current and spurious synchronous.

While some items are more basic than others and in many instances the level of performance becomes a matter of competitive trade-offs among several tubes or CCDs, they are all important in establishing the degree of success of the camera.

Although it has been argued by some that the 4.2 MHz video bandwidth limits the luminance resolution

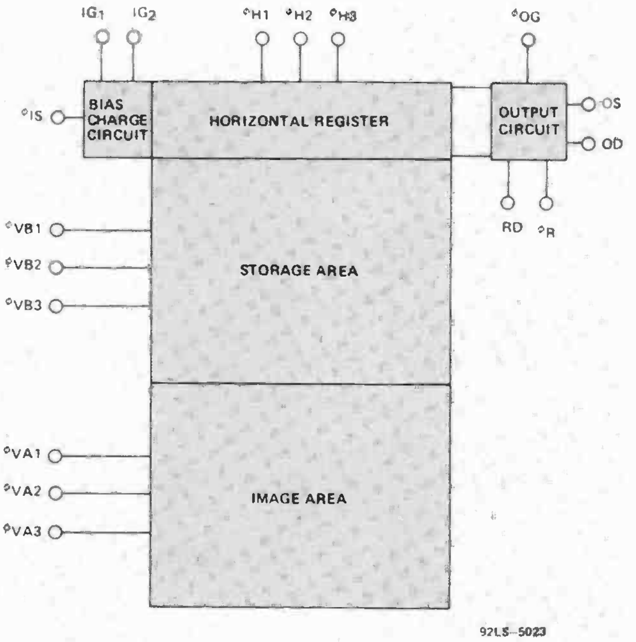

Figure 2

transmitted through the US NTSC system to 330 TV lines and/or the 3.58 MHz color subcarrier puts a further practical limit on resolution to approximately 280 TV lines, it has been generally accepted in the industry that substantial response from pickup tubes is required as far as 400 TV lines, with further enhancement possible with aperture correction circuits. This permits several generations of signal processing with a margin of safety that assures the final signal delivered to the transmitter to be the maximum in sharpness and resolution permitted by the transmission system.

The sensitivity of a broadcast camera should be such that a full rated output signal (*signal-to-noise, resolution, etc.*) can be achieved with less than 50 ft. candles of illumination on the scene.

The pickup devices must, of course, be sensitive to light throughout the visible spectrum. Compromises in colorimetry and effective camera sensitivity are usually related directly to the spectral performance of the pickup sensors.

Geometry and registration are closely interrelated. Registration accuracy is related to the consistency of the geometry of the sensor system among the three channels where basic geometry relates to good engineering practice of permitting little distortion of shapes in the reproduced picture.

Lag is a well known residual effect in photoconductor tubes that produces a trailing smear on motion which reduces dynamic resolution because of a short term memory in the photoconductor that prevents the charge pattern from being totally removed in a single pass of the scanning beam.

The transfer function describes the behavior of the electrical output of the sensor as a function of the optical input. Comet-tailing is a frequently encountered effect with PbO tubes when excessive heights (*speculars*) are encountered in the scene. The other basic problem is that of the three channels tracking over the full dynamic range of the camera while providing the chosen gamma (*transfer curve shape*).

Dark current may be the controlling variable that establishes signal-to-noise performance, but other spurious signals such as high rate sampling pulse crosstalk may set the effective noise level of the camera.

There are other key characteristics. Shading, for example, with sensitivity and dark current uniformity. Also spurious signal effects, image defects, size, weight and power, plus two scanning standards—525-line, 60-field and 625-line, 50-field.

Shading resulting from sensitivity variations or other additive signal discrepancies over the raster area present the biggest problem when the discrepancy occurs differently in the three channels since this results in a differential color error. Differential color errors are more readily apparent at signal black level.

Non-image related patterns such as mesh beat, or noise due to beats between the sensor cell pattern and image structure patterns are sometimes observable.

Image defects are usually spots and/or blemishes that are present in or on the photo sensitive area.

Scanning standards for broadcasting worldwide have not in general been a problem with beam scanned pickup tubes since the photolayer is continuous and homogenous over the scanned raster area. CCDs are discrete sensor cell arrays and thus must be made to be compatible with the scanning standard employed.

While size, weight, and power are not fundamental to making a good color picture, the desire for very compact, battery powered cameras make these characteristics very important considerations.

Before discussing the current performance of the SID-51232 CCD against the list of key characteristics, Figure 3 illustrates the picture capability of the CCD at the present time. It is a monochrome version of a color slide made from a color kinescope RGB display from three sensors.

It is perhaps better than one would expect to see from the imaging sensors at this stage of development. However, our current assessment of the resolution capabilities of the CCD sensor is that it is inadequate for broadcast purposes.

Vertical resolution is basically satisfactory, since there is a line of sensing elements for each active scanning line in the 525 line raster. However, in the horizontal direction there are only 320 elements per line. This results in only 240 elements per picture height.

In addition to the number of elements, a very important factor that establishes the effective aper-

Figure 3

Figure 4

ture response is that of charge transfer efficiency. Since the basic operation of the device involves the transferring of each charge packet from element to element until it arrives at the readout point, it is inevitable that there will be some loss of detail contrast since the transfer process is not 100% efficient. This loss of charge reduces the detail contrast and therefore affects aperture response.

Unlike a beam scanned tube, where the information is scanned along a television line in a continuous fashion, the CCD utilizes discrete sampling along the line as well as in the vertical direction. Such discrete sampling can give rise to beat effects which result when information density exceeds the sampling distribution. Should this happen, false information is generated. In this new technology, this effect has been called "aliasing". It is similar in effect and cause to the moire pattern which most of us are familiar with in looking at the horizontal resolution wedge in the test pattern.

Figure 4 shows a picture reproduced through a single channel CCD pickup. The effect of aliasing is quite pronounced. While there is an apparent sharpness to the small lettering, any attempt to read the letters is frustrating because the detail information is scrambled.

In Figure 5 you'll note in a listing of the key characteristics that the resolution is rated as a limitation for the CCD.

The second item of sensitivity is also assessed as being a limitation in the case of the CCD. Camera sensitivity is generally limited by one or a combination of factors such as noise level in the video amplifier, lag in the image sensor, spurious background signals generated in the sensor such as dark current, and the spectral sensitivity characteristics of the device as applied to a color camera.

The number three characteristic, spectral response, is also rated as a limitation at present. The CCD device is basically a silicon type of sensor which has an

KEY CHARACTERISTICS FOR COLOR CAMERA SENSORS

Resolution	Limitation
Sensitivity	Limitation
Spectral Response	Limitation
Geometry	Good
Registration	Good
Lag	Good
Transfer Function:	
Highlight Overload	Good
Color Tracking	Good
Signal-to-Noise:	
Dark Current	Acceptable
Spurious Synchronous	Acceptable

Figure 5

abundance of response in the infra-red, but is difficult to make with adequate response in the blue region of the visible spectrum. A major portion of the inherent sensitivity of the material is thrown away in a color camera when the infra-red response is filtered out.

The next two characteristics of geometry and registration are closely allied and depend very much upon the basic geometric construction of the sensor photo element pattern. Since this pattern is established by utilizing integrated circuit mask techniques, the uniformity from sensor to sensor is expected to be, and appears to be, extremely consistent. This makes both the geometry and the registration of the three sensors with respect to each other a very precise achievement when the proper mounting arrangement is used to position the sensors.

Since there is no electrical size or centering capabilities as is the case with beam scanned tubes, the sensor image areas must be positioned by mechanical means. Further, since there is no size adjustment of

OTHER KEY CHARACTERISTICS FOR COLOR CAMERA SENSORS

Shading:	
Sensitivity Uniformity	Acceptable
Dark Current Uniformity	Limitation
Spurious Signal Effects	Limitation
Image Defects	Acceptable
Size	Good
Weight	Good
Power	Good
Scanning Standards:	
525 – 60	Acceptable
625 – 50	Limitation

Figure 6

Figure 7

the scanned raster readily available, the optical image size in all three channels must be identical. This imposes a requirement on the optics of the camera which has not been stringent in the case of beam scanned pickup tube cameras since some adjustment of size, linearity and centering is quite readily accomplished.

The characteristic of lag, which in beam scanned tubes is an ever-present limitation, is non-existent in the CCD device. This is a fundamental requirement which must be met in order to make the device operate. Since it uses a charge transfer mechanism it is essential that close to 100% of the charge packet in a given elemental cell must be transferred to the next one in the course of operation; thus lag—practically speaking—is non-existent. This is a very good characteristic of the CCD.

Next, with respect to the transfer function, the SID-51232 has channel stops which completely surround each sensing site to prevent charge spreading, in the event of excessive highlights in the optical image. This anti-blooming function appears to perform very well. The transfer function of the sensor is also quite consistent, sensor to sensor, so that tracking among the three color channels over a large dynamic range appears to be very good.

The dark current in the CCD tends to be a substantial portion of the output signal current. This is partially a result of the silicon used in the sensor and also the presence of a dark current component which is called the "fat zero" level which is, by design, one of the parameters used for control of the charge transfer process. The second characteristic of spurious synchronous signal components, such as sampling pulse feedthrough, is not present to any appreciable degree in the SID-51232. In general, with respect to signal-to-noise, the CCD when operated at its optimum point, provides acceptable signal-to-noise ratios.

Figure 6 shows a continuation of the key characteristics list. The performance of the current SID-51232 with respect to sensitivity uniformity is generally acceptable. However, there remain uniformity problems at the dark current level which can lead to differential low-light color shading problems. Also, a limitation of the present devices is the spurious signal effects such as the aliasing or moire effects that have been referred to earlier.

Although there was great apprehension that the problem of fabricating CCDs with hundreds of thousands of sensor cells would be extremely difficult to accomplish without having a multitude of spots, blemishes, and similar defects, it has been the result that, through very careful preparation, sensors have been made with relatively few observable spot defects.

The present sensor with a limitation of 512 lines of raster scan is limited basically to a 525-line 60-field scanning system if one is to present a full raster of scanned information. For 625-line 50-field systems, this answer does not adequately fulfill the scanning raster requirements. One can, of course, fill as much of the 625 line scanning raster as can be done with 512 lines, but we feel this is generally unsatisfactory. The size, weight, and power of the CCD are perhaps its most attractive characteristics. It is obviously much smaller and much lighter than beam scanned tubes utilizing a comparable image size, and the power requirements for it are substantially reduced.

The alternative sensor for a small broadcast color camera is the 2/3″ beam scanned pickup tube. Figure 7 is a monochrome of a color kinescope picture made from a camera utilizing 2/3″ beam scanned tubes. In this case the color picture was NTSC encoded.

A comparison of the same scene made with the RGD CCD sensors (Figure 3) and the beam scan picture (Figure 7) show the relative general picture quality of the two approaches.

If we now look at the key characteristic chart, with this time an assessment of the CCDs and 2/3″ beam scanned tubes, Fig. 8 one can quickly see an overall comparison of the two as well as an assessment of the 2/3″ tubes for their possible utilization in a color camera.

KEY CHARACTERISTICS FOR COLOR CAMERA SENSORS

	CCD	2/3″ Tube
Resolution	Limitation	Good
Sensitivity	Limitation	Good
Spectral Response	Limitation	Acceptable
Geometry	Good	Acceptable
Registration	Good	Acceptable
Lag	Good	Acceptable
Transfer Function:		
Highlight overload	Good	Acceptable
Color tracking	Good	Good
Signal-to-Noise:		
Dark current	Acceptable	Good
Spurious synchronous	Acceptable	Good

Figure 8

OTHER KEY CHARACTERISTICS FOR COLOR CAMERA SENSORS

	CCD	2/3″ Tube
Shading:		
Sensitivity uniformity	Acceptable	Good
Dark current uniformity	Limitation	Good
Spurious signal effects	Limitation	Acceptable
Image defects	Acceptable	Acceptable
Size	Good	Acceptable
Weight	Good	Acceptable
Power	Good	Acceptable
Scanning standards:		
525 – 60	Acceptable	Good
625 – 50	Limitation	Good

Figure 9

First, it is apparent that no characteristic is assessed to be less than acceptable in the case of the 2/3″ tube. Resolution and sensitivity are indicated to be good, as are characteristics of low dark current, minimal spurious synchronous signals, good performance with respect to shading, and also good adaptability with respect to the scanning standards. All other areas are assessed to be acceptable. The performance in geometry and registration is not up to the performance capability of the 30 mm size tubes, but we have found that with careful design of deflection yokes it is possible to obtain registration that is acceptably good.

In the area of image defects, Fig. 9 spots and blemishes do tend to appear larger because of the smaller image size of the photo conductor and thus the relative size of the blemish is larger since the defect is probably mechanically the same irrespective of the size of the image on the photo conductor. The size, weight, and power requirements for a portable camera utilizing 2/3″ beam scanned tubes is undoubtedly larger in all respects than with CCDs. It is still, we feel within manageable limits so that a very acceptable and very useable camera can be designed utilizing small beam scanned tubes.

Figure 10

Figure 11

As a comparison, Figure 10 shows the aliasing reference picture (Figure 4) as reproduced by a 2/3" beam scanned tube. You can see that the lettering is much more readable when the beam scanned tube is used. The garbling of the fine lettering is essentially missing in the case of the beam scanned tube.

Figure 11 shows a color kinescope picture made, utilizing three 2/3″ beam scanned tubes looking at a general scene.

Although remarkable progress has been achieved in the development of solid state image sensors, the point of producing a broadcast color camera with performance competitive with small pickup tube sensors has not yet been reached. To fill the growing demand for small broadcast news gathering cameras that exists today, the designer's choice must be the 2/3″ pickup tube.

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Audio in Broadcast Television, an Overview of Current Techniques

Audio has come of age in the Television Industry. This has heralded the opening of a veritable pandora's box of techniques and tricks for audio signal processing which have hitherto been largely confined to the audio mastering and radio broadcast industries. With particular emphasis on magnetic recording, this paper will examine some of these techniques and highlight their applications in broadcast television.

There has been a rumor going around in Audio Engineering circles that Audio is no longer the Cinderella of the Broadcast Television Industry.

I am personally not entirely certain whether this has come about by osmosis with a very technologically active recording industry, or whether the extremely high quality of multiple generation video dubbing offered by the modern Quad VTR has so outstripped the audio as to be embarassing.

What is certain, however, is that more and more of the techniques, evolved and refined by the record mastering and radio broadcast industries, are being applied, piecemeal or suitably adapted, to the needs of the broadcast television industry.

It is perhaps pertinent, before reviewing some of these techniques, to examine some of the motivating forces which have brought this situation about.

Economics

The first and perhaps most powerful force is production economics. Whether one considers the problem of tying up two Quad VTR's to do post production audio sweetening or coping with the formidable task of doing justice to the audio for a high cost color musical show, the penalties of errors or lost time can be measured in dollars. Any process of technique which, for a given investment, permits an operator, (*whether a large color production house or a local station operator*), to maximize the utilization of equipment or minimize production time or cost, will draw attention from a purely operational economic standpoint.

Creativity

A second motivational force has been the desire of creative producers and artists to exercise more control over the total audio visual package delivered to the end user. This is understandable, since video as a communications medium has now matured.

Once the basic technological and artistic problems of controlling and adapting to a medium are overcome, a desire for increased sophistication is almost inevitable. The control which a producer today can exercise over the video image is enormous. The repertoire of effects which can be routinely called into play is sometimes overwhelming in impact.

However, the same advance has not always occured for the audio portion of the package. It has largely been in the recording industry that this creative flexibility has been honed to a fine edge. Small wonder that these techniques have in large measure been absorbed into the available arsenal of a video producers capability.

In addition there has been, (*also somewhat inevitably*), an increase in the sophistication of the consumer. The immediate novelty of electronic imagery in his living room has slackened, and he is now more responsive to the overall audio visual impact of a presentation.

In this sense, the use of modern audio techniques has a measurable value in terms of rating appeal.

Quality

A secondary spin off, resulting from the increasing sophistication of the technical application of modern audio techniques, is the opportunity for the broadcast television industry to materially improve the quality of the audio being transmitted. In Europe, this quality is actively sought by broadcasters to satisfy a generally more demanding viewer.

Technical quality in an audio signal often goes hand in hand with increased intelligibility, a not unimportant parameter in the delivery of a commercial message.

Quality for its own end is a difficult thing to quantize in economic terms, one can only reflect that perhaps the success of combined television and FM stereo presentations can in part be attributed to the pleasure of watching a superb color picture complemented by high quality stereo sound.

From this viewpoint then, there could be some measurable price tag on the benefits of quality on a television audio channel.

If the general quality of transmitted audio is good, the receiver manufacturer who responds to this, may well be surprised at the response given his product.

A typical multichannel mixing console.

Multichannel Mixing Consoles

The first to consider, the multichannel mixing console, is probably the most accepted in broadcast televison.

Here the flexibility afforded by having 20 or 30 or more inputs has it's biggest impact in live television shows. For example, the ability to close mike the orchestra permits good control of aural balance in what otherwise might be a very unfavorable acoustic environment. The ability to preset groups of microphones for each set relieves the need to frantically repatch and re-balance for the next shot.

In the recording industry the sophistication of these consoles has grown to the point where a record made from a recording done at a live concert, is difficult to distinguish, from a technical standpoint, from one made under the more ideal conditions of a recording studio.

A consequence of this is that the major recording artist making a tv show, need not fear that their particular "sound" which may well have been largely created in the recording studio, can not be very closely emulated in a live performance in front of tv cameras.

In addition, an artist may even bring their favorite mixing engineer to the tv studio to ensure similarity of their "sound" to that on released discs and tapes.

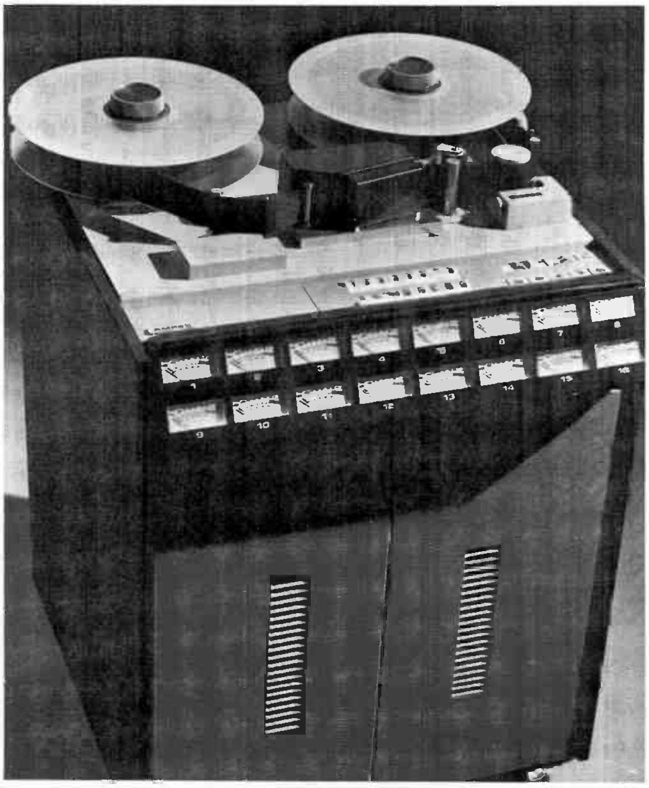
In this event the visiting engineer would feel more at home today than ever before in front of the mixing console in the tv studio, since it will have an extremely close familial relationship to the console in the recording studio.

This is of course not to deny that in some cases, for instance for active sound and dance routines, mining to a pre-recorded track may be more appropriate.

Probably the greatest impact which sophisticated mixing consoles give to the production house, is the ability to control the minutest segment of the audio and to be able to produce a far better and more appropriate sound. Because of this, the physical siting of an artist and musicians (for example) on the studio floor, becomes much more flexible, permitting a greater freedom for the producer, with less likelihood of undesired extraneous pick up.

The advent of lightweight high performance hyper cardiod or "shotgun" type microphones has played a significant part in keeping microphones off camera in wide angle shots. In the same vein, the greatly improved quality and reliability of radio microphones has also contributed to improved flexibility.

These last two improvements of course owe their genesis almost directly to the demands of tv production and to live stage shows.



The Ampex MM-1100-16 Mastering Recorder.

Multichannel Audio Recorders

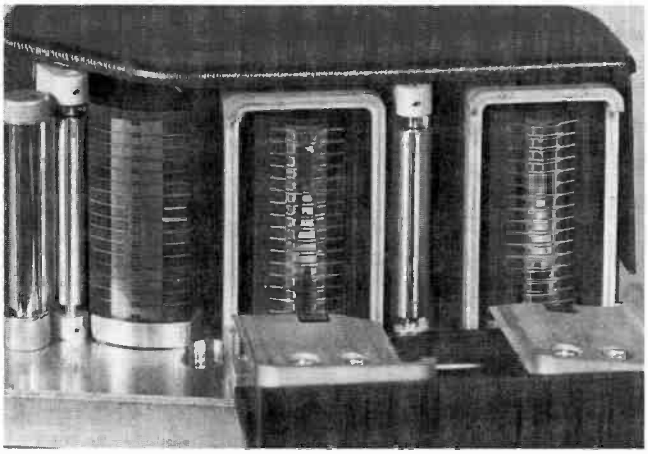
The second most basic tool in the record mastering industry today, is undoubtly the multichannel audio mastering machine. The industry standard density seems to have settled on 16 track on " tape or less commonly today, eight track on one inch tape.

The two-inch tape is, of course, formulated especially for longitudinal recording. (*Any video engineer who tries a modern two-inch audio tape on his Quad video tape recorder will find superb audio but rather scruffy video with many video dropouts and somewhat higher video noise*). Also in rapidly growing use today is the 24 track on 2" format and at least one manufacturer offers no fewer than forty tracks on 2" tape.

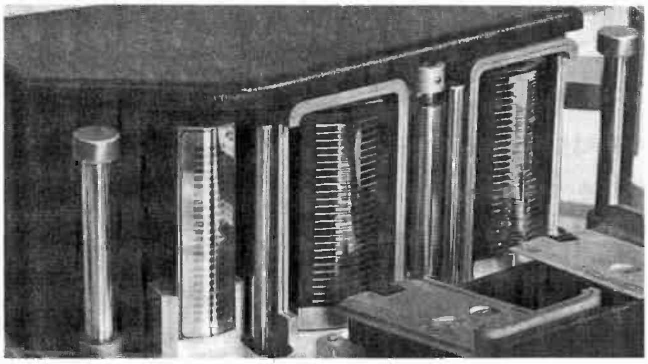
The latter two innovations have really only become practical in the last three years with improved tape characteristics. The 24 track machine of today using modern tape will have about the same performance as the 16 track machine of three years ago using older tapes.

The multichannel audio machine and the multi input/output consoles possess a rather chicken and egg relationship to one another.

The concept of either could have preceded the other but the outcome of the combination is an unprecedented flexibility in production. The producer and mixing engineer can lay down audio, one or more tracks at a time, building up to 16 or 24 tracks and then mixing them together as desired at any later time. This process is made possible by the technique known as "overdubbing".



Ampex MM-1100-16 Head Assembly.



Ampex MM-1100-24 Head Assembly.

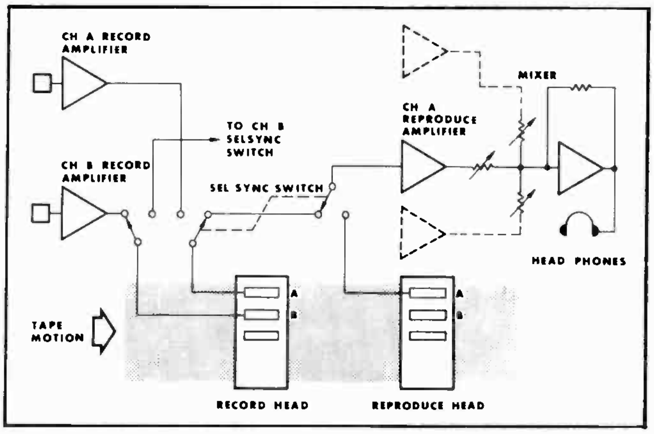


Figure 1.
Basic Ampex TM Sel Sync system for switching and application of overdubbing.

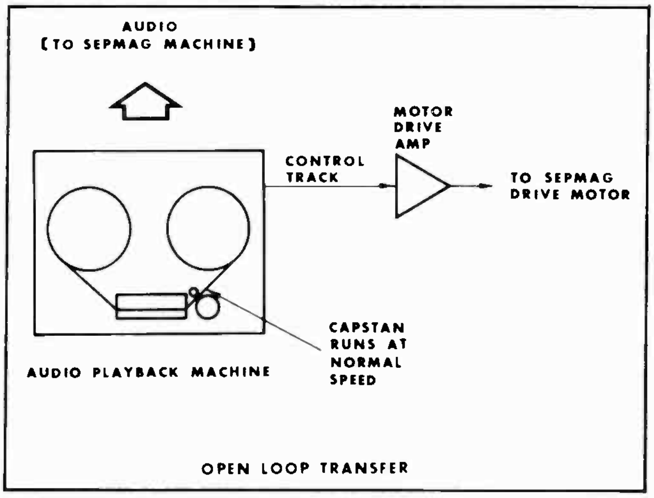


Figure 2.
A basic control track used to drive the external sel sync.

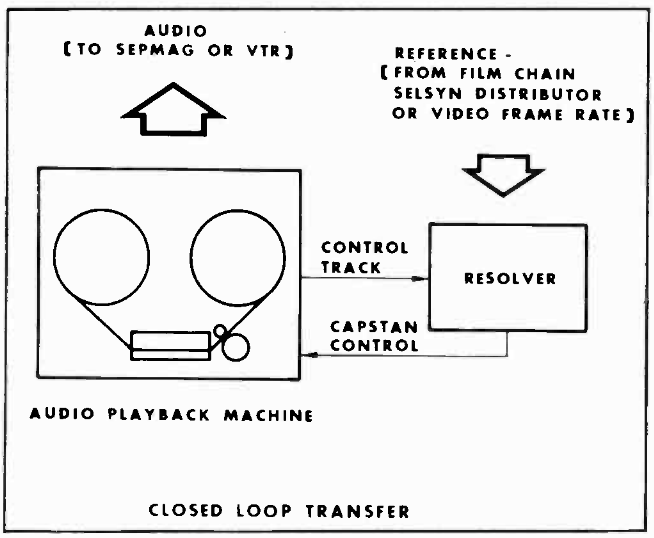


Figure 3.
Basic control track with resolving closed loop.

"Overdubbing" consists of using the recent head in a reproducing mode for those tracks already recorded. This is known as the Sel Sync mode (*Ampex TM*, Fig. 1). The output from these tracks is appropriately mixed and fed to the artist or artists by headphones. When adding a vocal to a background for example, the vocalist sings in time with the headphone signal, his singing being recorded on a separate track, in spatial synchronism on the tape with the backing material already recorded.

This technique permits multiple attempts at any single track, without changing any other track, and, most importantly, without requiring any generation build up in any other track.

The flexibility given by this approach has, in the recording industry, been virtually overshadowed by the artistic and creative possibilities opened up. This is perhaps amply illustrated by the impact which Les Paul in the mid-fifties had on the musical world. An entire selection was performed by Les Paul himself using the technique of "Sel Sync" and overdubbing. The vocal, and each musical instrument used, were recorded one at a time and finally mixed down to one or two channels for disc mastering or tape duplication. Les Paul was in this respect a pioneer of this technique.

Today it is commonplace to find a selection, with complex instrumentation and vocals, which has been made by a small group of perhaps four or five musicians, and in some cases even fewer. It is the multichannel audio mastering recorder then, that offers the video broadcast industry an entirely new equation for production economy and flexibility, as well as opening up new areas for creative artistry.

Double System Operation

The key to the marriage between audio recorders and visual media has been the revolution in technology brought about by the lower cost of integrated circuit logic.

The dropping of this economic barrier makes it possible to design low cost and highly reliable means of synchronizing two unsprocketed media together.

The simplest method, and the oldest, which will be immediately familiar to anyone with a film background, is to record a control track at the same time as the audio on a separate track of a normal audio recorder. The control track, in the case of the original film double system, is derived from the camera motor.

In the case of video/audio double system recording, the control track is related to the normal video signal, usually the vertical interval rate, and sometimes the normal video control track is used.

On playback the control track signal is recovered from the audio tape machine, (Fig. 2). It may have been recorded with any one of a variety of ingenious schemes to minimize either crosstalk of the control track into the audio signal, or to provide a good trade off between width of tape used for audio versus width of tape used for control track. The recovered control track is then used in a closed loop phase lock servo system, known generally as a resolver, (Fig. 3), where it is compared with a reference signal which is synchronized to the video signal or film chain. The error signal thus derived is used to drive a servoed audio tape machine such that the frequency of the recorded control track, conforms at all times to the reference signal.

There are many variants of this basic approach, even one which is not a closed loop system but which uses the recovered power line rate control track to drive the master selsyn of a film chain system directly.

The fundamental disadvantages of these systems is that while synchronous operation is easily obtained, a true lip sync condition must be sought manually, even if tight cueing is used to reduce the initial lip sync error. It is true that in many such nonsprocketed and sprocketed media transfers there is no real need for absolute lip sync. Some types of non visually related background material, for example, could be transferred as "wild" tracks, as much as 1/4 sec error being tolerable.

In the late sixties, and very early seventies, a number of systems were designed to try to overcome this problem.

SMPTE Time and Control Code

The introduction of the SMPTE Time and Control Code has provided a very powerful solution to the problem of obtaining automatic lip sync.

Since the SMPTE Time Code is well known in video circles, it suffices to note that it is an absolute time of day code, continuously laid along a track on any unsprocketed or sprocketed magnetic material, or indeed on any recording media.

The time code permits the identification of any part of the recorded signal by a unique address. The identification of the address can be made at any reasonable speed of the media, constrained usually to between forty times and one tenth of the originally recorded speed.

Since the advent of the SMPTE Time Code, the design of many video editing systems, both simple and complex, has been stimulated. The address code is by now well established as a means of performing automated or semi automated multiple or single video edits, with precise control of every video frame.

The concepts basically are that two video tape machines, for example, can be run up to lock together to exact frame congruence with one another and synchronous transfers subsequently made between them, whose start and end points can be controlled absolutely to within one frame. A three video tape machine assembly or insert edit procedure can also be implemented effortlessly by locking all three to the same master reference. It is usual in such systems to be able to offset any machine from any other by a specified number of frames. In other words, the addresses on the separate machines, do not necessarily have to be the same at lock up, but rather they may be separated by any amount, to an integral number of frames.

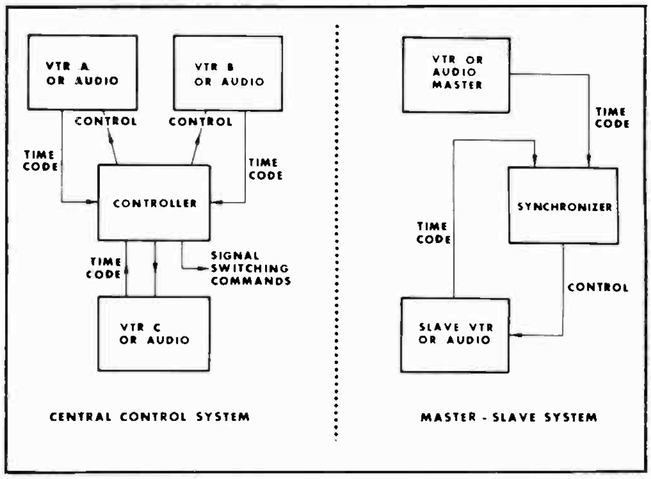


Figure 4.
The two basic control implementation systems.

Control Implementation

There are two basic control implementation of address code synchronizing schemes. (Fig. 4.)

The first is to slave each transport involved to a single electronically controlled source of address code. In such systems the control of all transports would fall naturally to a central control station.

The second, and more convenient for simpler systems, is to delegate one of two transports as a master and the other as a slave. The master transport is then operated normally and the address code derived from it used to drive the slave transport, via the synchronizer unit, into frame congruence. Control of the pair thus resides with the master transport.

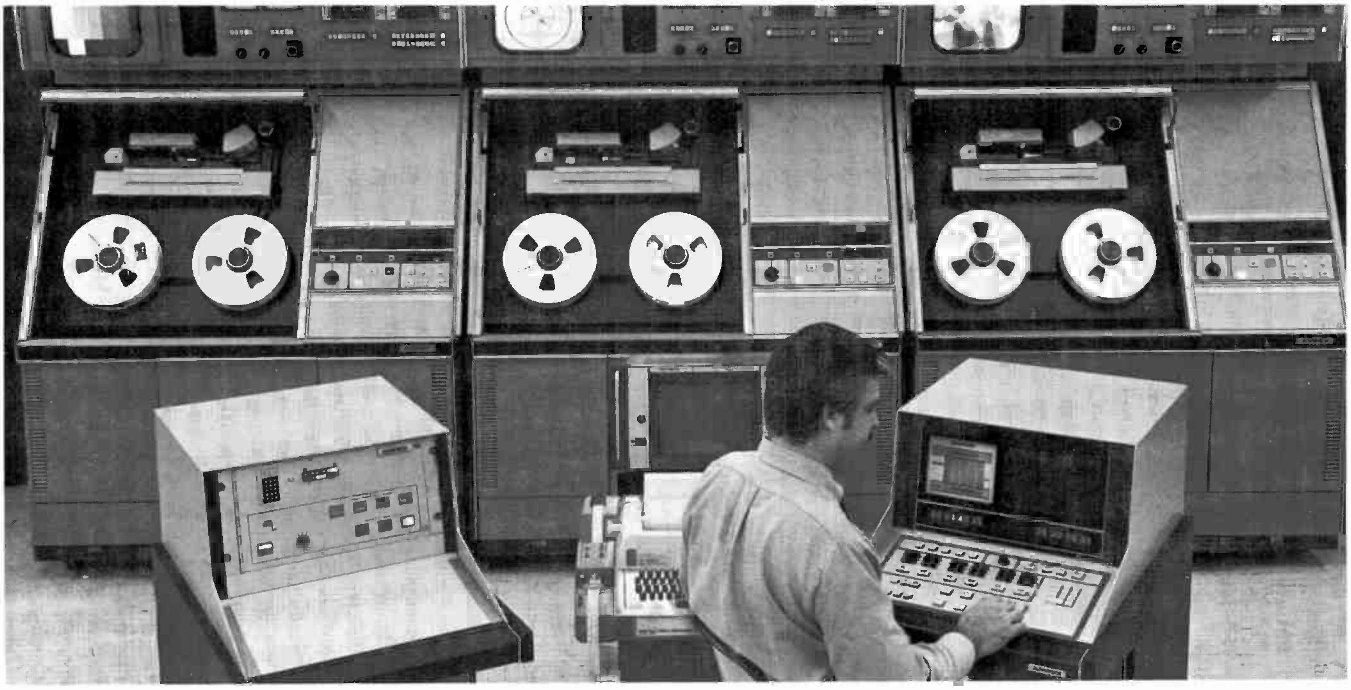
An example of the first type of system is the Ampex RA-4000 Random Access Controller, used, with an AVR-1 and two MM-1000 16 track audio recorders, to great effect, at both the Munich Olympics and the Winter Olympics at Sapporo, Japan. In this application the two MM-1000's provided thirty separate tracks for synchronous multilingual commentary. (*Each machine requiring one track for the time code*).

An example of the second type of system is the Ampex Synchronizer System.

Here, each pair of transport synchronized requires a separate compact synchronizer unit, so that expansion to say three or more transports slaved to one master, although perfectly feasible, may become more complex and less economic than a more sophisticated system such as the RA-4000 controller.

In both systems, provided suitable capabilities exists on a transport, be it video or audio, e.g. servo controlled capstan, remoteable transport functions, and a spare audio channel to carry SMPTE time code, then automatic synchronous operation is possible.

Given also the ability to provide incremental frame offsets between transports, the combined system can then be regarded as a truly sprocketless film and sepmag/dubber with all the flexibility that such chains have offered in the past for editing.



The Ampex RA-4000 controller, plus an RAC-200 ACSII Interface and an Ampex AVR-1.

The Ampex synchronizing system using SMPTE time and control code.

Application

While there are many ways in which time code controlled systems may be used, both for video and double system operation, the following examples serve to illustrate the major possibilities.

The video portion of a live production can be recorded (Fig. 5) on one or more Quad VTR's, laying SMPTE time code in the main audio tracks, and separately recording the entire audio portion of the studio shooting on a multichannel audio recorder. The same time code would also be recorded on one track of the audio recorder.

If the audio were complex, it could be recorded, in an essentially unmixed form, on several tracks of the audio recorder. A rough single channel mix could be fed, if desired, to the cue tracks on the VTR's, but the final audio mix could be deferred.

After the video is edited onto one tape, the time code, previously recorded on the main audio track, could be transferred onto the cue track, and the final audio mix synchronously transferred, retaining absolute lip sync, onto the normal audio track of the master video tape.

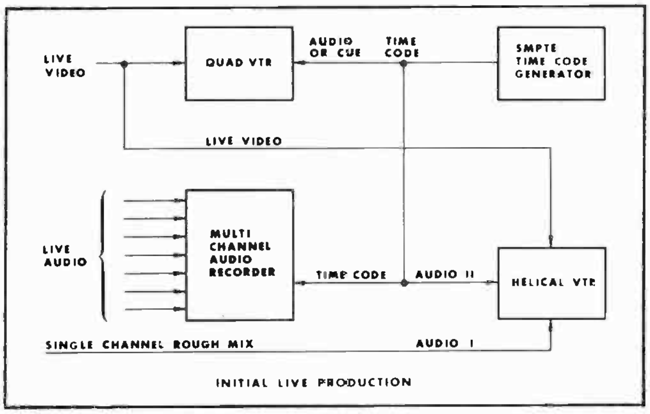


Figure 5.

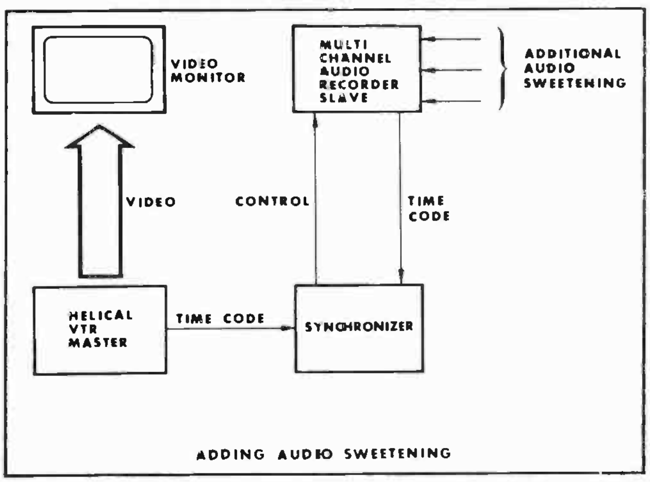


Figure 6.

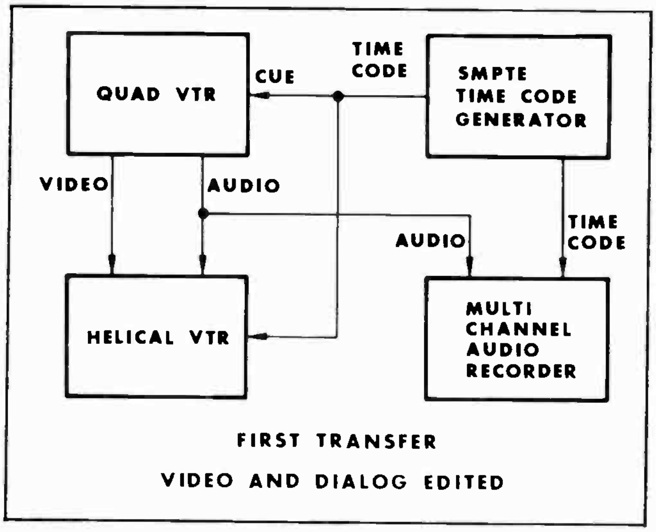


Figure 7.

If the video were edited using a system utilizing the SMPTE time code to define the edit points, prudent practice would dictate that the time code be originally recorded on the VTR cue tracks, with the rough audio mix being recorded straight onto the normal audio tracks. The final audio mix could still be transferred at a later time, still retaining absolute lip sync with the video.

Additional sweetening (Fig. 6) could be added to the audio quite independently from the video, or if desired, a single dub of the video and the time code made onto a lower cost helical video machine, to which the audio machine could then be locked, permitting the overall effect of added sweetening to be evaluated.

An alternate scenario for post production sweetening, might have the video and dialog track recorded and edited in Quad format. (Fig. 7) Time code would then be recorded on the Quad VTR cue track while also recording code on the Audio II track of a helical machine.

Simultaneously with this, the video is transferred from the Quad tape to the helical machine, (Fig. 8) and the audio from the Quad VTR track, and time code from the SMPTE time code generator, recorded on two tracks of the multichannel audio machine.

Having released the Quad VTR for other tasks and only requiring a single pass of the edited video tape, which by now represents a considerable investment, the audio sweetening can take place at leisure.

The final release track (Fig. 9) may then be mixed and recorded on a spare track on the multichannel audio machine, using a Sel Sync mode, and then, using the Quad VTR and master video tape, synchronously transferred back to the audio track on the video machine. Alternatively, as has been done, a secondary audio tape can also be created, the control track recorded on the secondary tape might either be color sync (*vertical frame rate*), for use with simpler resolvers, or the time code itself could be transferred.

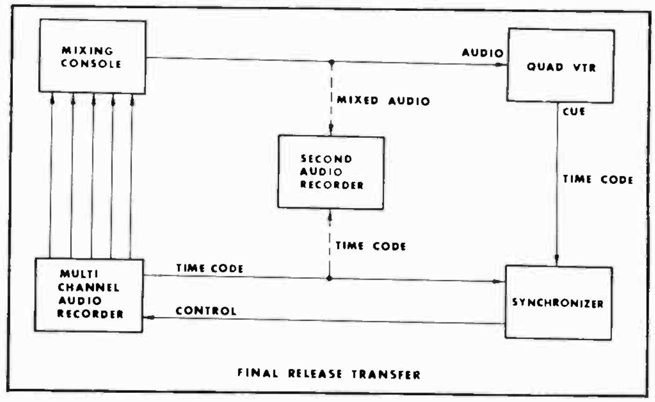


Figure 8.

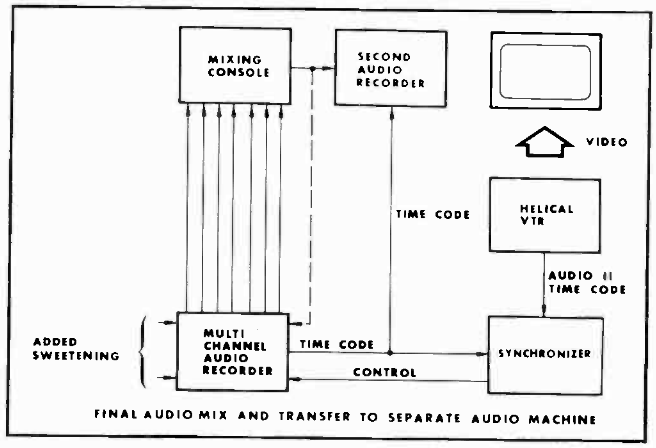


Figure 9.

Video Playback Heads

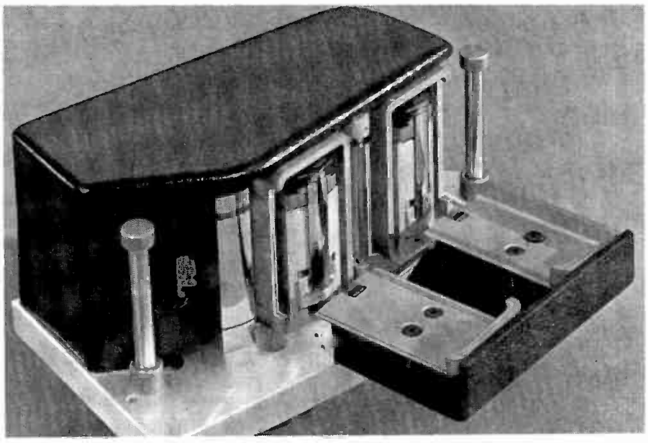
Today it is even possible, if more than one multichannel audio machine is available, to have a special head assembly with full video format, mounted on the audio machine and even avoid having to utilize the Quad VTR to make the final dub.

On the Ampex video layback head assembly for the MM-1100 mastering recorder, in addition to the normally expected audio and cue erase and audio and cue record/playback heads, there is an additional narrow gap reproduce head for both audio and cue tracks. This permits reproducing a previously recorded audio track with a better frequency response and signal to noise ratio than a normal audio record/playback head on a VTR.

If this assembly is used even for the initial transfer between Quad tape and audio machine prior to sweetening, the transfer can be made away from the relatively hostile audio environment of a Quad VTR. This process will result in even less degradation in the final signal.

Audio Insert Editing

This requires the provisions of a suitable delay between the erase and record/bias signals on a magnetic recorder. This delay prevents over recording when entering record and a gap when exiting record,



Ampex Video Layback Head assembly for the MM-1100 Recorder.

when making an audio insert edit onto a previously recorded track.

This feature, already provided on the Ampex AVR-1 and AVR-2, can now be added as an accessory, (PURC bias and erase card), to any or all channels on Ampex AG-440's, MM-1000's and MM-1100's.

This is a case where the special requirements of the broadcast television and film industry, have stimulated an addition to the capabilities of conventional audio recorders. The feature also has application in the audio mastering industry, but is most suited to applications where it will be under automatic control.

Quad VTR Audio Limitations

The first drawback the audio track on a Quad VTR suffers from, is the transverse orientation of the tape oxide. This orientation is optimal for video, but for a longitudinal recording causes a drop of 6-10 db in output for a given level of distortion. Unfortunately the tape noise does not follow suit, and the result is that the dynamic range, defined either as the ratio of saturated output to bias noise, or as the ratio of signal output at some arbitary distortion level to bias noise, is reduced to some six to ten db below the value which could be obtained using the same oxide were it longitudinally oriented.

This particular problem does not occur on helical machines, which use longitudinally oriented tape, but is replaced by the problem caused by the usually narrower tracks, and almost universal use today of thin oxide coats, these factors both reducing the available signal to noise ratio.

A second major drawback is that the rotary video heads contacting the tape at a 960 Hz rate (in 60 Hz power line environments), produces very large velocity variations in the longitudinal tape motion. This results in frequency modulation of any audio signal which may be present.

If an audio recording is reproduced on the same VTR, then the modulation is effectively cancelled. However, if the tape path geometry changes for any reason, for example, in interchange from one transport to another, then the cancellation will not be exact and the modulation sidebands will become evident.

This mechanism alone seriously degrades the audio quality. Unfortunately the phenomenon is not disclosed by normal flutter measurements, it is in fact not perceived as flutter, but rather as a "blurring" of the signal.

Over several generations, the modulation sidebands will themselves cause additional intermodulation products, and the final result become very objectionable.

A third problem is that the gap chosen for the combined record/playback head is, of necessity, a compromise between the requirement for a short gap to minimize gap loss in reproducing, and the longer gap required to optimize the record process. This problem becomes more acute at slower tape speeds, since for a given audio frequency the wavelength on the tape become shorter, and thus for given reproducing gap length, the high frequency losses are much greater.

The optimum recording gap length, on the other hand, is generally dependent upon the oxide thickness, shortest wavelength to be recorded, and on the spread in oxide particle switching fields. In general, for a given oxide type and coating thickness, the shorter the record gap, the more high frequency preemphasis will be required during recording.

Typical overall performance of the audio channel might approach 56 to 57 db signal to noise ratio, (*computed from three percent third harmonic distortion*), with a bandwidth of 50 Hz to 15 KHz ± 2 db.

On the MM-1100 for example, using modern tape, the signal to noise ratio, also measured from the three percent third harmonic distortion point, would typically exceed 66 to 67 db. If the previous generation of audio tape were used, this figure would degrade to approximately 64 to 65 db. The specified frequency response would be 30 Hz to 18 KHz ± 2 db, and typically would extend past 20 KHz within ± 2 db, with the band 100 Hz to 12 KHz having excursions no larger than ± 1 db from a mid band reference.

The improvement in signal to noise ratio is almost entirely due to the optimal oxide orientation for audio tapes, permitting higher recorded levels for the same amount of distortion, whilst the improvement in bandwidth is primarily due to the use of separate record and reproduce heads, which may now be separately optimized for their respective functions.

The differences noted above go a long way towards underlining the advantages of double system operation. Whilst the video signal might be taken four or five or more generations from the original with very little degradation, four generations through a Quad VTR audio channel will reduce the signal to noise ratio by at least 6 db for noise alone, and by 12 db for any coherent residual signals (i.e. horizontal rate or power line components).

Thus a 120 Hz component, which was buried below tape noise on the original generation, may be very obvious after four generations.

If the audio track, however, started it's life on an audio machine, the final release Quad VTR track might have a signal to noise ratio degraded by less than 3 db compared to a first generation track.

This takes on a new significance when it is appreciated that a high quality home cassette recorder, running at only 1/8 the speed, with tracks less than 1/3 the width, can, without benefit of any noise reduction, give a 3% to noise figure of around 46 to 48 db. This is about what a Quad VTR audio track might expect after 4 or 5 generations!

The introduction of compatible dual audio tracks on the Ampex AVR-2 to satisfy the need for both compatible stereo release capability, and for dual language dialog tracks, has inevitably caused a 3-4 db reduction in signal to noise ratio (when using Ampex 176 tape) when compared on a track for track basis with a standard width VTR audio track. (The overall degradation in signal to noise ratio of stereo dual track recordings, when played back on a standard width head, is of course less than the figures quoted above).

Usage of double system working can thus permit dual audio tracks, taken individually, to achieve release copy signal to noise ratios which are at least as good as practical standard width tracks, which had been taken through three or four generations without double system working.

The degradation in frequency response over the same number of generations is more closely controlled by the flatness of response in the audio channel(*i.e. 1 db error at 10 KHz could easily grow to 3 or 4 db*).

However, the audio only machine will still come out better over the same number of generations, since in general, adjustment for optimum flatness is far easier, and the initial bandwidth is wider.

An argument could be made that if modern noise reduction techniques were used, the degradation in noise in the single VTR audio track could be avoided. However, this would be to turn a blind eye to the obvious production flexibility and economy offered by double system operation. In addition, the same techniques could be, and are, applied to the audio machine.

Noise Reduction

The mastering industry has also had it's share of problems with dynamic range and signal to noise ratio restrictions.

It is due to these restrictions that noise reduction techniques have found such widespread acceptance.

In the professional audio field, two systems appear to have won wide acceptance, these being the Dolby A system and the newer DBX system. Both systems have their merits and their followers and an equal number of concommitant problems and detractors. The principal problem in both cases being that the channel gain accuracy and/or amplitude flatness, should be maintained to closer limits than normal, if satisfactory behaviour over more than two or three generations is expected.

However, the subjective improvement in dynamic range and signal to noise ratio usually outweighs the disadvantages and possible coloration, which with reasonable care can be minimized. The coloration moreover can usually only be easily discerned if A/B comparisons are made with the original. It should be emphasized that the two systems mentioned above are encode/decode systems, by which is meant that the incoming signal is encoded prior to recording and decoded during reproduction. For this reason the signal recorded on the tape is incompatible, and must be suitably decoded. By the same token, an unencoded signal will in no way be enhanced by passing it through a decoder, unless the listener has a predilection for the peculiar sound which would result.

In the last two years, considerable success has been reported on the application of a variant of the Dolby A system to film sound chains, both magnetic and optical. Particularly in the case of optical tracks, the resultant performance is very much closer to normally accepted audio performance of professional magnetic recorders.

The sheer number of optical systems throughout the world unfortunately will ensure that the process of upgrading optical sound channels with techniques of this kind, will take a long time.

Compressors and Limiters

Noise reduction systems are really only the special application of a class of devices which have much wider use, that of compressors and limiters.

Here again, the use of modern semiconductor devices has permitted the design of limiters and compressors with outstanding freedom from some of the undesirable effects which used to be associated with them.

In this field, however, the broadcast television and radio industry has itself been a prime mover in the development of suitable types of limiters and compressors, both for transmitter protection, increasing average "talk" power, to use a ham radio expression, and also, just as importantly, to prevent overmodulation and a visit from the FCC inspector.

In the recording industry the development of limiters has been mostly in the field of overmodulation protection whilst cutting disc masters. Compressors too, as in broadcasting, have many applications in the field.

However, with the improvements in the last decade, in all parts of the chain, there are now those who advocate not using any form of artificial limiting or compression, on the grounds of aural aesthetics. Their view is somewhat extreme, but it is still very common to hear the characteristic noise "pumping" of a signal far into compression, both on broadcast television and radio.

Were a video signal to be compressed by an equivalent amount, the results would be bizarre, to say the least!

Dynamic Noise Filters

Another class of channel processing devices, known as dynamic noise filters, have recently become available.

The range of dynamic noise filters, manufactured by Burwen Laboratories, and now being marketed by Ampex Corporation, are representative of this class of device.

Here it is not the overall channel gain which varies but rather the bandwidth. The object being in this case to exclude interference in higher frequency bands when no wanted signal is being transmitted in them. The most potent application of such devices is in the virtual elimination of various line crosstalk noises, on long telephone lines, and the reduction of heterodyne and adjacent channel interference on poor radio telephone or short wave links.

Judged subjectively, such devices seem to produce a very intelligible signal from what might seem a hopelessly chaotic mess.

Future Developments

It is perhaps important also to take a short time to contemplate the future.

When talking of the future in audio and video, one can not help but be influenced by the great proliferation of digitally based processing devices now available.

The British Broadcasting Corporation, for example, is currently using digital techniques to carry a full range audio signal in digital form, embedded in the normal video signal. The advantages of being able to transmit, switch and route both video and audio together over a single video baseband microwave link are many. Unfortunately, this format is unsuited for production work, for the very reasons which make it excellent for center to center transmission.

In the United States, similar usages of digital techniques are being considered, and it will probably be only a matter of time before we may see both the video and audio portion of a broadcast quality signal being routinely transmitted along microwave and global satelite links, in digital form. The economics of such methods are growing in persuasiveness.

In the field of audio alone there are now several manufacturers of digital audio delay systems, and there is even a digital system available to effectively replace the advance monitor head on tape to disc mastering systems.

In the meanwhile, the purely analog world of signal processing will continue to develop.

In the field of magnetic recording, advances in oxide formulation alone have provided almost 10 db improvement in dynamic range. Similar improvements have occurred for video tape.

The growing use of automated mixdown consoles foreshadows the day when the manipulation of an audio signal will be as complex in it's possibilities as the video signal is today.

Conclusion

This paper, while not claiming to be exhaustive, has covered many of the tools that are currently available for audio signal processing, both in broadcast television and in other areas.

Hopefully, although economics will still be the most powerful force, quality will be very close behind. If the rate of assimilation and cross fertilization of ideas and hardware continues to be as prolific as it is today, then assuredly quality will also assume a greater significance.

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Envelope and Synchronous Demodulation of V.S.B. TV Signals

Introduction

In order to evaluate the performance of a television transmitter, it is necessary to demodulate the signal as it appears at the output of the transmitter. The demodulated signal can then be used to measure the different parameters of interest. However, for these measurements to have any practical value it is essential that the demodulation process itself does not introduce any degradation of the signal to be evaluated.

As has been pointed out in the literature (1) (2) (3) envelope demodulation of a vestigial sideband signal causes non-linear distortion in the form of quadrature distortion and as crosstalk between the luminance and the chrominance signal.

Quadrature errors give rise to increased overshoots, loss in resolution and a change of the pulse ratio with square-wave signals. In the case of color transmission, quadrature errors cause desaturation of the color areas and phase modulation of the vision carrier, depending on the degree of modulation and on the instantaneous value of the vision carrier.

Non-linear envelope distortion can, however, be eliminated by use of a detector which responds only to the envelope of the in-phase component, since both the in-phase and quadrature components of the detected output are linear functions of the depth of modulation (4).

With the use of a product detector it should, therefore, be possible to correctly assess the transmitted signal provided the characteristics of the control receiver are close enough to the ideal characteristics. However, the tolerance limits on the Nyquist slope for both amplitude and phase have as yet not been defined in this country.

Nyquist Slope

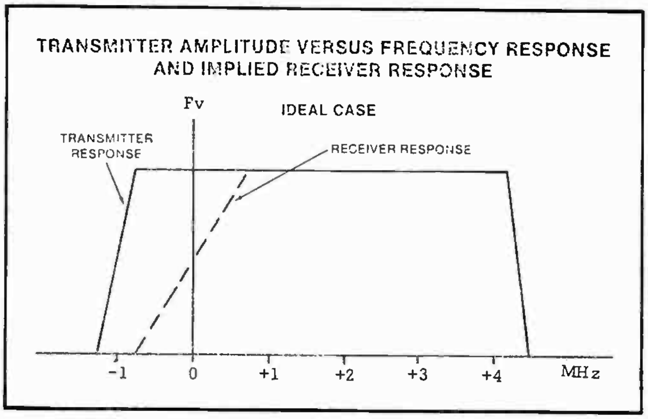


Figure 1.

The **FCC** standards of 1941 for monochrome TV transmission specify a transmitter amplitude versus frequency characteristic as shown in **Figure 1** with an implied associated receiver response shown.

We see that the response of the transmitter covers that of the receiver. Since the receiver response is at

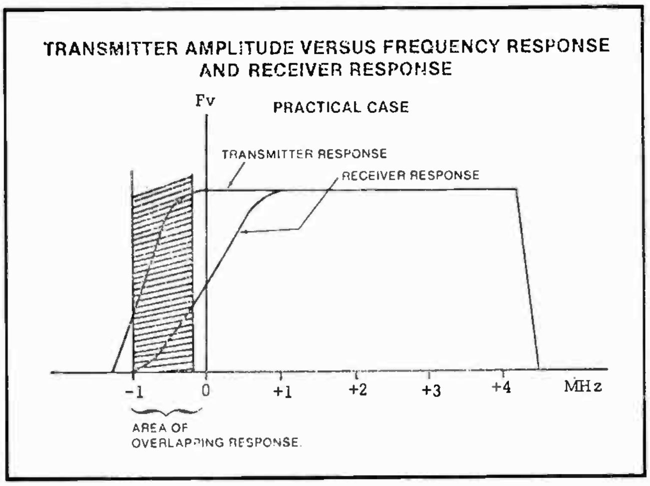


Figure 2.

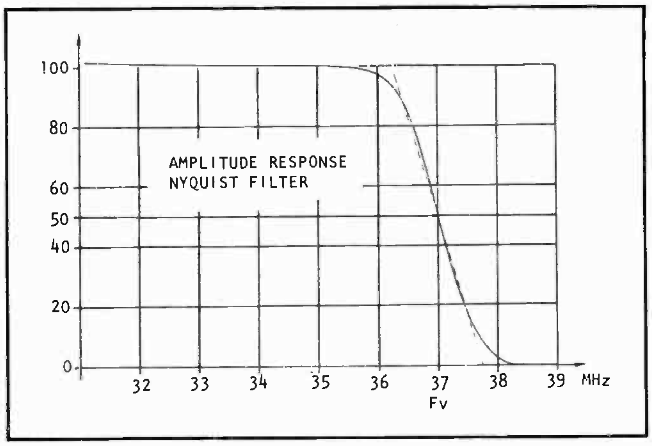


Figure 3.

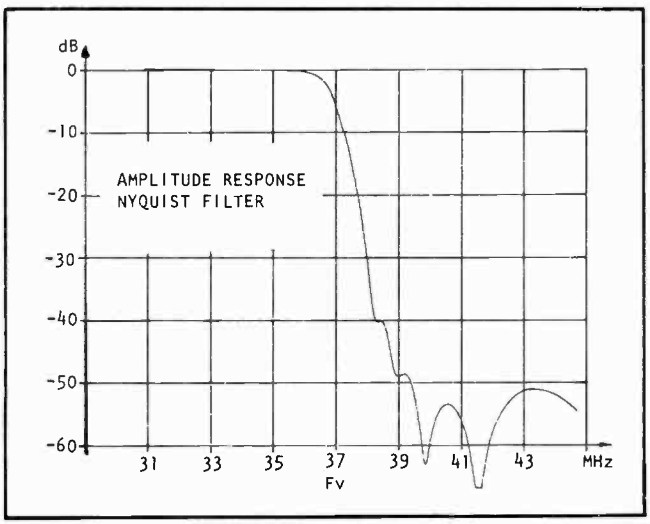


Figure 4.

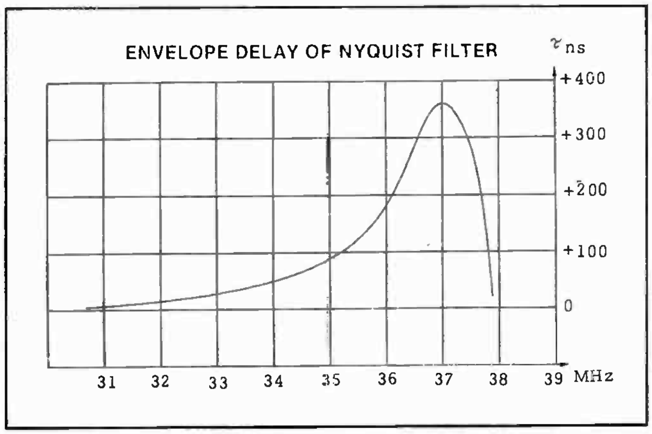


Figure 5.

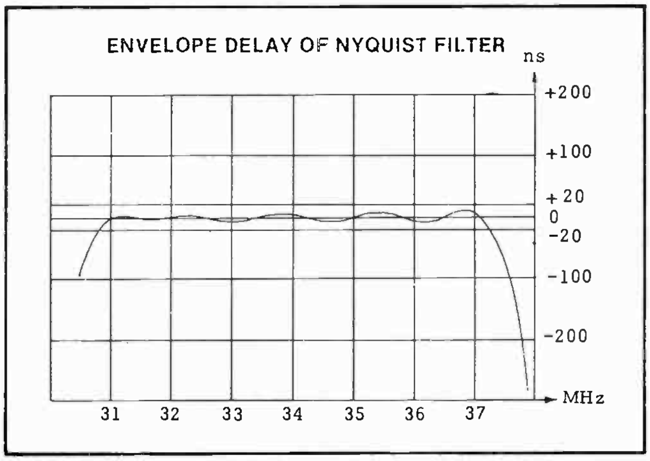


Figure 6.

0% where the transmitter response is still at 100%. This is the basic idea of **V.S.B.** transmission. The receiver operation should be the same whether the receiver is supplied with a signal from a true double sideband signal generator or from a transmitter in which the lower sideband is suppressed beyond the frequency at which the receiver has no response, provided the phase characteristics are the same. As long as the amplitude versus frequency characteristic of the transmitter and that of the receiver do not overlap, it is not necessary to define the exact attenuation curve of the transmitter or receiver (5).

Since this ideal case is not practical, the amplitude versus frequency response of the transmitter-receiver combination may be as shown in **Figure 2.**

It is obvious that variations in the amplitude or phase response of the lower sideband of the transmitter or receiver will affect the total system performance. For example, non-uniformity of the phase response in the vicinity of the video carrier frequency higher than 50 ns causes considerable distortion of the transient characteristics.

The Nyquist-filter designed for this demodulator therefore approaches the ideal response as indicated by **Figures 3 and 4.** Figure 3 shows the response on

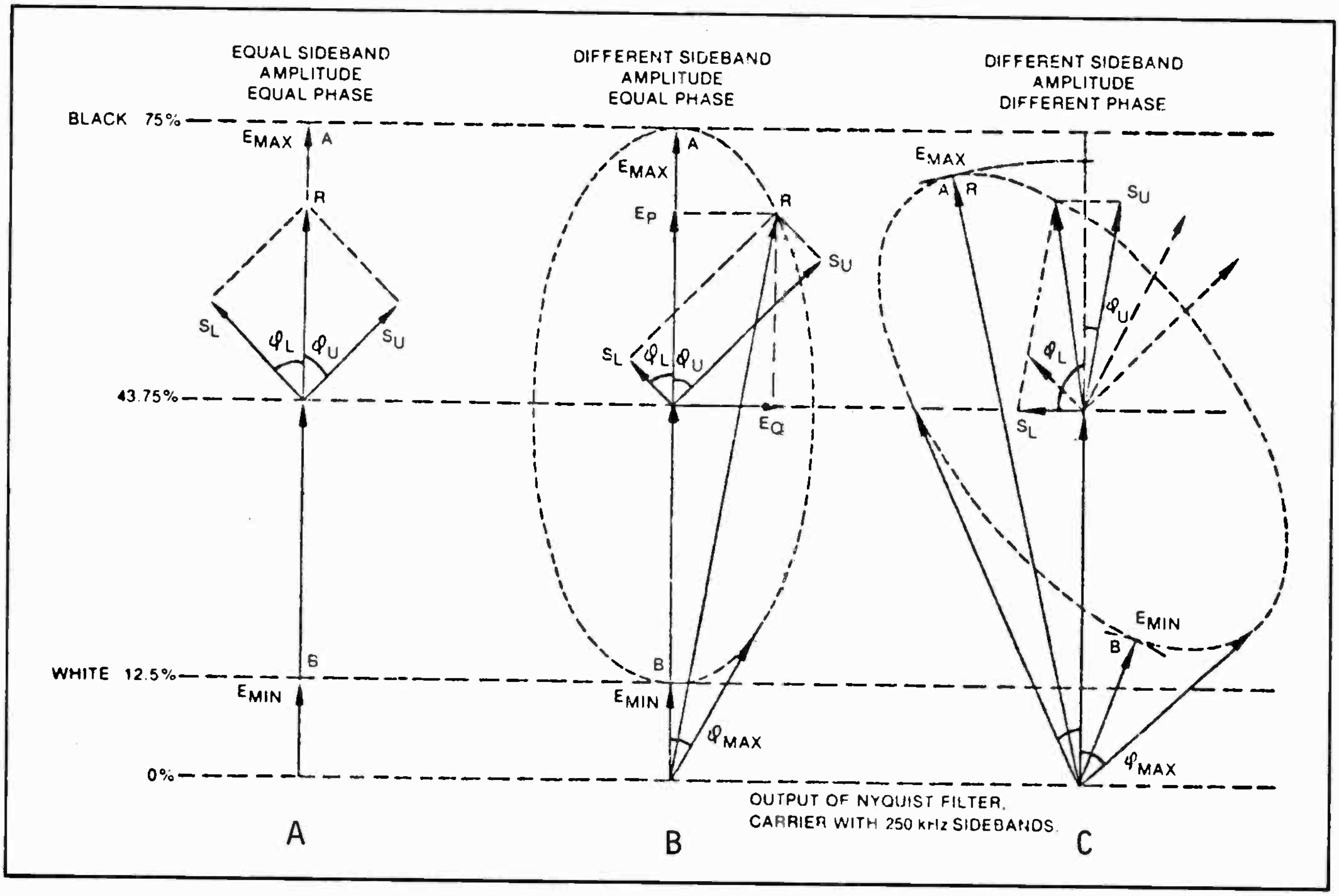


Figure 7.—A, B & C

the spectrum analyzer with the display in the linear mode. The ideal response is shown in dotted line. **Figure 4** shows the response in Log display mode. As can be seen, the response is down 40 dB for all frequencies beyond 1.2 MHz past the carrier frequency. The importance of this is that a double sideband test transmitter can be used to evaluate the performance of the Nyquist demodulator since the lower sideband (after conversion to **IF** the upper sideband) will contribute negligible error to the response after detection.

The envelope delay of this particular filter is shown in **Figure 5** before compensation and in **Figure 6** after compensation to a residual error of ± 10 ns from 31 MHz to 37.2 MHz. This is accomplished by the use of four all-pass sections at **IF** frequency.

The Nyquist filter is implemented as a 7-section modified Cauer Low-pass filter (Elliptic-Function Filter).

The importance of both the amplitude and its associated envelope delay response of the Nyquist slope can best be explained by phasor diagrams as shown in **Figures 7a, b, and c.**

Figure 7a shows the phasor diagram for a carrier modulated with a very low frequency sinewave after passing through the Nyquist Filter. Both sidebands are equal, the locus of the resultant is a vector in phase with the carrier with a maximum at A and a minimum at B.

Figure 7b shows the phasor diagram for a carrier modulated with a 250 kHz sinewave after passing through the Nyquist Filter. The locus is an ellipse with main axis equal to the sum and difference of the sidebands, with a maximum at A and minimum at B.

Figure 7c shows the phasor diagram for a carrier modulated with same 250 kHz sinewave after passing through the Nyquist Filter. However, this time there is phase asymmetry between the sidebands $\theta_L = \theta_U$. The locus is again an ellipse but a lopsided ellipse which reaches a maximum at A and a minimum at B.

From Figure 7b it can be seen that if the Nyquist slope is not close to ideal, the upper and lower sidebands will not add up to unity, and from **Figure 7c** it can be seen that if the phase response of the Nyquist Filter is not close to ideal the sidebands will not add up to unity. There will, therefore, be an error in the amplitude/frequency response and in the phase/frequency response after detection.

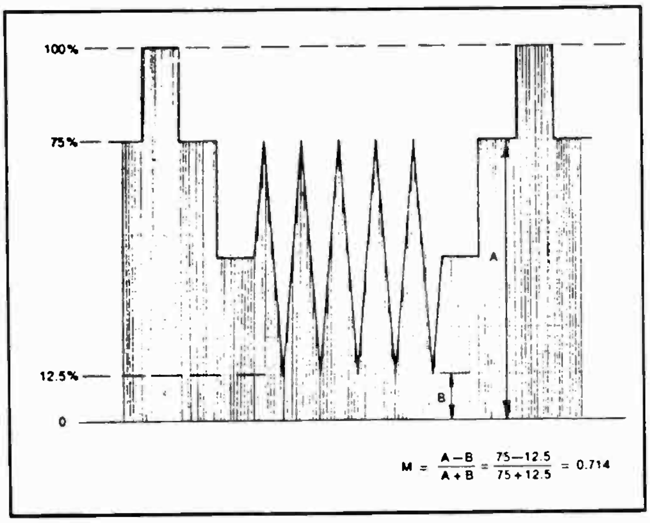


Figure 8.

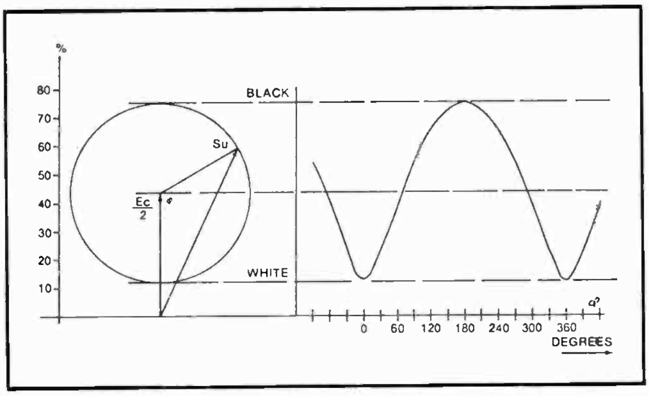


Figure 9.

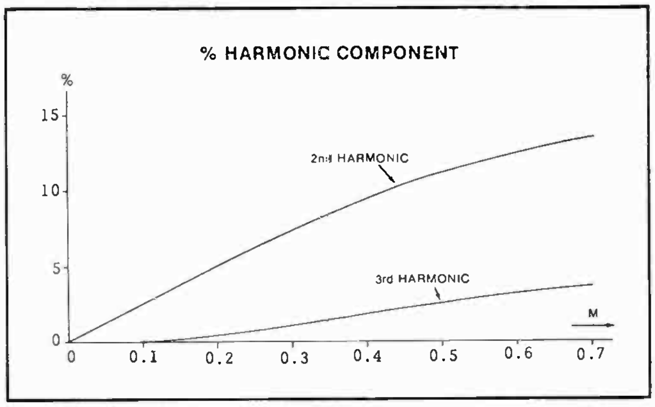


Figure 10.

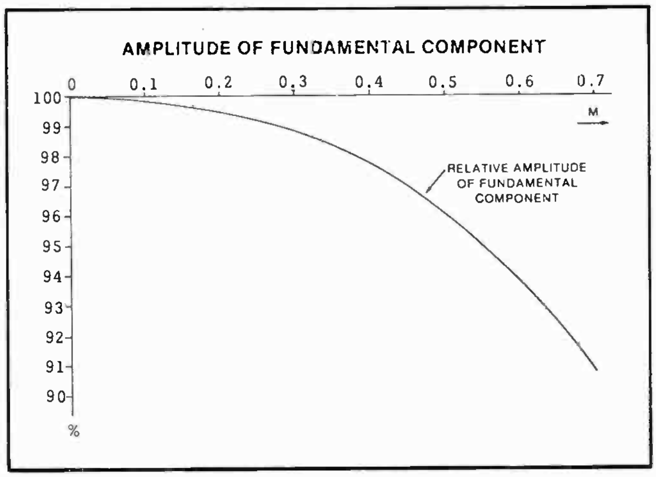


Figure 11.

Since an envelope detector will respond to the resultant vector R, phase modulation of the carrier will also occur at the modulation frequency. This is indicated by the phase angle ϑ.

Distortion Caused to a Sinewave Signal

In Appendix 1, it is shown that for different values of the Modulation depth M, the distortion will increase when M increases.

Consider the RF waveform as given in **Figure 8** where a sinewave of frequency greater than 0.8 MHz is modulated between the limits of 75% and 12.5%. This is the signal that will be available after the Nyquist Filter. The envelope of this SSB signal can be developed from its phasor diagram as seen in **Figure 9.**

At a modulation depth corresponding to white, the waveform is stretched where as at black, the waveform is rounded off. This is caused by the 2nd harmonic component which adds at white and subtracts at black from the fundamental component. This is also the signal that the envelope detector will produce if the relative phase of the fundamental and harmonic components is preserved.

The **DC** component has been shifted toward black from its original value (DC axis shift) and the magnitude of this shift will depend on the modulation depth M.

The amplitude of the harmonic components is shown in **Figure 10.** For M = 0.7 the amplitude of the 2nd harmonic, as shown in Appendix 1 is 13.6%, whereas the 3rd harmonic has an amplitude of 3.7%. These values decrease with M.

The fundamental component also changes with M as shown in **Figure 11** and for M = 0.714 its amplitude is reduced by 9.5%.

Distortion Caused to a Multiburst Signal

The information derived in Appendix 1 will now be used to discuss the transmitter/demodulator behavior when envelope detection is used. We will evaluate the results for different **IF** Amplitude/Frequency response characteristics of the demodulator, different Video Amplifier response characteristics and also the effect of the Low-pass filter following the envelope detector. We will use the Multiburst signal as modulating signal.

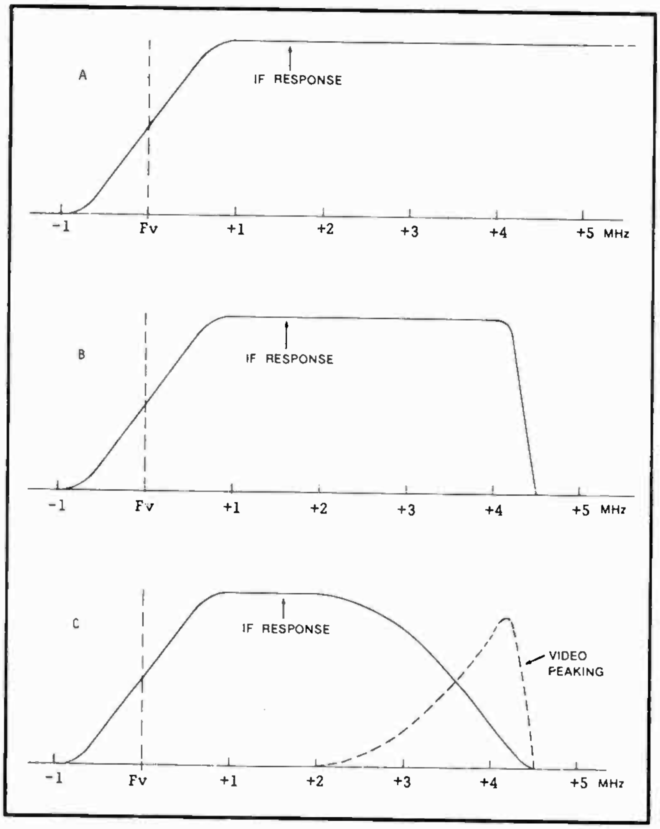


Figure 12.—A, B & C

a- Flat **IF** Response, no Video frequency response limiting. **Figure 12A.** All Multiburst frequencies above 1 MHz will have the same distortion components, equal amplitude of the fundamental component and equal amplitude of the harmonic components.

Figure 13 shows the 1 MHz component of the Multiburst signal after envelope detection and without any bandwidth limiting below 7 MHz. **Figure 14** shows the detected 1 MHz of the Multiburst signal.

b- Flat **IF** Response, Low-pass filter cut-off at 4.2 MHz. **Figure 12B.** In this case, the 1 MHz burst will be reproduced with all its important harmonics and will be very similar to that discussed above. The 2 MHz burst will have the 2nd harmonic component added to its fundamental and will therefore have appreciable distortion. All higher frequency bursts will only be present with their fundamental component and show little distortion. However, the amplitude of these higher frequency bursts will be lower than that of 1 or 2 MHz burst.

c- 6 dB roll-off at 3.58 MHz in the **IF** amplifier, video peaking to compensate for **IF** roll-off and 4.2 MHz low-pass filter. The 1 MHz component will not be affected much and will, therefore, closely resemble the waveform of **Figure 9.**

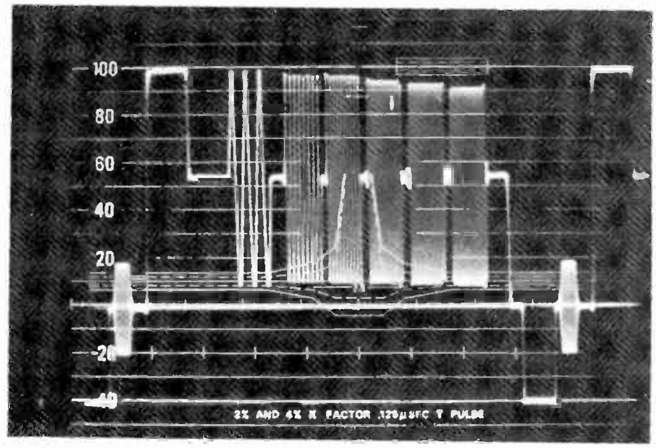


Figure 13.

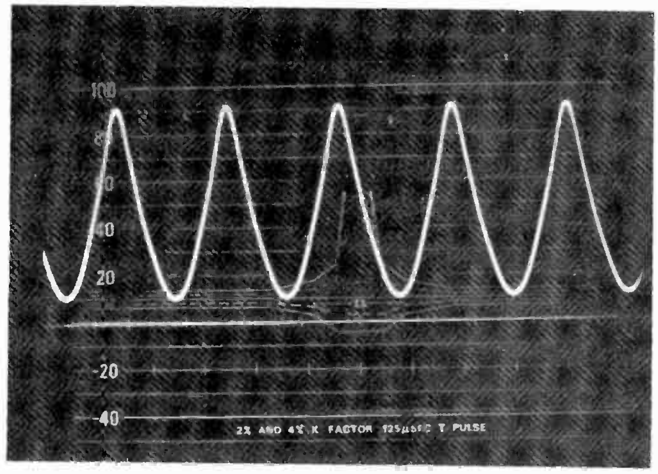


Figure 14.

The 2 MHz burst will have the same fundamental component as before; however, the 2nd harmonic created in the detection process will now be amplified by approximately 10 dB by the video peaking circuit and be added to the fundamental component. The 2 MHz burst will, therefore, show more distortion and its amplitude will differ from that of the 1 MHz burst. The 3.58 MHz burst is reduced by 6 dB before detection. This means that M has changed from 0.7 to 0.35, so the fundamental component will only be reduced by 2% instead of 8.5%. No harmonics will be added to this signal because of the low-pass filter and, therefore, the 3.58 MHz signal will be approximately the same amplitude as the 1 MHz. However, the dc axis shift will be more toward white than the dc axis shift of the 1 MHz burst.

From this brief discussion it can be concluded that the actual end result depends on the amplitude/frequency response curve of the IF amplifier, whether or not video peaking is used and what the actual cut-off frequency is of the video low-pass filter following the detector circuit. We have chosen the multiburst signal as this is most readily understood.

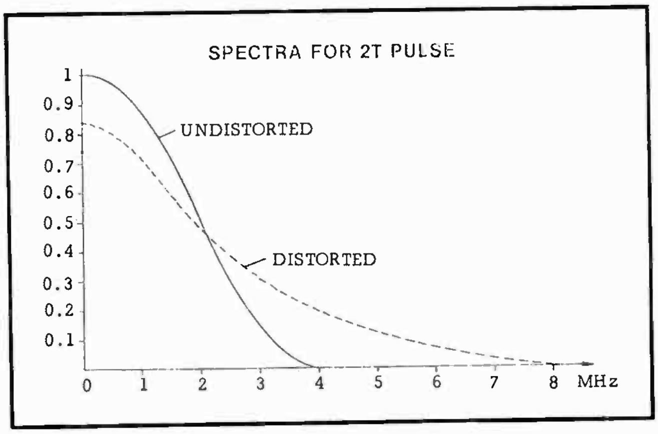


Figure 15.

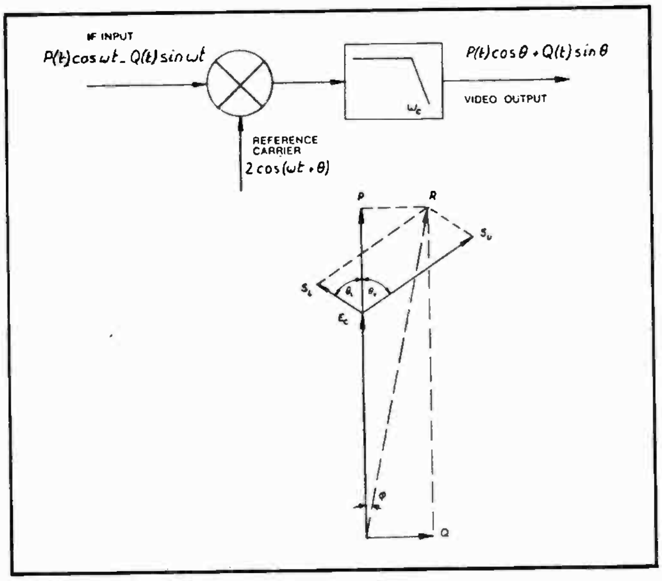


Figure 16.

The effect of envelope detection on the 2T pulse is shown in **Figure 15.** The normal frequency spectrum of the 2T pulse is 4 MHz. Because of the presence of the higher harmonic components, the spectrum resembles that of **Figure 13.** Therefore, the **H.A.D** will be narrower, and the actual width and amplitude of the 2T pulse depends on the cut-off frequency of the low-pass filter following the detector circuit.

Waveform interpretation is therefore seriously impaired by using an envelope detector unless we know exactly what the effects on the different signals are of the demodulator in question.

This is one reason why the Multiburst signal is not normally used at full amplitude. By reducing the bursts to 75%, the effects of quadrature distortion are reduced.

To distinguish between the effects of envelope delay and quadrature distortion, the 2T pulse is reduced by 10 dB so that the distortion of the pulse is mainly caused by envelope delay errors.

However, these are not very practical solutions. This is why for System I it has been proposed to use the original test signals but to employ criteria for assessment of distortion which are different from those normally used for linear systems. (6)

A 5.8 MHz phase-compensated low-pass filter is used at the output of the demodulator, and the mask normally used for K-rating assessment is modified to take the effects of pulse narrowing and overshoots into account. Similar steps have been suggested for other test signals including the stairstep signal.

Synchronous or Product Detector

(See Fig. 16)

If we consider expression (2b) from Appendix 1 and multiply this by a sinewave of the same frequency as the carrier frequency, but differing in phase by an angle θ, then the output of the detector can be written as:

$$e(t) = \left[P(t)\cos\omega t - Q(t)\sin\omega t\right] 2\cos(\omega t + \theta)$$

$$= 2P(t)\cos\omega t \cdot \cos(\omega t + \theta) - 2Q(t)\sin\omega t \cdot \cos(\omega t + \theta)$$

$$= P(t)\left[\cos\theta + \cos(2\omega t + \theta)\right] + Q(t)\left[\sin\theta - \sin(2\omega t + \theta)\right]$$

$$= P(t)\cos\theta + Q(t)\sin\theta$$

$$+ P(t)\cos(2\omega t + \theta) - Q(t)\sin(2\omega t + \theta)$$

If this signal is passed through a low-pass filter that will eliminate the two last terms in the expression, then the output of the product-detector reduces to

$$P(t)\cos\theta + Q(t)\sin\theta$$

When $\theta = 0$, then $\cos\theta = 1$ and only the in-phase component P(t) will be available. When $\theta = \pm\pi/2$, then $\sin\theta = \pm 1$ and only the quadrature component will be available.

Since $P(t) = \frac{E}{2}(1 + M\cos pt)$, no distortion will occur when $\theta = 0$. In the foregoing we have assumed that the reference carrier used is a "clean" carrier. Any amplitude modulation or noise on this carrier will be transferred to the detected output as noise.

Advantages of Product Detector

a- The ideal product detector is linear in the sense that the output with two signals applied at the input is simply the sum of the outputs obtained when the inputs are applied individually. This is in contrast with the envelope detector, where the output with two signals applied to the input will generate unwanted cross-talk products. For example, when both the color subcarrier and aural carrier are present, an unwanted 930 kHz product is generated.

In addition, there will be phase modulation of the weaker signal by the stronger and when both carriers

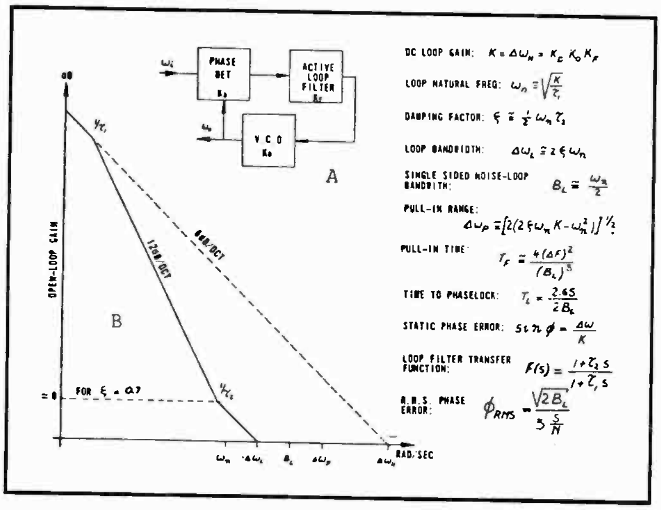


Figure 17.—A & B

are about equal in amplitude, there will be phase and amplitude modulation of each by the other (7).

b- The noise power spectrum at the output of the product detector is one-half that of the input signal. Product detection should therefore give a **3 dB S/N** advantage over envelope detection.

c- When a noisy signal is applied to an envelope detector, the noise will also be rectified and will produce a **DC** component which could prevent the detection of white picture information. The product detector does not have this limitation.

d- Since in an envelope detector the video information is separated from the carrier by **RC** time constants, the IF frequency cannot be arbitrarily low. With the product detector, a low **IF** frequency can be used since product detection can be considered as a shifting of the input signal in the frequency domain.

Regeneration of the Reference Carrier:
(8-13)

Since the reference carrier for the product detector is not normally available, this carrier has to be derived from the incoming signal.

Basically the system used in this demodulator is a **PLL** synchronized by a sampled carrier signal occurring at line rate.

Since numerous papers have been written on **PLL** we will limit ourselves to a review of the basic principles as they relate to our particular application. **Figure 17a** shows the components of a **PLL.** The system contains:

A **VCO** with a nominal frequency equal to the desired output frequency,

A **Phase Detector** which performs an integrating function. That is, it provides an output voltage proportional to the integral of the frequency error (phase-error),

An **Active Low Pass Filter** having a specific transfer function and which removes the time variant components from the phase-detector output before it is applied to the **VCO**.

The performance of a **PLL** can best be illustrated by its **Bode Plot.** A typical plot is shown in **Figure 17b** and all of the important frequency parameters are indicated in their approximate location. The **PLL** used is of the second order and the **Loop Filter** is of the proportional-plus-integral control type.

Desirable Characteristics of PLL:

a- The locally generated reference signal must have the same frequency as the transmitter carrier and be at a fixed phase relative to it. Further, this phase relationship must be preserved even when the transmitter carrier departs from its nominal frequency value.

Since the modulated carrier frequency can change by as much as ± 1kHz and the locally generated carrier can also change by ± 1 kHz, the worse-case frequency error can be as much as ± 2kHz. This error has to be reduced to a phase error of a few degrees in order to keep the phase variation between the incoming and locally generated carrier at a minimum. This **Static Phase** error can be reduced to any desirable value by increasing **the DC Loop Gain** of the system.

Increasing the **DC Loop Gain** will also increase the **Hold-in Range** since the higher the gain the more the input frequency can change before a ±90° phase error is reached between the two input signals of the phase detector.

The **Hold-in Range** is: $\Delta W_H = K_D K_o K_F$ and is determined by the conversion gain of the phase-detector K_D, the sensitivity of the **VCO** K_O and the **DC** gain of the Loop Filter Amplifier K_F. A given total **Hold-in Range** can be achieved with a large K_D and a small K_O or vice versa. However, in a practical system the input to the **VCO** is usually the most noise-sensitive point in the loop and therefore a large value of K_D and small value of K_O are desirable. This can be achieved with a high output phase detector and a "stiff" **VCO**. An oscillator whose frequency is hard to pull has usually good long and short term stability and produces a cleaner output signal than a "loose" **VCO**. For these and other reasons, the **VCO** in this demodulator is designed as a crystal oscillator (**VCXO**).

b- The **PLL** should also produce an output signal that has high **Dynamic Stability**. This means that it should generate a jitter free output signal even though the sampled input signal may be noise phase modulated.

The **RMS Phase error** can be made small by reducing the **Noise Bandwidth** of the loop filter. This is the purpose of the **Active Low Pass Filter**, which averages the phase information from a large number of successive carrier samples. Decreasing the **Noise Bandwidth** will, however, reduce the **Pull-in**

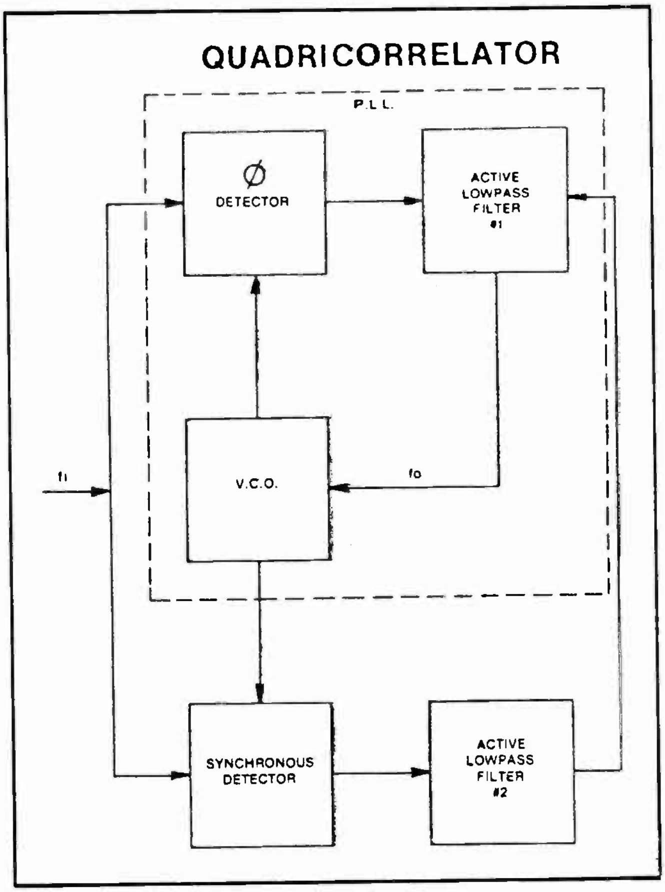


Figure 18.

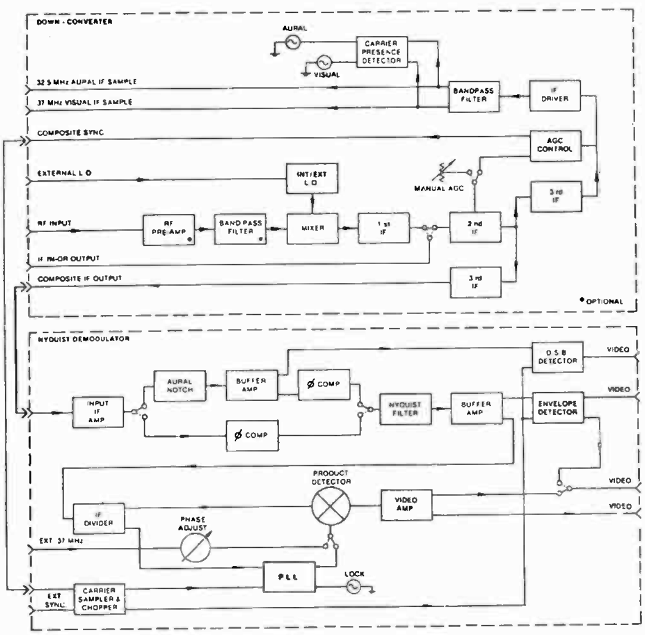


Figure 19.

Range and will drastically increase the Pull-in-Time.

c- The PLL should also minimize spurious spectral components. Because of the gated nature of the input signal, sidebands near the carrier frequency which are separated by integral multiples of the horizontal line frequency are also applied to the phase detector. Since the first sidebands are about the same in amplitude as the carrier frequency, sidelock could occur. However, the PLL can not be pulled this far so that sidelock cannot happen. However, spurious sideband signals can be generated by the PLL unless the time constants of the active low-pass filter are properly chosen so that the impulse nature of the input signal is completely smoothed out.

Improved PLL

Since good noise immunity and wide pull-in are not compatible, it would be best to use a two-mode **PLL** system whereby phase lock is achieved by one circuit and phase lock is maintained by another. The **DC Quadricorrelator** is such a system, and the design can be optimized for desired **Pull-in, Noise Bandwidth**, and **Static Phase Error**. A block diagram of the actual Quadricorrelator used is shown in **Figure 18**. The circuitry within the dotted line is the basic **PLL**; however, the transfer characteristic of the **Active Low-pass Filter** is switchable between a wide band mode and a narrow band mode.

When the system is not in-lock, no output is available from Phase detector #2, and the Active Low-pass filter is in the wide band mode. Therefore, the pull-in range is extended, and the **PLL** will lock-up. When this happens an output will be available from Phase detector #2 which will operate **FET** switch in the #1 Active Low pass filter, and its bandwidth will now be reduced to the desired range.

Down-Converter
(See Fig. 19)

The block diagram shows two units. The Down Converter and Nyquist Demodulator. Depending on the L. O. chain used, the Down Converter will accept any TV Channel from Channel 2 thru 63. The incoming signal is down-converted to the Gates standard IF frequencies of 37 MHz for Visual and 32.5 MHz for Aural.

The mixer used is of the double-balanced type and covers all TV Channels.

The first **IF** amplifier is a double-tuned coupled circuit followed by a Low-pass filter. The cut-off frequency of this filter is 42 MHz and the response is down 40 dB at Channel 2. The output can be fed to the 2nd **IF** amplifier or to the rear panel **BNC** connector.

The second **IF** amplifier is the **AGC IF** amplifier. The signal is amplified by 20 dB and attenuated by the **AGC** pin-diodes by a level depending on the **AGC** control setting and the level of the **RF** input signal. The range of the **AGC** is at least 40 dB. However, only 20 dB is normally used since over this range the amplitude response remains within 0.2 dB and the two tone **IM** products are down at least 60 dB.

The automatic AGC can be bypassed by a switch on the front panel and manual adjustment can then be used to control the input signal.

The third amplifier has two outputs, one which connects to the **Nyquist** Demodulator, the other to the **AGC** control circuitry and **IF** driver. Since keyed **AGC** is used, composite Sync is also available from the **AGC** circuit via a connector on the rear panel.

The **IF** drive feeds two Band-pass-filters which separate the Visual from the Aural **IF** carrier. The response of these filters is very narrow band—20 dB for 1 MHz bandwidth, 60 dB for 3 MHz bandwidth. A sample of these signals is peak-detected and used as carrier presence indicators. **LED's** are available on the front panel and dry-contacts are brought out on a terminal strip on the rear panel for an Alarm indicator.

IF samples from both the Visual and Aural Carriers are available on the rear panel for the Aural and Frequency monitors.

Nyquist Demodulator

The composite **IF** signal from the Down-Converter is first amplified in the **Input** amplifier and can then be applied to the Aural Notch or can bypass the Aural Notch. In the first case, the Aural frequency components are eliminated and the signal is then applied to a Buffer Amplifier-divider. One output of this divider feeds the Double Sideband Detector, the other feeds the Envelope-delay-corrector, which compensates for the Aural Notch and Nyquist Filter to the inverse of the Receiver-Equalizer curve. When in the Aural Notch bypass-mode, a second envelope-delay corrector is used which compensates for the Nyquist filter only.

The Nyquist filter output signal is first amplified and divided in a Buffer Amplifier. One output is applied to the Envelope Detector, the other to the Product Detector Circuitry. In the IF divider the signal splits and is applied to the Product Detector and to the **PLL.**

The composite Sync signal from the Down Converter is applied to the Carrier Sampler pulse and zero carrier reference pulse generator. In the **PLL** the incoming **IF** signal is sampled at line rate, and this signal is used to control the **PLL** as explained in Chapter 7.

This regenerated carrier is applied together with the modulated Visual **IF** signal to the Product Detector. The video signal obtained from this process is first applied to a Low-pass filter and then amplified in a video amplifier.

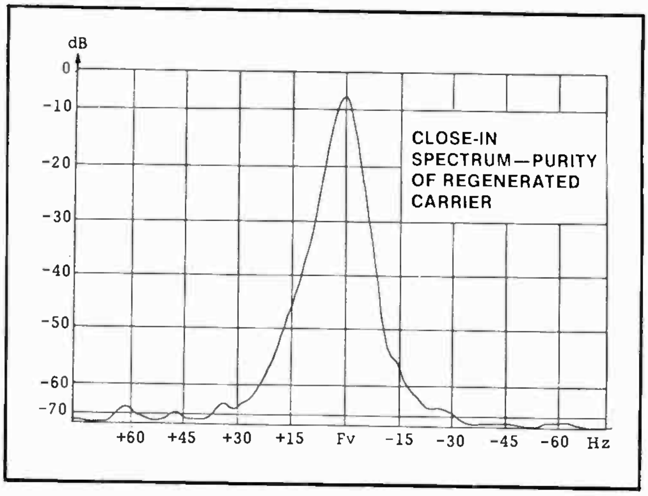


Figure 20.

A vertical rate zero carrier reference pulse is also generated in the Carrier Sampler and Chopper board and this pulse is applied to the Envelope and Double Sideband detectors.

If it is desired to operate the Demodulator from an external **LO** and **IF** frequency source then these signals can be applied to the Down Converter and the Nyquist Demodulator via connectors on the rear panels.

Test Data

We will present a limited number of test data in this paragraph.

a- **P.L.L.**

In order to achieve good noise immunity, the loopband width has to be very narrow. Since a sampled carrier signal is used to phase lock the locally generated carrier, a strong horizontal frequency component as well as weaker 60 Hz and 30 Hz components are measureable at the output of the phase detector. All of these components have to be rejected by the loop filter to avoid phase modulation of the **VCO. Figure 20** shows the spectrum of the output signal of the **PLL.** The vertical dB scale was calibrated using a switchable attenuator. Weak 60Hz and 30Hz components are present at carrier ±60 Hz and carrier ±30 Hz, but these are down at least 65 dB. When attempting to measure these components as AM and FM noise these components were down at least 67 dB with respect to 100% AM and 25 kHz deviation FM. This was close to the measuring capabilities of the test equipment being used. In fact, no measureable difference was noted when the sampling circuit was bypassed. When purposely phase modulating the visual carrier before AM modulating this carrier with video, it was noted that the **PLL** would reject this phase modulation by at least 20 dB, when comparing the phase modulation of the applied carrier to this of the regenerated carrier.

b- **Frequency Domain Measurements**

The amplitude and envelope delay response versus frequency are given in Figures 3 through 6

c- **Time Domain Measurements**

Figures 21 — 26 compare the Multiburst signal for envelope and product detection. The post detector amplifier had a flat response up to 7 MHz and was down 12 dB at 10 MHz. Both signals look much the same, but notice the waveform distortion on the 1 MHz component and compare the **DC** axis shift.

Figures 27 — 31 compare the composite signal for envelope and product detection. Note the difference in amplitude of 2T, 12.5T, and window signal and compare the 2T pulse response. When reducing the amplitude of the 2T pulse before modulating by 10 dB the result as shown in **Figure 26** is observed when comparing this with **24** we notice that there is still some quadrature distortion noticeable.

Conclusions

In conclusion it can be said that substantial improvement in waveform interpretation can be obtained by using a product detector in combination with an almost ideal Nyquist Filter that has been optimized for envelope delay at IF to within ±10 ns.

In this paper we have considered only two aspects of TV demodulators—the Nyquist Filter and the product detector versus envelope detector. We hope to present in the near future a study on Aural Notch shaping and its effect on the envelope delay, as well as an evaluation of the present **FCC** standard of the receiver-equalizer envelope delay curve.

It is hoped that our initiatives may result in reviving the interest in the Broadcast Industry in defining tolerance limits on **IF** waveshaping and that serious consideration should be given to the use of a product detector.

It is believed that this can only result in a more uniform optimization of TV transmitters which is beneficial to all of us.

Acknowledgment

The authors wish to acknowledge the co-operation of the many who have contributed to the development and testing of the equipment described in this paper—in particular to C. Smiley, G. Mendenhall, A. Spielbauer. Acknowledgment is also made to the Management of Harris' Broadcast Equipment Division for permission to present this paper.

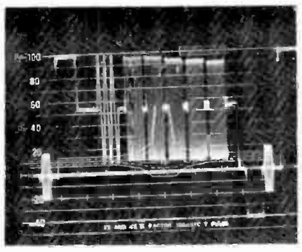
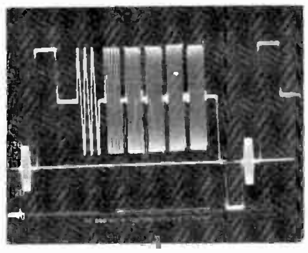


Figure 21.



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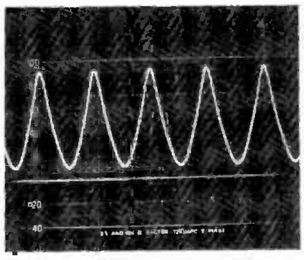

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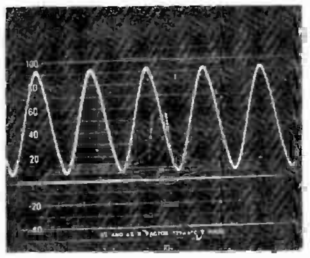

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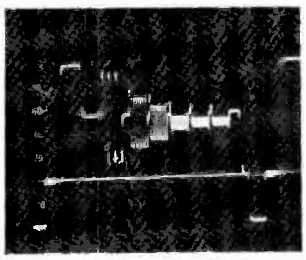

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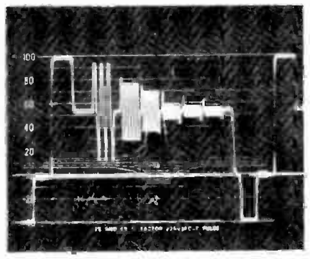
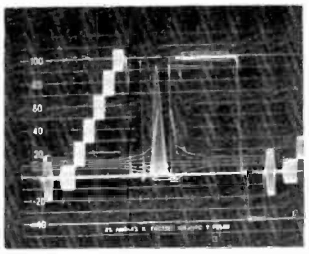


Figure 26.



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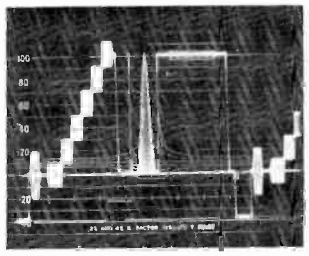

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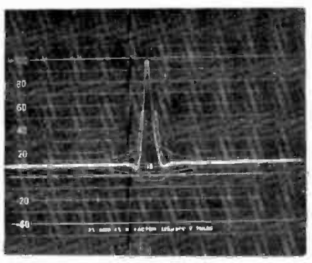

Figure 29.

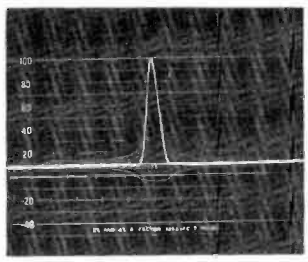

Figure 30.

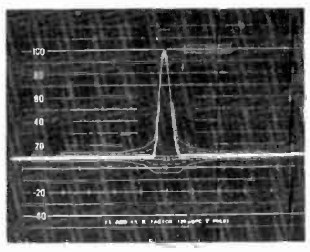

Figure 31.

Appendix 1—Distortion Caused by Envelope Detection

The modulation of a carrier of frequency ω by a voltage of frequency p can be expressed as:

$$V_m = E(1 + M \cos pt)\cos \omega t$$

$$= E\cos \omega t + \frac{EM}{2}\cos(\omega + p)t + \frac{EM}{2}\cos(\omega - p)t$$

If the lower sideband is removed and the carrier is reduced by 6 dB then:

$$V_m = \frac{E}{2}\cos \omega t + \frac{EM}{2}\cos(\omega + p)t$$

$$= \frac{E}{2}\cos \omega t + \frac{EM}{2}\left[\cos \omega t \cos pt - \sin \omega t . \sin pt\right]$$

$$= \frac{E}{2}\left[1 + M\cos pt\right]\cos \omega t - \frac{EM}{2}\sin pt \sin \omega t$$

$$= P(t)\cos \omega t - Q(t)\sin \omega t$$

In this expression, the first term is of the same form as (1) but reduced to half its original value; the second term is in quadrature with the first term.

The envelope detector will respond to the modulus of this expression or to:

$$\sqrt{\frac{E^2}{4}(1 + M\cos pt)^2 + \left(\frac{EM}{2}\right)^2 \sin^2 pt}$$

$$= \frac{E}{2}\sqrt{1 + M^2 + 2M\cos pt}$$

$$= \frac{E}{2}\sqrt{1 + M^2} \cdot \sqrt{1 + \frac{2M}{1 + M^2}\cos pt}$$

The last term in this expression is of the form $(1 + x)^n$ and can therefore be expanded into a power series as:

$$(1 + x)^n = 1 + nx + \frac{n(n-1)}{2!}x^2 + \frac{n(n-1)(n-2)}{3!}x^3 + \ldots$$

In this case $n = \frac{1}{2}$ and $x = \frac{2M}{1 + M^2}\cos pt$

Therefore, limiting the series to the terms of the 5th power (3) can be written as:

$$\frac{E}{2}(1 + M^2)^{1/2} \times \left\{ 1 - \frac{1}{4}\left(\frac{M}{1+M^2}\right)^2 - \frac{15}{64}\left(\frac{M}{1+M^2}\right)^4 \right.$$

$$+ \cos pt\left[\frac{M}{1+M^2} + \frac{3}{8}\left(\frac{M}{1+M^2}\right)^3 + \frac{35}{64}\left(\frac{M}{1+M^2}\right)^5\right]$$

$$+ \cos 2pt\left[-\frac{1}{4}\left(\frac{M}{1+M^2}\right)^2 - \frac{5}{16}\left(\frac{M}{1+M^2}\right)^4\right]$$

$$+ \cos 3pt\left[\frac{1}{8}\left(\frac{M}{1+M^2}\right)^3 + \frac{35}{128}\left(\frac{M}{1+M^2}\right)^5\right]$$

$$+ \cos 4pt\left[-\frac{5}{64}\left(\frac{M}{1+M^2}\right)^4\right]$$

$$\left. + \cos 5pt\left[\frac{7}{128}\left(\frac{M}{1+M^2}\right)^5\right] \right\}$$

From this expression it can be seen that harmonic components are generated which are a function of the modulation depth.

The amplitude of the harmonic components relative to the total signal amplitude can easily be determined from (4) for different values of M. This will result in the graphs of Figure (10) for the 2nd and 3rd harmonics.

In this Appendix we have only considered the harmonic distortion caused by detecting an SSB in an envelope-type detector. By intuition, we can conclude that the harmonic distortion for modulating frequencies along the Nyquist slope, where both sidebands are partially present, will decrease as the modulating frequency decreases. For a detailed study on this subject, we refer to Reference 14.

References

1. C.C.I.R. DOC XI/26E Oct. 1964
2. Telefunken Zeitung Vol 38 1 1965 pp 3-14
   Der Einfluss der Leuchtdichtesspruenge bei der Restseitenbandubertragung von NTSC, Pal und Secam Farbfernsehbildern auf die Farbflachen und Farbubergange. W. Bruch
3. N.T.Z. Vol 10 1965 pp 570-577
   Systembedingte, Durch die Restseitenbanduebertragung im Hochfrequenzkanal Hervorgerufene Pegelabhengigkeit der Phase des NTSC-Farbtragers. H. Lindenmeier, H. Eisenmann
4. RCA Review Dec 1965 pp 580-611
   Transient Response of Detectors in Symmetric and Asymmetric Sideband Systems.
   T. Murakami, R. W. Sonnenfeldt
5. RCA Review March 1949 pp 17-34
   Standardization of the Transient Response of Television Transmitters.
   R. D. Kell, G. L. Fredendall
6. IEE Conference Publication 69 pp 216-218
   The Interpretation of Waveform Measurements Made Through V.S.B. Transmission Networks.
   F. H. Wise
7. RCA Review Sept 1959 pp 455-472
   Transient Cross-Modulation in the Detection of Asymmetric Sideband Signals.
   T. Murakami, R. W. Sonnenfeldt
8. Proc Ire June 1959 pp 1137-1143
   A New Look at the Phase-Locked Oscillator.
   H. T. McAleer
9. Proc Ire Jan 1954 pp 106-133
   Color-Carrier Reference Phase Synchronization Accuracy in NTSC Color Television.
   D. Richman
10. Proc Ire Jan 1954 pp 288-299
    The DC Quadricorrelator
    D. Richman
11. Ire Transactions on Information Theory March 1955 pp 66-76
    Design and Performance of Phase-Lock Circuits Capable of Near-Optimum Performance Over a Wide Range of Input Signal and Noise Levels.
    R. Jaffe, E. Rechtin
12. Gardner F. M. Phase Lock Techniques Wiley, N.Y. 1967
13. IEEE Transactions on Communications Vol Com-21 #10 Oct 1973
    Burst Synchronization of Phase Locked Loops
    L. Schiff
14. Alta Frequenza Vol XXXVIII #5 May 1969 pp 385-389
    Modulation Envelopes. Tables of Contours and Harmonic Components. C. Egidi, P. Oberto

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Improving Microphone Utilization Techniques Or—Am I Using The Right Microphone?

Almost everyone involved with the use of microphones has a fairly good understanding of what a microphone is and some understanding of what it does . . . and most have their own favorites from among the many different types that are available. Few of us, however, have the occasion to really analyze the important effects of the application on a microphone's performance. The continuing desire for variety and innovation in programming leads to a variety of microphone applications . . . and often to frustrating audio problems. This presentation is intended to provide a little additional insight into some of the ways in which the subjective performance of a microphone is dependent on the application and, hopefully, to suggest some ways in which the utilization and choice of microphones may be more effectively accomplished.

To give an idea of what is meant by the relationship between subjective performance and application, consider the two frequency responses shown in **Figure 1.** A person with some experience in interpreting frequency response curves could quickly characterize the relative sound of the two microphones shown; the dashed response perhaps being described as "thin" or "shrill", or the solid response as "smoother" or "less bright". Actually, though, these two microphones are intended to sound as much alike as possible . . . when they are used in the intended manner. The microphone represented by the solid line is designed for use on a stand or in the hand in a normal fashion in front of the performer. The microphone represented by the dashed line, however, is intended to be worn as a lavalier, at a 90° angle to the performer's voice axis and against the chest; and the basic response is tailored to compensate for the high frequency roll-off and low frequency boost in this position. If both were used on a stand, the characterizations made from the response curves would be valid . . . but, when used as intended, these characterizations would not apply. The subjective performance, as you can see, is dependent on the application.

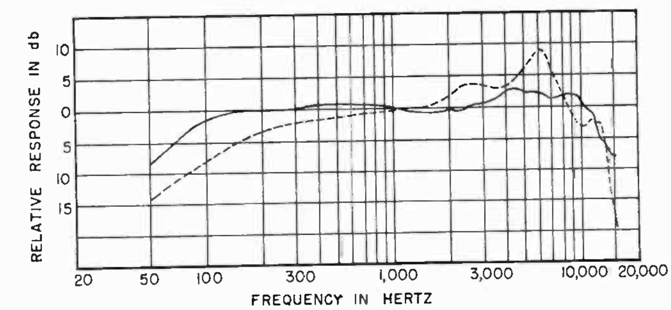


Figure 1.

The various factors relating to this performance versus application concept can, for purposes of this presentation, be grouped into three principal categories.

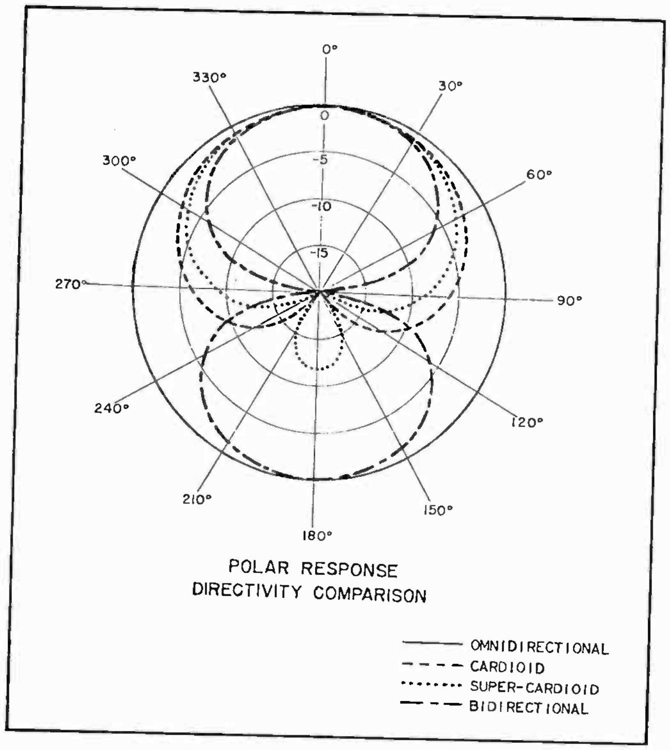


Figure 2.

DIRECTIONAL PROPERTIES

PROPERTY	OMNI-DIRECTIONAL	CARDIOID	SUPER-CARDIOID	COSINE
FRONT TO BACK(1) REDUCTION (ON AXIS)	0	INFINITE (THEORETICAL)	12 DB	0
RANDOM ENERGY(2) REDUCTION	0	9.6 DB	11.4 DB	9.6 DB
EFFECTIVE PICKUP ANGLE (3) 3 DB DOWN 6 DB DOWN	360° 360°	132° 180°	114° 156°	90° 120°
NULL ANGLE(4)	---	180°	126°	90°

NOTES:

1. PICKUP REDUCTION AT 180° COMPARED TO 0°.
2. REDUCTION IN PICKUP OF SURROUNDING SOUND -- COMPARED TO OMNI-DIRECTIONAL.
3. INCLUDED ANGLE AT MICROPHONE FRONT FOR 3 OR 6 DB DOWN AT EXTREMES.
4. ANGLE AT WHICH MINIMUM PICK UP (NULL) OCCURS. AS WITH CARDIOID (SEE #1) REDUCTION AT NULL IS THEORETICALLY INFINITE.

Figure 3.

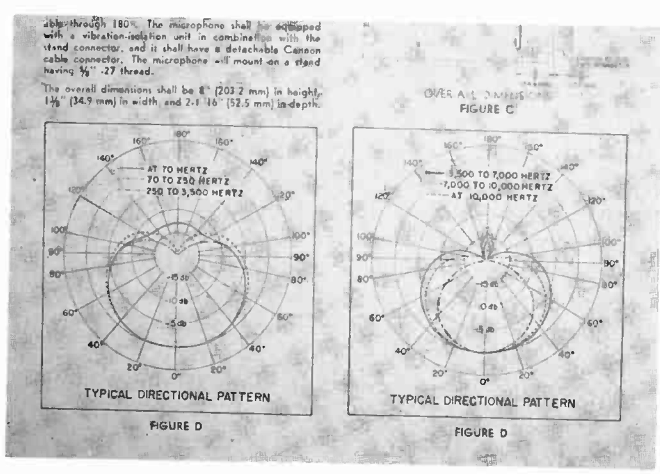


Figure 4.

Factor Relating to Directivity

Directional Types

The purpose of a directional microphone, of course, is control . . . control of noise from any of a number of sources, separation between vocalists and instruments, or between various instruments in music reproduction, and control of acoustic problems such as echos and reverberation. **Figure 2** shows the polar pattern (directional characteristic) for each of the most common types of directional microphones. Obviously, the different types are to provide optimum control of sound originating from different locations with respect to microphone and talker. In the early days of broadcasting it was quite common to see bi-directional ribbon microphones for many applications. The 90° rejection characteristic was very valuable with a microphone suspended over a desk facing toward the announcer and the dead area pointing down to discriminate against the script shuffling on the table. They were used on stands with the performer talking into the live side and the script held low, again to avoid the paper noise . . . as well as sounds produced by other performers standing to the side of the microphone waiting for their turn to speak.

The cardioid (uni-directional) has well known advantages in discriminating against sound originating from the rear, and variations of the cardioid with the minimum pick up angle moved around to between 120° and 130° also are useful in a variety of applications. The table in **Figure 3** suggests some of the differences between the various directional types and should provide some insight into what advantage a given type might have in a given application.

Obviously, the essence of using directional microphone types is to take full advantage of the maximum and minimum pick up areas . . . matching the action and the spurious sound sources in the best possible way.

Uniformity Of Directional Properties

Unfortunately, the selection and use of directional microphones for control is not quite as straightforward as it might appear from the idealized polar patterns in Figure 2 or the sterile numbers in the chart . . . and many of us have discovered this without really knowing why. For example, two different cardioid microphone models may be used almost interchangeably in one application but sound quite different in another. Why? Frequently this ambiguity is due to a factor called "uniformity".

A polar pattern for a microphone is generally measured by feeding a steady tone to the microphone and rotating the microphone with respect to the tone source. The resulting polar graph (such as those in Figure 2) represents the directional characteristic of the microphone at a given frequency. But what is its directional characteristic at other frequencies? If the directional performance is reasonably constant at different frequencies the microphone is said to be uniform, or to exhibit uniformity. Figure 4 shows the polar characteristics of a SM 56 microphone at a number of different frequencies. From 70 Hz up to several thousand Hz it will be seen that the variation in pattern is really only a matter of a few db. This is actually a very uniform microphone. By contrast, **Figure 5** shows the polar properties of an older version, the 556S, at several frequencies. Here there is considerable variation, to the point where the microphone scarcely resembles a cardioid pattern at all at 100 Hz. This characteristic of greatly reduced directivity at lower frequencies is very common in uni-directional microphone types . . . and helps explain a number of audio problems. It is apparent, for example, that the less uniform microphone would do a poorer job of reducing air conditioning or crowd noises which have large low frequency components. Band separation, particularly from the bass instruments, would certainly be different with this microphone than it would with the quite uniform model.

But this factor of uniformity relates not only to such matters as noise or unwanted sound rejection, but also to the sound of the microphone itself. To understand this, examine **Figures 6 and 7.** These are 0° (front) and 180° (rear) frequency response curves measured for these same two microphones. In the case of the uniform microphone, sounds originating from off axis (while their level will be reduced) will sound very much the same as sounds entering from the front. In the case of the non-uniform microphone, however, off axis sounds will have a distinctly "bassy" or "muddy" sound. In any fairly live studio or control room, then, the reflective sounds entering from the sides or rear of the uniform microphone will not significantly influence the sound quality. Such sounds, however, entering the non-uniform microphone will impart a "bassy" or "muddy" quality to the overall apparent sound of the unit.

This is an important concept in what is called the "reach" of a microphone. If these two examples were used at a distance such as six feet from a performer in a fairly live room, it is evident that the sound reproduced from the uniform microphone would be well balanced and would give a good feeling of presence. The sound reproduced by the non-uniform unit, however, would have a distinctly "distant" sound, due to the off-axis colaration.

The uniformity of a microphone can be rather easily checked subjectively. In a fairly quiet and dead room place the microphone on a stand and walk around it, talking at a uniform distance as you cover 360°. With someone riding gain (important, because of the Fletcher-Munson Effect)[1] record the test at a uniform level. With a uniform microphone the voice quality will be essentially the same at the various angles around the microphone. A nonuniform microphone should be quite recognizable, and the nature of the difference can be noted for future reference by subjectively analyzing the change in voice characteris-

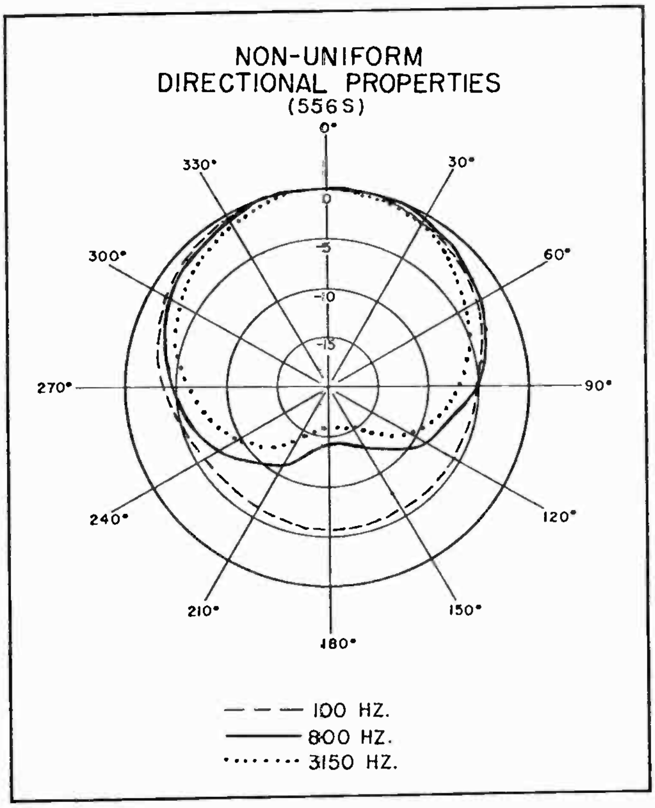


Figure 5.

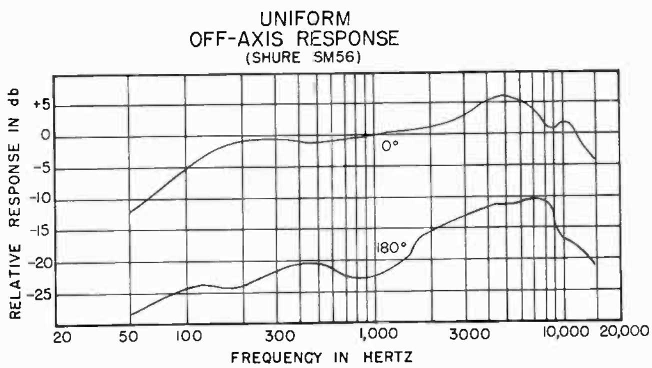


Figure 6.

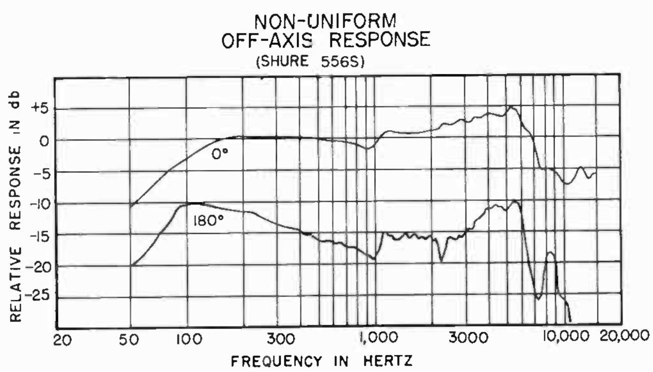


Figure 7.

[1] The Fletcher-Munson Effect related to a property of our hearing system which produces a disproportionate change in perceived loudness of different parts of the audio spectrum as the overall sound level changes. This is the reason for "loudness controls" in hi-fi systems.

tic. (This does not necessarily suggest that the microphone which exhibits a change be thrown . . . out but only that this is a factor which must be taken into account in making proper use of that microphone.)

Symmetry Of Directional Properties

A factor of similar significance has to do with the fact that the directional pattern of a microphone is actually three dimensional . . . that is, for example, the pick up of a non-directional microphone should be visualized as a sphere rather than a circle. Some microphones do not exhibit the same directional characteristics in every plane. For example, 90° off axis performance to the top, bottom, left, or right may be different. Of course, this departure from the expected directional performance would have effects similar to (but generally less severe than) the effects of non-uniformity. In general, lack of directional symmetry can be anticipated from any microphone which does not exhibit true geometric symmetry around its 0°-180° axis. This geometric asymmetry might be in the shape of the microphone's case or asymmetry in the sound openings, such as a longitudinal series of openings on only one side of the microphone.

Distance-Proximity Effect

With any directional microphone there is a tendency for low frequency output to increase disproportionately as the source moves closer to the microphone. This low frequency exaggration for close microphone-to-subject spacing is known as the proximity effect.

As shown in **Figure 8** this effect may occur in varying degrees but, again, it is a property common to all directional microphones (except parabolic). In the unidirectional types the effect is most noticeable with microphones using a single rear entry spaced fairly close to the diaphragm, and somewhat less in microphones using two or more rear entries with a considerable spacing between the diaphragm and the furthest of these additional entries.

As with the concept of non-uniform directivity, proximity effect is not arbitrarily good or bad, but rather an effect which must be recognized in using microphones most effectively. Frequently, proximity effect is used to advantage. The use of this low frequency boost to augment the low frequency content of the speaking voice is quite well known and very frequently used in both radio and television (although it must always be recognized you can't boost something that is not there . . . although there are many who try). It is also used to advantage by vocalists who will work very close to a microphone for a warm, intimate sound and farther away when a "punchier" sound is desired. This effect is also used for additional control of the tonal quality of musical instruments, both in recording and broadcasting.

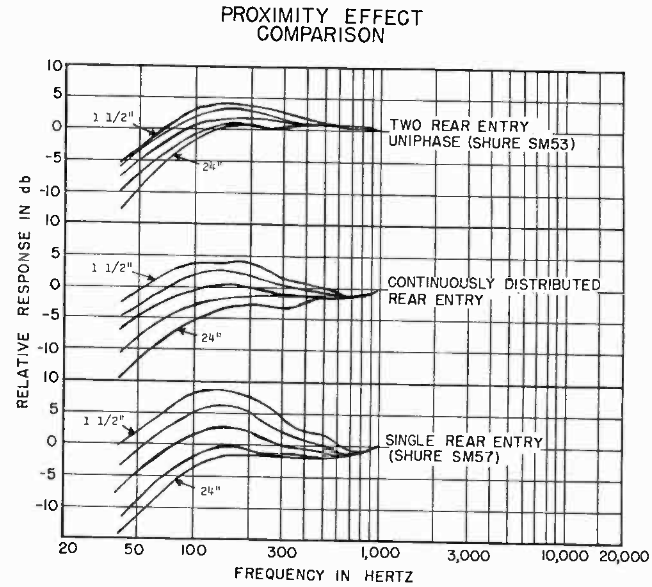


Figure 8.

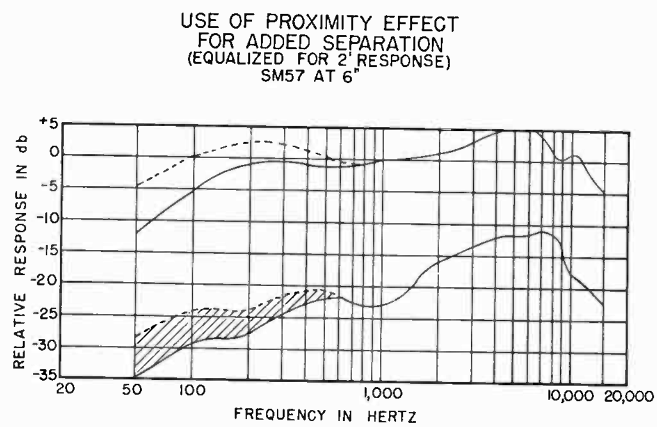


Figure 9.

The recording industry, and many broadcasters as well, also take advantage of this effect for added separation in music reproduction. **Figure 9** indicates the proximity effect of a microphone showing response both at 24 and at 6 inches. Frequently a microphone with a fairly high degree of proximity will be used even when the additional low frequency level is not wanted. In this case a high pass filter will be used to remove the unwanted low frequency boost. As shown in the figure, this has the effect of also reducing the off axis low frequency response by a like amount thereby adding additional separation for the low frequency instruments which are so hard to control.

Of course, unwanted quality changes can occur accidentally with microphones exhibiting a high degree of proximity effect and so models which minimize this property are also offered by many manufacturers.

Factors Relating To Phase Effects

Microphone Phase

The effect of a pair of out of phase microphones equidistant from a talker is probably quite well known . . . with well matched microphones the signals would essentially cancel, greatly reducing the combined output. Out of phase microphones generally cause cancellation problems of some sort when any common source is involved, such as multi-microphone pick ups of large musical groups or spaced microphones for audience reaction. Since effects of accidentally out of phase microphones are usually undesirable, it is good practice to check phase of all microphones and cables in your facility.

It is also fairly well known that there are some valuable uses for intentionally out of phase microphones. One common example is for sports pick ups. Many stations use separate microphones for play-by-play and color, separated from each other and 180° out of phase. As long as the two announcers are at least three times as far apart as they are from the nearest microphone there is no detrimental effect in operating this way. The out of phase microphones provide, in effect, a bi-direction array for distant sounds and have roughly the noise reducing effect that would be expected from a bi-directional microphone . . . an overall background noise reduction in the order of 6 db.

Another useful application for out of phase microphones is in the pick up of piano. There is a fairly common two microphone pick up technique in which one microphone is mounted above the strings toward the treble end of the instrument for the impact or percussive effect while the other microphone is located on the opposite side of the sounding board to provide the overall fullness or richness of the instrument. Since the two sides of the sounding board are out of phase the two microphones must also then be used out of phase. This not only provides the proper tonal spectrum but has an additional benefit of some overall noise or room acoustics discrimination as described before.

Signal Phase

Cancellations or reinforcements due to in phase or out of phase signals are also an important consideration in microphone applications. There are several ways in which out of phase signals may be generated in fairly common situations which deserve some mention.

Reflections

Figure 10 is a third octave measurement of the response of a microphone used in a desk top application. The two different responses are measured in two different microphone locations, as indicated in the figure; in one case (dashed) on a conventional desk stand and in the other (solid) located at the desk surface. (The curves were measured with an essentially uncorrected loudspeaker and pink noise source and, therefore, do not exactly represent the normal free field response.) The normal response of the microphone without the desk top in the measurement situation used in virtually the same as the solid curve. In other words, the use of the conventional desk stand to elevate the microphone above the desk top produces a significant deterioration in performance . . . particularly in the critical "presence" range.

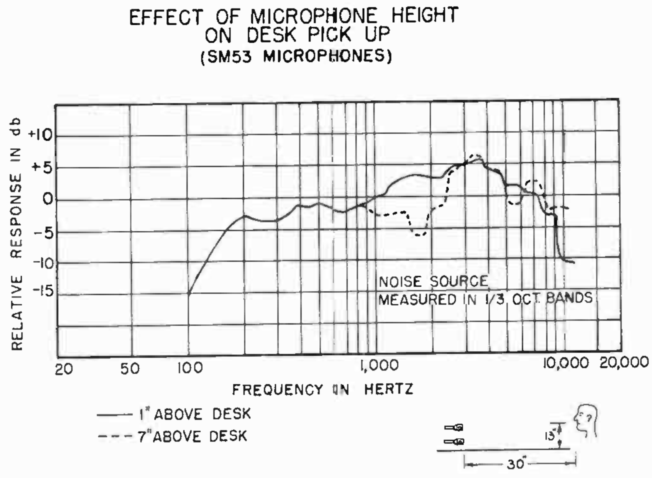


Figure 10.

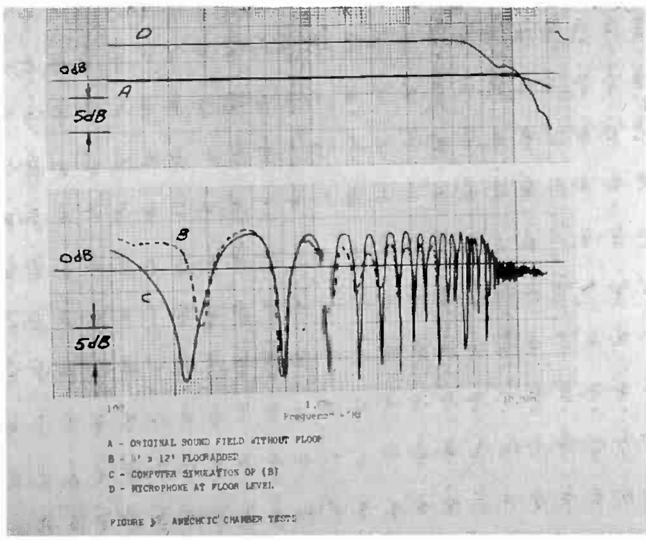


Figure 11.

The reason for this change is that the desk top acts as a second sound source by reflecting the talker's voice back up into the microphone. This reflected path is slightly longer than the direct path from the mouth to the microphone and, consequently, there are some frequencies at which the voice signal arrives in phase and some in which it is out of phase. This effect is shown graphically in **Figure 11.** Curve A represents a laboratory standard measurement microphone measured in an anechoic chamber. For Curve B the microphone was left in the same position but a hard plywood floor was introduced, paralleling the microphone to loudspeaker path. The resulting

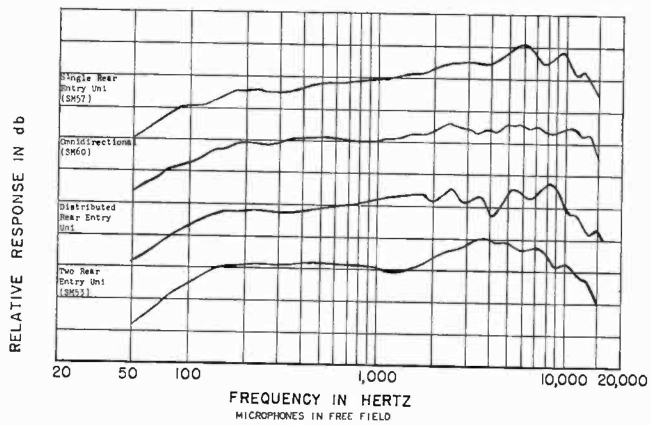


Figure 12.

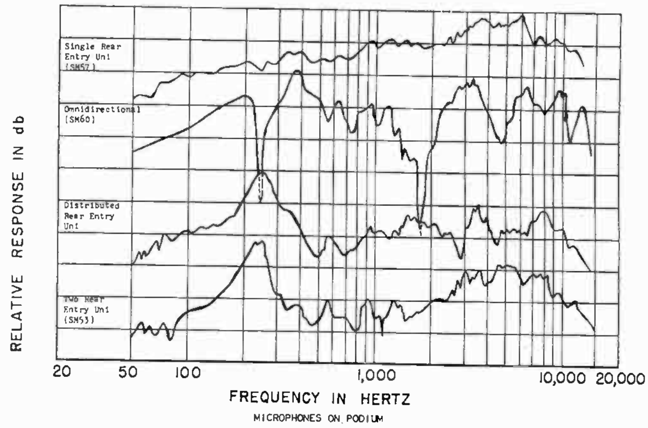


Figure 13.

phase interferences and additions due to the different path lengths of the direct and reflective signal can be plainly seen. (Curve C is a computer analogue of this set-up which was programmed for 100% reflectance. The cancellations are somewhat sharper but the agreement is quite striking.) Curve D is the response of the microphone with the floor still in place but with the microphone lowered to the floor surface. Note that the level is 6 db higher because of the two adding signals now essentially equal distance away.

This effect is strictly a matter of scale, and the frequency effects have to do with the ratio between the two path lengths. As a generality, effects in the audio range should be noticeable whenever the distance from the performer to the microphone is at least twice the distance from the microphone to the reflective surface.

As with the evident advantage to the low microphone mounting on a desk top application there may also be studio situations where, for example, it may be advantageous to place a boom microphone directly against a hard wall or even the floor for some difficult pick ups.

Resonances

Another situation which can produce not only out of phase signals but which can severely exaggerate some frequencies is often occasioned by "aesthetic" considerations . . . such as highly styled game show or news desks, specialized podiums and the like. **Figure 12** exhibits the frequency response curves for several different types of microphones. **Figure 13** shows the response of these same microphones mounted on an elaborate shrouded podium. The results are, to say the least, interesting.

The effects produced are partly due to reflections from the sides and surface of the podium, but the large disturbances are the result of a resonance occurring in a recess formed by a raised script surface and the shroud.

With the omnidirectional microphone (Second Curve) the resonance of this recess, and some harmonics, are clearly identifiable. The extreme dips in the microphone response are due to these exaggerated resonance tones from the "mouth" of the recess which are out of phase with the source at the microphone diaphragm. In the lower two curves for the multiple rear entry types these resonance signals at the "mouth" are sensed first by the lowest entries, which themselves must be out of phase with the front of the diaphragm (an essential provision in the directional system). Phase addition now occurs and large peaks are produced in the microphone response curve.

As can be seen from the top curve, the single rear entry unidirectional microphone is least affected by this resonance condition. The only rear entry position is significantly farther from the "mouth" of the resonant recess, and the exaggerated signals are rejected much as any sound originating from the rear. As a generality, the single entry microphone does prove to be most tolerant of this kind of situation . . . although, of course, the best approach is avoidance of such acoustical traps.

Microphone Intraction

Another type of situation in which out of phase signals can result in sound problems is in the use of multiple microphones to pick up a single source of sound. This situation can occur on talk or panel shows in which microphones are placed between the participants, on talk shows in which one microphone is used for the host and another microphone is used for guests, and on podiums where for one reason or another more than one microphone is used.

To visualize this effect consider the podium situation. **Figure 14** illustrates a somewhat common situation in which microphones are spaced at some distance apart on a podium, ostensibly to allow for motion of the talker. The solid line indicates the response of the microphone pair with the talker centered between them, and is essentially similar to the response of a single centered microphone. In the situation illustrated if the talker moves as little as five inches to either side the resultant response is shown by the dashed curve. If he moves ten inches to either side (directly in front of either microphone) the dotted curve is the result. The response effects are quite significant and will be readily detectable if you set up a similar experiment yourself. The changes in response, of course, result from the phase cancellations and additions of unequal signal paths when the source is not equi-distant from the two microphones, and are readily apparent whenever the distance from either microphone to the source is less than one-third of that between the two microphones. (Incidentally, this effect is a common trap in the miking of a grand piano with the use of two microphones above the strings and can seriously alter the reproduced sound of the instrument.)

The extremity of response change certainly indicates that this is a significant factor which must be watched with the use of more than one microphone to pick up a common source . . . and which certainly should be avoided in controlled situations such as a fixed podium.

On a podium or in situations such as a talk show with audience the effect on the directional properties of the microphones must also be considered. **Figure 15** shows the polar pattern resulting from the podium mounted microphone pair shown in **Figure 14.** In this case, the rejection at the rear is not seriously affected. (The irregularities at the front only explain the response changes previously seen.) However, the fact that the polar characteristic is affected at all may help explain some sound problems, especially when there is considerable background noise or potential feedback.

To eliminate this problem on a podium some have adopted the practice of using two microphones mounted very close together and angled so that their front pick up areas tend to overlap. **Figure 16** illustrates this situation and, indeed, the change in response in moving away from center is not nearly as severe as with the spaced pair . . . although the high frequency loss that results is still greater than would be encountered with only a single microphone mounted at the center.

Figure 17, however, shows that the deterioration in directional performance can be very serious. The two microphones used in this example are very uniform cardioid units and it is apparent that much discrimination has been lost.

In controlled situations such as podium when multiple microphones are felt to be necessary for redundancy it is generally far better to operate with only one microphone live . . . with the other connected but used simply as a stand-by. When the degree of operator control necessary to cover such an emergency is not available, the next best situation would be to center the microphones in an over/under configuration which greatly minimizes any possibility of this kind of phase interference. In either of these cases, the maximum effect of a position shift such as shown in these examples should not be more than about 3 db in level and the response change should be insignificant.

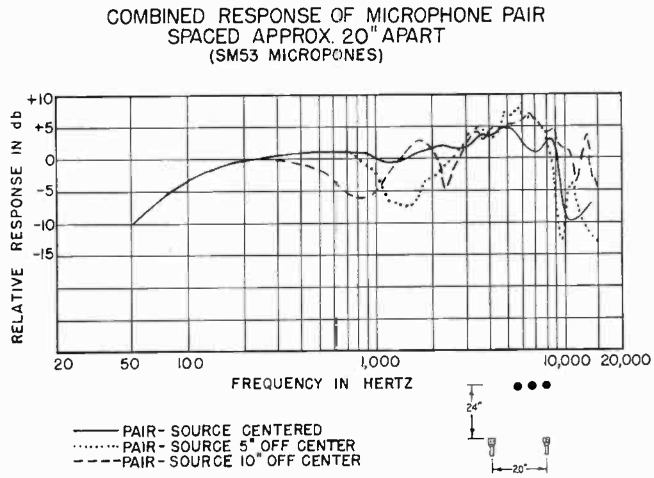


Figure 14.

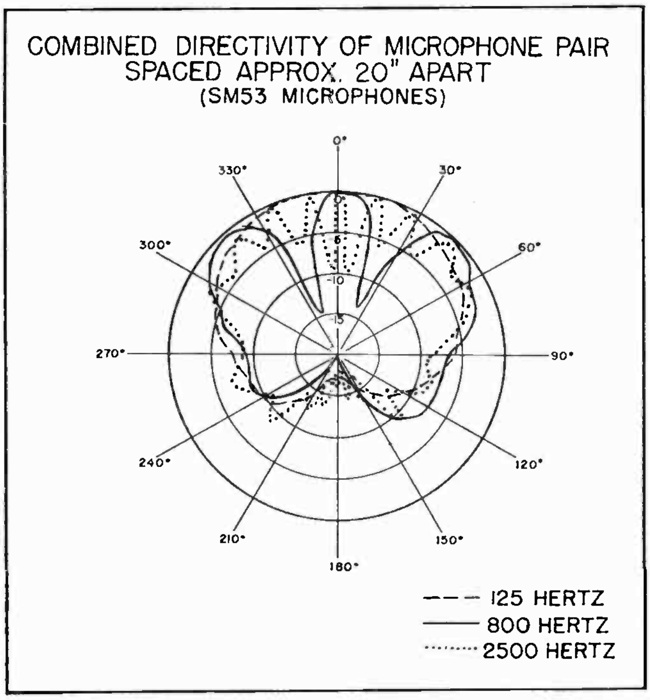


Figure 15.

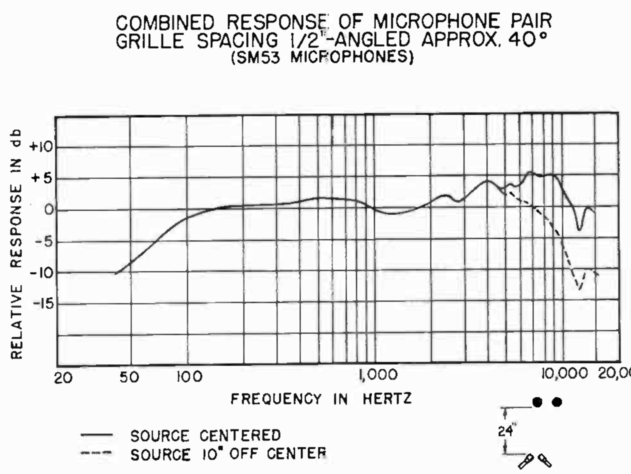


Figure 16.

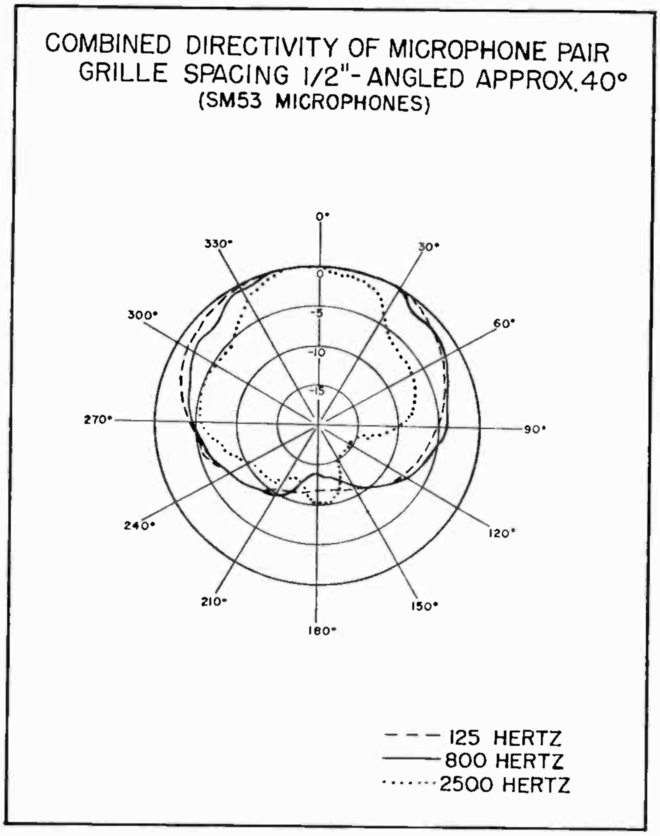


Figure 17.

Figure 18.

Factors Relating To Microphone Mobility

Microphones are used in a far greater variety of environments and in much different ways today than they were some years ago. The wide spread use for remotes, sports, and documentaries and location news exposed the microphone to a number of requirements which only a few years ago were the exception rather than the rule. Also the popularity of hand holding the microphone on the part of singers and many other types of entertainers has grown considerably and added a new series of requirements for the microphone. We will turn our attention to several of these factors now.

Mechanical Noise

With the tendency for hand holding and the use of microphones mounted in a variety of ways in a number of different environments, the problem of mechanical noise has become much more significant than it was some years ago. The continuous desire for smaller microphones has not been of much help either.

Mechanical noise is really a compound problem. We are all probably familiar with the bumping type of noise when a microphone is accidentally struck or when a stand is kicked, and have heard the effects of a vocalist or stand-up comic man-handling (person-handling?) an improperly selected hand held microphone. The noises produced are, of course, distracting and demand a proper provision.

An additional undesirable aspect of mechanical noise involves the use of compressors and limitors. In such instances the mechanical noise may not be as noticable as is the gap in program material while a compressor or limitor recovers from the thump which has been accidently produced. This is not an uncommon effect and can be even more distracting than just the impact noise alone.

Actually, there should be little reason today for living with mechanical noise problems. There are several readily available microphone types which have remarkably effective shock isolation built in . . . and these microphones (particularly omnidirectional types) need not be large, as shown by the example (SM61) in **Figure 18.** For boom or fish-pole use there are also several choices available with very effective internal isolation.

Additionally, there are external shock isolation devices manufactured to fit different types of microphones. Such a device (A53B) is shown in a boom configuration in **Figure 19** and in a conventional stand application in **Figure 20** (A53M). Such devices can be extremely effective and, as can be seen in the illustrations, need not be very large or conspicuous. A word of caution, though, is in order in using such devices. Conventional microphone cable is fairly rigid and transmits mechanical noise a bit too well. Therefore, the use of an external isolation mounting with a conventional cable may be only partially effective in preventing mechanical noise transmission. Whenever such a device is used, the microphone

Figure 19.

Figure 20.

connection should be made iwth a very flexible cable . . . preferably a limp, cloth covered type such as frequently used with boom or fish-pole connections. Such cables are offered as an accessory for external shock mounts such as these shown and are readily available.

Most of the mechanical isolation systems (both internal and external) are based on a very compliant suspension for the microphone cartridge or the entire microphone. The result, as shown in Curves (a) and (b) of **Figure 21,** is that mechanical noise pick up over most of the audio range is greatly reduced. There is, however, a very low frequency resonance between the very compliant mounting and the microphone or cartridge which can be noted in Curve B. When this secondary resonance is noticeable (by sound, meter fluctuation or compressor reaction) a high pass filter may be introduced, as shown in Curve (c). Some of the small in-line types, such as the A15HP, can be quite effective for this use.

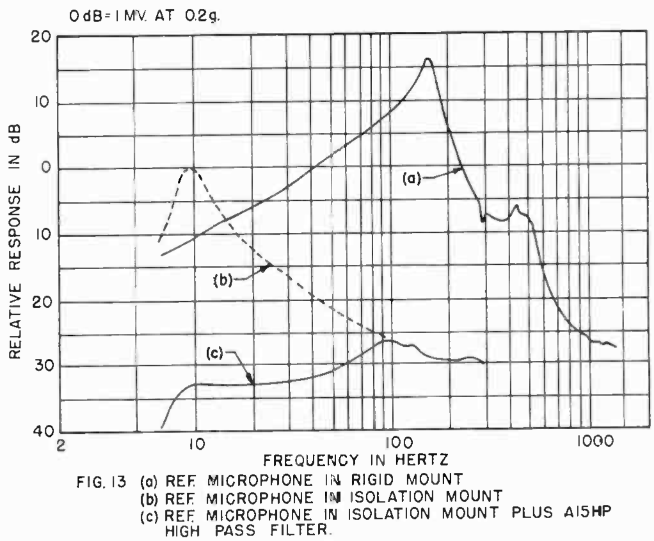


Figure 21.

Pop

This is another all too familiar effect, produced when explosive speech sounds ("p", "t", "k", or "th", for example) are produced near the microphone. The frequent result is a "blasting" or percussive sound produced by the reaction of the microphone diaphragm to the sharp pressure change.

To heighten the problem, there is a fairly consistent "worst pop condition", which happens to be the same for almost any type of microphone . . . this is with the source about three inches from the front of the microphone and at 70° to 45° off axis. If you observe how most performers use a hand held microphone, you will see that this happens to be the most common use position.

Again, though, there is really little reason for a severe pop problem today. There are a number of microphones available which have good to excellent pop protection built into the microphone grille . . . and often (particularly in the case of omnidirectionals) this internal pop filter is fairly small. The microphone already shown in **Figure 18** is a good example and **Figure 22** illustrates a small unidirectional (SM62) with very good internal pop protection. Most manufacturers also supply, for at least some of their microphones, auxiliary or slip-on pop filters which may be used to minimize this problem. Again, many of these are not awfully large to produce an adequate effect. The use of a high pass filter as mentioned in the previous section is also often helpful in minimizing the effect of pop in severe situations.

Figure 22.

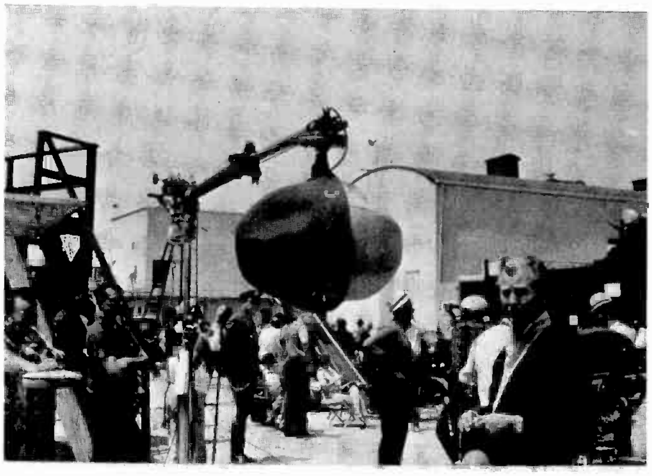


Figure 23.

Wind

This is another problem related to increased microphone and broadcast mobility. The effect of wind is really twofold: wind "rumble" occasioned by fluttering of the diaphragm due to changes in wind velocity or eddy currents around the microphone, and the "hiss" or "whoosh" component produced by noise generated in the microphone housing as the wind passes across.

For moderate wind conditions microphones are available which have fairly effective integral wind screening. For more severe conditions these microphones may be augmented with a slip-on wind filter and many will function quite well in fairly severe wind conditions. The microphone shown in **Figure 17**, for example, functions quite well in moderate winds and even in severe wind conditions with its accessory wind screen.

In extreme conditions, however, such as very strong and gusty winds or windy locations where the microphone must be some distance from the performer it must be recognized that there is no substitute for size in providing an effective wind screen. **Figure 23** indicates the kind of size which the motion picture industry has found necessary in the extreme situations which they encounter. It is no secret that producers are often horrified at seeing a sizeable wind screen on a microphone, but the alternative of the very distracting audio results which may result from inadequate wind screening must certainly be recognized.

There are, of course, some who have tried to circumvent the size question. A long standing "home remedy" has been the use of small rubber, and more recently, plastic membranes placed or stretched over the microphone grill. This approach is audio disaster! To illustrate this point to a devotee of this wind screen approach, the measurements shown in **Figure 24** were made. The unquestionably bad effect of the membrane on the microphone performance is obvious . . . both from a standpoint of frequency response and discrimination. The individual for whom these measurements were made insisted that this could not be correct and that it was necessary to slightly inflate the membrane when installing it. This was done, with the results shown in **Figure 25**. It is

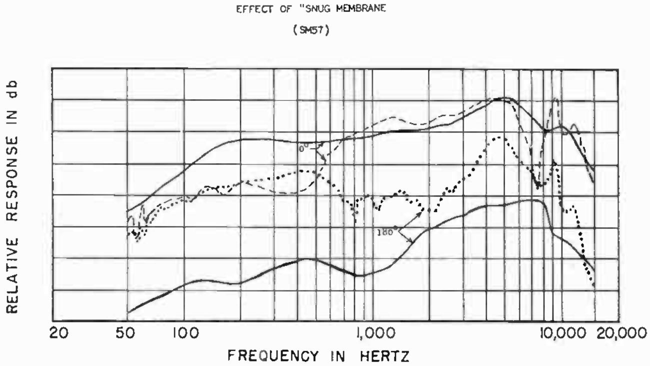


Figure 24.

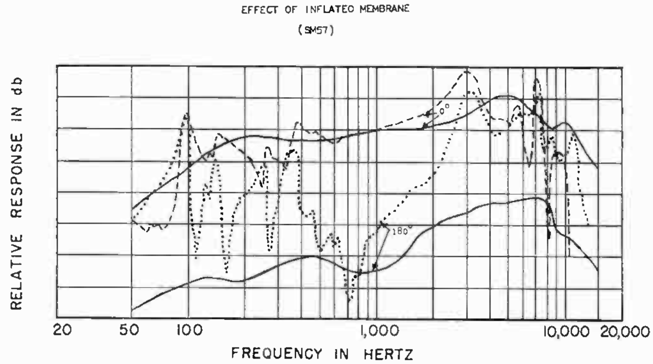


Figure 25.

obvious that some difference in wind noise would result, of course, simply because of the gross change in the overall frequency response of the microphone. This hardly, however, represents a really suitable alternative.

In the case of multi-rear entry microphones (those microphones having additional sound entries along the microphone handle, for example,) it is necessary to realize that these additional entries are indeed sound entries and that any wind screening of the microphone must also include these additional openings.

The low frequency (flutter) component of wind noise can often be further reduced by a simple high pass filter as has been mentioned in the **Mechanical Noise** and **Pop** discussions.

Long Microphone Lines

This is the last factor for consideration in this presentation and is included because it marks an area in which there is an innovation to report.

There is frequently in sports, documentary, and news programming the need for using a microphone at some distance from the remote console or truck. Often it is just not feasible to supply another remote amplifier at the end of this (perhaps) several hundred feet of additional cable and so a microphone level signal is used . . . but with the hope that exposed connectors will not provide ground loops and that interfering electrical signals will not be too bad.

With these applications in mind, a new microphone (the Shure SM82) has been developed which combines the desirable features (good response, extremely low handling noise, cardioid directional pattern, and good pop rejection) for a hand held remote microphone with a line amplifier, so that the long line may be fed at line level.

This microphone also features an internal limiter to prevent any possibility of overloading the battery (internal) powered line amplifier.

The entire unit is self contained and quite handleable (although somewhat longer than a normal microphone). It does provide an extremely convenient answer to the problem of the long remote microphone line.

The line output also makes it possible to directly feed a telco line . . . preferably, of course, through a Voice Coupler (Type 30B, for example) although the microphone's output configuration is suitable for direct connection in an emergency.

The SM82, its block diagram and the limiter characteristics are shown in **Figures 26, 27, and 28** . . . and this microphone is now available.

This presentation is certainly not exhaustive . . . either of the factors which have been mentioned or of the many more which could be. It is hoped, however, that what has been presented does provide a little additional insight into some of the considerations in the use of microphones . . . and perhaps may suggest the solution to some audio problem for you.

Figure 26.

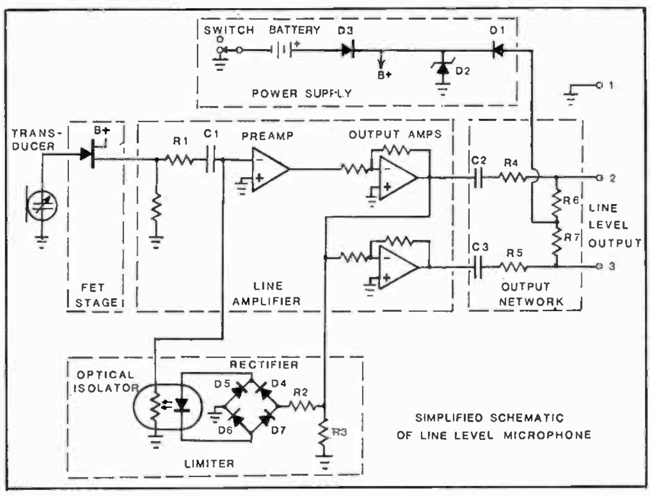
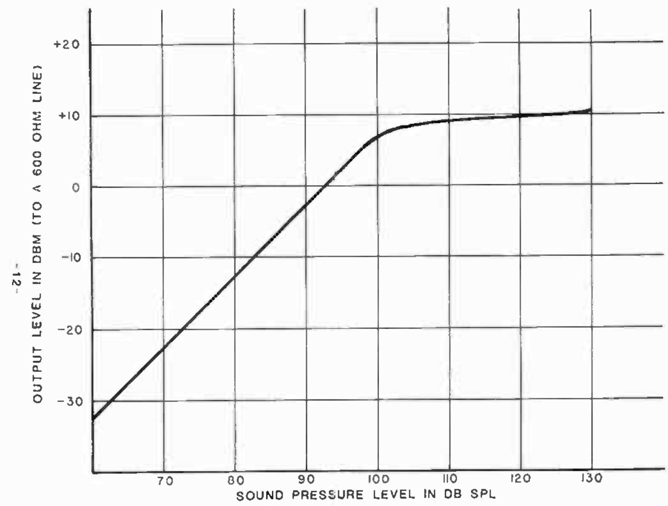


Figure 27.



INPUT-OUTPUT LIMITING CHARACTERISTIC OF LINE LEVEL MICROPHONE.

Figure 28.

B. VanBenthem
(and R. H. McMann, C. Smith)

CBS Laboratories
Stamford, Conn.

A Practical Method of Automatic Color Balance Correction

Design Philosophy

In the middle of 1970 CLD introduced a manual encoded signal color corrector for NTSC or PAL signals and since that time our Laboratory has been investigating to find a practical way in which to operate the manual corrector in a semi-automatic or better yet automatic mode. As you probably know by this time the manual color corrector allowed for color modifications or correction of an incoming encoded signal by adding a correction vector to the program signal rather than demodulating—correcting—remodulating the total signal.

Although the units proved very successful in the field, they still lacked automatic operation! For instance, during panic periods such as insertions into news shows with several short commercials back-to-back, it became apparent that the manual corrector could not be used for painting, i.e., commercial-to-commercial matching. The available time was too short and the operator needed too much time in order to know which one of the six (6) knobs RGB black level and gain was needed for correction. If the material to be corrected had never been seen before, correction was a very difficult and a nerve-racking task. Therefore, it became apparent to CLD that if a unit could do color balancing of such a signal, even semi-automatically, with just one push-button instead of six (6) controls, this addition to the arsenal of signal improvers would be most welcomed by the operators.

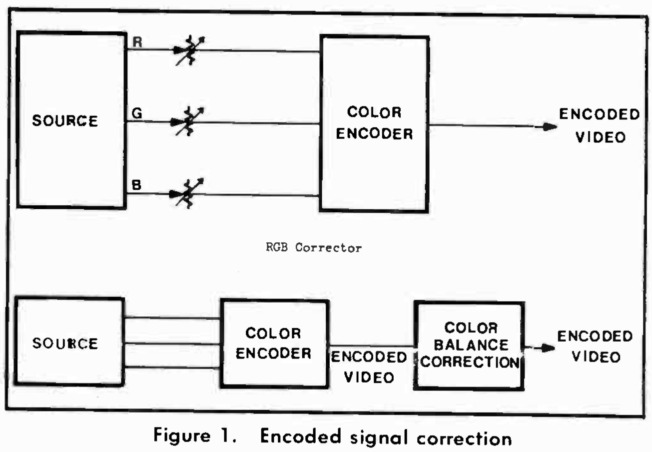


Figure 1. Encoded signal correction

We started to look into the possibility of developing an automatic or semi-automatic color balance corrector. But, could electronic circuitry replace the human operator? What was the operator doing? He was looking at picture highlights and blacks and examining them for any residual coloration. Often, he would do this with the chroma increased on the monitor. This indicates that the human operator needs increased loop gain to operate effectively. It is the decision-making process of the operator which had to be replaced with electrical components. As input, an NTSC encoded color signal is used. In this example, we will use NTSC, although it should be understood

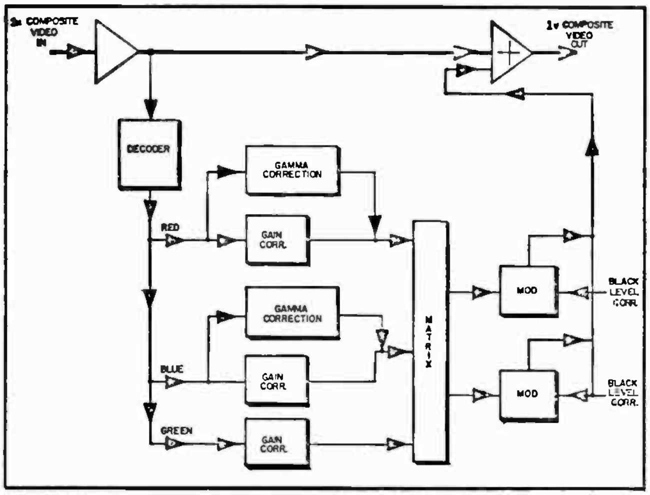


Figure 2. Simplified Block Diagram of Model 5500A NTSC Color Corrector

that the same arguments hold for PAL encoder signals. A properly encoded signal has the following characteristics that can be considered as the basis for an automatic corrector.

1) A true black signal does not have any luminance level other than set-up and no chrominance components.
2) A true white signal has a maximum luminance level and no chrominance component.

With these parameters in mind, we will examine a color signal for maximum and minimum luminance levels, while at the same, we perform a check for chrominance at these white and black levels, and if a chrominance component is present we can establish its relative phase and amplitude and using these parameters, we thus can generate a chrominance signal, that, if summed with the original signal, could eliminate the chrominance component. As a result, we could color balance the incoming encoded signal. Now let us look closer at this.

First, by definition, a white signal has a 100-IRE units luminance amplitude and a black signal has 5,7-½ or 10 IRE units of luminance depending on utilized set-up standards. Throughout this discussion, we will use 7-½ IRE units for the set-up level. An experienced operator, however, will tell you that

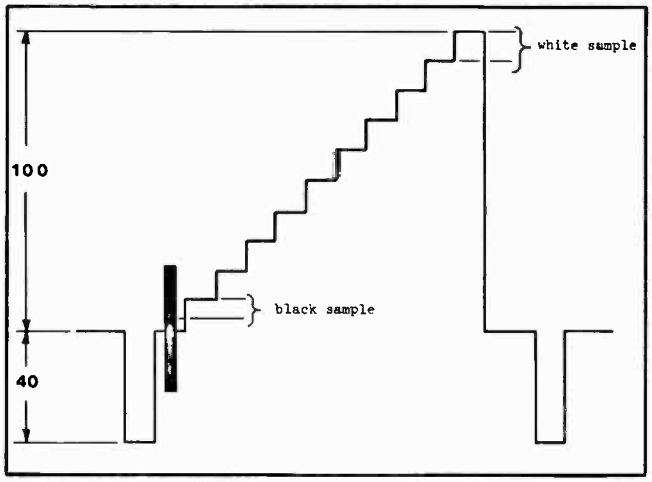


Figure 3. Luminance sample areas

these levels are not always present and that the blacks may have a greater or lower luminance level due to set-up standards or set-up level variations due to improper settings of the signal origination equipment. Similarly, the whites in a signal are not always exactly at the levels they should be. Improper operation of auto-white in a film chain or improper gain settings are just a few of many causes for this. Clearly, it became apparent that under practical conditions we cannot just look at the 100 IRE units of luminance or at 7-½ IRE units of luminance for detection of whites and blacks in a signal. Experiments have shown that we can get good results by looking at the top six or seven IRE units for establishing whites and the lower 5 IRE units for blacks. At the same time we are looking at the chrominance levels in these black and white areas.

In practice, if whites or blacks are present a sample pulse is generated which in turn opens an electronic gate which then routes the chroma error signal to an integrator circuit. This circuit then develops a dc potential which is analog to the amount of the error. Thus we are taking a calculated risk that a very high luminance level plus a small chrominance component rarely appear in a properly encoded NTSC signal. After extensive field studies, we found that this is a valid assumption. The signals which were most troublesome in our earlier experiments were on the extremes of the B-Y axis and either deep dark blue or high pastel yellow. It is easy to see that both of these signals meet our conditions of having very low or very high luminance level plus chrominance.

In order to cope with these signals, we designed a circuit which detected the extremes at either side of the R-Y and B-Y axis. In this manner, we developed a limit logic for these extreme colors. We thus instruct the corrector to ignore certain colors in the high or low luminance area. In the extreme, of course, we would ignore all colors and all we would have left is a very expensive distribution amplifier! An important reason why in practice we can operate successfully with colored objects at white or black is that we have four feedback sampling loops each containing an integrator that stores all of the samples taken over the whole frame and acts on the average to determine the proper correction.

Before we go further with this discussion, we will take a look at a far more serious problem. We cannot assume that there will always be white in the picture, or that the white will not rapidly come and go. If the incoming signal levels are not properly maintained, the unit may never detect white. If the level is too low there is not much you can do about it because this might have been the intention all along and if you increase the gain of the signal so that the unit has an average white, it could make a major mistake! If the white level is too high, we can bring it down by use of an AGC amplifier, however, it was felt that the color balance corrector should not affect throughput program levels so we decided to change the sampling gate level in proportion to the white level, that is to "AGC" the gate, which allows us to track whites above maximum leve. This floating gate will follow the video if the levels are too high and as much will guarantee that we are sampling in the right part of the signal. If the level falls below the minimum video level to be sampled (93 IRE units), we can no longer obtain a white sample and all further white color balance correction stops. We now have one of several choices:

1) We can maintain last correction until new sample comes along.
2) We can return to zero correction.
3) We can wait for several seconds, then return to zero slowly.

Changes in correction due to the loss of white sampling or acquisition of sampling should not result in immediate correction since fast responding corrections in these cases lead to color flashing which is very disturbing even to the non-technical viewer. Holding correction until a new sample comes along, regardless of how long, can lead to serious errors. For example, during the last few frames of a program, a film, for instance, may require correction. When we switch to another film which has no whites, we maintain the previously established correction and as you can easily see, we would be in serious trouble. Therefore, a mode was selected in which an error correction is made and maintained for a certain length of time after which a slow return to zero will follow.

Recent field test indicated that an additional mode of holding correction until next correction is called for, should be made available. But as discussed earlier this method of operation should not be done in a fully automatic mode.

A single push-button control allows automatic correction sampling only when the operator engages it. We are now back to a mode of operation in which an operator is in control and he can judge if a correction is needed and make that correction by simply pushing a button. It is certainly true that it is far easier to push one button than to work with six controls to obtain

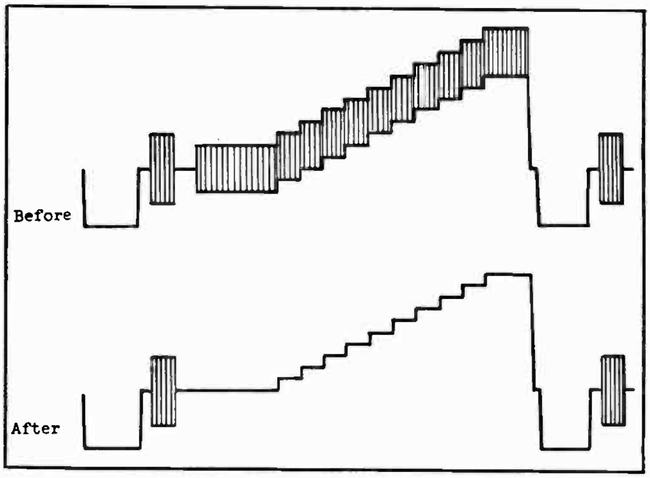


Figure 4. Effect of balance correction

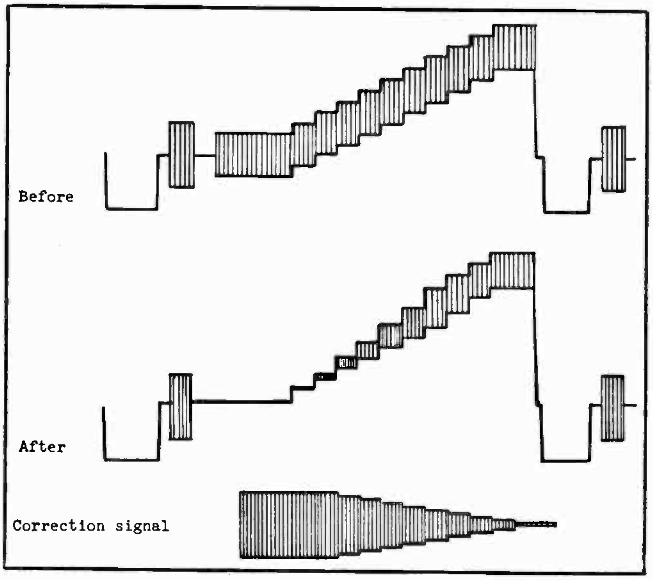


Figure 5. Balance correction with black only sampling.

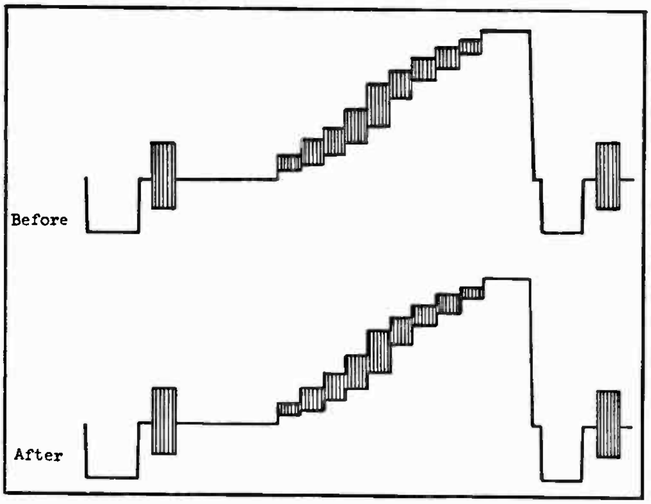


Figure 6. Auto balance correction does not effect a gamma corrected signal

proper balance. Also, if a constant error is present, which quite often occurs in films, only single correction is needed at the beginning, and the unit will obtain proper correction until told otherwise. Thus the unit we have described has basically three modes of operation:

a) The unit can be used as a fully manual color balance corrector.
b) The unit can be used as a semi-automatic color balance corrector by utilizing a single sample-and-hold button.
c) The unit can be used as a fully automatic color balance corrector, that is the unit will continuously look on its own for balance errors and make corrections as necessary.

In the fully automatic mode, there are two operating logic modes available, return-to-zero when not sampling and non-return-to-zero. The choice depends on whether there is expected to be a constantly changing error or a constant error as encountered in film stock. Let us now take a look at the application of this unit. Since the color balance corrector is basically performing a quality control function, its logical location in a video system is where a quality control function is normally performed:

1) At the CCU position of a camera chain.
2) At the output of a tape system.
3) At the output of a studio system.
4) At the output of Master Control.

If the unit is utilized at the control position of a film chain, it allows for instant painting without disturbing the basic set-up of the film chain. Since the unit can be preset to take care of severe errors (in manual mode) with the flick of a switch, we can apply immediate correction where needed. If the unit is operated in the automatic or semi-automatic mode, no preset is needed. The greatest affect is seen in the black region where "muddy" blacks and colored shadows are reduced or eliminated.

Although the color balance corrector could be used anywhere in the video system, experience proved it to be most successful in correcting the output of Telecine chains. This is due to the great variations in film material. Live cameras produce far fewer color errors in practice since they generally have an operator already making color correction decisions at the CCU.

The color corrector should not be used if the program material consist of more than one video source such as special effects, chroma key, etc.

Since the system averages the color errors for several fields in order to avoid "color flashing" the above mentioned signals can consist of a "good" signal and an unbalanced signal. I guess I don't have to mention the unpleasant result of this.

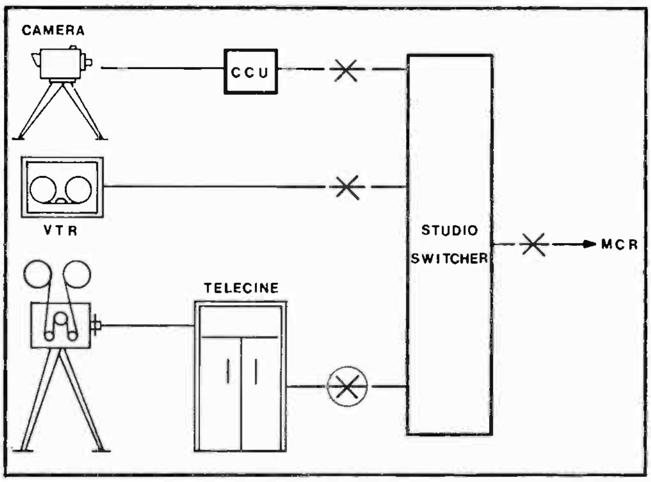


Figure 7. OPTIMOD 8000A—Simplified Limiter Block Diagram

Description of A Practical Version of the Auto Color Correction System

Figure 8 shows a block diagram of the total system. The left side of this diagram shows the block functions of the manual color corrector, while the other side shows the sensor. As can be seen program video enters the unit, then passes through a multiple output amplifier and a correction summing amplifier. The first amplifier is used to add the correction vectors. The total system is also protected with a copper to copper bypass mode. The correction signal is derived from decoding—encoding process. After decoding we are able to operate on the RBG components of the signal and modify RBG. Black level—RBG gain—R—B gamma and overall chroma gain. After remodulation we develop an error signal consisting of subcarrier with its relative amplitude and phase so encoded that when vector summed with the original program video, produces the required color corrections.

The correction signals are blanked out during H and V blanking in order to avoid erroneous addition of chroma in these areas.

Let us now look at the other side of the diagram, the sensor. As discussed earlier, this part of the system has been designed to automatically or semiautomatically operate the color balance functions of the manual corrector, i.e. the gain and black level controls. As said earlier, the sensor searches for black and white regions in the signal and then looks for chrominance components within these regions.

The sensor unit has several "inputs" which come from the manual corrector:

1) Luminance input
2) R – Y input
3) B – Y input
4) Corrected video input
5) Sync input

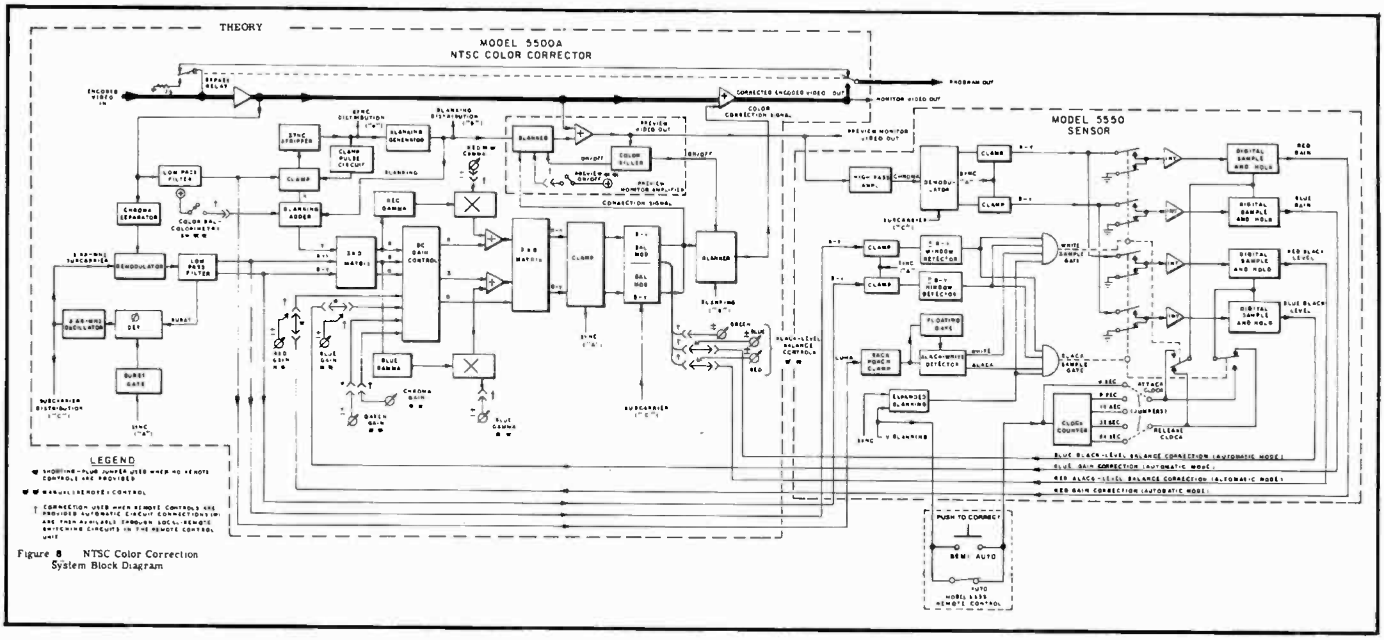


Figure 8. OPTIMOD 8000A—Simplified Stereo Generator Block Diagram

The luminance input is utilized to establish black and white sampling gates. This circuit has internal references for these parameters. If the luminance component reaches a level below 11 IRE units a black gate pulse is generated and above 93 IRE a white gate pulse is generated. If the signal goes over 100 IRE units a floating gate reference circuit comes into action. This circuit guarantees that above 100 IRE units of luminance we only derive a sample gate pulse during the top 7% of that signal, thus avoiding program AGCing.

The R − Y and B − Y signals are utilized to develop inhibit pulses at the extremes of these areas. These circuits make it possible to handle low luminance, high chrominance signals such as deep blues or high luminance, high chrominance signals such as bright yellows. These circuits can be adjusted to special programming needs if required.

The corrected signal is passed through a high pass filter and demodulator. The resultant R − Y and B − Y signals are sampled by the black and white sampling gates for their chrominance contents. If during these gating pulses a chrominance component is present in either the R − Y or B − Y signals this signal is then fed into an integrator circuit which has an integrating time constant of approximately ten fields. The output of the integrator produces a dc level analog to the measured error. The following digital circuit consist of a comparator, a counter, and D-A converter, i.e. this circuit counts up or down depending on the input dc level until its output dc level is equivalent to the input. If no error is present, the error voltage is 0 and the counter will return to 0. The correction "on" and correction "off" times can be individually selected as we utilize a separate attack and release clock.

As can be seen in this block diagram the clock counter is triggered from V blanking. In this line we have included a number of contacts. In one mode the "auto" mode we always correct, in the second mode "semi-auto" we only change correction when the push-button is activated. It should be noted, however, that in either mode the sensor keeps sampling for error.

The output of the digital sample and hold circuit is routed to the proper input of the manual color corrector which then will result in the required correction. The corrected video is continuously fed into the sensor which completes the correction loop.

Sync routed into the sensor is also used to generate extended blanking. The extended blanking is timed such that we stay away from the picture edges in our error sampling process and thus avoiding improper correction due to possible imperfections in the areas such as yoke ringing, cornering, etc.

At all times the corrector can be turned off (inhibit summation) or the unit can be returned to a manual mode if accurate matching of parameters, other than black levels and gains is required.

Al Martin

Director of Engineering
Kaiser Broadcasting Co.
Southfield, Michigan

The Need for a More Efficient UHF Transmitter

Kaiser Broadcasting Corporation operates seven UHF stations in the top 10 TV markets in the country. Our 1974 transmitter power bills exceded half a million dollars. In 1975, it is projected they will be $621,000, which is an increase of 24 percent over last year. In light of the current on-going concern over the country's energy shortage and the ever present pressure on all businesses to reduce operating costs, we at Kaiser Broadcasting are becoming increasingly concerned over the poor efficiency inherent in the state of the art UHF transmitters. It is essential that UHF transmitters develop very high output power to reach the viewers with an adequate signal, and to compete with well established VHF competition. In line with that, Kaiser Broadcasting recently completed a project of installing 120 KW transmitters in Boston, Philadelphia, Detroit, Cleveland and San Francisco. Our Chicago and Los Angeles stations are using 50 and 55 kilowatt transmitters. The klystron tube has been the answer to this very high power output requirement. However, coupled with the high power output of the klystron comes the low efficiency which brings us to the subject of this paper.

Kaiser Broadcasting is presently experiencing between 25 and 30 percent efficiency with our transmitter systems, and approximately 32 percent klystron efficiency.

Varian Associates, the manufacturer of UHF television klystrons used in most of our transmitters has recently introduced an improved version of the klystron which has been dubbed the "high efficiency" tube. Several of our stations have one or more of these tubes and we are experiencing approximately a 5 to 6 percent improvement in efficiency. Specifically we can achieve the same output per tube with a reduction of DC input power of about 6KW to reach a 30 KW peak output with the new tubes. This improvement is slight and certainly appreciated but to be painfully blunt we need efficiency figures much higher than that. A little arithmetic shows that the following increases in tube efficiency will result in a power saving of:

Tube Efficiency	*Power Savings Over Current Usage*
37—43%	15%
43—50%	26%
60%	37%

Several manufactures are actively pursuing various higher efficiency projects at this time. RCA and Varian are both deeply committed to UHF manufacturing and a recent idea between the two companies has resulted in a concept which has been developed that sounds encouraging. The idea is to pulse modulate the klystron. RCA is actively investigating this approach. The general concept is as follows.

The present design of UHF transmitters applies a steady high voltage potential to the modulating anode resulting in a continuous klystron input power of up to 90KW DC power for a 30KW klystron. Please notice that I said steady. However, this amount of input power is actually required for only 17 percent of the duty cycle. This is the time during which horizontal blanking, including sync and color burst are present. During the remaining 83 percent of the duty cycle, which is active video, it is computed that 60KW of input power would be adequate to properly amplify the TV signal. Theory then would say we would "pulse modulate" the klystron with a "pulse" of 90KW input power during horizontal blanking only and reduce DC power input to approximately 60KW during active video. That overly simplifies the concept because it's not that easy in practice, but RCA believes it can be done. 90KW for 17 percent of the time, averaged with 60KW for the remaining 83 percent, averages 65KW. That's a reduction of 26% of input power. That now approaches 50% tube efficiency. Converting this to dollars would mean a reduction in power bills of $161,400 dollars a year in our group alone.

The cost of this pulser may make it impractical for 30KW transmitters, at least for a short term pay back. But it appears that power savings on larger transmitters (50KW and up) would permit acceptable pay backs in power savings realized.

In our discussions with RCA it appears that the anode pulsing concept is a promising one, and barring unforseen problems this feature may be available to the broadcasters by approximately mid-1976.

A further development that should interest everyone involved in this problem is that International Telephone & Telegraph (ITT) recently purchased RCA's klystron manufacturing system and also some tube manufacturing and design licensing from the Sperry Co. I understand that they now plan to manufacture television klystrons relatively soon and they feel that tube efficiencies up to 50% are feasible with present mechanical parameters. Efficiencies higher than that, in the neighborhood of 60% could probably be obtained but would require additional changes in existing transmitter equipment and would probably result in a gain reduction in the order of 10db. If these efficiency figures can be reached, UHF broadcasters could very well welcome ITT to the market place with open arms. Their research should be strongly encouraged by broadcasters in everyway possible.

At the close of 1974 there were 344 UHF stations operating with another 93 pending. Arbitrarily assuming that all those stations were operating a 60KW transmitter at 30% efficiency for 16 hours a day, and if we could reduce their power consumption by 25% it is estimated we could save 342,000 KWH per day. It is further estimated that the average residence uses 625 KWH a month. That means that we could furnish electric power to a town of 16,000 homes every day that we saved that 25%. Granted, this is a play on numbers but I believe it shows in a rough way that we are talking about a very sizable power saving with an efficiency increase that seems to be generally accepted in the industry as entirely within reason and available technology.

From a practical standpoint, we realize that appropriating R&D budgets for such a project as increasing efficiency in UHF transmitters is a problem in and of itself. A company such as Varian spends a minimum of $100,000 to develop a more efficient tube. Varian has accountants as we all do, and they must be judicious in allocation of such funds. Therefore, to further pursue this project we need a conserted effort between UHF broadcasters, broadcast organizations and ETV local and national groups to ban together in an industry-wide approach to prod and encourage manufacturers to achieve a goal of transmitters that are at least as efficient as VHF transmitters. I think that should be our minimum goal.

Our country needs this kind of energy conservation and UHF broadcasting certainly needs to reduce its operating costs. Those are two of the best reasons I know to continue the pursuit of better efficiency in UHF television transmitters.

I would like to take this opportunity to charge the manufacturers represented here today to continue to pursue this problem. Contrary to what may appear on the surface, we are far from content with present day UHF transmitter efficiency and $100,000 a year power bills at our station.

In our opinion, this problem should be at the top of the agenda on industry committees. We at Kaiser Broadcasting will continue to pursue this project.

This paper has been about transmitter efficiency. However, there is another portion of the complete system which needs considerable improvement, and that is the UHF tuner in the TV receiver. The signal to noise ratio and sensitivity of UHF tuners can be improved. We have the technology. We now need pressure brought to bear on receiver manufactures to incorporate that technology into their tuners. Increased s/n ratio and sensitivity in tuners would allow UHF Broadcasters to reduce their ERP's and not degrade the quality of the picture at home. A reduction in ERP would result in additional savings of power and costs to the broadcaster and our country.

I would be unfair if I did not mention that many people, organizations, and companies have put a lot of time, effort and money into this project already and have also greatly assisted me with information. I had hoped to have a panel of these people but time would not permit. My thanks to Terry Hickman and Hans Bott at Harris, Gates Div., Bob Schmidt at Varian Assoc., John Romaine at ITT, Dan Wells and John Wilner of PBS, and last, but not least, Ray Desrochers at RCA.

Moderator:

Albert H. Chismark

Director of Engineering
Broadcast Div./Meredith Corp.
Syracuse, N.Y.

Electronic News-Gathering Workshop

Participants:

Robert Mausler, *Senior Engineer*
NBC Television Network, New York, N.Y.

Julius Barnathan, *Vice President, Broadcast Operations and Engineering, American Broadcasting Co., New York, N.Y.*

Thomas M. Battista, *Vice President and General Manager*
Television Station KMOX, St. Louis, Missouri (representing CBS)

Ralph Hucaby, *Vice President for Engineering*
Television Station WLAC, Nashville, Tenn.

A. H. Lind, *Manager, Studio Equipment Engineering*
RCA Corporation, Camden, N.J.

Plus, Questions and Answers

Robert Mausler

Senior Engineer
NBC Television Network
New York, N.Y.

It can be said without equivocation, that electronic journalism is the most powerful, dynamic force now in television broadcasting.

When we think of "News", in reportorial terms, I'm sure that the impact of "On the Spot Coverage", in reporting "News", has not been lost on the "News" profession. Historically, in one of the earliest "on the spot" systems, reporters were ensconced in a watch-tower.

From the beginning of tower to tower and message runner to relay man communication, not forgetting the medieval jongleur of news in verse and song; to the Graham McNamees, Bill Sterns and Ed Herlihys—with microphone in hand, delivering eye witness reportage to its audience, via Radio, (and can we ever forget Orson Wells's description of the Martian invasion?), to John Cameron Swaze and the Film Camel News Caravan, each night, we now arrive at Direct to Air coverage of the event, as it happens.

What we might call, people to people communication.

Now to consider how to deliver this "News"—with a spontaneity, timeliness and live quality, which is the sine qua non of Electronic Journalism. It is convenient to note that an electronic journalism delivery system is not dependent on:

A. The purchase and use of film stock
B. The purchase and use of film processing equipment
C. The purchase and use of film laboratory services

Accordingly, some ancillary consequences of reducing or eliminating dependence on film naturally follow, such as:

A. The elimination of film processing delay which means late "News" can be aired.
B. Reduced manpower requirements which translate to significant dollar savings a year.
C. The economy of reusable tape vs film's one time use; which means real savings in addition to elimination of the waste on the cutting room floor.

The key development in the evolution of electronic journalism is the availability of modern reliable, high quality portable color cameras. At NBC we are presently using the Phillips PCP90, the Ikegami model HL-33, and the latest Fernseh model KCN. Some of the important operating parameters for these cameras, are as follows:

A. The head without lens varies from 15½ to 18 lbs.
B. Control unit back packs vary from 18 to 32 lbs.
C. The standard lens is a 10 × 15 f2.0 Cannon.
D. A battery pack provides 2 hours operation (without r.f. equipment).
E. Light requirements vary between 75 foot-candles (Fernseh) to 300 footcandles for the ikegami.

The corollary development, and perhaps the most important one, which has given impetus to electronic journalism is of course, the portable VTR. It is this development of low cost VTR equipment which is the essential factor in replacing film.

We use the Sony VO-3800 with its ¾'' cassette format, configured with SMPTE Time Code and Camera Trigger operation. The unit uses a 20 minute cassette and can operate in excess of one hour on each battery pack. The combined weight of VTR and battery pack is approximately 31 lbs.

With the provision of a Time Base Corrector (CVS504) and microwave equipment, the system is capable of Direct-to-Air playback. We may note that our latest microwave equipment, the Microwave Assoc. MA-13CP (0.1 Watt) is portable with up to two hours battery life, and operates at 13 GH_z—suitable for short hop operation. In New York, for relay to the Empire State Building, both 2 GH_z and 7 GH_z are presently used.

Transportable "news"s field equipment packages also include Sony VO-2850 Record/Playback/Edit VTR's, therefore, used with a Time Base Corrector and with signal transmission to an Earth Station, fully edited "news" segments may be aired via Satellite. In fact, this mode of electronic journalism was demonstrated by network coverage of both the Japan and Martinique trips of President Ford.

Figure 1

Figure 2

For inplant editing, in New York, we presently operate two Electronic Journalism rooms complete with Sony MX-14 audio mixers and with an Announce Booth available to each facility.

A pair of Sony VO-2850 units, with a Sony RM400 Edit controller are the basic tape editing equipment. The controller and VTR units are modified for Pre-vue and Videola operation, both NBC developments.

Figure 3

The pre-vue modification allows the operator to select his edit point and to pre-vue the result without committing to the edit unless it is satisfactory. This feature can save a considerable amount of editing time.

The Videola modification permits a tremendous speedup in the editing process, allowing the editor to use Forward or Reverse at twice normal speed, with accompanying sound and picture; providing editing capability similar to that widely practiced in film editing with a Movieola.

The final edited copy is played back to air using a third VO-2850, a Time Base Corrector and Image Enhancement (CBS 8010). Thus editing may continue while the first tape plays back to air.

Datametric SMPTE Time Code Readers (SP425R) are available at each VTR position. The rooms also have a cassette receiving and storage facility with provision for a constant recycling of the cassettes. In addition to the daily editing a weekly summary recording based on all On-Air material is also packaged.

The cassettes now in use are the Sony KC20S in the field, and either KAC60 or the 3M UCA10 and 30 minute units for editing.

I will now show NBC's Electronic Journalism story. The story began in 1939. (**Figure 1**) shows history in the making—an early electronic journalism. It's a pickup of former RCA Board Chairman, Gen. David Sarnoff, on April 20, 1939, at the New York World's Fair. This was the world's first "news" event coverage by television. An Iconoscope camera is shown.

Figure 2 shows an early mobile relay van used at our Burbank facility, circa 1955. The antenna mast extension system for support of the microwave dish is noteworthy.

Figure 3 depicts portable electronic journalism of recent vintage—a Phillips PCP90 portable color camera, an Ampex VR3000 portable quad VTR and an RF link. This combination has been noted for producing fine quality pictures.

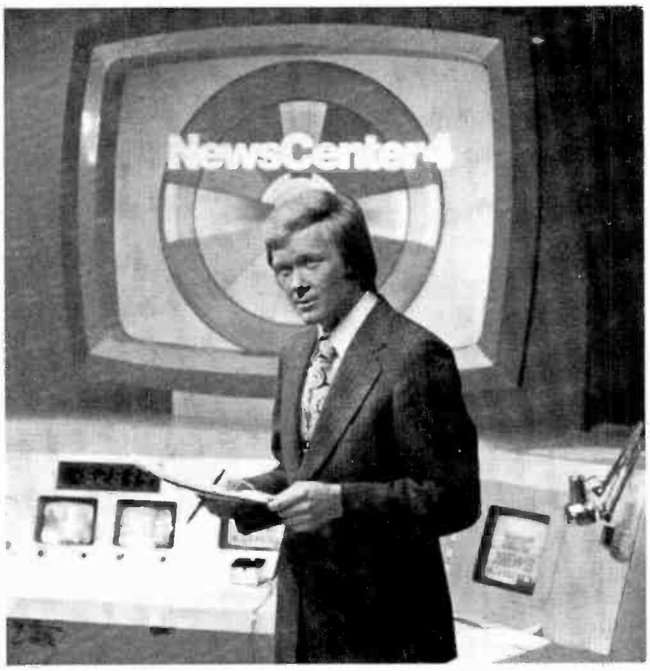


Figure 4

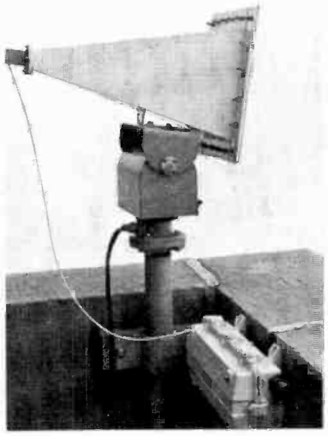


Figure 5

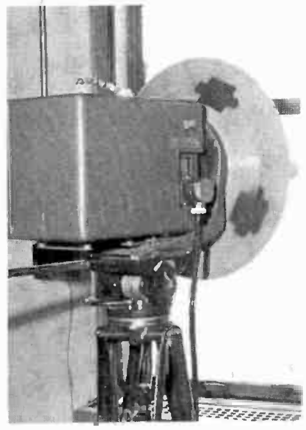


Figure 6

Figure 7

Figure 8

This story continues, now focused on electronic journalism activity scheduled on February 13, 1975. Two crews, operating in lower Manhattan and remote Brooklyn, conducted interview segments with two-way interaction between the locations and a panel of experts within the "News Center 4 Show", that day, "live". One application of the electronic journalism concept.

News Center 4—WNBC-TV's premier two-hour "news" show with host Chuck Scarborough is shown in **Figure 4**. The electronic journalism story was integrated live, on a two-way basis, within the show of February 13, this year.

Figure 5 shows one of the four horns on the 86th floor of the Empire State Building—target of the microwave link to each remote location. Each horn has a 150MHz converter and remote panning mechanism. (Azimuth and Zenith). The horns have a one foot aperture, 25″ length and a gain of 25db, assuring pickups from up to 45 miles distance. They operate at 7GHz.

Figure 6 is a through the window shot at the Empire Transmitter, a 2GHz dish. It's a Microwave Associates MA-2B unit.

Figure 7 shows a Microwave Associates MA-7B operating at 7GHz atop a supermarket roof in lower Manhattan. **Figure 8** is of the same dish showing an ideal line of sight shot to the Empire State Building.

Inside the supermarket (**Figure 9**) an Ikegami HL33 and backpack is used in covering an interview. The close up (**Figure 10**) shows the support hardware and Canon lens.

Figure 11 depicts the mobile "News" van at a Broadcasting location. The 7GHz microwave (with a 20 Watt booster) bounces its radiation off the apartment house for pickup at the Empire State Building. One of the van's crew is shown in **Figure 12** with a Sony VO-3800 portable VTR (not in use for this direct pickup). Within the van note the Time Base Corrector, CVS504, and of course, the waveform monitor and other video equipment.

Figure 13 shows the storage inside the van of the Sony AC Adaptor, equipment racks, and another housing for dish control and microwave equipment. What space is left is for storage.

Inside the supermarket (**Figure 14**) the Fernseh KCN camera, on a Tripod, records an interview conducted by NBC's reporter, Robert Potts. The Fernseh chain with its Backpack/Camera control unit at the tripod base is shown in **Figure 15**.

Figure 16 shows one of the electronic journalism edit rooms at Radio City. Note the Master and Slave Sony VO-2850 Cassette VTR's and the Playback VO-2850. The rack contains Sync Generator, Time Base Corrector, Video Distribution Amplifiers, Audio Monitoring and Communication facilities. The announce booth window is at the left.

Figure 9

Figure 10

Figure 11

Figure 12

Figure 13

Figure 14

Figure 15

Figure 16

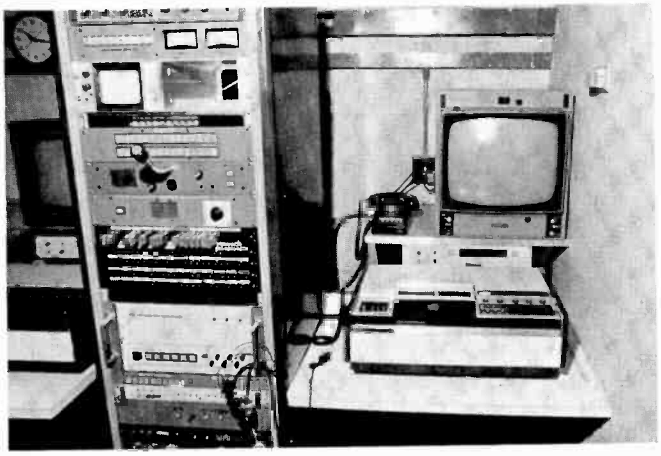


Figure 17

Electronic journalism room number 2 (**Figure 17**) gives a view of the playback position. The VO-2850 and a Time Code Reader is shown. Note that playback to air does not interfere with the editing position. It is from here that the edited cassette PGM is played to air.

Figure 18 shows the editing position with the Sony audio mixer and editing control panel, and the Datametrics Time Code Reader. The edit controller has been modified with the NBC Pre-vue and rapid viewing features (Videola).

Figure 18

Julius Barnathan,
Vice President

Broadcast Operations and Engineering
American Broadcasting Company
New York, N.Y.

I'm going to take a different approach to explain what we're doing at ABC. I'm not here just to talk about ABC, but I want you all to learn from the mistakes we made.

First, electronic journalism has been here for many years. We've been through this at the national conventions and everywhere else. Let's not say it just started. What we are talking about now is something that we call mobile electronic journalism. Does everybody know what the word mobile means? It means you can carry it. We're talking about mobility.

We have an excellent film camera called the CP-16, or its other counterparts, that does a first class job. That's taken 20 years to get to where it is. It's a good piece of equipment and it'll be around a long time. But there's one thing that film doesn't do that electronics does, and that's basically "go live".

Going any further than that, depends upon your own setup in terms of what are you going to do from here on in—namely, "is it better to go electronic than to go film?" That argument will not be decided today. I think you'll hear some cases for and against. But the argument for that is not what we air now.

Let's define what we're trying to do. Compare the 35 millimeter film discussions during which we went through the same thing. The 35mm cameras were too heavy to walk around with, so we decided on 16mm. The quality of 16 is not 35, but it's acceptable, very acceptable and it has improved over the years.

We're talking about the same thing now in terms of quadraplex vs. helical scan.

What we have to do now is to compare electronic news to 16mm film, and that's what we've got to get down to. Can it be competitive to 16 millimeter film, give us the fact that we can get on the air late and also give us the flexibility of going live?

At ABC we sat back and waited. We did not go through the expense that NBC went through—PCP-90, the VR-3000, which together cost $170,000, and then to other cameras. We bought two Akai cameras for $4,000.00, installed them in Washington. Unfortunately, there were two other networks there and they were running around with PC-90's, so my news department was very unhappy with me.

After a year and a half of waiting, they finally convinced me to go ahead and buy big cameras. And we did. We bought the cameras, but at about the same time we were working on tape. We knew CBS was working on a camera, so we figured we'd concentrate on tape, and that's really where we spent our time. Developing and improving time-base correctors. Coming up with the 504 and working on the Ampex time-base corrector.

When we got started we went over and saw what CBS had done, and like good Japanese, we bought the same camera. And then the tape machine.

I would like to show you very quickly, the equipment that we just talked about.

CAMERA	TAPE RECORD
1 Tube	¼ Inch
2 Tube	½ Inch
3 Tube	¾ Inch
Vidicon	1 Inch
Chalnikon	2 Inch
Plumbicon	

Figure 1

All right, here's the confusion factor. **Figure 1** shows one, two, three tubes. What do you like? We've got something for everybody, and this is all part of the confusion that has existed for the last few years, and it's going to continue to exist.

There are two forms of electronic news-gathering systems (**Figure 2**). One is on tape. A little truck with a little tape recorder. That's the 16 mm film concept. The other case, live from a remote location.

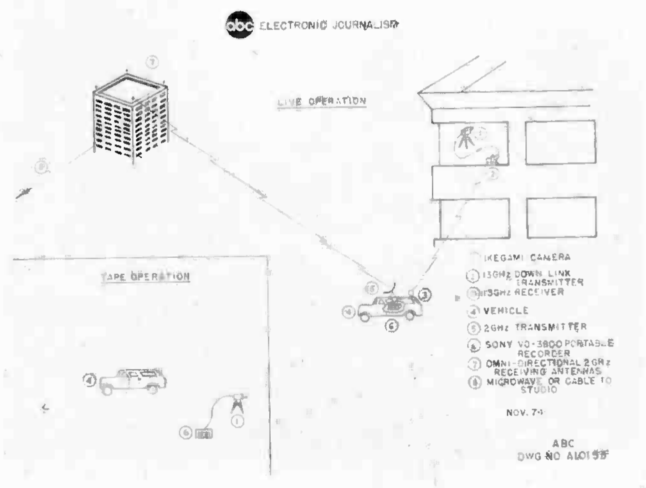


Figure 2

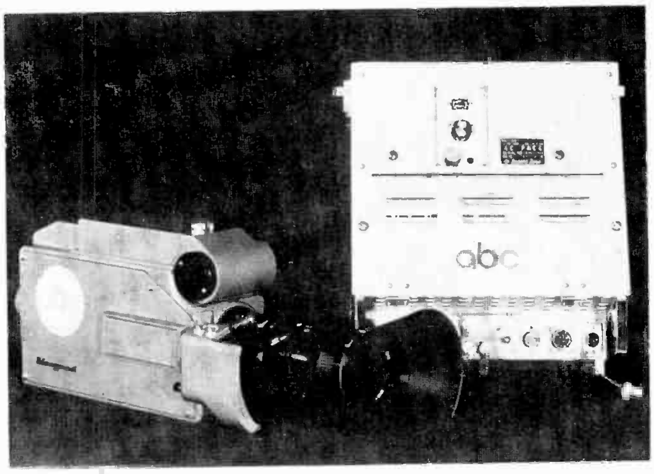


Figure 3

Figure 3 shows the Ikagamai camera. We bought two for each location instead of one.

Figure 4 shows the small tape machine for the field without the record attachment. This is not built for broadcasting but for the institutional market. We're trying to adapt it at the present time. And hopefully within the next eight months, Sony will come out with a more rugged portabalized packaging for broadcasting.

This is the camera (**Figure 5**) sitting on a tripod. On the flood is the backpack that goes on the back of a man.

There they are together (**Figure 6**). You can see that the tape recorder is almost the same size as the camera control unit.

That's the van (**Figure 7**). We bought a suburban so that we could go on all the highways, so you're not restricted as a commercial vehicle.

Now this is the small microwave (**Figure 8**) that goes with the camera. **Figure 9** is the dish that shoots up to the top of the building. **Figure 10** gives you an idea of how it all goes together.

There is the four-sided receiving antenna (**Figure 11**). It picks up the strongest signal and relays it back to the studio.

Figure 12 shows a simple editing system. It's a cut from one to the other. Very simple, very crude, and should be improved upon, but this can do quite a bit of work just the way it is. It has audio-follow-video and two soundtracks. Obviously, this must have monitors and a few other assorted devices.

That's the same recorder (**Figure 13**) which you would use for playback. To air you must have a time-base corrector (**Figure 14**). This is a digital device, which makes the time-base correction, which is really the breakthrough to be able to use small recorders.

These are the basic elements that you need. How you start, depends on your own situation.

Figure 4

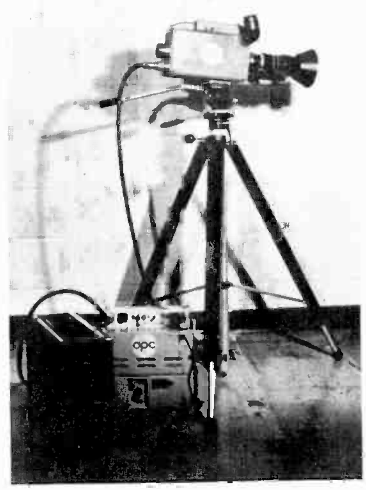


Figure 5

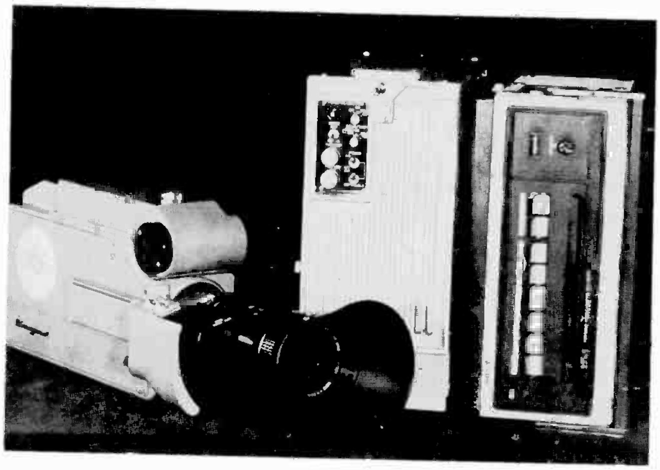


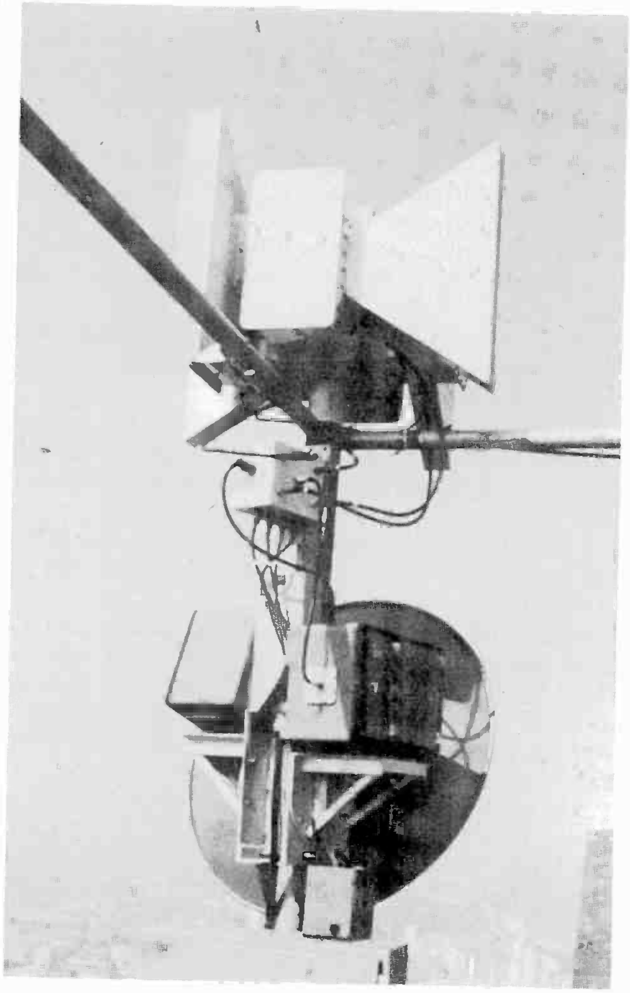
Figure 6


Figure 7


Figure 8


Figure 9


Figure 10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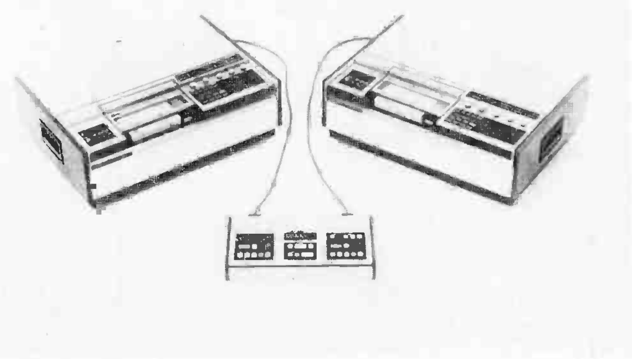

Figure 12

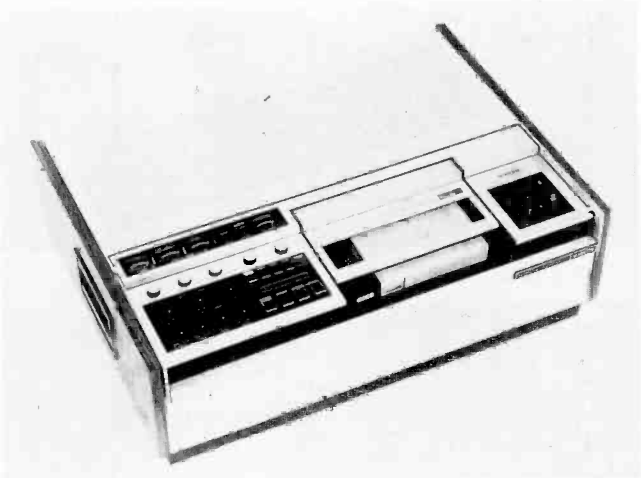

Figure 13

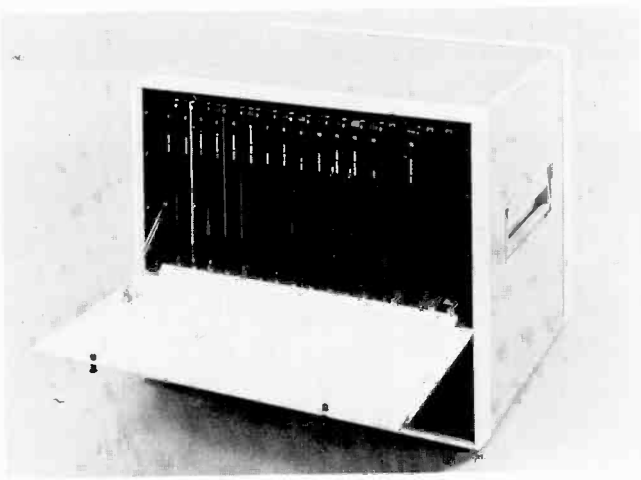

Figure 14

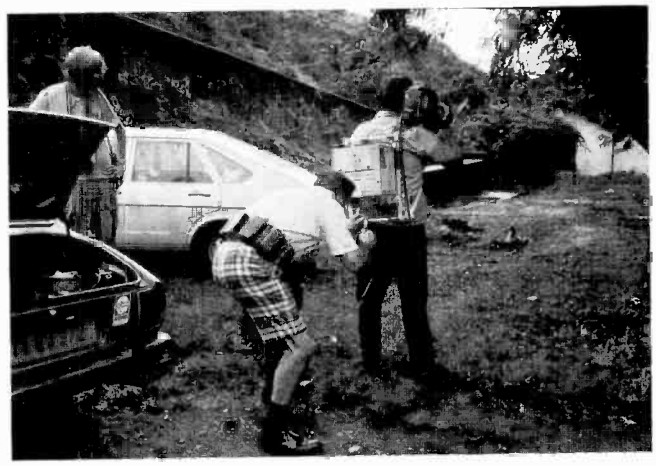

Figure 15

Figure 16

In a presidental pickup from Martinique we used a two-man crew as shown in **Figure 15**.

Figure 16 shows the crew with the producer.

There is the three man crew **(Figure 17)** with the correspondent.

This is interesting and I did it on purpose **(Figure 18)**. It represents the flexibility of the two-man crew.

Both men have to be able to do everything **(Figure 19)**, so you don't have just the camera expert. They must know both positions so that they can relieve each other. They are standing in those positions for several hours, and since they're only two men, they have to know how to help each other out and both be able to operate the equipment.

Here is a typical three network pickup **(Figure 20)**. ABC has a silent camera to protect us. CBS Ikagami, ABC Ikagami and NBC Furnsay. We all stand in the same location and that's very important. They will give you a position and you must shoot from the same distance as film, so therefore you need the same focal length.

One of the advantages of electronic cameras over film is that the producer can sit **(Figure 21)** and take

Figure 17

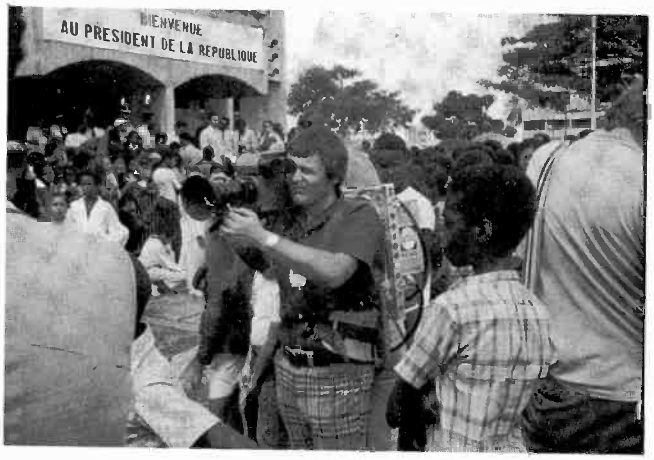

Figure 18

notes so he knows exactly what he's going to use right on the air. That's a real plus.

Now that's our little Martinique portable editing unit (**Figure 22**), with the little device in the middle and the monitors. That's basically how it goes.

In concluding I want to make one point. If you're not in electronic journalism, dip your toe in the water. It can be as small a price as the smallest camera. You have to learn how to go out. You have to learn all the problems. You have to learn how to go out. You have to learn what it's all about. During the great experience we had working with the small cameras, we found out all the problems we have no matter what cameras or tape machines you have.

I think it's very important that all of you dip your toe in the water on a small basis. That doesn't mean you go out and get the top-of-the-line, but get in and do something. Thank you very much.

Figure 19

Figure 20

Figure 21

Figure 22

Thomas M. Battista

Vice President and General Manager
Television Station KMOX
St. Louis, Missouri

What is electronic news-gathering? Well, it's the gathering of the news of the day, news which can be tragic, joyous, tender, formidable and above all, informative about actions in government, actions in our community, and about sports, weather and all the events going on about us. But what makes this electronic news-gathering is that it is done entirely with electronic tools. Electronic cameras have replaced film cameras. Videotape has replaced film and film processing. Even the editing is performed electronically. No one physically cuts or splices one frame of film.

This is a first. Ours is a first. The first all-electronic news-gathering.

The hub of the gathering process is the newsroom. It bears little resemblance to the old print-media newsroom. And as a matter of fact, it bears little resemblance to the traditional broadcast newsroom. Each reporter's desk is wired for picture and sound. They prepare and broadcast the news of the day right from their own desk.

But what makes it all work? People, skilled news people who are using electronic tools to do a better job.

That's our electronic news-gathering operation, but let me tell you how it all works.

In one corner is what we call the ENC, or executive news coordinating console. We've combined the functions of the traditional assignment editor's desk with those of the executive producer's and added a complete ENG communications system. Every story begins and ends with ENC's.

The executive on duty posts assignments on the board. All the while, he is listening to police radios and observing another story coming in on his monitor.

Microwave receiving channels are located adjacent to the newsroom. As soon as an ENG crew reaches its assignment, it ordinarily begins transmitting a picture signal so that it can be checked here and simultaneously fed to monitors and recorded on these one-inch helical scan videotape recorders. More about this in just a moment.

Our van arrives at the location of the new assignment. Our reporter is there with her guest. Despite a very light rain, she decides to keep the interview in front of house.

Our camera is comparable in size and weight to a 16 mm camera. As soon as the camera is focused and framed, the interview can begin. Unlike the film camera, the electronic viewfinder of this camera displays the actual picture being taken. The camera operator knows exactly when it's focused without having to wait for the film to be developed. The camera we use is an Ikagami with a 10-to-1 electronic zoom.

The picture is recorded on small helical scan videotape recorders in the van, but at the same time the story is being transmitted via microwave to the studio. The reporter can replay the van recording and review the coverage while she is still on location.

When it is not convenient to run cables to the van, as in the shooting of interiors or moving the camera some distance for following action, a small, battery-powered Sony videotape is used, with the camera operator and the tape operator working as a single unit, independent of the vehicle.

We have a tape assembly room with the microwave receiving channels. There are pictures on two channels. One is carrying a house restoration interview. The operator receives a signal, and as he records it on one of the master video units, it is also available at the ENC's desk for his viewing. This allows him to stay involved with the progress of the story.

Simultaneously, it is recorded in one of the EDV, or electronic decision video in the back of the newsroom, immediately providing the writer with a videotape playback. He reviews the house restoration story. Had the story been shot with a film camera, then sent back to our news center, on arriving, it would have had to be processed and he would not have seen it for at least another hour.

On reviewing the incoming videotape, the writer talks via radio-telephone with the reporter at the side of the interview.

He's on the readout below the monitor. By pushing the hold button, he can stop the readout and enter the number on a log sheet. These are precise frame numbers showing the exact points where he wishes to make edits.

After the decisions have been made and the log completed, he hands it to a technician in the adjacent tape assembly room. After he receives the log sheets on the editorial decision, a technician in the tape assembly room can now assemble the tape.

This is simple assembly equipment with the master recording on the left and the assembly taking place on the right. By setting numbers which correspond to the time code numbers on the log sheets, the tape is automatically edited according to the decisions made by the writer. In contrast with film, the tape is not physically touched by hand. There is no problem such as film dirt and scratches. The time code numbers are clearly visible on the readout as compared with the tiny numbers placed on the edge of a motion picture film. Also, unlike film, which is physically cut and edited, the master tape remains intact, it may be used again and again any number of times for editing.

As I said earlier, all stories begin with the ENC and end with the ENC. And once a story is approved by him, it is transferred to a two-inch videotape cassette. By means of keyboard, cassettes are coded to identify story and length. This provides the flexibility of deferring final decisions till the last minute on which stories run where.

And that's how we at KMOX-TV call in news electronically—the electronic cameras, recorders, microwaves, editors.

There is one thing that only a total electronic news-gathering system can do, and that is go live anytime during the broadcast day.

It may not seem like a big deal to you—one reporter talking to a couple of guys, but it was a major story here in St. Louis, and we had it first, hours before anyone else.

ENG enables us to cover more stories and more broadly. We are the first all electronic news-gathering television station. I have a feeling we won't be the last. Thank you very much.

Ralph L. Hucaby
Vice President for Engineering
Television Station WLAC
Nashville, Tenn.

In 1971, our management became interested in electronic news-gathering—particularly as applied to live telecasting of major news events with a small mobile unit on a "crash" basis. A thorough engineering and cost analysis was made, which established the fact that hardware costs were too high at that time to make the project feasible; so the plans were shelved.

Early in 1974, three years later, we decided to take another look. The state-of-the-art in equipment design and configuration had advanced considerably. Costs had also come down. Significant advances in microwave antennas, light-weight, battery powered, television cameras, digital time base correctors, and the like, made the possibility of electronic news-gathering much more practical. A decision was made in April, 1974, to make a large scale commitment to ENG, and work was started at once.

The basic objectives of this commitment were two-fold. The first, which remains unchanged from our 1971 study, was to enhance a feeling of viewer participation in *unedited* "news-as-it-happens" by: (1) the interruption of regular programming for the telecast of *live* news stories of *major* proportions, and (2) by the insertion of live news stories into every newscast.

The second objective was to decrease the cost of news-gathering, chiefly through the use of reusable videotape instead of non-reusable film for the majority of all *non-live* stories.

In 1973, WLAC-TV shot and processed over 1,093,000 feet of 16mm color film which cost approximately $90,500. The cost of operating the film processing lab for that year exceeded $37,000. Although News film did not account for quite all of these costs, it certainly made up the major part of them.

Obviously, electronic news-gathering can do little toward reducing the personnel required for today's sophisticated newscasts, but it can do a great deal to reduce the cost of the medium by replacing some of the film, and its processing, with videotape. Some things, such as documentaries and stories sent in by "stringers", will probably continue to be done on film for some time to come. Some of our initial stock of

Figure 1

videotape, 125 twenty-minute cassettes costing $3,150, is still in use after 6 months. We do know that we have run six months on a tape cost equivalent to what we once spent each month for film. I think the cost reduction facet may be best illustrated by pointing out that WLAC-TV has invested about $350,000. In electronic news-gathering, and that station management is fully expecting the savings to pay for the system in a maximum of six years.

Permit me now to take a few minutes to describe our Live Action Cam hardware, and to explain how it is organized into a flexible, efficient, news-gathering system.

First, it was determined that WLAC-TV could effectively gather news within its primary area with two, identical mobile vans (**Figure 1**), which are operated six days a week. Presently, they are not used on Sunday, giving time for vehicle and equipment maintenance—a *very real* necessity!

The crews consist of a photographer, a combination driver/security man, and sometimes a reporter—although he often travels to the scene in one of the "pool" automobiles. Special training was given to several of the newsroom personnel to enable

Figure 2

Figure 4

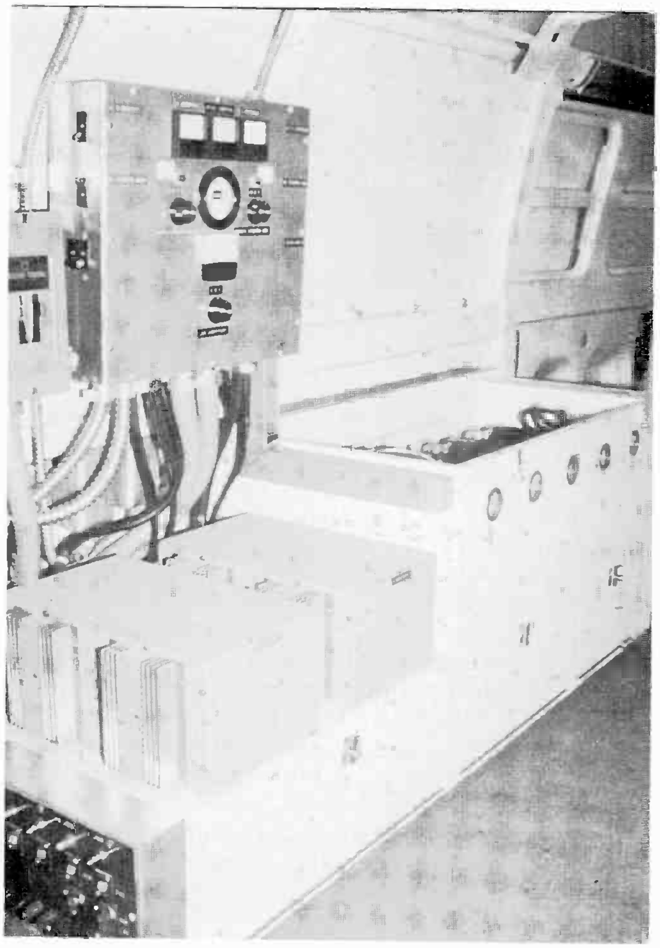

Figure 3

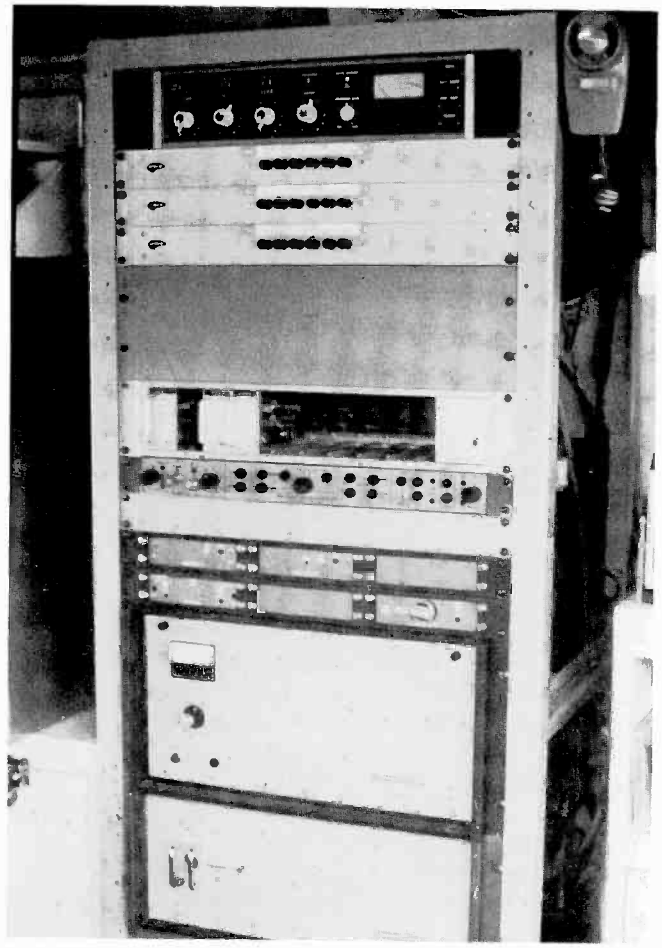

Figure 5

them to obtain the FCC Second Class Radiotelephone license which is required for those operating the microwave transmitter on board.

Two Ford Econoline 200 vans were modified by our engineering personnel for the purpose (**Figure 2**). A top platform was added to support a hinged microwave antenna assembly which can be quickly raised and oriented. The cameraman can also work from this platform, if desired.

A power source was installed consisting of standard 12 volt automobile batteries (**Figure 3**) whose output is inverted to 115 volts AC.

Each van carries one Ikegami HL-33 color television camera which delivers pictures of excellent quality with amazing stability (**Figure 4**).

The microwave system on board the van (**Figure 5**) consists of a Microwave Associates transmitter capable of 12 watts output power which feeds into a Nurad

Figure 6

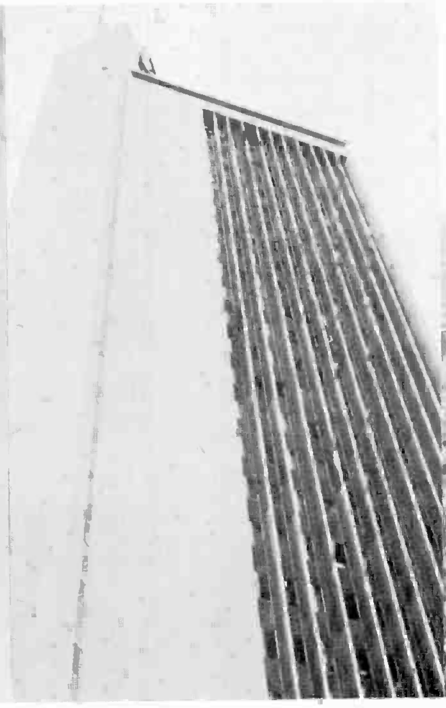

Figure 7

Figure 8

circularly polarized antenna on top of the vehicle.

The 2000 mc frequency, coupled with the relatively high power (**Figure 6**), makes "bounces", and even multiple bounces, feasible when a direct path does not exist.

A unique microwave receiving antenna system is located on the roof of a 33 story building in the center of Nashville (**Figure 7**).

This Nurad system consists of four "horns" (**Figure 8**) each having a 90 degree horizontal acceptance angle, and each mounted to receive from one of the four major directions. These horns feed a coaxial switch, remotely controlled from the newsroom, where the engineer on duty switches to the horn giving the best pictures and sound from the van.

Once the proper horn has been selected, its polarity can also be remotely switched for further improvement in picture and sound reception (**Figure 9**).

Before we leave the subject of microwave, I might mention that we also have two Microwave Associates low power units operating in the 13 KMC band at 0.1 watt, which we will use when possible between the camera location and the van—thus eliminating the necessity of stringing cable.

Helical videotape equipment has also made giant strides in recent years. The invention and marketing of the digital time base corrector has made it possible to use relatively inexpensive helical machines without violation of FCC Standards on time base stability.

Since videotape was to be handled extensively by non-technical news personnel, we felt strongly that

Figure 9

tape cassette machines would be much preferable to reel-to-reel types. Fortunately, the Sony Corporation chose this time to introduce two new helical machines designed around their familiar ¾ inch U-Matic cassette (Figure 10).

Their VO-3800 is portable (30 pounds), battery operated (one hour) and accepts a cassette twenty minutes in length. It is this unit that is carried on board each mobile unit for recording at those locations where a microwave path cannot be obtained or where camera cable cannot be run.

The Sony VO-2850 was the machine chosen for use in the newsroom for recording, editing, and playback. This is a larger machine which has capstan servo, thus lending itself to frame accurate editing and to inter-

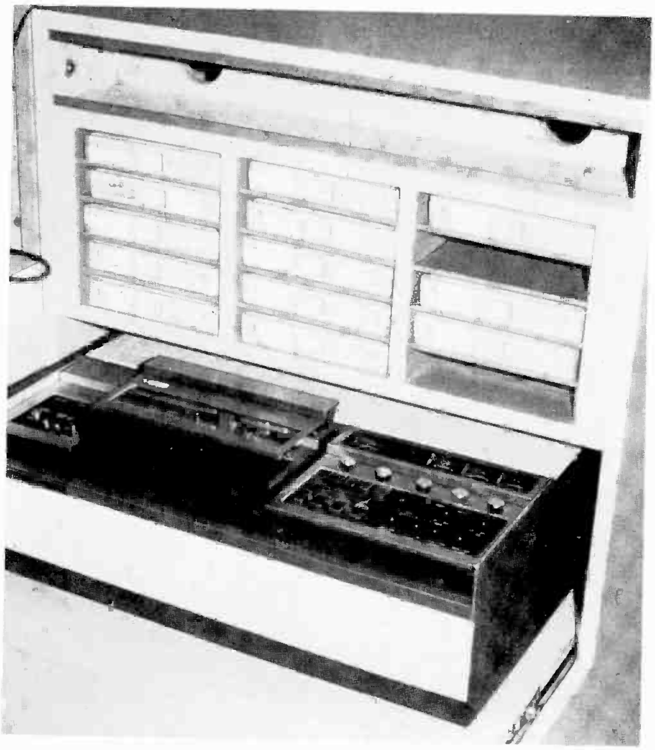

Figure 10

Figure 11

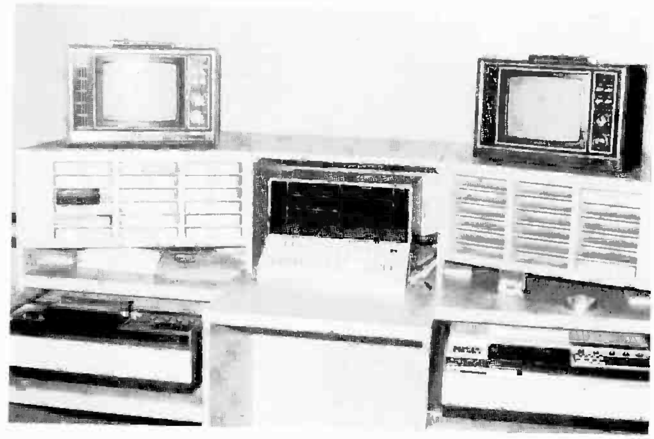

Figure 12

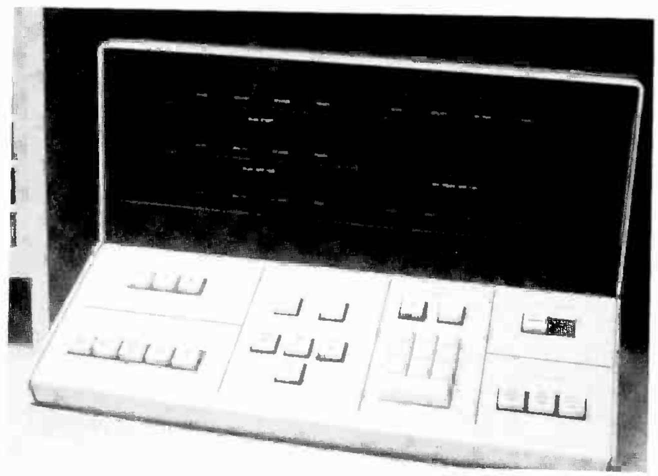

Figure 13

face with the studio system. Both the 3800 and the 2850 process the chrominance portion of the color signal via the heterodyne method, but they have a very acceptable signal-to-noise ratio, and the final result is most often superior to the average 16mm news film.

The routine microwaved news story, as received in the newsroom, is recorded on the master helical recorder (**Figure 11**) along with a SMPTE time code signal that is logically generated. If the story is to be aired "Live", the master recorder may be bypassed, and the signal fed directly to our routing switcher for routing to the proper point of origination.

Upon completion of the master recording, the cassette is given to news personnel for editing.

This is done at a special console equipped with two helical machines interfaced to a Datatron Time Code Editor (**Figure 12**).

The Datatron control panel arrangement (**Figure 13**), and number of readouts, which display tape position and all edit points, make it especially suitable for operation by non-technical news personnel.

Playback of the final (**Figure 14**), edited news stories on helical tape cassettes is done on two more Sony machines which alternately feed the time base

Figure 14

corrector. Let me emphasize that at no time do we use our quad machines in the ENG system!

The studio intercom system was expanded to tie the newsroom to the director of the newscast and to the master control room. Even the two-way radio system is interfaced to the director's intercom so that he may talk and listen to the live action cam personnel while doing a live feed. This two-channel, 450 mc radio system, licensed in the remote pickup service, is used to coordinate the efforts of all those involved in the news-gathering operation (**Figure 15**).

How well is it working? What are the good points? What are the disadvantages? Well, six months of operation may not be enough to be able to answer these questions in depth, but it is long enough to get some good indications. An average taken over a recent six week period shows 70% of our news stories

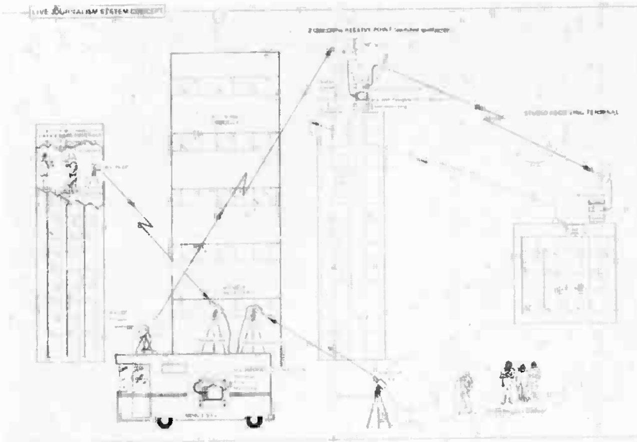

Figure 15

either live or on videotape, with 30% on film. Film footage processed has come down from a pre-ENG average of 105,000 feet per month to an average of 68,000 feet per month.

Live action cam photographers, knowing that their tape is reusable, tend to shoot more footage on each story—giving the editor a better choice of material with which to enhance the story. The fact that a story microwaved in at 9 a.m. can be edited at 9 a.m., thus avoiding the traditional "panic period" editing following a film processing run, makes possible more leisurely and thoughtful editing. Of course, it goes without saying that we can telecast later breaking stories than we would when we were saddled with film processing. We usually do one or two live stories during each newscast—even if they aren't big ones! This helps to fix our live capability in the minds of our viewers. They know that if a really big story breaks—live action cam will be there!

One of the most exciting advantages of ENG is to be found in the opportunity for active participation by the news director, assignment editor, or others in news management, in each story being received by microwave. They can see and hear the story as it happens, and by two-way radio, they can evaluate the worth of the story, suggest interview questions, or even relay late breaking events, unknown to the reporter on the scene, that may affect the story in progress.

I am being quite candid in saying that, so far, we have not found any real disadvantages to electronic news-gathering. I have been with WLAC-TV since its inception twenty years ago, and I can say without reservation that we have never before initiated any new type of service that has brought so much favorable comment from our viewers. Thank you.

A. H. Lind
Manager
Studio Equipment Engineering
RCA Corp.
Camden, N.J.

It's pretty obvious from all of the comments made that there's a great deal of activity in the electronic news-gathering field. But also I think, from the comments made, it's quite evident that a great deal more work needs to be done in the equipment and apparatus area. For electronic news-gathering to become widely used by television news departments, the equipment available for this purpose will have to become more competitive with 16 mm film equipment. The equipment which the news crew is expected to carry must be highly portable, which means lightweight and compact. However, it must also be very reliable, easy to use, and as automatic as possible. Of course, it must be battery-powered and capable of extended periods of operation on a single battery charge. The cameras must be capable of making excellent pictures under conditions of available light for most normal lighting conditions.

For most news situations, a means of recording the signal is essential. The recording could be done at the studio if wireless links or cable links are available. However, in general, it is necessary to have recorders in the field, either in a van or in a portable form, brought along with the camera to the site of the pickup.

At the present time, the portable equipment intended for use on location is contained in packages weighing about 30 pounds each. Some units, of course, are lighter.

I would like to review very quickly three camera systems that range from portable production news-gathering to a future camera that holds promise for a very lightweight, small equipment that will be highly competitive with 16 mm cameras.

The first camera system is the TKP-45, which you may recall was introduced at the 1974 NAB Convention in Houston. This camera was tied to the TK-45 CCU, which is a rather large unit for any portable operation, although in transportable field cases it is quite readily operable in the field. This year we're introducing a battery-powered CCU portable unit which converts this system into an entirely portable battery-powered one.

A wide spread of camera systems are possible using one single camera, which can form the nucleus for a very versatile and relatively universal system. There's a wide range of lenses available, all the way from large zoom lenses to the order of 34-to-1 in focal length range, down to fixed focal length lenses of the 35 mm camera type, by utilizing a suitable adapter on the camera. In addition, there's quite a variety of mounting arrangements for carrying on the cameraman's person; a lightweight, readily carried tripods; or on pedestals with adapters for using the very large lenses. Two different sizes of viewfinder, three different methods of operation, utilizing the TK045 transportable KCCU, a triax cable adapter which permits 5,000 feet of separation between the camera and the CCU, the battery-powered portable pack with remote control panels available, etc., etc. This forms an extremely flexible system which can be utilized in many situations.

The TKP-45 is a full-performance camera which offers studio-type pictures and yet facilities for field location operation. Its primary designed purpose is to provide a portable production camera system. However, in its minimum package configuration, it becomes a very usable high-quality news-gathering system.

The second camera is the TK-76 which is being introduced this year. It represents a very substantial step forward in the reduction of size and weight of the camera system. This camera chain consists of two packages, the first one of which is the camera head with its viewfinder and zoom lens, and the second is the power pack.

The third camera system is one which will employ solid state sensors, frequently referred to as Charge Couple Devices or CCDs. In the RCA exhibit there is a demonstration which will show you on an RGB display basis the kind of pictures that are possible to obtain utilizing CCDs today. A camera employing these devices, but similar in many respects to the TK-76, I am confident will result in a complete camera chain which will weigh under 20 pounds. The small size of the CCD sensor permits new choices in the configuration of the camera head package, which among other things, should lower its profile to make possible a package that is a smaller visual obstruction to the cameraman. Although results with CCD sensors is very encouraging, it is still a number of years before we expect a practical deliverable camera product to become available.

As electronic news-gathering becomes a more widely employed technique, the equipment market for highly portable but excellent performance equipment will of course grow. I see great potential for portable television pickup equipment developed and manufactured to serve the market as completely and with the same degree of satisfaction which the studio and field production types of equipment that are currently available do. Thank you.

Questions and Answers About Electronic News-Gathering

Following the presentations, authors answered questions from the audience. The exchange follows:

Q: How do you handle the archiving of your stories?

Mr. Hucaby:

Obviously, if you save everythng, you haven't gained anything, because most of your savings are built around reusing videotape. We started out by saying there shall be no archiving, and then it dawned on somebody that license renewal is coming. So what we are doing now is archiving only what goes on the air each day which turns out to be seven or eight days on a one-hour cassette.

Mr. Battista:

As far as our archiving is concerned, we have a couple of ways that we work. Immediate access—everything, the entire show as broadcast is taped on a Sony and that's kept for 60 days. It is determined by news people what they perceive to be important pieces, and they are lifted out of either the master or out of the Sony and then put on two-inch, labeled and timed accordingly. And that's how we do it. And we have I think a 90-minute reel that we just have isolated for a morgue.

Mr. Barnathan:

I think it's pretty much the same problem that we have over here. We haven't at the moment got enough tape in the house yet, but I want to tell you, it's going to be the nightmare of all times, because what happens is, unless you're able to get your news people to give up that tape and make sure that you do it right up front by logging it properly, you're going to have the same problem we have now with the two-inch stuff. It's just going to lay around and nobody's going to make any decisions.

Q: I'd like to know what has happened to your costs pre-ENG and now that you're into it.

Mr. Battista:

You can take a cost figure and do all kind of tricks with it. Pure and simple, before we went into electronic news gathering we had 56 technical people. That's a combination of two different unions. We now have 50. We are literally out of the film business. We don't process in-house one frame of film. In fact, we sold our processor.

We are spending around 85 to 90 thousand dollars a year processing film, plus the technical people involved. We no longer have the cost. Of course there's the videotape cost, but it's peanuts compared to what we were spending for film.

There's another cost that you have to take into consideration. It's kind of a reverse trick on numbers, and it's this: You put together a system. You develop a news broadcast that is better than the competition. And if that broadcast is accepted by the viewing audience, you can forget about costs because you're going to make all kinds of money. And I think ENG does that for us. There is a cost savings—a lot.

Q: Could you give us some idea of the setup time once you've reached the site?

Mr. Hucaby:

I think our shortest time has been something like two minutes. Our longest time has been something like 25 or 30 minutes. The average run something like 10 minutes.

If you make a card file for each location as to where you pointed the dish and where you park, and you go back there a second time, all you've got to do is pull the card. It's not too bad. It's amazingly good.

Mr. Battista:

We have another system besides the card file that helps us setting up time. If you drive through many of the metro areas of St. Louis you'll see little red spray can marks on the curbs. What we do is we line up our back tire to those little things and we can get it to you in a minute and a half.

One other thing we keep weekly, monthly tabs on the efficiency or inefficiency of the operation. Last March we covered with the same—now, we had three film crews then and we now have three ENG crews. We covered 20% more stories last month than we did a year ago, against film. So that—you can take that as a time factor. We're doing it faster and better.

Q: Mr. Battista, how did you arrive at the present setup? How many years and how many facilities did you go through?

Mr. Battista:

Well actually, we worked hand-in-glove. There are a couple of guys here in the room that worked with us on it. Joe Flaherty, who is with CTN Engineering and Ray Snyder.

I have to tell you something it was the first time that I really felt a single direction by many people of different divisions and different groups and different departments moving in one direction to get this thing done. How long was it? A year and a half.

Q: I believe Mr. Hucaby said he's not using technical people on his crew. Mr. Battista are you?

Mr. Battista:
Yes. I'll tell you one place where we get burned, and this is not a problem that we can't work out. The entire ENG crew, are IBEW people, as well as in-house people, the editors and so forth and so on. Where we do get burned is the meal period. Because they're used to taking their meal period in three hours or whatever it is; news doesn't wait for that. So right now we're paying through the nose, but we hope to work it out.

Q: Do the photographers or cameramen do their editing, or does someone else do the editing?

Mr. Battista:
The editorial decision is a newsroom decision. It could be a producer, a writer-editor, it could be any one of the members of the news management team. They make the editorial decision. The in-cuts and the out-cuts and any kind of cue lines, a technician does the physical editing.

Q: What is your overtime situation? Are you running into a lot of maintenance time?

Mr. Hucaby:
Not too much. The equipment has been amazingly good. In fact, I don't think in 20 years I've ever seen us put into service any system of this complexity that gave us so few start-up gremlins. The stuff is holding up very well and we're getting pretty darn good support from the suppliers, too. I think they have an interest in seeing it work. I don't know.

Q: How do you control the microwave links?

Mr. Hucaby:
The microwave is controlled by remote control. The microwave receiving antennas are controlled by tone-operated remote control. And the technician on duty in the newsroom can select the proper one of the four horns. Once he has selected the proper horn, then he can select the proper polarization of that horn to give the best pickup. And it's all done with a wire line. It could also be a radio link if you wanted.

Mr. Barnathan:
I'd like to add that the later microwave designs are automatic. You don't even need the person in the studio to do it. I think that as it goes along, the need's there and it'll be filled.

I might also point out union-wise you've got to find out what you can and cannot do within your own staff.

For example, editing, a person sitting there with a tape machine editing, could not be done by the editor himself. It would have to be done by a technician running the machine. The decisions could be made by him, but the technician would have to run the machine to determine the edit points, not for screening.

What I'm telling you is to get out and put your foot in the water before you go all the way, so that you can solve the problems and also find where all the red marks are.

Q: What type of lighting equipment do you carry on the crews?

Mr. Battista:
We don't cover any night stories.

Mr. Barnathan:
We use the same pack that we use for our news film crew. I might point out again, lighting is going to be another factor that could be a great problem in some of these operations, because remember, regardless of when there's a story, all three stories will be there, or whatever they are, and they'll come in, and their lighting may be done in a center and may be done by one union and your union is claiming the other. In fact, we have this problem between the three networks in Washington right now in terms of who does the lighting. And if you think that's not going to be fun, come join us.

Q: What about weather problems? How does it affect your operations?

Mr. Barnathan:
I might point out the cameras, in terms of the state-of-the-art, are farther ahead than the tape. The tape machines were just starting. The cameras are further ahead.

To give you an idea, we did a pickup a few weeks ago in a macaroni factory, and we got spaghetti in our tape because of the humidity. So, you know, we're learning as we go. There are problems.

Q: Since you sold your processor, how are you handling documentary work now?

Mr. Battista:
We're in the process of shooting two documentaries right now, which doesn't really come out of the news department; it comes out of the production, and they are being shot with ENG.

Interesting, we're not rolling that whole red truck to do that, because documentaries are kind of a walk and talk and, there's no reason to roll that truck just for the microwave equipment. So we have, in effect, put a Sony tape, an Ikagami and the lighting staff in a station wagon and we're doing it that way

So, we're really out of the film business.

Q: How many stories do you average on an ENG in a half-hour show?

Mr. Battista:

On a daily basis we do three separate half-hours. We do a 5:00 p.m. half hour and then we do a 6:00 half hour and then we do a 10:00 p.m. and on the weekends we do just the 6:00 and 10:00 p.m., two half hours.

Last month it was 402 stories. It was right around 400 in March.

Q: How do you coordinate microwave uses with two, three and four ENG trucks?

Mr. Barnathan:

This, of course, is going to be the real problem and all of us are going to have to work together. First to get the Commission to make decisions and secondly to coordinate our activities in the field of microwaving. There are only seven channels in the two giga-hertz frequencies. You use the same one. Basically, your answer to your question is you use one. You either shoot one or the other. You only use one at a time. I don't believe you normally will go to the other. You can switch from one to the other. But at the present time, these frequencies are used primarily in STLs and other local channels, and it's the problem when all three of you want to go to the same spot and start shooting, you're all in trouble.

Mr. Hucaby:

For two units, we use one frequency. And the old days when you had two microwave equipments, you wouldn't have dared turn it off after you once got it on. But the new solid state stuff, you say, "Number one, turn yours on. Number two, turn yours off," and it works.

Q: What has been your experience in pickup for fast action?

Mr. Hucaby:

Well, pretty good from our standpoint. The camera has a six-db-gain switch on it, which of course brings in some noise. But the noise that you get with a six-db-gain switch is no worse than the grain you would get in push processing 16 mm film. And I understand the Ikagami is here at the convention with a higher sensitivity camera. I haven't seen it yet.

Mr. Battista:

No, we haven't had any problems.

Q: What is your number of maintenance engineers that you carry on your crew?

Mr. Hucaby:

A supervisor and three people they're pretty well specialized. One of them takes care of all studio cameras and all Ikagamis. The supervisor is pretty good with the tape machines, and another guy handles microwave. It's pretty well specialized.

Mr. Chismark:

Well, we're just about at the end of our time and I want to thank you gentlemen for coming here.