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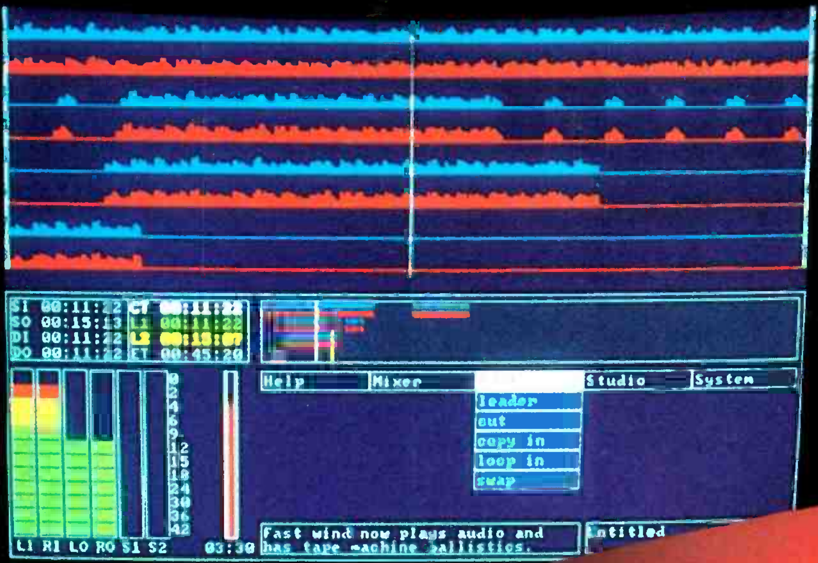

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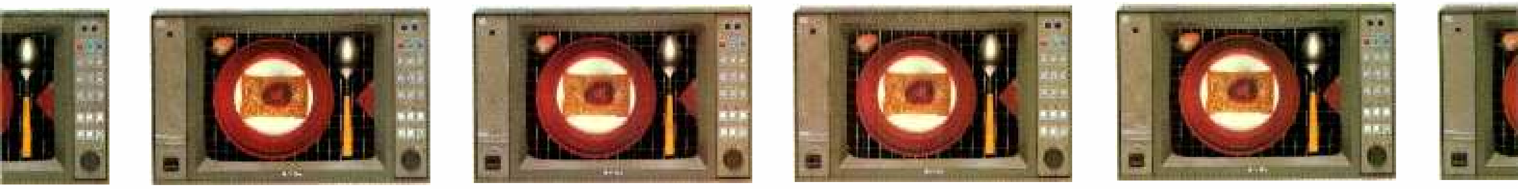

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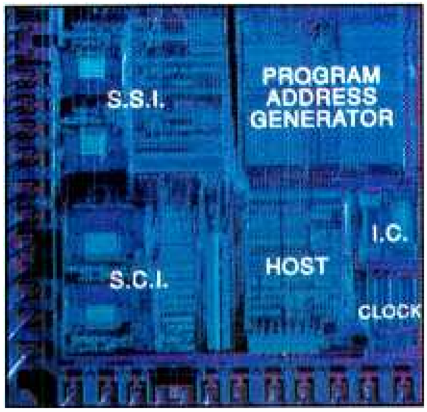

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AUDIO TECHNOLOGY UPDATE:

Broadcasters must continue to improve their station's audio systems. Consumers now have a wide variety of high-quality signal sources available to them. The competition from these sources requires that broadcast engineers continually re-evaluate the station's audio system to ensure that the audience is receiving the best possible audio. This month's feature theme looks at several ways engineers can improve their facility's audio systems.

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Digital audio is forging inroads for higher-quality audio systems in the '90s. (Cover credit: Kim Bracken, *BE* graphic designer. Console by Auditronics and monitor workstation screen display by AKG Acoustics.)

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Family tree.

1720 Vectorscope
The world's most popular line of high-performance signal monitors starts here: the 1720 delivers advanced features and easy operation with an unbeatable price-to-performance ratio.

1730 Waveform Monitor
The 1720 finds the perfect partner in this versatile, value-packed NTSC waveform monitor with similarly unmatched economy. They're ideal for side-by-side mounting.

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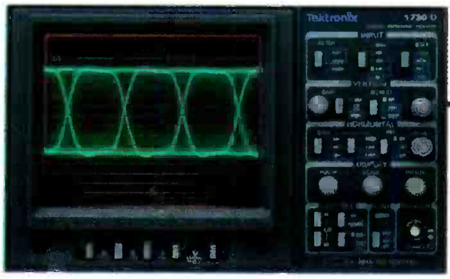

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News

By Dawn Hightower, senior associate editor

NAB urges WARC allocation of spectrum for DAB

The National Association of Broadcasters said that it wants the FCC and the United States to propose the allocation of spectrum for digital audio broadcasting (DAB) at the 1992 World Administrative Radio Conference (WARC). NAB further opposed WARC allocation of spectrum for wideband, satellite-delivered high-definition television (HDTV).

In its comments, NAB told the FCC that L-band spectrum (1.5GHz) is "strongly preferred" for DAB because its use of S-band spectrum "would significantly increase costs to consumers, even to the extent that the DAB service might never be established at any time in the near future." NAB said that aeronautical telemetry, a service currently located on the L-band, could be moved to the S-band. The association further asserted that it is not necessary for DAB to use the same spectrum in all parts of the world, so the support of the S-band (2.4GHz) for DAB by other regions should not influence what happens in the United States.

NAB urged the FCC to ensure that any spectrum allocated at WARC for DAB be allocated on a co-primary basis for BSS (Sound) (radio via satellite) and terrestrial broadcasting. "The adoption of co-primary status for BSS (Sound) and broadcasting would preserve maximum flexibility for the United States and retain all options to consider different forms of DAB in the future," NAB said.

The association said it also opposed the proposed spectrum allocation for the broadcasting-satellite service at the frequencies 24.25-25.25GHz. NAB cited its previous comments demonstrating that, especially in light of proposals for digital terrestrial transmission of HDTV in 6MHz channels, a wideband HDTV service is "unnecessary and unjustified."

NAB asserted that progress in digital compression is being made extremely rapidly. The association's previous comments also noted that "bandwidth efficiencies achieved through compression techniques may permit the quality envisioned for wide RF-band HDTV to be achieved with the channel bandwidths of the current BSS plan." For these reasons, NAB urged the commission not to allocate frequencies anywhere in the 22-24GHz band for broadcast-satellite service.

NAB asks Congress to maintain survival of free TV

The nation's system of free television and the service it provides the American public are in jeopardy, and only Congress can provide broadcasters with the tools they need to survive in the dramatically changing TV marketplace, according to the NAB.

In testimony before the Senate Communications Subcommittee, Edward O. Fritts, NAB president and CEO, said Congress and the American people are all too familiar with the problems caused by cable's monopoly power.

Fritts urged the committee to incorporate into its cable legislation a "Marketplace Improvements Act of 1991." As outlined by the NAB board in January, such legislation would ensure that:

- local stations are assured of access to their markets;
- broadcasters have access to new technologies, such as HDTV;
- broadcasters have the right to control the retransmission of their signals, their only commodity; and
- restrictions on telco ownership or control of video and audio programming are maintained.

Free employment services for SBE members

The SBE has installed a job line listing potential new employment opportunities for SBE members. This new member service provides a listing of employment opportunities throughout the country, basic details about the position, location and desired qualifications. The service is free to SBE members.

A job number is given with each listing. By using this number, SBE members can obtain detailed information about the opening from the SBE office.

Employers also can take advantage of the SBE job line. Technical openings for broadcast and media related positions can currently be listed for a nominal fee. Contact the SBE office for details at 317-253-1640. The SBE job line number is 317-253-0474.

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BROADCAST ENGINEERING is edited for corporate management, engineers/technicians and other station management personnel at commercial and educational radio and TV stations, teleproduction studios, recording studios, CATV and CCTV facilities and government agencies. Qualified persons include consulting engineers and dealer/distributors of broadcast equipment.

BROADCAST ENGINEERING is published monthly (except in the fall, when three issues are published) and mailed free to qualified persons within the United States and Canada in occupations described above. Second-class postage paid at Shawnee Mission, KS and additional mailing offices. POSTMASTER: Send address changes to **Broadcast Engineering**, P.O. Box 12960, Overland Park, KS 66212.

SUBSCRIPTIONS: Non-qualified persons may subscribe at the following rates. United States and Canada: one year, $50.00. Qualified and non-qualified persons in all other countries; one year, $60.00 (surface mail); $115.00 (air mail).

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Editorial

It's a great place to visit, but I wouldn't want to live there. Well, that's not quite true. My trip to the ITS Convention in Montreux, Switzerland in mid-June was an enlightening and enjoyable experience.

Because of the show's location (it has been held in Montreux for the past 30 years), the ITS show is often referred to as the Montreux convention. The gathering is an important European event for teleproduction equipment users and manufacturers. Although we in the United States view NAB as the primary yearly convention, the Europeans use IBC and Montreux as yearly gauges of industry news and health. This year's show was well attended and full of new announcements.

I was told to expect a rather relaxed, laid-back show. Former attendees told me to plan on having some time to experience the beautiful country and scenery. After planning such a schedule, I was disappointed to find absolutely no time to take in the beautiful surroundings and Swiss culture.

It didn't take long for the fast-breaking events to begin. The show got off to an encouraging start at the first manufacturer's press conference. There were the typical announcements of new products, improved products and the glossy presentation we press have come to expect. But one new technology that was described was repeated by two other manufacturers within two days.

The event I'm talking about was the announcement by three industry giants of future releases of new digital tape formats. Call them D4, D5 and D6 or whatever. Wow! In a matter of a few hours we discovered the future of videotape may lie in one, two or even three new digital tape formats. The reason we don't know how many formats will finally be released is that the technological specifications weren't announced. It's not clear whether there will be a single industry-approved standard for the new formats or if we'll be forced to endure three incompatible new formats.

The exciting news didn't stop with videotape. Touring the exhibit floor provided plenty of evidence that the other manufacturers haven't been sitting on their hands. The products ranged from dramatic 3-D graphics and character generators complete with every type of flip, warp and twist you can imagine to high quality 16:9, 625-line PAL transmitted images.

In one manufacturer's booth, a 16:9 image was relayed from Paris to the show floor via satellite, microwave STL and finally actual terrestrial transmission. The images displayed on the wide screen receiver were impressive. It's clear that the Europeans are running in fast forward toward improved image quality and HDTV-format viewing.

Although many U.S. broadcasters have not had the opportunity to visit the ITS show, I strongly encourage you to do so. You will find a wealth of new products, truly knowledgeable people and just maybe a hint of what you'll see at the next NAB.

If you go, you should plan for a rush of new products and features, but don't forget to allow time to enjoy scenic Lake Geneva and the friendly Swiss people.

Brad Dick, editor

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NAB Broadcast Engineering 1991 Hit Pick

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FCC studies multiple station ownership

By Harry C. Martin

In May the FCC issued a Notice of Proposed Rulemaking seeking comments on possible changes in its multiple ownership rules and policies. The changes propose the following:

- The revision of the current national ownership limit of 12 AM and 12 FM stations by substituting a numerically based or market rank-based approach.
- The revision of the contour overlap rule that prohibits ownership of more than one AM and one FM station in the same market (the "duopoly rule") to permit ownership of multiple stations in a single market that would be limited by cumulative audience share.
- The adoption of a policy that would encourage joint ventures and cooperative arrangements between competing stations.
- The adoption of rules or policies to govern time-brokerage arrangements.
- The raising of the minority ownership limitation from 12 to 20 stations per service.

In relation to these ownership limitation revisions, the FCC is proposing to raise or eliminate the current restrictions on how many AM stations a licensee may own. In this way, a single owner could own an unlimited number of AM stations, but no more than 12 FM stations. A variation of this rule would permit the ownership of an additional AM station in lieu of an FM station, as long as the total number of stations does not exceed the limit. For example, if the station limit is 30 and the FM limit is 12, a group owner could own 18 AMs and 12 FMs, or 22 AMs and eight FMs.

The agency also is considering relaxing the radio duopoly rule to allow a single owner to control any number of AM stations in the same area if the cumulative local market shares of those stations are at or below a given percentage. Other options include imposing a numerical cap in conjunction with an audience share limitation, or adopting a higher percentage share limitation for smaller markets.

With regard to the FCC's proposal to encourage broadcasters in the same market to participate in joint ventures, the commission is seeking comments on:

- whether such joint arrangements may be undertaken only by a limited number of stations in large and diverse markets to ensure that price and service competition will remain effective and robust.
- how to ensure that licensees will retain a sufficient degree of control over their facilities so that they will comply with all commission rules and public interest requirements.
- which mechanisms could be used to permit prompt termination of joint venture arrangements when the individual participants believe they are no longer consistent with the public interest or their responsibilities as licensees.

FCC amends settlement cap rules

The commission has clarified and modified the rule changes it announced in December, which eliminated settlement payments after a hearing begins. The commission will now permit recovery of expenses at any stage in the comparative hearing process for new stations. Furthermore, the settlement limitations will not apply to the bona fide mergers of competing applicants. These modifications will become effective Aug. 1, 1991.

In clarifying its position on mergers, the commission said merger proposals would be reviewed on a case-by-case basis in order to ferret out proposals designed to circumvent the new settlement limitation rules. In particular, the commission will reject merger proposals in which a dismissing applicant receives cash, either up-front or on a deferred basis, and the payment is guaranteed regardless of the outcome of the business venture.

New comparative hearing procedures

The commission has also clarified the hearing procedures and discovery rules it announced in December as part of a comprehensive effort to streamline the comparative process. These clarifications include:

Hearing fees. As of July 1, 1991, the hearing fee payment for all commercial broadcast applications for new stations not previously accepted for filing will be due on the date specified in the public notice announcing the acceptance for filing of the application. Applicants will be able to have the fee refunded only if:

- the application is dismissed, voluntarily or involuntarily, before a hearing designation order is issued.
- the application is granted without being designated for hearing.
- there is a settlement before the notice of appearance is due, or only one applicant files a notice of appearance.

Applicants accepted for filing before July 1, 1991 must have paid the hearing fee before July 15 if the application had not been designated for hearing by that date. If an application was designated for hearing before July 15, the fee was due with the notice of appearance.

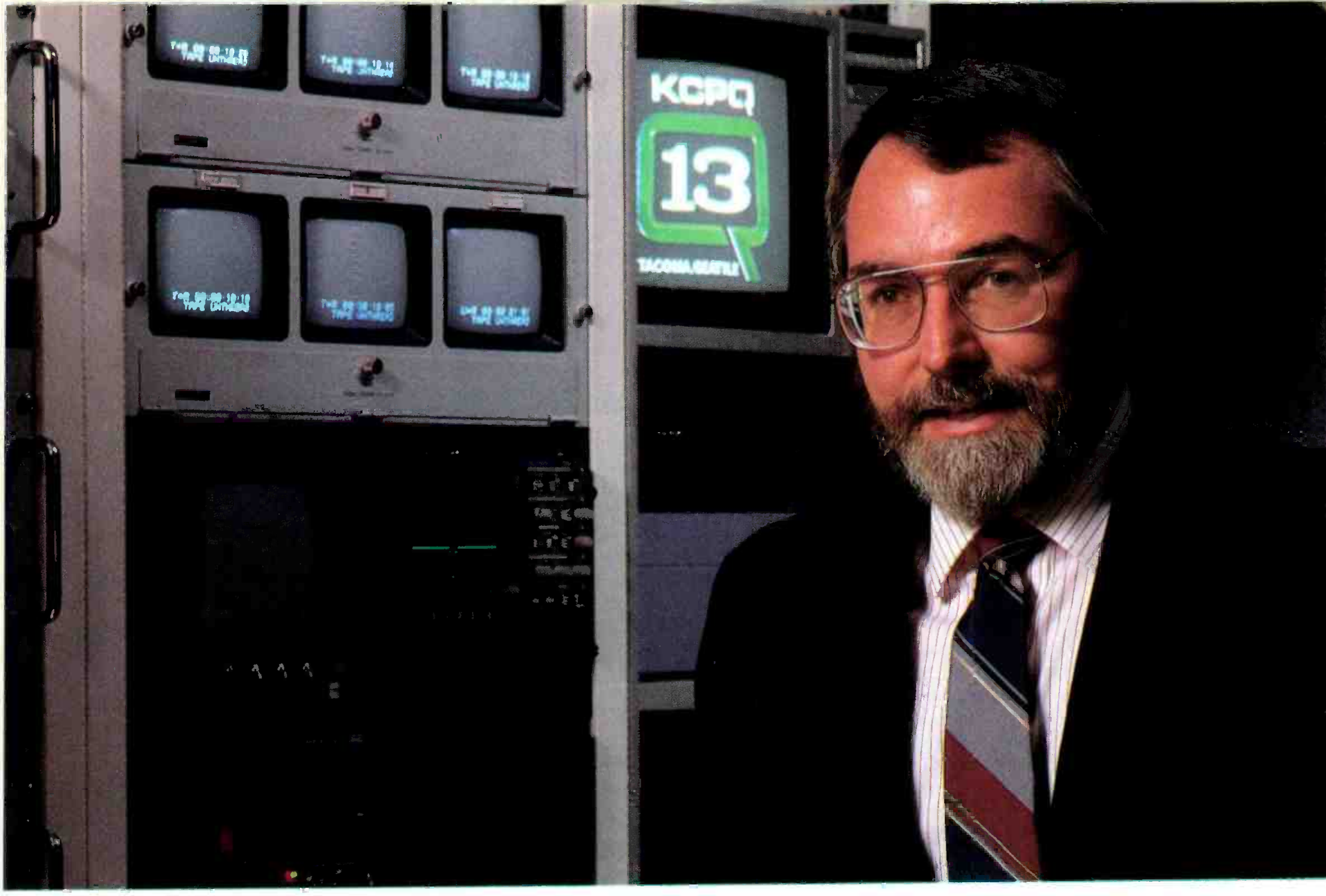
The "Ruarch policy." The commission extended the time for filing settlements that propose the withdrawal of comparative promises (for example, to integrate into management or divest another station) to the date set for exchanging direct case hearing exhibits.

Discovery/integration statements. Also as of July 1, 1991, hearing designation orders will set the fifth day after the filing of notices of appearance as the date for the initial exchange of discovery documents and integration statements.

Policing comparative grants. The commission clarified its requirement that successful applicants report later deviations from their comparative promises. This requirement is now limited to those applicants granted after consideration of their comparative merits in hearing, or according to post-designation settlements.

Pioneer's preference. The commission declined to make any changes to its comparative criteria, such as the establishment of a "pioneer's preference" for applicants who locate a frequency and have it assigned through a rulemaking proceeding.

Martin is a partner with the legal firm of Reddy, Begley & Martin, Washington, DC.



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Larry Brandt, *Chief Engineer*
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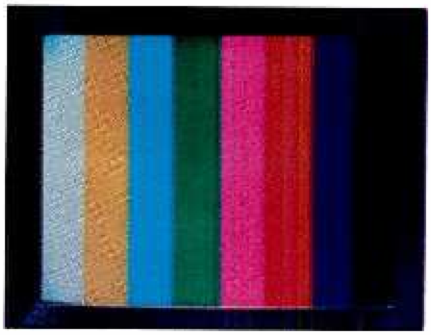


High-definition audio coming to TV

Digital audio compression

By Rick Lehtinen, technical editor

Last month, this column discussed the use of digital switched telephone networks as a means of bringing top-quality remote audio back to the station. The Switched-56 telephone network was introduced as a convenient means to accomplish this. Switched-56 is a precursor to tomorrow's phone system, the Integrated Services Digital Network (ISDN). Under ISDN, expect TV digital audio to leap into prominence.

ISDN is a switched system with two 64kbyte/s signals and a message channel per line. However, adding both of the ISDN's 64kbyte/s channels still yields only a fraction of the 1.5Mbit/s bandwidth required by CD-quality audio. There must be further processing to make the audio fit the media. This is where the techniques of digital audio compression come into play.

Squeeze down the pipe

Once the audio is in digital form, special signal-processing algorithms can shrink the amount of data that needs to be transmitted to the available bandwidth. The principle is the same as that of writing shorthand — greater volumes of information can travel down the same channel (pencil and paper) if an extra coding and decoding step is added. In shorthand, letters and sounds that are not necessary to reconstruct the original message never make it to the paper. It is the same with digital audio compression schemes.

Once the audio is in a digital form, special signal-processing algorithms can be applied to shrink the amount of data that needs to be transmitted to the available bandwidth.

But what is necessary and what is not? This is similar to the quandary faced by the originators of NTSC color. Some guidelines were needed for allocating colors to either the full-bandwidth I channel or the reduced-bandwidth Q channel of the 3.58MHz color subcarrier. At length, NTSC's designers solved the problem by testing groups of people to find out which colors they could see most acutely. Then, the most important colors were put on I, the less important colors were put on Q, and NTSC's 18° axis shift was added to make it easier for the system to reproduce flesh tones. Some color detail got left out, but few people noticed.

Extensive subjective studies have shown that the ear doesn't really hear all that is presented to it...[In some audio compression schemes] portions of the audio signal that fall into a masked area are not transmitted.

Designers of digital audio systems have followed a similar tactic. Extensive subjective studies have shown that the ear does not hear all that is presented to it. Take the case of high-amplitude tones. A powerful tone of a given frequency will capture the ear and completely swamp out other tones of lesser amplitude at nearby frequencies. This *frequency masking* effect is one of the opportunities exploited by some digital audio compression schemes. Portions of the audio signal that fall into a masked area are not transmitted.

A second opportunity occurs because of *temporal masking*. Loud sounds capture the ear for a short period on either side of the sound. Because they cannot be heard, some compression systems cut out sounds that occur in this masked period.

Temporal masking comes in two forms: *post-event* (just described) and *pre-event*. Pre-event temporal masking apparently occurs because loud noises appear to take the high road to the brain's perceptive centers. Loud sounds can apparently bypass subtler, hence, slower sounds. (This may be the reason a cracking limb on a quiet night can sound like a gunshot, while the same noise amidst daytime sounds scarcely garners a second glance.) Some compression designers use the pre-event temporal masking as a convenient hiding place for the pre-ringing associated with digital filters.

This leads to a final powerful data-reduction technique. Sixteen-bit linear PCM systems avoid noise by having great dynamic range. Some digital compression schemes use clever bit allocation schemes that allow the noise floor to float upward until it is just below the information level. This works because the same phenomena that mask audio also mask noise.

TV audio products

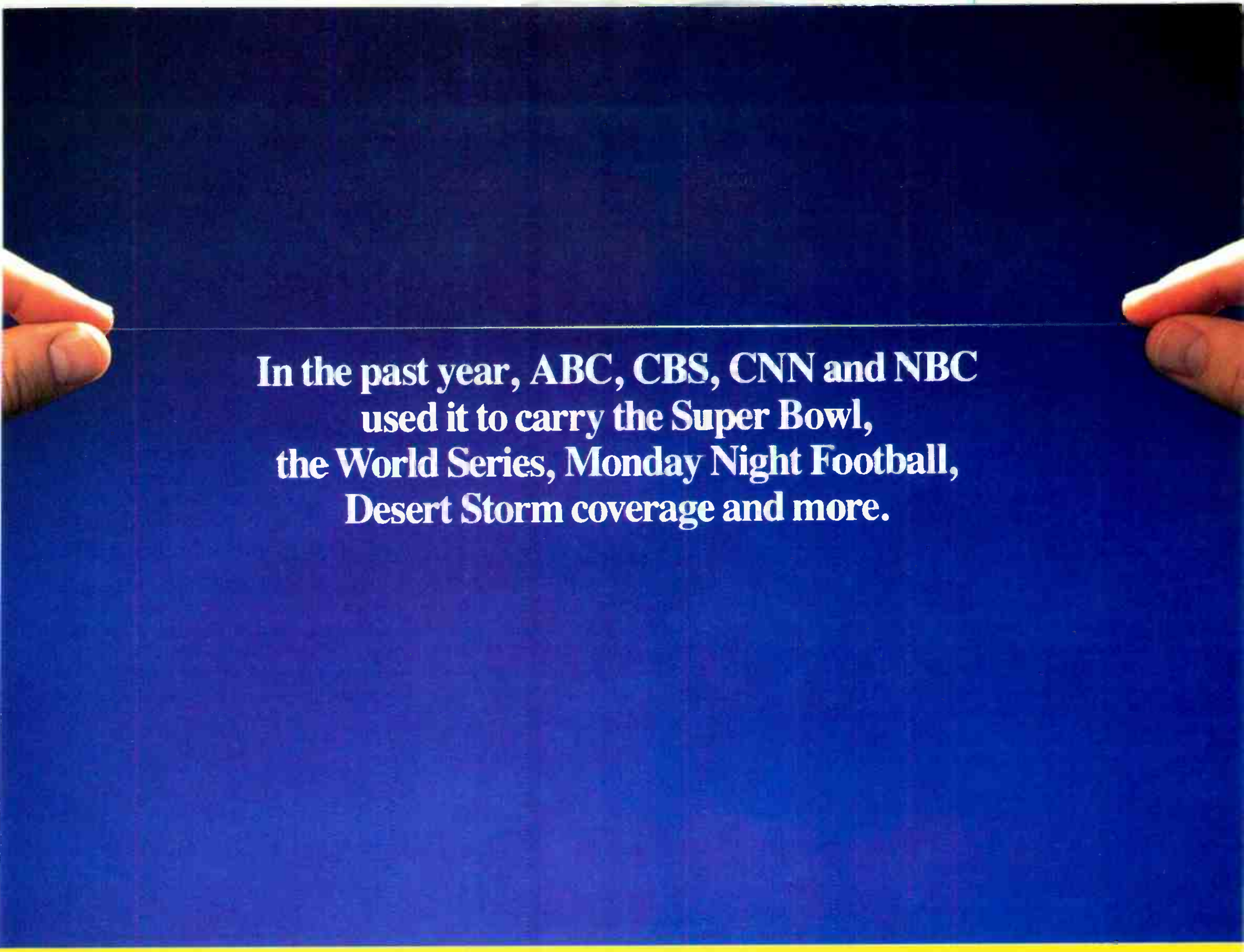
The fact that these compression techniques have application in television has not been wasted on the equipment designers. Some manufacturers plan to release TV products soon. But how to get this improved sound through the system to the viewer is a tougher question. One suggested technique encodes digital audio into sync tips, or onto the color burst or subcarrier.

Digital audio techniques will radically upgrade the audio quality of TV signals. It will take a major effort to improve today's system of transmitters and receivers. However, the areas of audio signal acquisition and in-plant audio processing are open for digital improvements. There is every reason for today's engineers to begin taking advantage of the benefits offered by digital techniques.

Under ISDN, expect TV digital audio to leap into prominence.

Acknowledgment: The author wishes to thank Dr. Larry Hinderks, vice president of engineering, Corporate Computer Systems, Holmdel, NJ, for help in preparing this article.

Editor's note: For a thorough discussion of the NTSC encoding system, see "Enhancing NTSC," *Broadcast Engineering*, April 1989. BE



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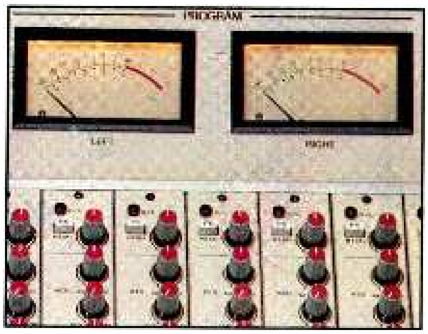

Applying vectors

Complex vectorial operations

By John Battison, P.E.

For many of you, dealing directly with vectors is not a part of your everyday workload. You don't often need to whip out a calculator, or pencil and protractor, and derive a resultant. But if you become involved in circuit design, vectors will play an important role in your designs. Probably the most important circuits, and certainly the most prevalent, are series and parallel combinations, which produce the resonances that critically affect your operations.

Anti- or parallel resonance

It is a common belief that the impedance at resonance of a parallel circuit is infinite. The vectors in Figure 1 illustrate this kind of complete cancellation. However, the cancellation is not actually complete because resistance is present even when the L and C currents cancel. There is no current along the zero axis, and only voltage appears. Figure 1 also shows an idealized circuit without residual resistance.

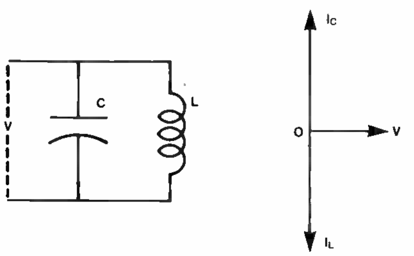

Figure 1. *The simple parallel circuit and a corresponding vectorial notation.*

Figure 2 shows a circuit with resistance (R) in series with the inductance. The voltage (V) that exists across the coil and capacitor is shown by vector OV at 0°. The current I_C leads V by 90°, as shown by vector OI_C on the positive Y axis, which is 90° ahead of OV. Correspondingly, vector OI_X depicts the 90° lag of the current I_L behind V.

This is a problem that circuit designers frequently face. When resistance exists in the circuit, there must be *less than 90°* of current lagging or leading. As the circuit in Figure 2 illustrates, the resistance is in series with the coil, so there will be less than 90° of current lag in the inductive arm of the circuit. The lower the Q of the coil is, the greater the effect will be. If Q were high, the effect of R would be much less, and I_L would have almost 90° of lag behind V. In Figure 2, it is assumed that Q is high, so the effect is small, and OI_L is shown with approximately 20° less lag. Because the voltage across it must lead, vector OV_L is drawn with a 90° lead.

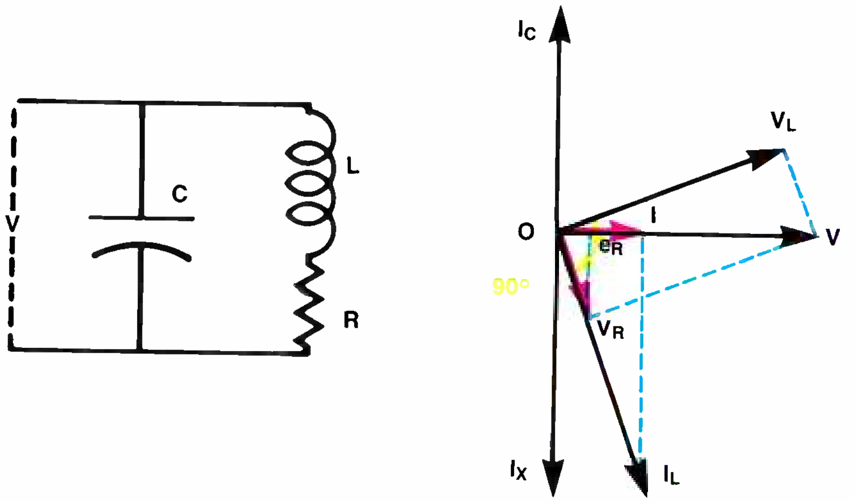

Figure 2. *Parallel resonance with R present. Note that I_L and I_C do not completely cancel because of OV_R.*

The voltage drop across R in the coil is in phase with the current in the coil. Therefore, OV_R, the voltage drop vector, is drawn along OI_L to the point e_R.

When resistance exists in the circuit, there must be less than 90° of current lagging or leading.

But now there is another complication: Addition in vector form will produce the wrong answer. If OV_L and OV_R are added in this way, OV and the current (I) will be zero. This ignores the resistive element included in C. To calculate the true resultant currents in L and C, the branch currents OI_L and OI_C must be added. The result of the two branch currents is OI in phase with V.

Now we can consider what happens at true resonance. The Y axis components should be equal in magnitude and 180° out of phase. This is shown by OI_C and OI_X in Figure 2. When these two are added, they will cancel, and only vectors OI_L and OI will remain at coincident angles.

In the three parts of this series, I have only touched the surface of vectorial applications. An understanding of vectors and what they can tell you can often clarify a problem, and help you find a quick solution. I'd like to end the series with this caveat: Be sure to put arrowheads on all vectors. If they follow each other around the figure, you have made a mistake.

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SBE announces slate of candidates

By Bob Van Buhler

The SBE nominations committee has presented its official slate of candidates to the board of directors. The committee's members included chairman Bob Van Buhler, former SBE president Roger Johnson and current secretary Paul Lentz.

Richard Farquhar, SBE's vice president, has been nominated for the presidential seat. Farquhar has served the organization as vice president for two years and as secretary for two years. He has also been a member of the certification committee and has held other key positions in the society.

The SBE vice presidential candidate is Jerry Whitaker. Whitaker, a former California broadcast engineer, is a well-known author and editorial consultant. He has published several books on broadcast electronics and has worked extensively on broadcast engineering subjects. He also is a recognized authority on RF transmission and AC power systems. Whitaker has played a major role in the production of many SBE national conventions.

Director Bill Hineman is the candidate for the secretary position, which is currently held by Paul Lentz. Hineman has served on the SBE certification and executive committees, and has strongly supported the SBE national office in Indianapolis. For example, Hineman was often called to help with the telephone and computer systems in the national office and was responsible for helping coordinate the relocation of the SBE office.

Treasurer Robert Goza has been nominated to serve another term. Goza, a broadcast engineer at KMOV-TV in St. Louis, and an active member of his local chapter, is convention committee chairman and has been involved in the financial management of SBE's national conventions.

Under the policy introduced by SBE president Brad Dick, the meeting attendance, voting record and committee participation of all incumbent candidates are reported to the membership annually so that SBE members can make more informed decisions when voting on their officers.

Van Buhler is manager of engineering at KNIX-FM/KCWW-AM Phoenix.

Candidates' requirements

For the first time, candidates will be requested to publicly commit to serve the society. All candidates for office will be asked to sign a form stating their agreement to fulfill to the best of their ability the requirements of office. This commitment form will symbolize their intention to attend all board and committee meetings and perform their assigned duties.

According to Dick, members should seek election to the board with their eyes open to the great amount of work they will be undertaking. Dick said that holding an office carries a financial and personal price, and that all candidates should understand that they will be held publicly accountable for their actions. Because there are only 12 directors and four officers, each must be fully prepared to accept these responsibilities.

For the first time, candidates will be requested to publicly commit to serve the society.

Directors are expected to pay their own travel and administrative costs, and receive no discounts on SBE convention admission. Considerable telephone, mailing and travel expenses are also a part of board membership. Board meetings are held twice each year, once at the October SBE convention and later at the site of the next convention. Officers can expect to spend several thousands of dollars per year just in travel expenses.

All directors are expected to serve on at least one committee. They must also juggle their time during their work hours to accommodate SBE-oriented telephone calls because like many professional organizations, much of SBE's work must be done during business hours. Because directors cannot expect to fulfill all of their responsibilities after work hours, they must have a supportive employer who understands the value of SBE.

Convention sites approved

The SBE board of directors recently approved convention dates and locations for the next two national SBE Conventions. The 1992 convention will be held Oct. 14-17 in San Jose, CA and the 1993 convention will be held Oct. 13-16, in Richmond, VA. These dates were chosen because they did not coincide with many other industry events, which will allow a greater number of broadcasters and exhibitors to participate in the convention.

These two sites were also chosen in response to the expressed desire of SBE members. The membership survey showed that members strongly support moving the convention around the country. In this way, local members, who might not be able to travel long distances, can attend the convention. San Jose and Richmond have the advantage of being a reasonably short distance from other major cities while still offering low-cost accommodations.

Congressman Ritter to speak at convention

Pennsylvania Congressman Don Ritter will deliver the keynote address at the 1991 SBE Convention in Houston. Ritter, who is no stranger to engineering, is one of the few congressmen with any technical training. He is also the only member of Congress who holds a doctoral degree in an engineering field. Ritter is a member of the House Telecommunications and Finance Subcommittee, which makes him well-qualified to speak on broadcast issues.

Ennes Foundation receives donation

The SBE Ennes Foundation announced the receipt of a $1,000 donation from Intertec Publishing, the Kansas-based company that publishes *Broadcast Engineering* and *Video Systems* magazines. The donation will be used to support the foundation's educational programs. Current Ennes projects include the SBE certification program, scholarship awards and intern and cooperative work programs. Another on-going educational activity is the presentation of workshops at the SBE Convention.

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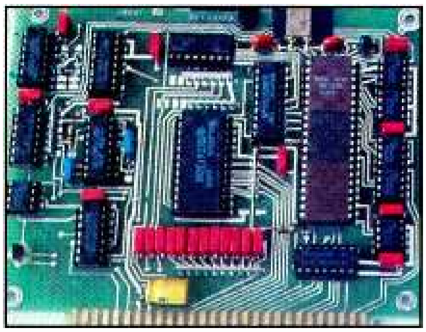


Building with microcontrollers

Z-8 as a remote control

By Gerry Kaufhold II

This article will discuss the Z-8 as a remote-control element for a 3/4-inch tape machine.

Many machines' back panels have one big and one small remote-control connector. On the larger connectors, each pin commands one function by sensing a relay closure. The simple Z-8 microcontroller described in Parts 1-10 of this series can easily drive such a relay-closure type of remote control. The smaller DB-style connectors usually operate on an RS-422 serial data communications loop. Although this connector provides a more versatile interface, the communications format and the details of the command sequences are often proprietary information.

Making remote connections

Obtain a connection diagram of the machine you wish to control. Note that some pins are for commands and others are for status.

The commands are inputs to the VTR. A change in a voltage level will cause the tape machine to take an action, such as going into the play mode or fast forward. The documentation will indicate if these pins are asserted by a logic high or low.

The status pins are outputs from the VTR. In many machines, these pins are passive unless the machine's status input is asserted. When the machine reaches a given state, say thread end, it asserts the appropriate status pin, but only if a "request for status" has been asserted.

The key to keeping the programming simple is to use a parameter list.

The Z-8 samples the signals on these pins to determine what the VTR is doing. For example, it is important that the Z-8 be able to verify that the record mode has been entered following the play/record command. To accomplish this, the Z-8 program outputs a 1 to the status pin, and then reads the current status level (either a 1 or a 0) from the record status pin.

A personal computer makes a good control panel for the Z-8. A simple BASIC language program is all that is needed at the computer.

Going with the flow

A personal computer makes a good control panel for the Z-8. The computer can control the Z-8 by connecting the Z-8's RS-232 interface to the computer's COM1: port. The Z-8's port 2 and port 3 I/O bits can then implement the "relay closure" remote control.

A simple BASIC language program is all that is needed at the computer. The first part of the BASIC should assign COM1: as the serial I/O port, and configure it to talk to the Z-8 card. Then give each of the desired VTR functions a 2-digit number, ranging from 01 to 26. To control the VTR, the BASIC program will output the appropriate number code to COM1:. The Z-8 will then close the appropriate relay on the VTR control connector.

The key to keeping the programming simple is to use a parameter list. When the operator of the personal computer requests the play mode, the BASIC program can use the 2-digit number for play mode as a pointer to several characters of data that are stored in the parameter list.

The first character of the parameter list is a 1. (Later on, this character can be used to select a given VTR when controlling more than one.) The second character of the parameter list is the I/O port number of the Z-8 that will be used for output. The third character is the bit that is to be controlled on the Z-8. The fourth character tells whether to assert the I/O bit (make it a 1) or negate it (make it a 0).

If the fifth character is a 0, then the command terminates. Otherwise, the fifth character indicates how long to wait before accessing a status pin of the VTR. The sixth character identifies the Z-8 I/O port and the seventh character identifies the bit of the I/O port that will be read for the status. The eighth character will be a 0, to indicate the end of this command string.

After locating the parameter list, the BASIC program outputs the string to the Z-8. If the parameter list was an output-only function (indicated by a 0 in the fifth character position), that is all that happens. If the fifth character is not a 0, then the BASIC program must monitor the Z-8 communications channel and wait for reception of the status information.

The assembly language program on the Z-8 can be simple. First, it must decode each character as it is received from the PC. The Z-8 program then selects the appropriate I/O port and the I/O bit of the selected port, and then toggles the selected bit high or low.

If a status input is to be read, a short wait loop should be performed before reading the selected I/O port. The bit can then be translated into an ASCII character and passed through the UART to the personal computer.

It is easy to apply the Z-8 microcontroller "core" to a wide variety of remote-control interfaces.

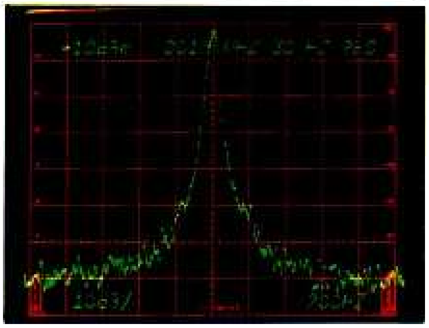
No keypad needed

The simplicity of this circuit makes it easy to apply the Z-8 microcontroller "core" to a wide variety of remote-control interfaces. By putting the intelligence of the program on the PC, the Z-8 can become a universal PC-to-anything interface.

Kaufhold is an independent consulting engineer in Phoenix.

DAT maintenance

Symptoms and solutions



By Richard Maddox

Most of the problems that engineers encounter with DAT machines are caused by mechanical, rather than electronic, malfunctions. But many of these same problems are caused by the tape, not the machine itself. Tape shedding, poor mechanical construction of the cassette shell, low RF levels recorded on the tape and uneven magnetic coating, for instance, can all cause the two most common DAT problems: audio dropouts (muting) and noise spikes (digital glitches). Troubleshooting can be difficult because of this. But armed with the proper test tapes, test gear and procedures, a good technician can find the solution.

Head drum issues

The first troubleshooting step whenever dropouts or noise spikes occur is to clean the head drum and capstan. Clean the heads manually using a light side-to-side motion with a chamois or deerskin cloth dipped in head-cleaning fluid or ethyl alcohol, or use a "wet" DAT cleaning tape. If a dry cleaning tape is used, reclogging or permanent head tip damage could occur.

The first troubleshooting step whenever dropouts or noise spikes occur is to clean the head drum and capstan.

After cleaning, find a good tape and play it on the machine. If the head had been clogged, it probably caused the playback problems. On a 2-head machine, however, recently recorded tapes may have high error counts if one or both heads were clogged. These tapes should be checked for playback problems on a properly functioning machine.

If the problems clear up for a short time and then reappear, the head drum probably needs replacement. Typically, head drums need to be replaced between 750 and 2,000 hours of use, depending on the model.

If the DAT machine does not have an elapsed time meter, you may want to install one when you change the head. Curtis Instruments (Mt. Kisco, NY) offers an LCD elapsed time readout that can display head record/play usage as accurately as one-tenth of an hour. The time meter can be easily interfaced with any DAT deck that has a play LED indicator. The meter is driven from the play LED driver logic signal, and uses +5V from the DAT's power supply for the display.

Maddox is technical manager at Media Management Associates, Lynnwood, WA.

Any binding or sluggishness in the guides that load the tape may prevent them from falling out of the way when the tape is unloaded.

Tape eating

Any binding or sluggishness in the guides that load the tape may prevent them from falling out of the way when the tape is unloaded. This causes the machine to "eat" the tape. When a tape is eaten, an inch or two of tape hangs out of the DAT shell, and then is crunched by the lid as it closes when the tape is ejected.

The eaten tape is not playable, unless it is damaged only at the head or tail of the tape, which can be removed and the remaining tape spliced to the clear leader. Therefore, it is a good idea to rewind the tape to the head (or fast forward to the tail) before unloading it. You should also leave 30 to 60 seconds of blank space at the head of a tape when recording, just in case this happens.

After a tape is eaten, you must find and clean the sluggish guides. In order to remove and clean the guides' mounting posts, the transport must be disassembled. The guide that most often exhibits a problem is the one next to the capstan, so the capstan must often be removed as well. Fortunately, no alignments need to be made because the capstan assembly is screwed into the transport chassis.

Worn or degraded brake pads or tension bands on the take-up and supply reel table also cause the machine to eat tapes.

Worn or degraded brake pads or tension bands on the take-up and supply reel table also cause the machine to eat tapes. When the tape is loaded, one reel is left freewheeling to allow the tape to be pulled out, while the other reel is held in place by its motor or brake. If the brake pad on the freewheeling reel sticks to the reel table, the tape does not have enough slack to be properly loaded, and will be pulled up and over the slant guides. This problem occurs most often during the first loading of the day, after the deck has been inactive overnight in a cool studio.

If this happens, the machine senses a loading problem because there has been no reel table movement. The machine will then unload the tape, and attempt to take up the slack by momentarily hitting the fast forward and rewind modes. If the brake pads are still stuck to the reel table, the tape will not go back into the shell, which means it will be eaten when it is ejected. To prevent this problem from reoccurring, the brake pads should be replaced.

Worn out pinch rollers can also cause the tape to stick or slip during play. DAT pinch rollers wear faster than head drums, and usually need replacement after a year or so because plastic binder and other tape elements build up, making the roller's surface hard and slippery. Because pinch rollers generally cost less than $20 and are easy to obtain, they should be replaced frequently.

Next month's column will cover DAT part replacement. Find out which parts wear out and and how much it will cost to replace them.

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Part 3

Management for engineers

Time management

Protecting your time

By Judith E.A. Perkinson

Last Monday, I had to finish a report. Although I only needed about four hours to complete it, I scheduled nothing else for the day. By 8:00 that night, all of my coworkers had left and I was still sitting in my office writing the report. When I finally got home, I asked myself why this had happened.

Here is what I discovered. Almost everyone I had called during the past two weeks called me back. Co-workers constantly came into my office to ask me questions. I had to respond immediately to two items in the mail and my secretary asked me to help her with a computer glitch. Three people dropped by to chat. Lunch took longer than it should have and my husband called me at least six times.

I had allowed every time thief in the book to steal my day. The price I paid was working until 10:00 that night to finish the report that should have been completed by noon. I had no one to blame but myself.

It is important to understand that I didn't waste my time. There is a great difference between wasting time and allowing it to be stolen.

In order to start protecting your time, you must:

- Understand how it is taken away from you.
- Believe you have a right to protect it.
- Develop techniques to protect it without offending the thieves.

Types of time thieves

We all agree to do tasks when we shouldn't. (See "Management for Engineers," April 1991). It is a hard habit to break. But you must break it because it is a treacherous time thief.

Most time thieves are not malicious. Rather, they are just people acting on their own needs and schedules.

Phones. Most people cannot let a phone call go unanswered. People make phone calls at their convenience. Unfortunately, those on the other end of the receiver are usually being interrupted. If you don't want the phone to disturb you forward your calls, have your secretary hold them, turn on your answering machine or just unplug the phone.

Remember, not answering your phone is not a crime. If the call is truly important, the caller will reach you.

Interruptions. Like phone calls, interruptions are the products of other people's needs. You broadcast to your co-workers how interruptible you are. If you want them to stop interrupting you, you must send the right signals. To do this:

- Close the door.
- Don't look up.
- Don't make eye contact.
- Put up a "Do Not Disturb" sign.
- If they are persistent, tell them you will handle it later.

These are all ways of building an invisible wall around you that says you don't want to be disturbed.

The key to protecting your time is knowing which situations are important and which are only a waste of your time.

You may occasionally run into someone who simply does not respect any of your signals. When this happens, you must explain to this person that you do not mean to be unresponsive, but you will not tolerate constant interruptions.

Your right to protect yourself

Being needed makes you feel important. Working until 10:00 at night makes you feel used. Protecting your time does not make you less important, but it will make you a better manager.

Before you can effectively use any of these techniques, you must believe that you have a right to protect your time.

The foundation of this conviction is a belief that:

- What you do is important.
- Your time is as valuable as the time of anyone who interrupts you.
- Being unavailable is not a crime.
- Few things are so important that they cannot wait. You will need this conviction to successfully protect your time.

Protection without offense

It is natural for people to care more about their needs than yours. When you implement techniques to protect your time, you will probably find that some people are not happy with your new-found protection.

Their reactions will vary from quiet disappointment to open hostility. Much of the reaction is simply the frustration people feel because their needs are not being met. Many of these time-protection signals are the same signals used to broadcast conflict or rejection.

Therefore, it is important that you tell these people that you are not rejecting them. They will be less resistive if they understand that their requests are having a negative impact on you.

You must also give them some estimate of when their needs will be met. To do this, you can leave a message on your answering machine telling them when you will be available or hang a sign on your door indicating when you can be contacted.

You cannot expect the time thieves to change overnight. For a while, you may need to be stern. But most important, don't send mixed signals. Mixed signals only broadcast that you will relent if they are persistent.

Unavoidable time thieves

There will always be circumstances that take you away from your planned schedule. Most of these are legitimate and unavoidable. The key to protecting your time is knowing which situations are important and which are only a waste of your time.

Of course, there will be times when you will slip, like I did last Monday. When this happens, you must determine how much responsibility you must bear for allowing the theft of your time. Remember, the ability to control your time is an essential managerial skill.

Next month, I'll deal with a subject that is vital to everyone — procrastination. It's guaranteed to help you.

Perkinson is senior member, the Calumet Group, Inc., Hammond, IN.

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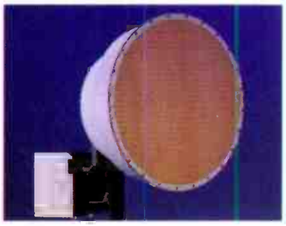

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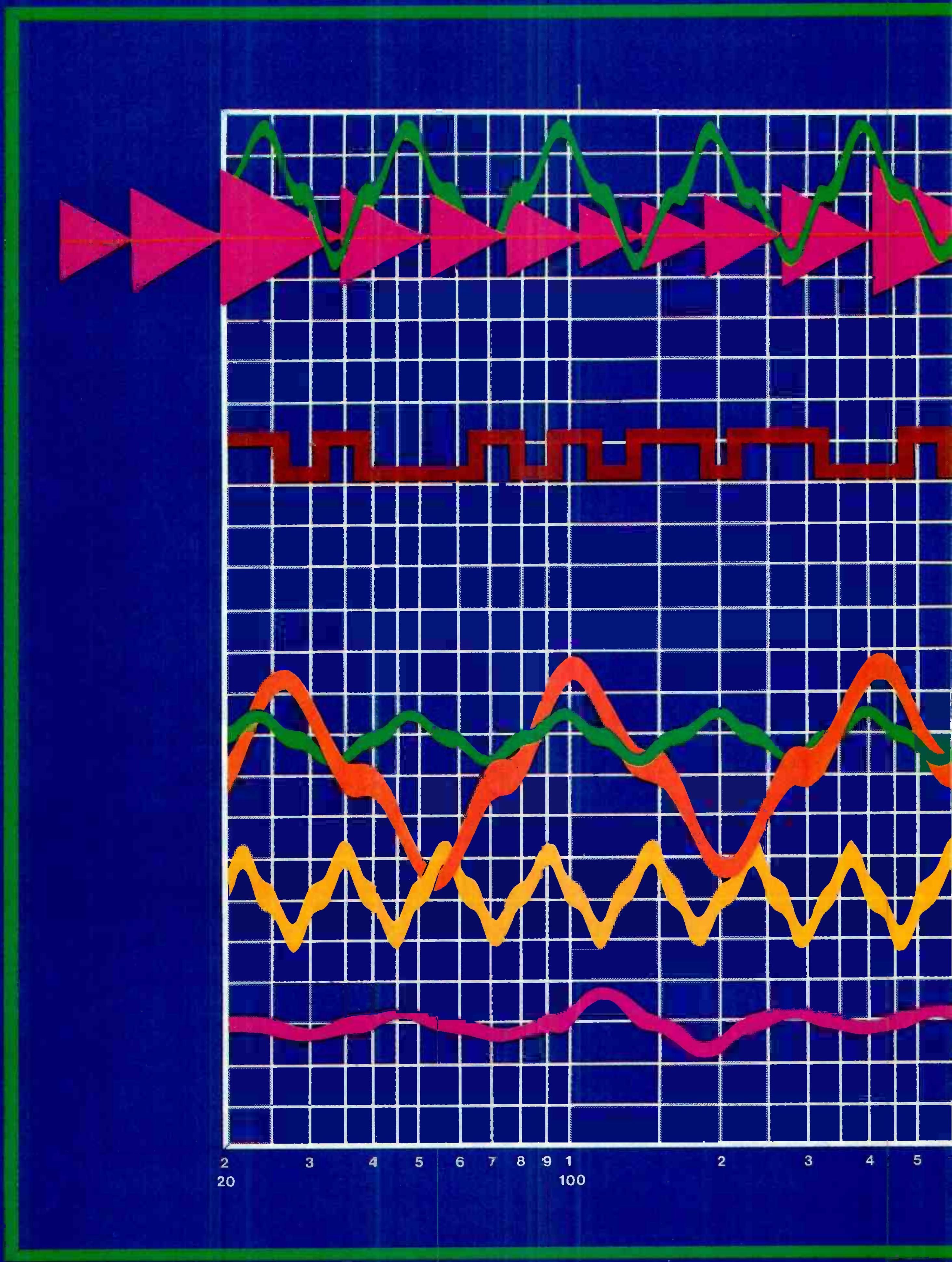
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Audio technology update

It's not just coming out of a 4-inch speaker anymore.

Come to think of it, even 4-inch speakers sound pretty good these days. And with the advent of the "personal stereo," high-end car audio systems and stereo/surround TV audio, radio and TV audiences are hearing broadcast programming with better audio quality. Thanks to digital technology, an audio revolution is under way, and it is showing little sign of abating. As a result, those audiences' aural tastes continue to move forward. Broadcasters can join in the parade, or stand still and be trampled by it.

The digital revolution also means that consumer systems aren't the limiting factor anymore. There was a time when broadcast facilities led the way in audio quality, and consumers lagged far behind. Not so today — it is often the broadcaster who is trying to catch up.

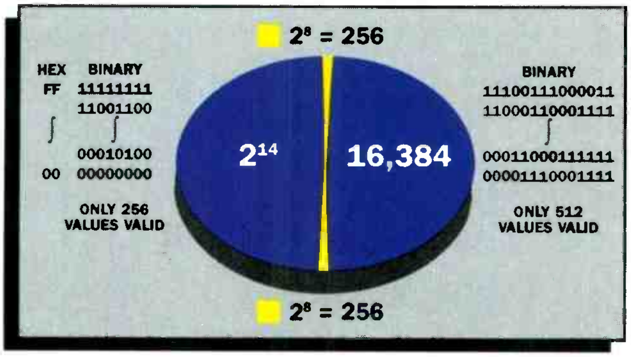
In the TV world, audio has always been the poor relation. Although video equipment costs more, and producing the visual part of a program requires a larger staff and budget, psychologists claim that more brain power goes into processing *sound* than visual images. After all, we only see an oval-shaped area in front of us, but we hear from all around. Our range of perception in the frequency domain covers approximately *10 octaves* for sound, while the visible light range is slightly *less than one octave*. In digital systems 16 bits of resolution are arguably required for quality audio, but eight bits may be enough for video (a 256:1 ratio). On a pure communication level, in most cases, consider which conveys more information from a TV broadcast: turning off the sound and leaving the picture or vice versa?

As with art, the audience doesn't know much about what they are hearing, but they know what they like. In one recent test where the same video was played first with a mono and then with a stereo soundtrack, the audience overwhelmingly responded that "the *picture* was better" on the second (stereo) sample. We are a visually oriented society. When we understand something we say, "I see." A big idea is called a "vision," and so forth. It's no wonder that aural issues are misunderstood or get short shrift. So it's up to broadcast professionals to do their best in keeping audio priority and quality high, and that's what this month's features are all about. Today and in the future, good sound implies *digital*, so we will present digital techniques that can be applied to audio processing, measurement, routing and broadcast transmission.

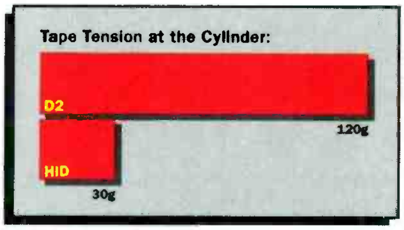
- "Digital Radio: The First Five Years" . page 26
- "Testing Digital Audio Devices" 38
- "Digital Audio Processing" 46
- "Digital Audio Signal Distribution" 54

Skip Pizzi, technical editor

DIGITAL TH



ROBUST 8-14 CHANNEL CODING



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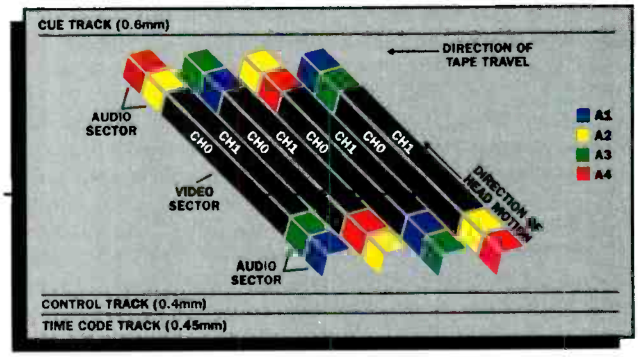

AUDIO EDITING AS IT SHOULD BE

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Digital radio: the first five years

A look at where things stand today, and how they got there.

By Skip Pizzi, technical editor

Digital radio made its debut Aug. 1, 1986, when WGBH-FM, Boston, began simulcasting its programming in the EIAJ pseudovideo PCM (F-1) format over its sister station WGBX-TV. It didn't appear to be such a big step at the time, and to no one's surprise, it worked. EIAJ format converters had been in use for several years as a way to record digital audio onto videotape. The converters took stereo analog audio in, sampled and quantized it into a 16-bit PCM datastream, and then encoded that onto monochrome NTSC (or PAL) video frames. The converter's video output plugged into any VCR's video input, so why couldn't it also be broadcast through the visual section of a TV transmitter? The signal was decoded back to analog audio at the receive end by another EIAJ converter fed by the video output of a VCR or monitor/receiver tuned to Channel 44, WGBX's operating frequency.

Although it was soon acknowledged as a milestone, the experiment also revealed the chief failing of digital audio systems of the day — they took up a *lot* of RF spectrum. No one expected much success from a system that used the better part of a TV channel's bandwidth to deliver two channels of audio, no matter how good they sounded. However, it was a necessary first step, and an appropriate one in the days of linear PCM.

Enter Eureka Project 147/DAB (digital audio broadcasting). Its first on-air demonstration in September 1988 showed that spectrum requirements might not be an insurmountable hurdle. Here was a way to squeeze digital broadcasts of audio and data signals into the types of channel bandwidths broadcasters were accustomed to with FM service. The multipath elimination and on-channel boosters that the system offered seemed almost too good to be true.

Broadcasters probably heard for the first time the terms *source coder* and *channel coder*, and realized the depth of development that the European Eureka consortium had undertaken.

However, as U.S. broadcasters began to consider the application of such a system, which was developed for European use terrestrially and via satellite, problems became evident. The Eureka system, although efficient, still needed a new spectrum. Where would it come from? Economic concerns also surfaced. Who would be the licensees of new digital radio stations? And what role would satellites play? (The Eureka system had originally been developed to reduce power requirements and improve performance for hi-fi audio delivery via satellite.)

As the NAB and others embraced the Eureka format, alternative systems began to surface. Another new term, *in-band*, was introduced, describing formats that

Eureka 147/DAB antenna (left) and FM comparison channel antenna on the roof of the Las Vegas Hilton for the first U.S. on-air demonstration during NAB '91.

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would place digital radio service within the existing AM and FM broadcast bands. Meanwhile, the 1992 World Administrative Radio Conference (WARC) meeting — where international spectrum agreements for digital radio will be hammered out — rapidly approaches.

That brief history of digital radio indicates why the early 1990s could be regarded in future years as the dawn of digital radio broadcasting. That history is similar to today's reflections on the early 1950s for FM and the early 1930s for AM. At present, however, it is hard to say how the end of this story will read.

Format wars

At the 1991 NAB Convention in Las Vegas, digital radio's *formats du jour* were examined. Of the eight digital radio format proposals announced, two require new spectrum, and the remainder use existing radio broadcast spectrum in one form or another. These in-band systems are further distinguished by whether they put a station's digital signal within its own current channel (in-band/on-channel [IBOC]), or if they use other unassigned frequencies within the AM and FM bands (the so-called "interstitial" approach). In either case, no new band of spectrum is required, but only in the IBOC case would no new *allocation* be required.

Although each system is different, they do share some common elements. Each system uses a *source coder* for audio data compression, and a *channel coder* for preparation of the signal for modulation onto RF carriers. Today's source coders typically take a 16-bit linear PCM stereo audio datastream (requiring a path with a data rate of 1—1.5Mbit/s), and reduce its data rate to approximately 256kbit/s. This is achieved by eliminating redundant data (numerical compression), by coding only the *change* in data between samples (delta-modulation), and most recently, by incorporating psychoacoustical elements (perceptual coding).

Perceptual coding

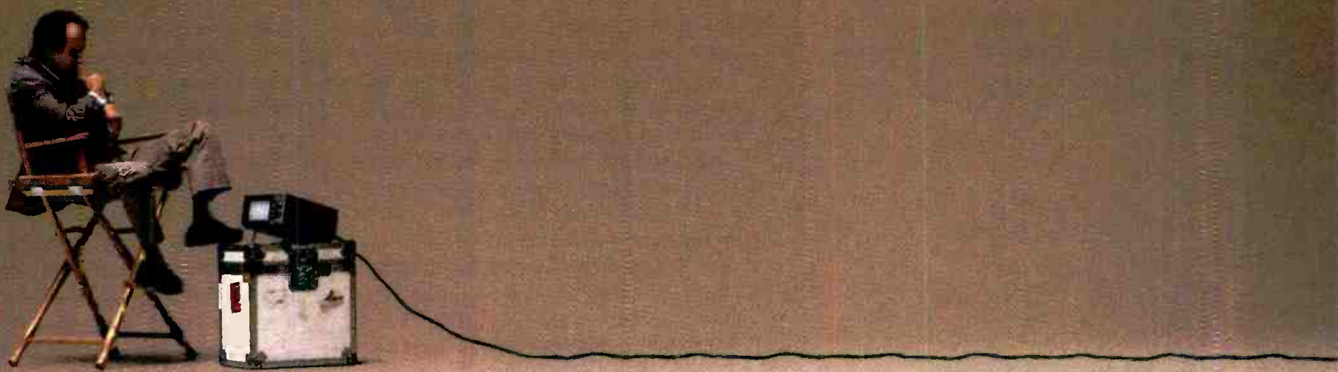
Because a digital audio signal's data rate is a product of sampling rate multiplied by data word length (number of bits), reducing either element will reduce the overall data rate. Dropping the sampling frequency carries the necessary consequence of reducing audio bandwidth, according to the Nyquist theorem, which states that the sampling frequency must be at least twice the highest audio frequency desired. In theory, cutting the word length reduces dynamic range, primarily by increasing the noise floor 6dB for every bit removed. This is where psychoacoustics comes in.

The studies of human hearing perception that psychoacoustics undertakes have shown that audio program signals can mask (render inaudible) lower-level interfering signals (noise) that fall near in frequency or time to the program signals. (See "Digital Radio: Promise and Perils," December 1990.) So source coders hold sampling frequency constant, preserving audio frequency response, but dynamically reducing resolution — based on an assessment of the current signal conditions — to as low as possible, while still keeping quantization noise inaudible because of masking by the program audio.

Two methods are used for this assessment and bit allocation process: subband coders and transform coders. Each offers its own advantages. Subband coders break the signal into small spectral chunks and analyze each independently in the amplitude domain. Transform coders run blocks of audio through an FFT or other transform into the frequency domain, and perform their analysis. In either case, psychoacoustically based algorithms dictate the bit allocations after considering this signal analysis. The only significant difference between coder types tangible to broadcasters is the slightly longer throughput time required by transform coders. (In some cases, it is long enough to be disturbing to a live announcer monitoring off-air.)

Source coders are the subject of some heated debate within the audio industry, and several national and international "shootouts" have been held in an (as yet) unsuccessful attempt to designate a standard system. Additional tests are scheduled for later this year. More specific decisions are expected from these, rather than a single global standard. For example, one upcoming test will subdivide decisions on the basis of how many compression and expansion cycles (transcodings) can be tolerated. The thinking here is that original recording or gathering of sounds might experience several more transcodings, while distribution of signals from a central (network) source to multiple broadcast outlets might still see one or two more generations, while actual broadcast algorithms need not stand up to further transcodings downstream. Another distinction is how low a bit rate is required for the application. Digital radio proponents seem satisfied with the current minima of 128kbit/s/channel codecs, while telephone operations are pushing for 64kbit/s systems to fit their existing data networks. Speech vs. music quality is also a factor.

The solution may be a multirate or variable-compression coder, although nothing of the kind has been introduced by any manufacturer. One recent advance, however, is the acknowledgment of the amount of redundant information between the two channels of a typical stereo signal. The next generation of one popular codec system will incorporate this



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redundancy to allow what is claimed to be wideband, near-CD quality at 128kbit/s or 192kbit/s for *stereo* (equivalent to 64kbit/s or 96kbit/s/channel). So far, this level of compression has been unobtainable without audible degradation. However, it has only been previously attempted with 64kbit/s coders operating independently on each channel of a stereo pair.

Channel coding

Digital radio systems route compressed digital audio from a source coder to a channel coder for restructuring of the datastream into a form suitable for robust transmission. Although digital radio formats share a fairly common approach to source coding, they vary more widely in their treatment of channel coding. However, one theoretical element is shared to some extent among all proponents — the *diversity* concept.

Combating the flat and frequency-selective fades caused by multipath conditions at the receive antenna is the primary challenge that digital radio systems seek to overcome. (Flat or "stoplight" fades are those that occur from a strong, single, specular reflection that completely cancels an entire FM channel's bandwidth; more common are the narrower comb-filter-like notches of frequency selective fades.) Most digital radio channel coders use some form of diversity to solve this problem. The simplest approach is frequency diversity, in which source coder output data is encoded on several carriers that appear at widely spaced locations across the receiver's channel spectrum. In this way, if a fade is experienced, it cannot wipe out every iteration of the data, because the same data appears elsewhere in the spectrum. A refinement of this concept is called *adaptive equalization*, whereby the receiver learns about the multipath conditions of the moment from a training sequence at the beginning of each transmitted block of data, and adjusts its receive sensitivity across its channel spectrum accordingly.

Time domain diversity can also be used under the assumption that some multipath conditions (those caused by moving reflectors or experienced by mobile receive antennas) are changing over time, and that the portion of the spectrum where cancelation occurs at one moment will not be canceled shortly thereafter.

Finally, space diversity may be used to create multiple time windows for the receiver by placing two or more receive antennas at different places, and allowing the receiver to select the better signal at the moment. This assumes the antennas are properly spaced, so that a fade at one point on the spectrum at one antenna has a minimum likelihood of occurring at the same point on the spectrum at the other

Digital radio formats compared

Note: "Advantages" are claimed by developers; "Disadvantages" are reactions from observers in the industry.

Acorn DAB (FM)

Description: *In-band, on-channel system encodes and QPSK-modulates data onto 21 (or more) carriers across approximately 200kHz, encrypted 30dB below FM signal, with 1.6bit/s/Hz efficiency. Frequency reuse module allows FM and digital carriers to occupy same channel without interference.*
Source coder: MUSICAM
Channel coder: CPVDM (coded polyvector digital modulation)
Spectrum used: Existing FM channels
Current status: Prototype hardware, first public demos held 4/91.
Developers: USA Digital Radio (Gannett, CBS, Group W, Stanford Research Institute, Corporate Computer Systems), Los Angeles
Advantages: No new spectrum or allocations, and minimal new transmission hardware required; highly power efficient; easy implementation
Disadvantages: No on-channel boosters or gap-fillers possible; may not be as multipath-resistant as wideband systems.

ADR DAB

Description: *In-band FM system, using multiple carriers and dispersing data from multiple stations across them. Three to 10 stations will share a transmitter and antenna. Operates independent of present FM stations, on multiple adjacent channels. Design philosophy foresees incremental replacement of FM stations in large markets to free spectrum. Existing AM and FM stations will be accommodated on new system, which will operate within 88-108MHz.*
Source coder: Unspecified; possibly ASPEC
Channel coder: ADR
Spectrum used: FM band
Current status: Paper; first demo projected for NAB '92
Developers: American Digital Radio, Haddon Heights, NJ
Advantages: Provides multipath resistance of wideband system without requiring new spectrum; incorporates AM stations in some markets immediately.
Disadvantages: Requires new transmission hardware; requires common transmitter/antenna approach; potentially cumbersome transition in larger markets.

Digital FM-S

Description: *In-band, on-channel system using four FM subcarriers on standard FM baseband for data transmission, and an active diversity antenna system to combat multipath.*
Source coder: Unspecified, pending development of 128kbit/s stereo compression algorithms by several manufacturers.
Channel coder: FM-S
Spectrum used: Existing FM channel
Current status: Paper; first public demos projected 2Q93
Developers: Synetcom Digital, Hermosa Beach, CA; Radix Technologies, San Jose, CA
Advantages: On-channel system that requires no authorization; operates under existing FM SCA regulations and emission designations for mono-FM operation; new stereo generator is only station hardware required to implement; could be compatible to existing composite STLs.
Disadvantages: Requires FM mono operation and eliminates most existing SCAs (FM stereo operation possible, but not under current regulations); uses complex receive antenna system; needs twice the audio data compression of most other systems.

Eureka 147/DAB

Description: *Uses hundreds of 15kHz-wide, orthogonally spaced carriers to carry frequency- and time-interleaved data from multiple data-compressed audio channels. (Current plan for U.S. implementation puts six stereo signals on 1.5MHz.) DQPSK modulation employed, with guard intervals between symbols to reduce intersymbol interference (ISI). Convolutional coding and Viterbi maximum-likelihood decoding provides error correction.*
Source coder: MUSICAM
Channel coder: COFDM (coded orthogonal frequency-division multiplex)
Spectrum used: North American proposals range from approximately 50-75MHz, L-band
Current status: Second-generation hardware
Developers: Eureka Partners (EBU, CCETT, IRT); Eureka Project office (U.S.), Darien, CT
Advantages: Allows on-channel boosters; highly power and spectrum efficient; multipath immune.
Disadvantages: Requires new spectrum and new transmission equipment; requires common transmission point for groups of stations; may require gap-fillers in urban areas.

LinCom DAB (FM)

Description: *This in-band system has only been vaguely explained to date, but uses frequency diversity and DMSK modulation with 1bit/s/Hz channel coding of 10:1 compressed data. Output fits within an existing 200kHz FM channel, but fully occupies the channel (not an on-channel system). No mention is made of specific adjacency, interference or other transitional and applicational issues.*
Source coder: Proprietary
Channel coder: Proprietary
Spectrum used: 200kHz per station, within FM band
Current status: Paper; 2Q93 projected for first public demos
Developers: LinCom Corporation, Los Angeles
Advantages: Unknown
Disadvantages: Unknown

MFM (multifrequency modulation)

Description: *An existing FM station's audio data is modulated on 192 orthogonally spaced 1kHz-wide carriers and broadcast on a first-adjacent FM channel from the same tower used by the station's FM antenna, at 16dB below the FM signal level (1/40th power). Uses trellis coding for high efficiency.*
Source coder: Not specified; can accommodate any 256kbit/s stereo coder
Channel coder: MFM
Spectrum used: 200kHz per station, within FM band, on first-adjacent channels, or elsewhere in VHF spectrum
Current status: Paper; first public demos projected for 4Q91 or 1Q92
Developers: Mercury Digital Communications, Monterey, CA
Advantages: Extremely high data efficiency (roughly twice that of Eureka 147), with good mobile reception characteristics; can carry up to six 16kbit/s auxiliary data channels.
Disadvantages: In-band — but not on-channel — system, requiring second FM transmitter (low power) and antenna; may not be as multipath-resistant as wideband systems; on-channel boosters may not be possible.

Power multiplexing

Description: *An in-band, on-channel system in which the digital carrier is mixed with the existing FM carrier at a lower level. Standard FM receivers capture only the FM carrier, while in new receivers, a hard limiter demodulates the dominant (FM) carrier, sending it through a PLL and into an auxiliary loop, in which a polarity-reversed image of that carrier is generated. A second hard limiter receives the original signal and the pol-rev carrier, canceling the FM carrier and leaving only the lower-level digital signal for decoding.*
Source coder: Unannounced
Channel coder: Proprietary
Spectrum used: Existing FM channels
Current status: Prototype hardware completed
Developers: Kintel Technologies, San Jose, CA
Advantages: No new spectrum or allocations required; three or possibly more carriers (of various modulation types) may be stacked on the same channel; extremely power efficient with simple receiver design.
Disadvantages: Fully compatible on-channel boosters or gap-fillers may not be possible; may not be as multipath-resistant as wideband systems.

Stanford Telecom DAB

Description: *Uses frequency hopping of QPSK-modulated, orthogonally spaced carriers (typically 268.5kHz carrier bandwidth with 2ms, 12-hop cycle across 3.5MHz channel) plus Viterbi coding, interleaving and adaptive equalization to fight multipath and ISI.*
Source coder: Dolby AC-2
Channel coder: D-SCPC (dynamic single-channel-per-carrier)
Spectrum used:20-38MHz, L-band
Current status: Fully developed on paper; first public demo 4Q91
Developers: Stanford Telecommunications, Santa Clara, CA
Advantages: Frequency hopping requires only one carrier to be demodulated by receiver at any time; designed for high channel capacity with low power, from satellite and terrestrial sources; multipath-immune, allowing on-channel boosters.
Disadvantages: Requires new spectrum and equipment; may require urban repeaters; requires terrestrial broadcast at L-band and from common transmission points.

antenna(s). Some formats could allow *transmit* space diversity, where the transmitted signal emanates from two or more antennas, and a single receive antenna recovers the least canceled signal. Because either of the space diversity solutions involves additional hardware, larger space and more complex installation, most proponents are attempting to develop their systems without relying on them.

Channel coding has significant impact on receiver complexity, but this area has yet to be fully explored. Consumer electronics manufacturers have not been greatly involved in digital radio discussions, although several have kept a close eye on the issue. Existing services' interference to and from digital radio will probably become another high-priority topic as this dialogue is engaged.

Most digital radio formats also make provision for auxiliary data transmission, making these systems far more than digital audio broadcast systems. Anticipated applications include those now performed by FM SCAs and the RDS system, plus still-video ("radio with pictures"), commercial/promotional or public service messages in text or graphics, artist information for the musical selection currently on-air, background text or bibliographies for news or public-affairs programs and so on. Receiver makers and broadcasters may find these features useful in marketing future products and services.

Economic and policy issues

Concerns in the broadcast industry regarding digital radio are not focused on technical, but on ancillary issues affecting the viability of such service and the fiscal context within which it must fit. Systems that require new spectrum are especially problematic. The following issues are currently under consideration:

- If new channels must be allocated (even under an in-band but not on-channel scenario), licensing issues must be considered. Will all existing broadcasters be granted a new digital channel? Will spectrum fees be levied? Will the marketplace be opened to new competition? Will auctions or lotteries be used?

- The only likely spectrum areas for digital radio service are in the 1,500 (L-band) and 2,400MHz (S-band) ranges. (Another possibility, the UHF TV spectrum, is considered unavailable because of needs for advanced TV simulcast spectrum.) Even with the power efficiency of digital radio systems, broadcasting in either of these bands will require high-power transmitters and/or a large number of small gap-filler transmitters approaching a cellular situation. Beyond power efficiency, keeping operating frequency low is also advantageous for terrestrial applications of digital radio systems in terms of reduced Doppler shift under mobile reception, and increased benefit from constructive echoes. The sole advantage of higher-frequency application is a slight reduction in man-made noise.

- Of those two potential bands, only 1,500MHz is considered to be barely within the limits of feasibility for digital radio broadcasts. However, this band is currently in use by the U.S. military (and others) for aeronautical testing. The Air Force has filed comments citing the band's use for the development of the recently renowned Patriot and Tomahawk missiles and the F-117 Stealth fighter, and has served notice that they will not give up the band without a fight. Meanwhile, the International Association of Broadcasting has endorsed the use of this band for digital radio.

- Exactly how much power is required for digital radio systems is also under debate. Are the F(50,50) criteria used for analog broadcasts appropriate? Or does digital radio's rapid degradation in the fringe or under poor reception conditions mandate a higher figure, such as F(90,90) or even F(99,99)? If so, the power levels required increase dramatically. On the other hand, a "graceful failure" function (degrading slowly to noise) is now being included in some system proposals.

- Based on recent official and unofficial comments from FCC staff, it is likely that digital *satellite* radio broadcasting in the United States will become a reality within the next decade (as it already has in Japan). Furthermore, it will probably end up in one of these same parts of the RF spectrum. If terrestrial and satellite digital radio were to coexist in a common band, manufacturers could provide a single consumer receiver to tune in both services. This would put terrestrial broadcasters in the ironic position of helping satellite services to penetrate the marketplace with receivers, which has long been a primary impediment to any DBS system's success.

- So-called wideband systems require several stations' signals to be bundled into a single datastream and transmitted on a common multiplexed carrier set. This will require either unprecedented cooperation among broadcasters or a new common carrier type of operation (which will expect to be paid for its services) to handle the multistation transmission system. In either case, a station will probably no longer have unilateral control of its transmission hardware.

- Any new allocation to an existing broadcaster may involve some dislocation from its present coverage area. Depending on the specific situation, this may or may not be advantageous to a station.

- Regarding the cost/benefit ratio of transition to digital radio, existing broadcasters must realize that digital transmission would only be an enhancement to their service, not new service *per se*, at least in the short- to mid-term. It will generate no new revenue until a sufficient number of receivers exist in the marketplace to make viable the termination of simulcasting, and the operation of the digital channel as a separate service, as in the AM-FM transition. That model notwithstanding, it is unlikely that many broadcast owners will have the foresight and patience (let alone the risk capital) to invest heavily in any new service that will not begin to pay off for several years (if at all). The less costly a system is to implement, the more likely stations are to take a chance and set rolling the snowball that ensures a new format's success. Conversely, if only a few stations institute the service, there will be little incentive to listeners to buy digital radios. Therefore, the format will fail to reach the critical mass required.

- A scenario that frightens some broadcasters is one in which all existing AM and FM broadcasters receive equal weight in the allocation of new digital channels. With a completely leveled playing field, the advantage that successful FM and clear-channel AM stations hold over their competitors may be lost. The upheaval in station market values that this would cause has already sent some initial shock waves through the industry. This so-called *parity* issue will loom large in upcoming discussions of digital radio. Some feel the most pragmatic approach involves all stations moving forward in roughly proportional increments at about the same time, such that a significant benefit to the public interest is assured, but without disturbing the economic balance within the industry. Whether this methodology is adopted, or possible, remains to be seen.

In-band inroads

Regulators and some broadcasters have already taken note of these and other difficult issues barring the way of a digital radio service's establishment. They have also realized that many of those problems disappear if an in-band, on-channel (IBOC) system can be made viable. Such a solution is so appealing that some have expressed a willingness to even trade some of the technical benefits of a system that requires new spectrum for the ease in transition an IBOC approach engenders. There is even speculation about the Eureka 147

Continued on page 36

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Evaluating data-compression artifacts

By David Schwartz

Does it really take "golden ears" to hear data-compression artifacts in digital audio bit-rate reduction systems? No, not if you're willing to spend a little time training your hearing system. What must be learned is the "signature" of the compression noise. Because this is a new type of aural artifact for most broadcast engineers, it may not be apparent upon first listening to a compression system.

For ear-training purposes, select a music or voice recording that is familiar to you, and make a 10- to 20-second loop. (A CD player with A-B repeat is good enough, even with the recue time.) Listen repeatedly to the loop, alternating between the playback source's output and the data-compressor output. Try headphones and near-field monitors. Be patient. Listen in sessions that are separated by a few days. It may take awhile, but at some point, barring physiological hearing impairment, you will start to home in on the noise characteristic. Artifacts will vary from system to system, and between different types of program material or throughput levels.

Objective measurements

What about quantifiable benchmarks for these impairments? Given most broadcast facilities' budgets, test benches tend to be underequipped for the rigorous test procedures required. If you can afford it, a digital signal processor-based test system on a PC platform is excellent for testing a compression system. But you can get a good look at some compression artifacts with a 20MHz 2-channel scope and a reasonable audio signal generator.

First, calibrate your gear, and make sure your tone generator makes a clean sine wave. Look for symmetry and noise, especially at 15kHz or 20kHz. If there is a little "hair" on the sine wave from oscillator noise (at maximum scope gain), it is still all right. In fact, it will be interesting to see what the compression system does with that noise component.

Schwartz is senior member of the technical staff at Tandy Electronics Research Center, San Jose, CA.

If the compression system under test has a bypass mode, verify unity gain for the setup through it. Ground-loop or DC-offset problems should also be fixed before proceeding.

Noise spectra test

Start with a 10kHz test tone. (You can assume that 1kHz will look fine, because compression system designers almost always optimize that traditional test number.) Display the uncompressed 10kHz signal on one scope trace, and the compressed and decompressed signal on the other. You will need to use the horizontal trace position adjustment on the scope to align the two signals in time, because of the processing time delay of the compression system. Note what the time delay is; anything longer than a few milliseconds will require reworking of your air monitor signal path.

In a 16-bit system with a peak-to-peak signal of 2V, one bit's worth of audio signal resolution is approximately 30μV. If the noise is fattening the processed signal trace by a couple of bits' worth, it will be obvious. Turn the intensity up and take a close look at the processed trace. Ideally, the noise should be white (without any specific harmonic content). If so, the trace will be simply an undifferentiated band in the exact shape of the unprocessed trace, only thicker. Because the signal is spread by the noise, the processed trace will appear less intense.

If the compression system is introducing harmonic distortion, the processed sine wave will be modulated by the spurious harmonics. Sometimes this looks like a second sine wave of a higher frequency laying right on the peak of the test tone waveform. The triggering of the scope may make this form crawl on the test tone trace. This effect can be more pronounced with low-frequency test tones. It is not uncommon to see a combination of wideband noise products and harmonic distortion.

Boundary condition tests

Next, look at the system's limits or its boundary conditions. Boundaries of interest are idle-channel signal, low-level signal and clipping levels. To look at idle-channel effects, use the stereo bus output signal from your on-air mixer as the source, powered up, but with all faders down. On your scope, at maximum gain, compare this noise signal with its processed version. You may notice that the compression system has imposed a pattern on the noise, or there may be glitches every now and then. If so, the patternistic and/or "popcorn" noise may or may not be acceptable — it's a judgment call.

For low-level effects, look at processed and unprocessed tones. You should be looking for the small-signal performance of the compressor as it ramps up from idle to about idle plus 0.1V on the scope. Ideally, a sine wave should grow smoothly out of the noise floor. (Getting the scope to trigger properly for this test can take some setup time, but it is worth the trouble.) Does the processed signal pop up out of nowhere sometime after the unprocessed sine wave is on the screen? If so, low-level granularity may be a problem. Try this test at various input frequencies. If granularity is evident, listen to a long, slow fade on music to judge acceptability.

Performance of a compression system just below clipping will determine what practical system headroom will be. Because compression artifacts are sometimes non-linear with respect to frequency, you'll be looking for the maximum level of unclipped signal over a range of frequencies. For reference, drive the compression system into clipping at 1kHz, noting the level where the waveform starts to flatten. Then look at the 1kHz signal 6dB down (half amplitude) from clipping. Now see if the signal looks the same at 15kHz and 300Hz at this level as it does at 1kHz. You may find that usable headroom is much smaller at one or both ends than at 1kHz.

For more detailed analysis, a precision distortion analyzer and a spectrum analyzer should be considered. Some tests that will reveal more performance characteristics are continuous-tone frequency sweeps, THD+N, twin-tone response and single-tone full-spectrum plots.

Continued from page 32

team returning to the drawing board to devise an in-band implementation of their scheme. The historic precedent favors an IBOC method, because almost all prior U.S. broadcast format enhancements have been downwardly compatible, on-channel systems.

There are some disadvantages to an IBOC system, as presently configured. It is constrained to a relatively narrow channel bandwidth, therefore reducing its multipath immunity compared to more spread-spectrum techniques. (IBOC proponents claim that approximately 200kHz of channel bandwidth may be enough to combat most, if not all, FM multipath conditions. However, some have hinted at the possibility that a bit of spatial receive diversity may be required to completely solve the multipath puzzle.) Furthermore, on-channel boosters and spatial transmit diversity will probably not be accommodated in IBOC systems. Finally, although several of the FM in-band proponents listed here have also developed (or are developing) IBOC or other types of AM digital radio systems on paper, it seems likely that at least some AM channels will need to be relocated, perhaps in the expanded AM band.

Whether such a system can be fashioned, or whether good ideas from multiple proposals can be merged, it is certain that events will continue to unfold rapidly. Such is the pace of a new radio format's birth.

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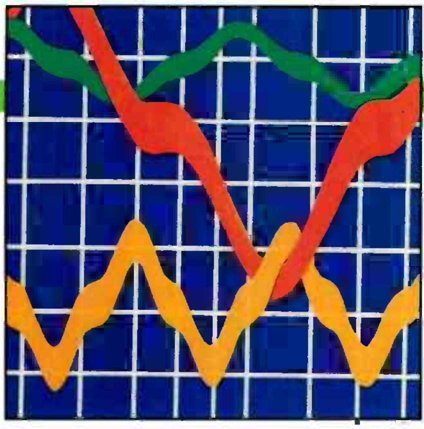
Testing digital audio devices



As digital audio takes over, users need new methods of comparison and testing.

By Richard C. Cabot, PE

Modern digital audio equipment is now available for a wide array of functions. Among other equipment, you can buy digital tape recorders, mixing consoles, delay units, reverberators, equalizers, STLs and FM stereo generators. A few years ago, most of these devices were exclusively analog. Users employing well-known techniques could compare the merits of one design over another, and trace problems to a given circuit.

If the digital equipment directly replaces an analog function, users can perform some tests implementing existing procedures. However, if the device has only digital inputs and outputs, or if the testing is for diagnostic purposes, the solely analog approach will not work. This article explains how measurements in the digital domain differ from their analog counterparts, and when it is appropriate to use digital measurement techniques.

Testing digital with analog gear

If digital equipment has analog inputs and outputs, analog equipment can test it. However, if the digital equipment is malfunctioning, such end-to-end testing does not allow you to pin down the trouble spot. Users must access the digital signal path.

A starting point is to isolate the problem to either the A/D or D/A conversion sections. One approach is to use a reference A/D or D/A to translate signals between the analog and digital domains. (Such a converter may be one-half of a known-good RDAT recorder.) This allows users to take measurements with existing analog test equipment, and to correlate those measurements with comparable analog devices.

This technique requires the conversion system to have significantly higher performance than the device under test. This may be the case when testing low-performance digital equipment. However, professional equipment employs the highest-performance converters available.

Another technique is to single out either the A/D or D/A by introducing digital gain or loss. Shifting bits of the digital word left or right by N bits produces a gain or loss of 6NdB, respectively. This works easily when testing a console, because digital gain controls are readily available. It may not be as simple with other types of digital equipment.

Putting digital gain between the A/D converter and the D/A converter increases the system's analog output. This emphasizes low-level non-linearities in the A/D converter. It uses the full dynamic range of the D/A to reproduce a small input signal. This makes A/D non-linearities a larger percentage of the signal than those of the D/A.

Putting digital attenuation between the A/D converter and the D/A converter decreases the system's analog output. This emphasizes non-linearities in the D/A, because it converts signals using the full dynamic range of the A/D.

Adding digital gain or attenuation limits the dynamic range of the measurements. However, because the converter noise floors are independent of the digital gain, the residual floor of distortion measurements is also reduced.

Apples and oranges

The anti-alias and reconstruction filters in digital equipment introduce response irregularities. One filter's response masks the other's. It is possible to subtract amplitude and phase response errors if they are sufficiently repeatable. Users should determine each filter's response using the analog test equipment, then apply this data to correct the measurements on the complete system.

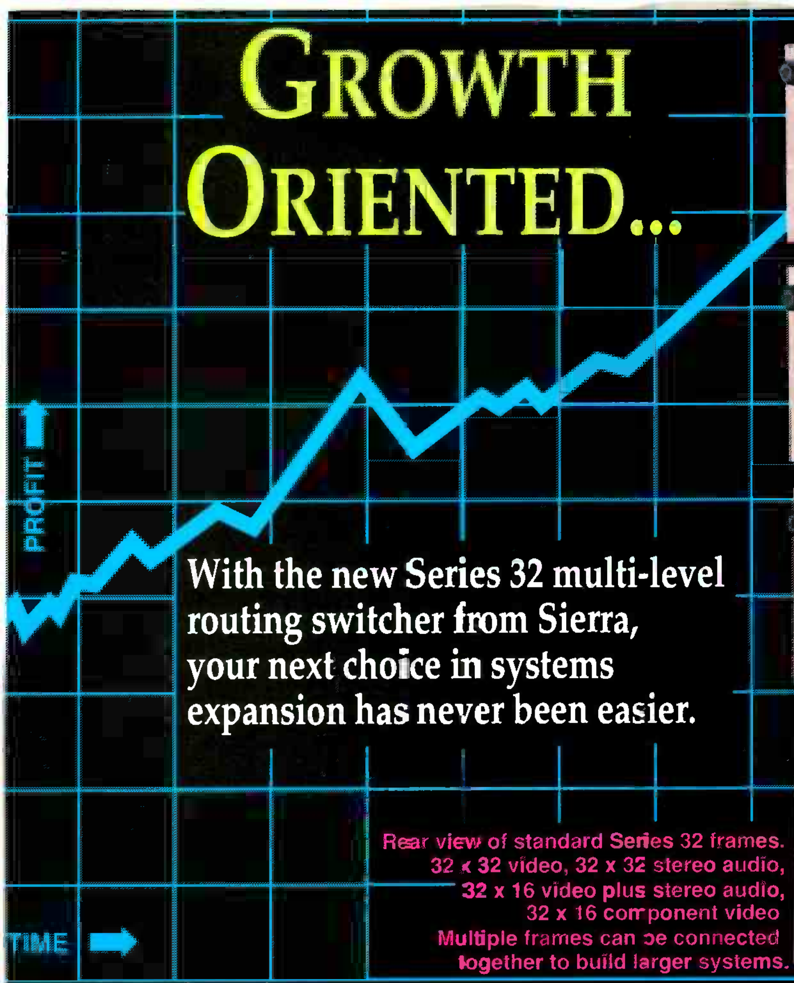
In addition to linear errors, such filters also introduce non-linearities and noise. Compensating for these errors is difficult. Designers must configure the filter in order to avoid them.

Other digital errors, such as sample and hold problems, de-glitcher problems and clock jitter, may also complicate the measurement process.

Another problem is that some parameters of interest to digital designers are difficult to measure in the analog domain. These include measurements of differential non-linearity, integral linearity and missing codes. Other digital-only measurements include error rate, error-correction efficiency and interpolation accuracy.

Making measurements in the digital domain could avoid these problems. Special-

Cabot is principal engineer at Audio Precision, Beaverton, OR.



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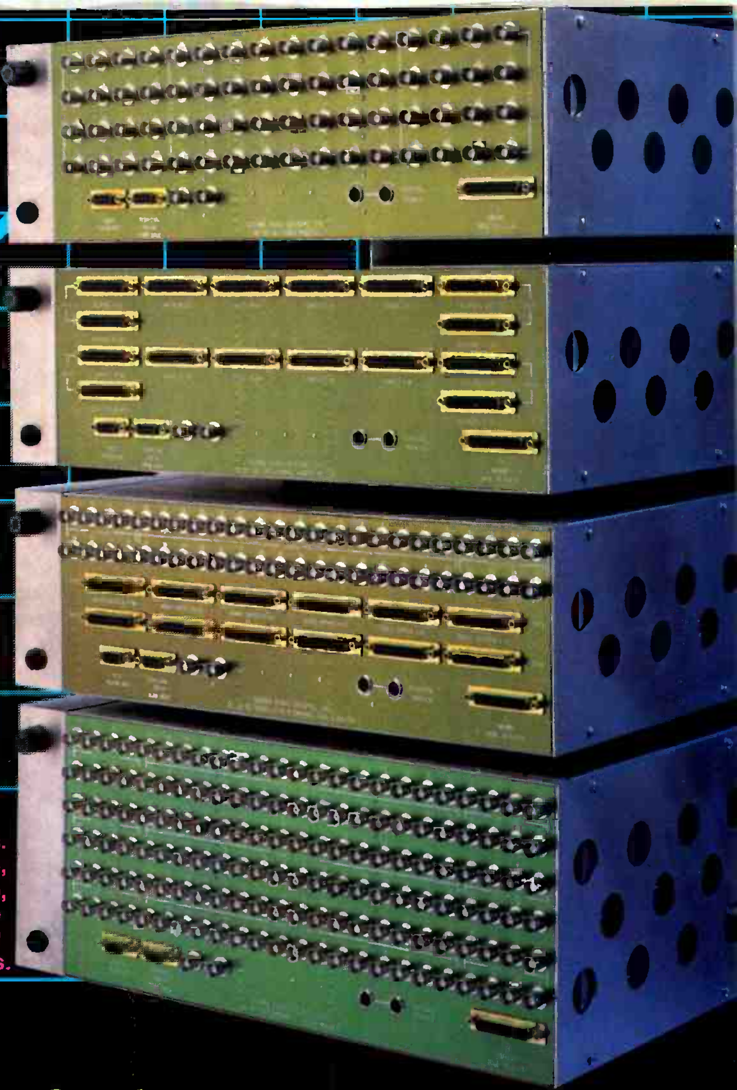

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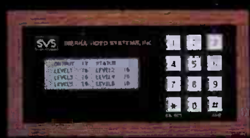

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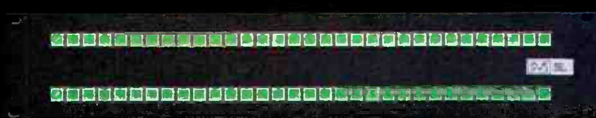
Rackmount keypad control panel



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ized hardware that can perform standard audio measurements directly on digital signals is available.

Comparing signals from digital and analog systems requires making the measurements in a similar manner. For example, suppose the THD+N at the analog output of a digital tape recorder is unacceptably high. An FFT distortion measurement taken at the machine's digital output will show if the problem is in the record or reproduce section. However, there is no way to correlate the measurement with analog readings. The measurements are not comparable. Therefore, it's good if a digital signal measurement system can simulate a conventional analog system.

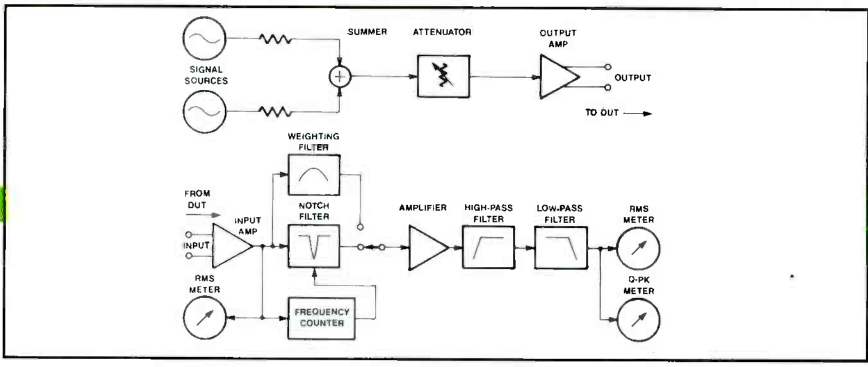


Figure 1. Block diagram of a basic analog distortion analyzer.

Digitally implemented techniques

Figure 1 shows a typical analog signal generator and distortion analyzer. Summing one or more signal sources creates the test signal. For harmonic or intermodulation distortion tests, the sources are low-distortion sine wave oscillators. The resistors that sum the signals also determine each component's amplitude. After summation, attenuators set the signal to the desired level. The signal goes to the device under test (DUT) via a balancing amplifier or transformer.

The measurement hardware buffers and converts the DUT output to a ground-referenced signal for processing. The signal then feeds many different measurement circuits: an rms level meter, a frequency counter, a noise-weighting filter and a notch filter.

The rms level meter provides the level reference for distortion measurements. It also provides general-purpose level measurements and measures frequency response when the stimulus is a sweeping frequency.

The notch filter removes the fundamental when measuring THD+N. Following any necessary amplification, the remaining signal is bandlimited by high-pass and low-pass filters. This filtered signal goes to a second level meter, which displays the distortion component amplitudes. Dividing the amplitude of the distortion products by the amplitude of the input signal yields distortion as a percentage or decibel value.

The frequency counter measures the incoming frequency and tunes the notch filter. This allows measurements with test tapes at the receiving end of broadcast links or other situations, where the input frequency cannot be precisely predetermined

In noise measurements, the weighting filter simulates the response of the human ear. Various standards recommend different weighting filters and different types of level meters. American and Japanese standards are written around "A-weighting" networks, and rms or average-responding meters. European standards specify a CCIR-weighting filter with a quasi-peak level meter. You must switch the filters and detectors to change between noise, distortion and response measurements.

Advanced digital signal generators and analyzers operate in much the same way. The software in high-speed digital signal processors (DSP) configures the system to implement — as needed — signal generators, filters, frequency counters and rms meters, which can simplify construction. Instead of having to switch in and out of a bank of filters, the software loads appropriate coefficients into one multistage filter. The filter then performs the required measurements — as a multipole notch filter for measuring THD+N, or a weighting filter for noise measurements.

Digital domain measurements

Measurements, such as error rate testing, are specific to digital systems. Occasional glitches from a digital tape recorder or console are hard to find using conventional measurement equipment. However, a digital test system might feed the DUT a pseudorandom digital pattern, comparing the output data to the input data. The system could then count up and display errors over a desired period.

Such input/output data matching allows users to detect signal problems. For example, the equipment may produce constant interpolation errors, or the "bypass" mode may not really bypass the processing at all. One popular portable RDAT recorder processes all digital inputs through a 0.5Hz digital high-pass filter. Therefore, it is incapable of making exact digital copies of any signal. Furthermore, this high-pass raises the noise floor above the −98dB theoretical 16-bit value, and modulates it up and down by 3dB at the 0.5Hz rate.

Testing the digital interface involves another set of measurements unique to the digital domain. The AES/EBU interface and its consumer equivalent, SPDIF, are the most common I/O formats in professional digital audio systems. (See the related article, "Digital Audio Signal Distribution p. 54.) However, the definitions of status bit functions are vague. Various manufacturers require that their equipment receive some status bits before operating. Some equipment performs in different ways, depending on what status bits it receives. Other manufacturers have simply left off some of the status bits, which results in many pieces of equipment that will not digitally interconnect.

For example, one portable RDAT recorder requires the presence of the status bits that indicate the sample rate. Many manufacturers drop these bits, because most of today's devices use PLLs to automatically lock to the incoming signal. Nevertheless, this recorder requires these bits, which means it cannot record from most digital audio sources.

The interface data rate is approximately 3Mbit/s. The specification defines the transmitter waveform amplitudes and timings and the receiver minimum sensitivity requirement. Although manufacturers should adhere to the specification, some of them stray widely. For example, one popular small digital mixing console appears to need digital input signals of greater than 1V for proper operation. The standard, however, specifies 0.2V as the correct minimum receive level. Needless to say, this has caused many interconnection problems in the field.

Typical digital measurements

Users need digital test equipment in order to work with digital signals. Such equipment can verify digital audio signal parameters into and out of pieces of digital audio equipment. This equipment can also test the error rate in the cables that

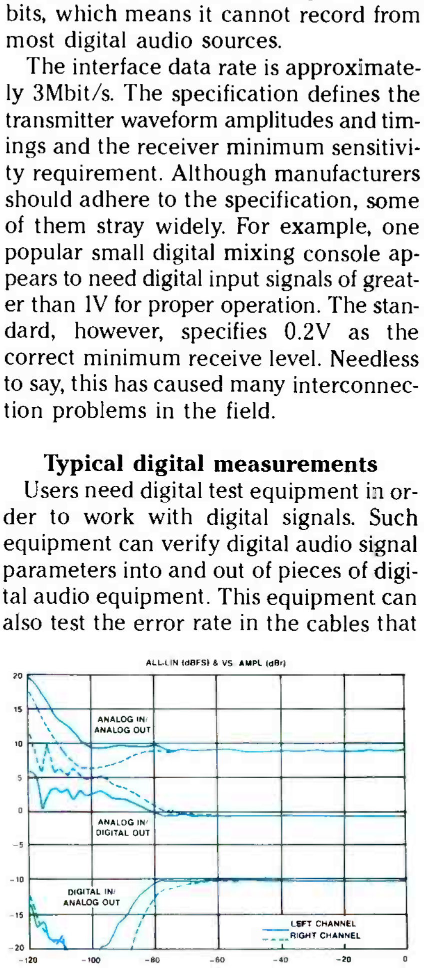
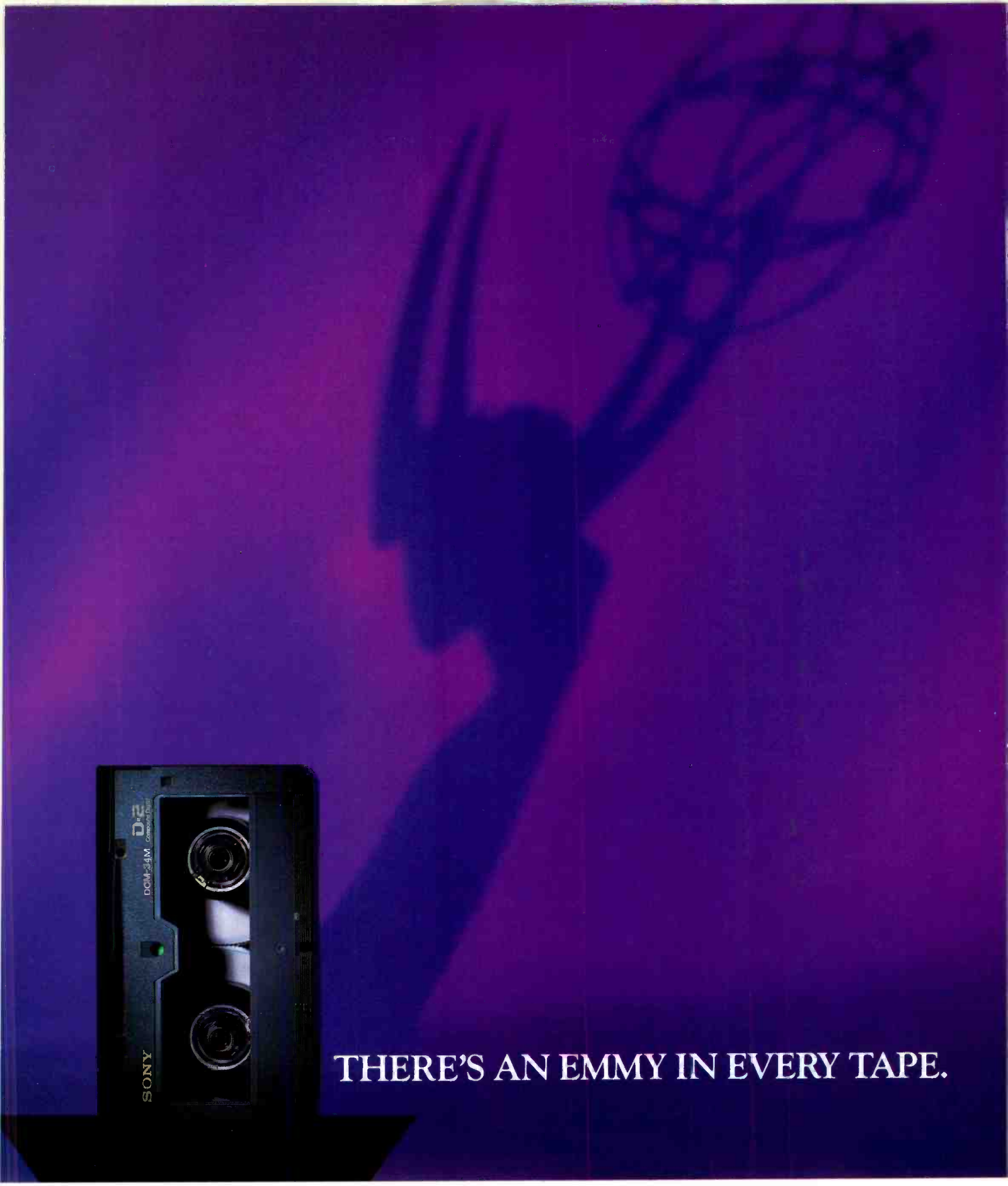


Figure 2. The level linearity for an RDAT recorder, measured three ways. An ideal recorder would have a flat plot.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interconnect such gear. To find errors introduced by rolloff, reflections and noise on long digital interconnect cables, the equipment sends the proper test data down the line. It then measures the results at the other end.

Converter linearity is measured by plotting conversion gain as a function of level. Stimulate D/A converters with a digital sine wave. Using an rms meter, measure the results through an analog bandpass filter. Swap the analog and digital elements to measure A/D converters. Stimulate the converter with a low-distortion analog generator. Program the digital measurement device to bandpass filter the converter's digital output, and compute the rms amplitude.

Figure 2 shows the level linearity for an RDAT recorder, measured three ways. The upper trace plots analog in to analog out. The middle trace is analog in to digital out. The lower trace is digital in to analog out. An ideal recorder would have gain constant with input level, resulting in a flat plot.

In this example, the all-analog measurement shows relatively good performance. However, the two hybrid measurements look significantly worse. This may show that the deck's record and playback sections have errors that compensate for each other. If a tape made on this recorder was played back on a different machine (or vice versa), the error would not be compensated for. Significant distortion would occur. Similarly, future readjustment of this recorder's D/A converter could result in distorted playback of previously recorded tapes.

Figure 3 graphs the frequency response of an RDAT recorder, measured three ways. The upper trace is the response of the record section, with the input signal applied to the analog input and measuring at the SPDIF output. The middle trace is the playback response of driving the SPDIF input and using an rms voltmeter at the analog output. The bottom curve is the response from the analog input to the analog output.

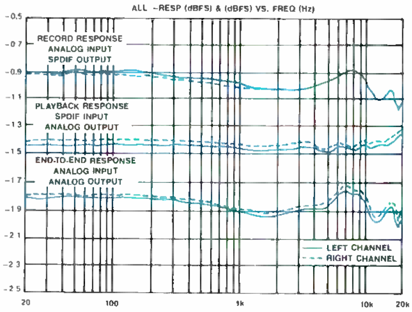

Figure 3. This is the frequency response of an RDAT recorder, measured three ways. Notice the variations in response at higher frequencies. The playback response is apparently compensating for some of the record response anomalies.

Notice the variations in response at higher frequencies. Although it is hard to determine the source of the response ripples using this measurement alone, the graph reveals that the playback response is compensating for some of the record response anomalies. This is especially so at around 8kHz and above 15kHz. This performance and compatibility problem may not have been revealed by making all measurements "end-to-end."

Figure 4 graphs distortion vs. frequency of an RDAT recorder, measured four ways. The lower curve measures distortion from the digital input to the digital output. As expected, it measures as an ideal 16-bit system. The second curve from the bottom plots the playback portion driving the SPDIF input and using an analog distortion analyzer at the analog output. The third curve (at 1.2kHz) is the response of

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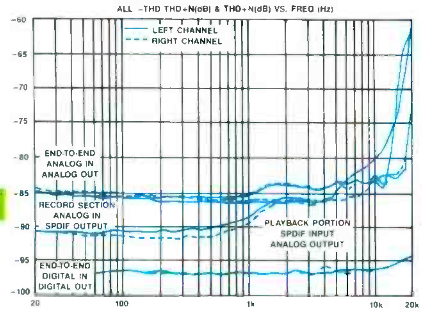

Figure 4. *This is the distortion vs. frequency of an RDAT recorder, measured four ways. The bottom curve behaves close to what would be expected for a 16-bit ditigal system.*

the record section, with an analog input and measuring at the SPDIF output. The top curve (at 1.2kHz) is the response from the analog input to the analog output.

Figure 5 shows the results of an alias rejection test, which is unique to digital devices. The tested device is an oversampling A/D converter. A sine wave feeds the converter. It is set at a level 6dB below full scale to prevent overdriving the converter's input filters, and swept in frequency from 10kHz to 200kHz. The graph shows the rms digital output level. This provides a plot of the response of the converter to in-band and out-of-band signals. All output above the Nyquist frequency of 48kHz represents aliased components. The alias products, which the converter allows through, graph clearly at approximately 50dB below full scale, 44dB below the input. They are mirror imaged around 96kHz (2X the sampling rate) and 192kHz (4X the sampling rate). There is also a strong alias product between 24kHz and 28kHz. This is because of the 1/2-band topology of the last stage of decimation filtering in the converter.

The dashed line shows the signal frequency at the converter output. At 73kHz analog input, there is a significant alias product at 23kHz (96kHz-73kHz). As the input frequency increases, the alias frequency drops. As the alias product reaches approximately 7kHz, the noise floor obscures it.

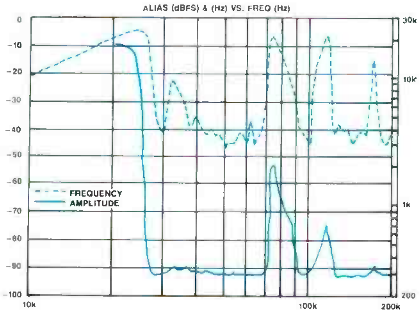

Figure 5. *A plot of an alias rejection test for an oversampling A/D converter. The dashed line shows the signal frequency at the converter output. The solid line is signal amplitude. Any amplitude above 24kHz indicates an alias product.*

Measuring up

As digital audio takes over from its analog predecessors, digital test and measurement techniques will become more commonplace. In the mean time, users should be familiar with both systems. This allows them to use whichever is most appropriate in each situation. It also allows meaningful comparison of measurements made in each domain.

Editor's note: For further information, please refer to the following:

1. Cabot, R.C. "Audio Measurements." *Audio Engineering Handbook*, K. Blair Benson Ed., McGraw-Hill 1988, Chapter 16.
2. Cabot, R.C. "Modulation Noise in Digital Audio Systems or Who Wrapped the Mics in Sandpaper?" *Recording Engineer/Producer*, July 1988, pp. 20-24.
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4. Finger, R.A. "Review of Frequencies and Levels for Digital Audio Performance Measurements." Journal of the Audio Engineering Society, vol. 34, January and February 1986, pp. 36-48.
5. Finger, R.A. "On the Use of Computer-Generated Dithered Test Signals." Journal of the Audio Engineering Society, June 1987, vol. 35, pp. 434-444.

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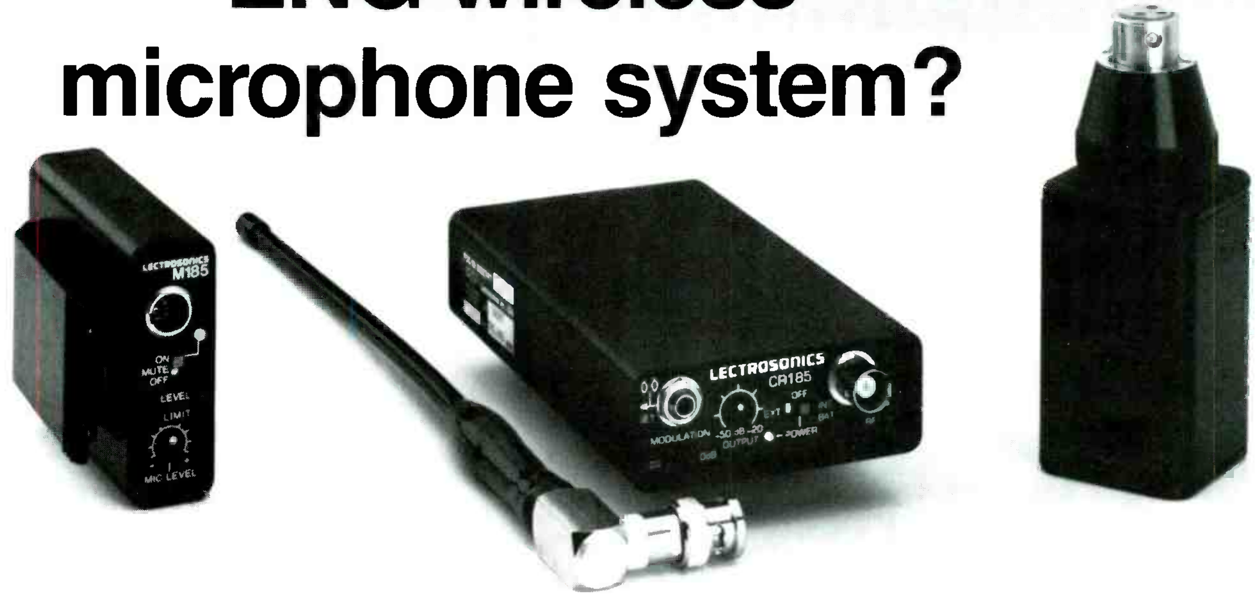

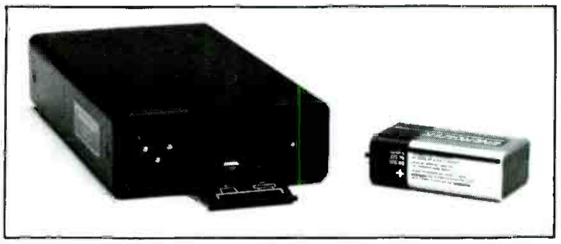

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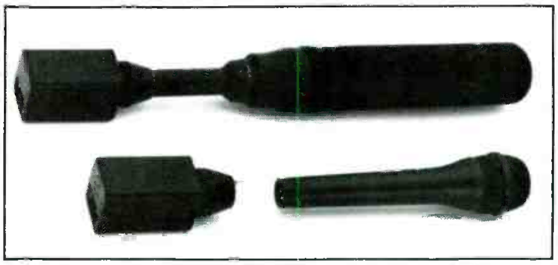

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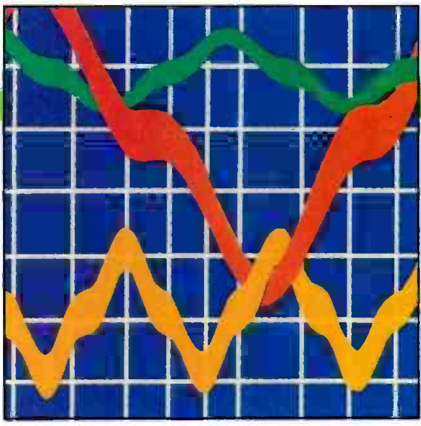
Digital audio processing



Manipulating audio signals in the digital domain has its advantages — and its drawbacks.

By Robert Orban

Digital audio processing can provide its users with many functions that are unavailable from analog designs. It can emulate familiar analog processes, such as equalization, limiting, compression and de-essing. It also can perform operations that require large time delays for realization — functions that are essentially impossible to achieve with analog circuitry. Digital devices are easily programmable, so the user can store and later recall control settings.

However, digital audio processors have their disadvantages. If they are interconnected with conventional analog circuitry, each processor in a chain must perform an analog-to-digital (A/D) conversion before its digital processing section, and then must convert the processed signal back to analog at its output. The converters at both ends are usually the weakest link in the chain. Because of improved topologies many digital-to-analog (D/A) converters boast true 16-bit accuracy, although this is rarely achieved in low-cost signal-processing equipment.

However, on the A/D side, true 16-bit accuracy remains expensive and difficult to achieve, and is rarely found in low- or moderate-cost equipment. Digital compressors and limiters are particularly challenging, because their A/D converters must handle extremely wide dynamic ranges, so today's analog compressors are likely to yield better signal-to-noise performance.

Orban is chief engineer at Orban, a division of AKG Acoustics, San Leandro, CA.

A solution, of course, is to minimize the number of conversions by keeping signals in the digital domain. Some processors permit this, but even when available, digital audio interfacing remains somewhat challenging to the user.

Other potential digital processing pitfalls are less obvious. Unless carefully done, digital processing can audibly distort the signal in various ways. A processor's output can have far less than 16-bit resolution, particularly if the internal digital arithmetic within the processor is limited to 16-bit words.

If that's the case, the loss of resolution can result in additional noise and unpleasant-sounding distortion and modulation noise. Processors that change the level of the signal in the digital domain (like compressors, limiters and mixers), or that filter the signal at low frequencies are particularly subject to problems unless design criteria can ensure that any loss of resolution appears simply as a higher noise floor and not as added distortion. For this reason, 24-bit digital signal processing (DSP) chips have become popular in high-performance digital processors; they are less subject to resolution-loss problems than DSP chips that limit internal word lengths to 16 bits.

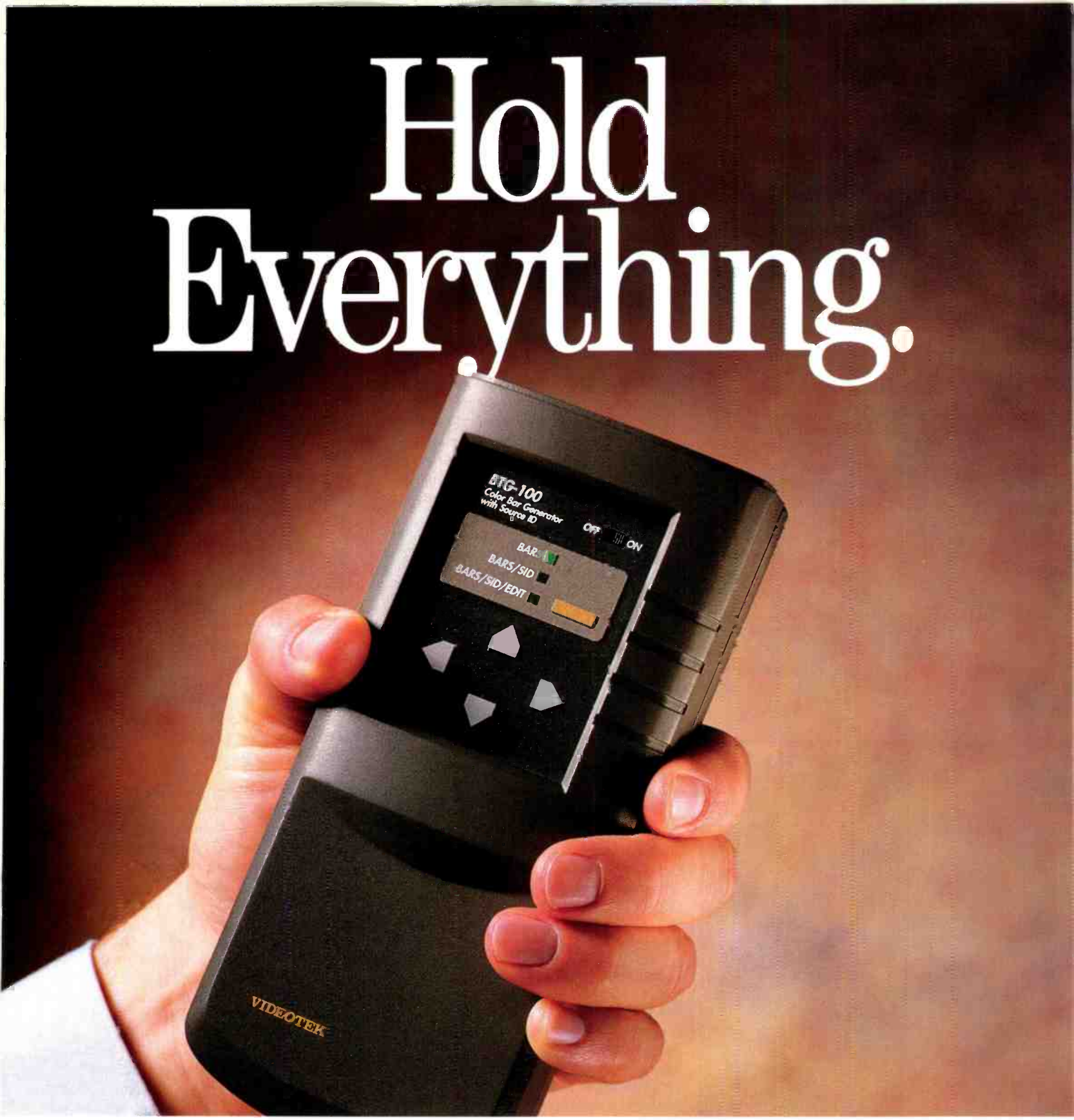
Delay lines

One of the most straightforward digital audio processing functions is simple time delay. Here, the signal is converted to digital form, applied to a digital memory, and read out after a desired interval. In radio, delay lines are most commonly used as profanity delays for call-in shows. In television, they delay the audio for resynchronization with video that has been passed through a framestore or time base corrector.

The audio quality of a delay line is usually determined by the number of bits used to represent the audio signal internally, and by the quality of its A/D and D/A converters. Because delay lines usually do not filter the signal or change its level, little can go wrong in the digital section. In profanity delays, manufacturers use various techniques to build the delay back to its full value following a dump into real time. The user should evaluate such schemes for naturalness and the amount of time required to re-establish full delay. In audio-for-video delays, the user's job is simplified if the delay line is easily coupled to its corresponding video framestore, so that it automatically tracks the delay of the framestore without operator intervention.

Pitch shift and time compression/expansion

Pitch shift is the process of changing the pitch of a signal while keeping its duration constant. As usually understood, a pitch shifter performs *geometric* pitch shift, retaining the ratio between the pitch of the fundamental and its various harmonics. In contrast, *arithmetic* shift moves all frequencies by the same number of hertz, thereby putting harmonics out of tune with the fundamental. Arithmetic shift is much easier to achieve, but is typically not useful for anything other than "robot voice" effects, or telephone-line fre-



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quency extension.

Although there are several ways to perform geometric pitch shift digitally, what's important to the user is the smoothness and naturalness of the result. Older or poorly designed pitch shifters have a tendency to glitch and make thumping, clicking and burbling noises. These problems usually get worse as the amount of pitch shift increases.

When pitch-shifting stereophonic program material, be sure that relative phase shift between channels is not introduced. This will affect the frequency response of the L+R signal, causing unintended effects for the mono listener. Successful stereo pitch shift is tricky, and few manufacturers have fully mastered it.

Time compression and expansion combine pitch shift with variable-speed playback of the original program material. For example, to compress a 31-second spot to 30 seconds, the original is played back at 31/30 times normal speed. This raises the pitch, so the audio is then pitch-shifted to lower the pitch by a factor of 30/31, restoring the original pitch but retaining the faster playback. The most useful devices integrate playback-machine speed control with the audio pitch-shifting function.

Reverberation

Simulating reverberation is one of the oldest applications of digital audio processing. Today's digital reverbs usually have programs to simulate three basic types of ambience: *halls*, *chambers* and *plates*.

A hall program simulates a large space. This means that there is a considerable time delay between the original sound and the onset of the first reflection, and that the early reflections tend to be sparse. These are best for non-percussive sounds or classical music.

A chamber simulates a classic acoustic echo chamber (a small room with reflective surfaces). In this case, there is little time delay between the direct sound and the onset of reverberation. Early reflections are usually dense, making the chamber better suited for percussion than a hall program.

A plate program simulates the classic plate reverbs, and produces a lively, dense reverberation that works well with vocals, acoustic guitar and percussion. Chamber and plate programs have an overall brighter sound because they are not emulating the higher *air loss* (the absorption of high frequencies by the air) caused by the large volume of air in a hall.

Some reverbs also have specialized programs (room simulation) designed to create acoustic spaces that would not ordinarily be considered to have good acoustics for music or voice. These are used subtly in film or video production to create a sound that is congruent with a given visual environment.

Digital reverbs can also provide nonlinear decay programs (also called gated reverb), which truncate reverberant "tails" to provide dense reverb on instruments, such as the snare drum (the "Phil Collins" sound), without muddying the mix with long decays.

Special effects

Manufacturers offer a mind-boggling assortment of special effects in the digital domain. These include various types of echo, flanging, chorusing, resonances, distortions, auto-panning and other cross-channel stereo effects.

Exciters are another type of effects processor, designed to add clarity and definition to the program material. Some units add harmonics to the audio; others are fixed or program-adaptive high-frequency equalizers. Digitally realized exciters may add high-pass-filtered, time-delayed signals to the original. Others provide more than one of these functions. As usual, there is no substitute for listening tests to determine whether a particular exciter's effect is satisfactory.

A further, recently active area of DSP application is spatial or 3-D sound processing. In addition to multichannel surround systems for consumer use, processors that encode some spatial information into *two channels* have stimulated some interest (and debate) in the professional broadcast and recording communities. (More on this subject in the September issue.) DSP's versatility and processing power make it appropriate and easily programmable into these and other processes, much like the op-amp has been for analog circuit applications.

Equalizers and filters

Compared to analog equalizers, digital equalizers produce far more precise and repeatable curves, and usually have programmable memory so that these curves, once set, can be recalled on command.

Digital equalizers and filters can be designed to simulate their analog counterparts' behavior in amplitude and phase. On the other hand, digital equalizers can also be made to affect amplitude response, but delay all frequencies *equally*. If this constant delay is ignored, the latter equalizer design could be considered to have no phase shift. Although theoretically possible, that kind of equalizer is impractical to build with analog circuitry.

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"Little Death" created by Matt Elson for Symbolics. Computer-Symbolics. "Chinese Man" created by Livoni Design for Abekas Video Systems. "CNN Newsource Extra" created by Michael J. Lizak at Digital Post and Graphics. for Turner Broadcast Systems. Computer-Symbolics.

and *finite impulse response* (FIR) are often used in digital circuit design specs. IIR uses digital feedback loops to create equalizers of the traditional analog (phase shifting) type. FIR designs create equalization by adding many delayed versions of the input with different delays and gains. If these delays and gains (coefficients) are chosen correctly, an FIR equalizer can indeed have constant delay at all frequencies.

You might think that an equalizer without phase shift would sound better than a conventional analog-style equalizer, but this is not necessarily true. First, the constant time delay produced by the second type of equalizer can be significant and can cause comb filtering if the equalizer's output is mixed with the same unequalized (undelayed) signal. Second, if an equalizer is used to correct poor frequency response of something earlier in the audio chain, that "something's" frequency response error might have been accompanied by a complementary phase shift.

An analog-style equalizer could correct the phase and amplitude response simultaneously; the constant-delay equalizer would leave the phase error intact. The moral: You must always trust your ears and audibly compare the effect of both types of equalizers to hear which sounds better in a given application.

If the equalizer uses IIR circuitry, test its low-frequency (around 30Hz) band(s) carefully with low-amplitude, low-frequency signals, and listen for distortion. Many things can go wrong in digital equalizer design, making them probably the least ideal of all digital audio processing devices. *Caveat emptor.*

At this writing, there are few pure digital compressors and limiters available. Those that exist are usually designed for CD mastering and accept only digital inputs. There is a good reason for this. Because the A/D converter in a digital dynamics processor must accept the full dynamic range of uncontrolled analog input signals, the user must allow an extra decibel of headroom in the converter for every decibel of gain reduction range desired. This quickly raises the converter noise floor, and makes a wide range digital compressor (with the >40dB gain reduction range often required in recording studio applications) an impractical proposition, given the state of the A/D art.

Peak limiting requiring only a few decibels of gain reduction *is* practical in the digital domain, however. One such approach follows the classic BBC analog delay-line limiter model. It smooths the gain-control signal with a low-pass filter, and then delays the audio before the gain control element, to prevent the delay in the control voltage (caused by its low-pass filter) from causing overshoot in the peak-controlled audio. Downward expanders (including noise gates) are also eminently practical in digital, and can have the usual advantages of programmability, memory recall and computer control.

For highest audio quality, the signal should be re-dithered in the digital domain whenever its level is reduced below the level at which it is applied to the input of the dynamics processor. (*Dither* is the addition of white or frequency-shaped noise to the audio signal to prevent it from becoming distorted or disappearing when its level approaches the least-significant bit [LSB], which is the smallest level that can be resolved by the digital code. The random nature of the added noise signal prevents a low-level coherent signal from toggling between the LSB and an "all-0s" condition, thereby reducing distortion.)

Carefully audition any digital expander or noise gate for an audible increase in distortion with increasing gain reduction. Any device exhibiting this problem probably does not re-dither the signal correctly.

Transmission signal processing systems

A competent transmission processor ordinarily limits instantaneous peaks to an absolute level while controlling audio output spectrum for AM broadcasting to NRSC (or EBU) standards, or for FM, such that the pilot and subcarriers are fully protected.

A transmission processor also may include multiband compression, high-frequency limiting (to protect the system from pre-emphasis-induced overload), and equalization to color the signal as desired. If the entire processor is digitally realized, it can be readily reconfigured in milliseconds to change almost any aspect of its topology, such as the number of bands in its multiband compressor. Control settings can also be stored and later recalled by local clock, remote control or computer, to implement daypart processing. The processor can readily generate test and signaling tones, facilitating tests of the transmission system and the generation of EBS alert tones.

A competent transmission processor can be interfaced to any common STL system. To accommodate digital STLs, the transmission processor should have provisions for digital inputs and outputs (most commonly AES/EBU).

Above all, a digital transmission processor should have sound quality equal to or better than its analog counterparts. Without a fully competitive sound, convenience features are of little value. Achieving this in a digital processor requires a marriage of art and mathematical design more rigorous than anything in the genesis of its analog ancestors. Although DSP provides many new possibilities beyond analog circuitry's capacity, some common analog processing functions (such as clipping) are much more difficult to accomplish digitally.

With careful listening, it is possible to reject an incompetent processor in a few minutes. But taking the time to fine-tune a competent one, then fully evaluating it with many different types of program material and radios, requires days, if not weeks. With patience, digital's overwhelming advantages will manifest themselves as clearly here as they have elsewhere in the audio-processing arena.

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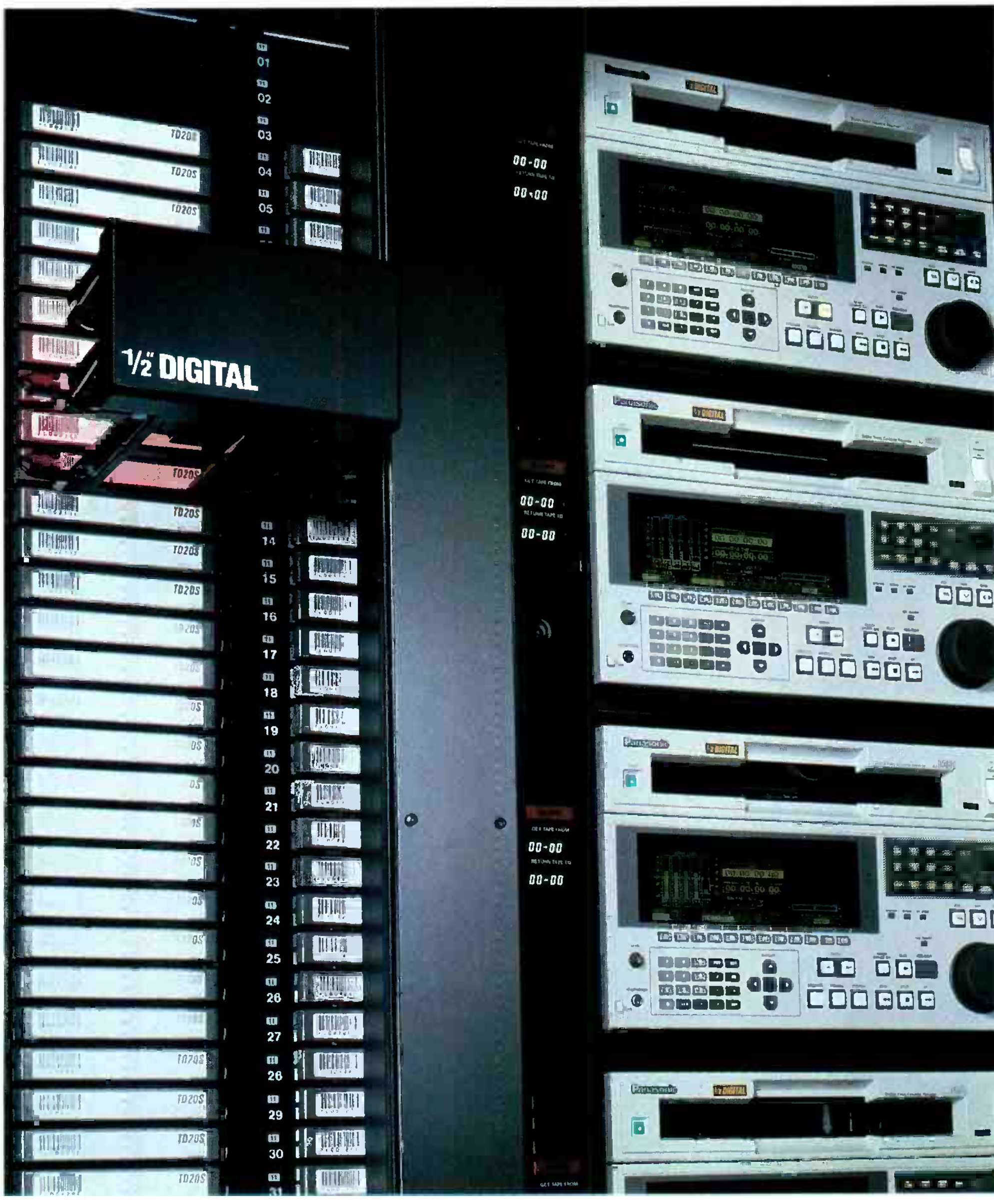

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Digital audio signal distribution

Charting a path through the maze of digital audio technology.

By David L. Bytheway

Watch out — digital audio may be coming to your teleproduction or broadcast facility. And although digital audio techniques are far from mature, it is now possible to do an entire production in the digital domain.

Some people believe that adopting digital techniques means an end to signal-distribution problems. However, digital signal distribution may present new, and sometimes, difficult issues to the engineer. This article will discuss some of the important considerations associated with plantwide digital audio distribution and offer some solutions.

Why digital audio?

Digital audio systems are superior to analog for signal storage and long-distance transmission. However, analog methods are simpler, and in some respects, better for in-the-plant distribution and certain processing functions. So why would anyone want to use digital audio for signal distribution? Because the final signal quality will be higher if it is not converted between the analog and digital domains more often than absolutely necessary. This conversion process is a primary source of signal degradation. Therefore, as tape machines and other audio hardware become digital, digital signal distribution becomes an important key to preserving signal quality.

Bytheway is a principal design engineer responsible for audio product design at Broadcast Television Systems (BTS), Salt Lake City.

The AES/EBU standard

The AES/EBU standard was jointly developed by the Audio Engineering Society (AES) and the European Broadcasting Union (EBU). The standard is a stereo signal protocol that allows digital audio of three different sampling rates. It can carry up to 24 bits of audio data in each channel. The main audio data usually occupies from 16 to 20 bits. Four auxiliary bits can carry cue channel information, or they can add to the 20-bit audio data — providing up to 24 total bits. (See Figure 1.)

The AES/EBU standard specifies that data be sent least significant bit (LSB) first. This is opposite to the way audio is usually stored on tapes or CDs. The data is also sent in two's complement form. This means that numbers with a leading "0" represent positive waveform values, and numbers with a leading "1" represent negative waveform values.

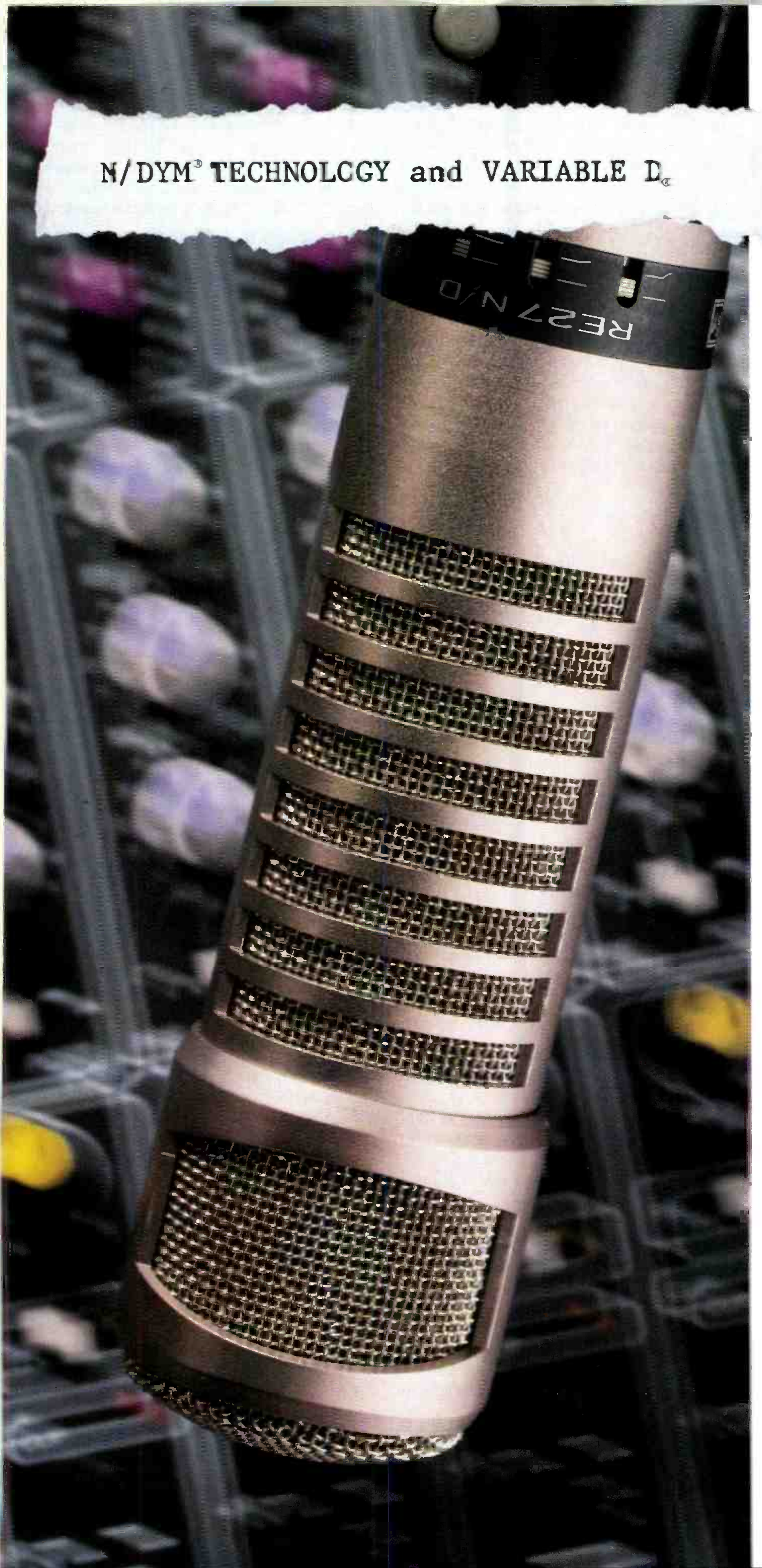
Four preamble/synchronization bits are associated with each audio sample. In addition, there are four auxiliary databits: the validity bit, the parity bit, the channel status bit and the user bit.

The validity bit and parity bit are the only error-detection bits associated with each audio sample. The user and channel status bits provide auxiliary information available to the user. Because there is only one status and user bit for each audio sample, a block of 192 consecutive frames (one frame is a left and right channel sample) provides 24 bytes of channel status and user data. At the maximum sampling rate supported (48kHz), a status block transmits every 4ms. Error-correction circuitry calculates a cyclic redundancy check character (CRCC) on the entire status block and transmits it in the last byte of the status block. This error check can verify correct transmission of channel status, but not of individual audio sample data.

The standard implementation of AES/EBU protocol supports three sampling rates: 32kHz, 44.1kHz (as used in CD players) and 48kHz (usually considered the professional standard). Data is encoded using a modified Manchester encoding method. This results in a biphase signal that has no DC component. Therefore, it can be AC coupled. This is advantageous. Designers can use simple pulse transformers for input and output coupling devices.

The protocol uses standard twisted pair cable. This enables the use of existing wiring. The signal is designed for twisted pair cable with an impedance of 110Ω. At the maximum sample rate, the AES/EBU signal produces a data rate of 3.072Mbits/s. Work is currently under way to define a coaxial version of the standard.

Equipment following the AES/EBU standard is available from many manufacturers. Also, there are many different implementations of the AES/EBU protocols. Engineers may sometimes have difficulty interconnecting devices that claim to be AES/EBU compliant. The AES continues to further define the standard, and now recommends a minimum standard and full implementation of the interface. Nevertheless, there are still cases where devices equipped with AES/EBU interfaces may



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fail to work with each other.

Some consumer equipment, such as CD players with digital audio outputs, use a simpler version of this protocol. This standard, which is sometimes called the Sony/Philips Digital Interface Format (SPDIF), does not have extensive status and user bits, but still encodes the audio samples similarly to the AES/EBU standard. In many cases, users can interconnect consumer and professional equipment.

Problems in AES/EBU signal distribution

The following problems are common in AES/EBU signal distribution:

Impedance matching.

The encoding method used by the AES/EBU signal has significant energy beyond 6MHz. This makes it an RF-type signal, much like analog video. Therefore, AES/EBU signal handling requires RF-type as opposed to audio techniques. Additions to the AES/EBU standard now require 110Ω source and load impedances. The standard has no definition for a bridging load, although loop-through inputs, as used in analog video, are theoretically possible. Improperly terminated cables may cause signal reflections and subsequent data errors.

Use of distribution amplifiers.

Every source must drive only one load. This means using a distribution amplifier if a single source must be connected to more than one destination. For correct operation, all cables must be impedance matched and correctly terminated, especially if they are more than a few feet long.

Distance.

The relatively high frequencies of the AES/EBU signals cannot travel over twisted pair cables as easily as analog audio. Capacitance and high-frequency losses cause high-frequency rolloff. Eventually, signal edges become so rounded and low that the receivers can no longer tell the "1s" from the "0s." This makes the signals undetectable. Typically, cable lengths are limited to a few hundred feet.

Receiver equalization can improve the rise times, making the edges more defined. The AES/EBU standard includes an equalization curve for use at the receiver, which has the potential to increase the usable cable length. However, this equalization is rarely used in today's equipment. Workable cable length can be greatly increased by using low-capacitance cable specifically designed for data transmission.

Jitter and skew.

The accumulation of jitter or skew in the signal is particularly troublesome. (See Figure 1.) It can cause data degradation and even prevent correct decoding. The encoding method used by the AES/EBU signal puts the data in the transition edges of the signal. This means that the location of the edge within the signal is more important than whether the signal is high or low. At the 48kHz sample rate, each AES/EBU bit time or *cell* is 325ns wide. After the receiver circuit locks to the incoming signal, it checks to see if a transition occurred within the current cell. If the transition is within the center of the cell (with a tolerance of $\pm$20ns), the receiver considers the data to be a 1. If no transition occurs, it considers the data to be a 0.

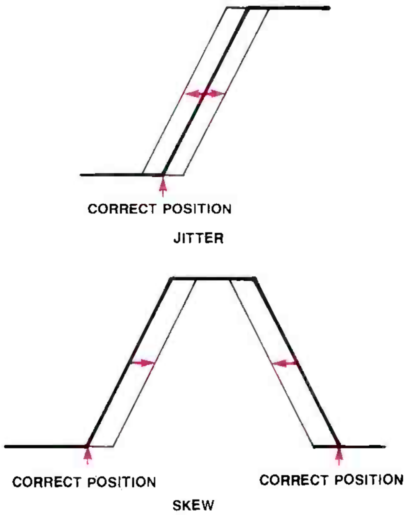

Figure 1. *Jitter and skew can corrupt AES/EBU digital audio signal data. Jitter occurs when the signal transitions come either before or after their correct positions. Skew occurs when edges permanently shift position, changing the waveform's duty cycle.*

When a signal acquires jitter, the position of the transitions will vary. This causes the transitions to be late or early, compared to where they should be. The standard AES/EBU specification calls for jitter to be less than $\pm$20ns. If it is greater than this, the receiver circuits may have trouble locking to the signal or may make errors interpreting the data. In extreme cases, jitter may compromise the signal to the point that it becomes unusable.

Another similar problem is *skew*, which is the change of duty cycle in the signal. If a device causes an offset in the position of the rising and falling edges of the AES/EBU signal, the transitions can be skewed. Skewing causes errors similar to jitter.

These problems are similar to those experienced in serial digital video distribution systems.

Synchronization.

Signal synchronization is a virtually unknown problem in analog audio systems. The methods of switching or mixing analog audio are simple and straightforward when compared to their digital counterparts. When switching between two audio signals, there should be no audible disturbance or click. Properly designed analog switches can easily accomplish this. However, for clickless switching of digital audio, the two sources must be synchronized. The frames must be aligned so that the individual signals' bits arrive at the switch at the same time. In addition, the switch point must occur at a point that presents complete samples to the output. Failure to do so can cause incorrect data, which may produce an audible disturbance.

Two separate issues exist: synchronization and timing. Synchronization means that the two sampling rate clocks are phase-locked. Timing means that in addition to the signals being synchronized, the frame boundaries are also aligned and occur at the same moment in time.

The solution is to resynchronize the signals, which is really a form of sample rate conversion. Conversion from one sample rate to another is fairly straightforward for signals of widely different sample rates. However, the process is more difficult for signals of close sample rates. Such is the case for two signals that are the same sample rate but are asynchronous with each other.

The AES recently published a draft specification on the synchronization of digital audio studio equipment. This specification outlined a practice of correcting timing differences on all inputs and bringing all signals into phase (frames aligned with a reference signal). To do this, each input circuit should provide a ½-frame (approximately 10.8μs) buffer. This way, signals that arrive early can be delayed long enough to match the last signal to arrive. Output signals must align within $\pm$5% of the reference timing point. This is to prevent output timing errors from exceeding the buffer capacity of a downstream device. If a device creates more than one frame of processing delay, the specification requires that the delay be clearly marked on the equipment.

In order for this automatic timing to work, all signal sources must be synchronized. This means that a complete digital audio distribution system will require sync systems to align the frame boundaries, similar to those used in analog video. (See Figure 2.)

Distribution system methods

There are several different approaches to providing plantwide distribution of digital audio signals. Using distribution switchers and amplifiers is the most common approach.

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nal distribution system should be independent of either sample rate or synchronous signal requirements. This means that the output of an unsynchronized CD player could pass through an entire distribution system without needing to be sample-rate converted, resampled or reclocked.

Reclocking is the reconstruction of a signal by detecting and re-encoding the data. To reduce jitter, the sample clock must be recovered from the incoming signal and regenerated, usually with a low-jitter phase-lock loop. There are other methods for reclocking. Some lend themselves to integration into a single IC. Future developments may greatly simplify the reclocking process.

Some people in the industry advocate reclocking at every distribution amplifier, switcher and other device in the system. This practice may cause problems in large systems. First, the reclocking circuit must be able to detect every anticipated standard and sample rate that will ever pass through the system. This complicates circuit design. Second, the system will only be as good as the poorest reclocking circuit. If one weak reclocking circuit makes an error, then the signal will be unrecoverable by all subsequent stages.

Some conventional analog switchers and distribution amplifiers may pass, equalize or switch the signal while introducing minimal jitter and without reclocking. Such systems can be somewhat independent of standards and sample rates. Non-standard variable-speed sample-rate signals could pass through such systems, as could asynchronous signals.

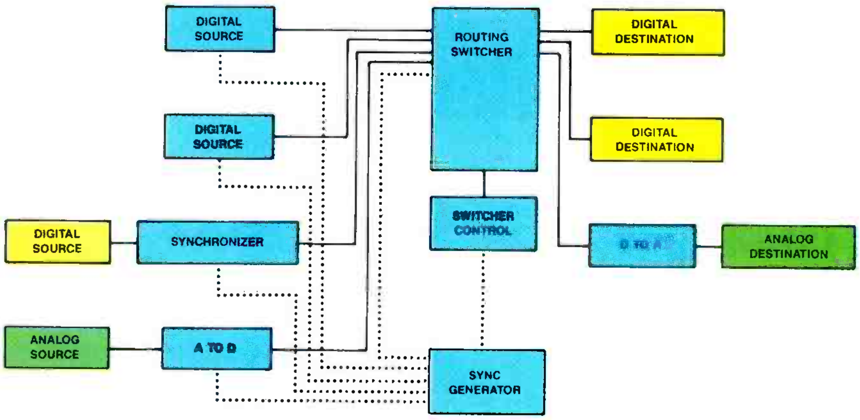
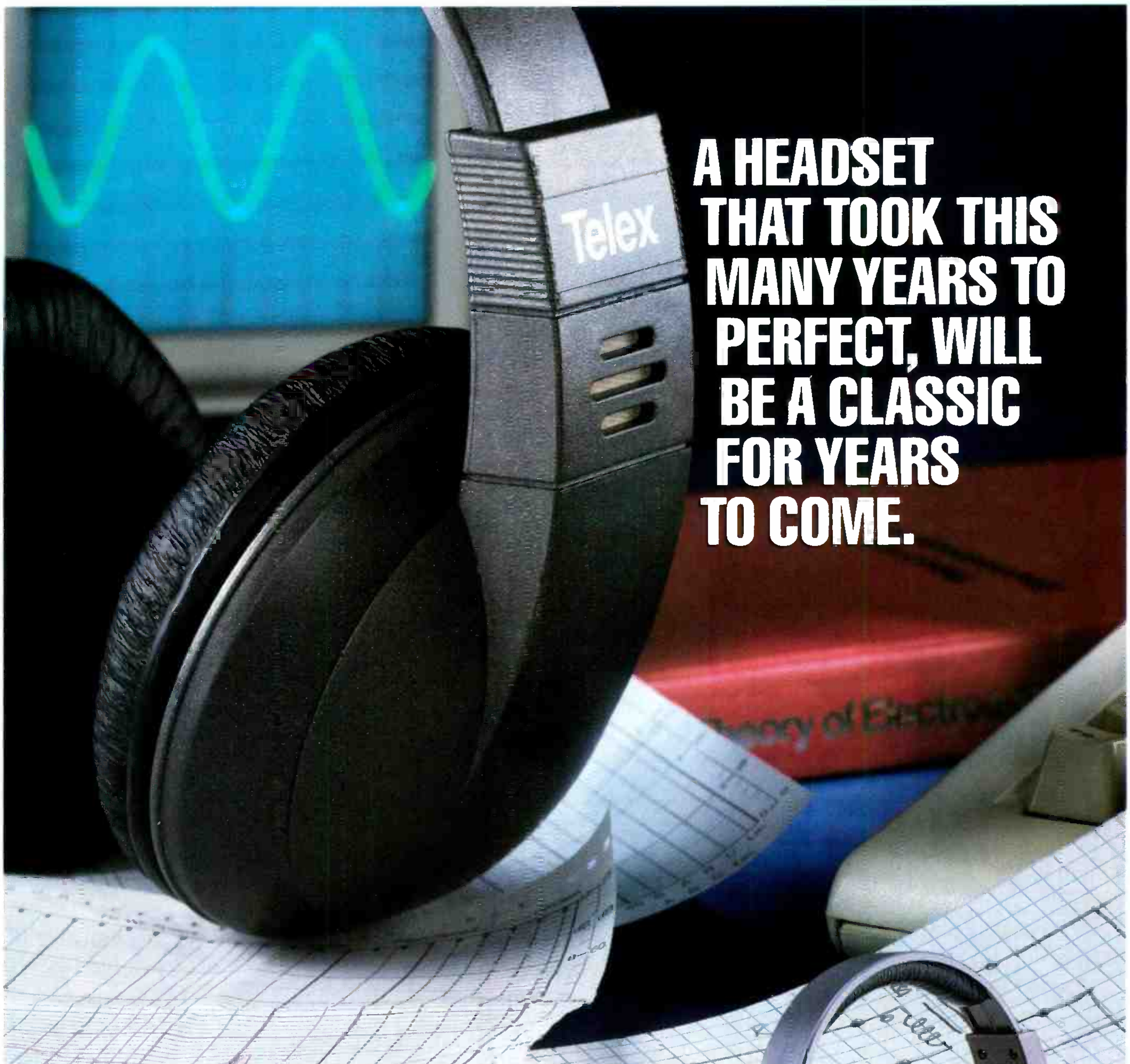


***Figure 2.** An AES/EBU distribution system showing the synchronization system.*

Switcher topologies

Several different switcher topologies are available for AES/EBU signal distribution. (One is shown in Figure 3.) One system uses time division multiplexing (TDM) to reduce switcher complexity. This method creates a large internal bus, which carries several input signals. The system decodes the input signal and changes it into an internal parallel digital signal, which occupies a specific time slot. The destination



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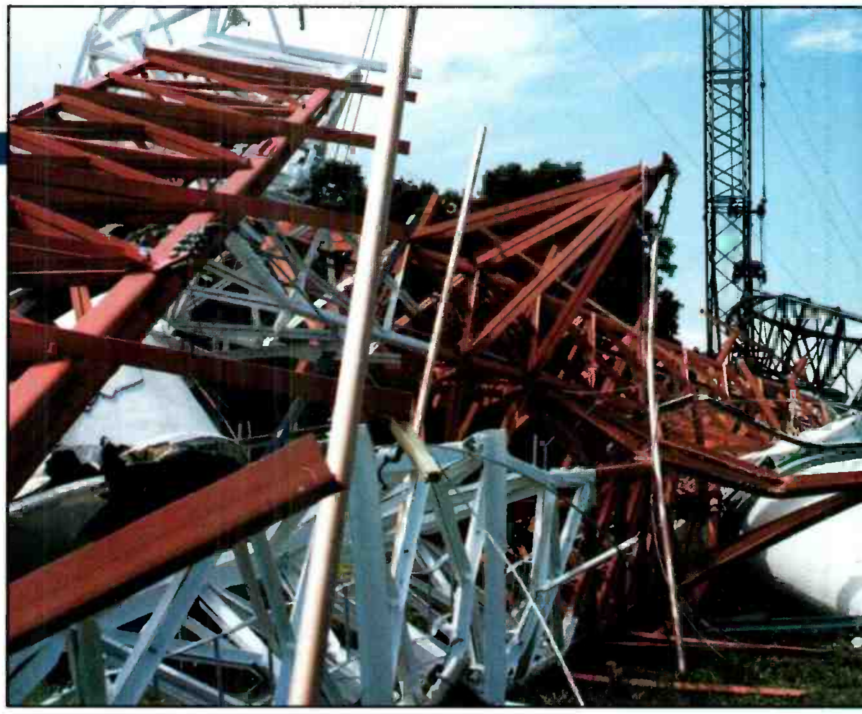


Courtesy of Tower Technology

Calculating the dollar cost of disaster.

By Martin Sacks

A 3-alarm fire roared through the high-rise building that housed the WGAY-FM/WWRC-AM studios in Washington, DC, this past February. After the fire had been put out, station personnel were relieved to find that their facility had suffered little damage beyond a dusting of soot and smoke. However, the fire caused so much structural damage in the affected areas that the entire building was condemned. The staff was prohibited from returning to its studios and offices for four days. Fortunately, station personnel had previously prepared a contingency plan to help make it through such a disaster. Neither the FM nor AM facility was off the air for more than a few minutes. The station maintained a program and commercial schedule, even though the staff was locked out of the facility. This contingency plan prevented what could have been a tremendous loss in revenue and business goodwill. This article will explain how to develop such a plan, and the financial benefits of doing so. (See "Hardening Broadcast Facilities," March 1990, and "Picking Up the Pieces," April 1990.)

Out-thinking doomsday

Preparing a disaster plan is not an easy task, but it should not be overlooked. Because there are so many kinds of emergencies, there is a strong possibility that your station will face one at some point.

Management should regard the time spent in disaster planning as an investment, because lost airtime is lost money. A station can insure facilities, equipment and even revenue, but no amount of insurance can recapture market share after listeners have tuned away and advertisers have taken their business elsewhere. Calculating the true dollar cost of disasters can quickly generate support for developing a disaster preparedness program.

Sacks is chief engineer, WGAY-FM/WWRC-AM, Washington, DC.

Developing the plan

The following four steps are a guide to disaster preparedness planning at your facility.

Step 1: List scenarios

Carefully consider all the possible disaster scenarios and their potential consequences to your operation. What is your facility's proximity to geological faults and flood plains? How often is your area affected by tornadoes, floods, hurricanes, crippling ice or snow? Also, consider your station's potential vulnerability to crime, civil strife or terrorism. Account for different types of emergencies. What problems could be caused by a short-term emergency that only lasts one or two hours, such as a leaky gas main or a bomb threat? What problems could be caused by longer-term emergencies of more than a day's duration? Don't be surprised if it takes a while to work through this process.

Don't build your list in a vacuum. Collect input from all departments, and don't limit the discussion to department heads. Often, the employees closest to the task can provide important information that someone higher up, hence further removed, may overlook.

Consider problems created by extended loss of electrical power, telephone service, heating and air-conditioning, water and sewage. Review the side effects of having to operate from an ENG van or from a "doomsday studio" located at the transmitter. How will you get by without telephone hybrids, remote controls, modems and other communications tools?

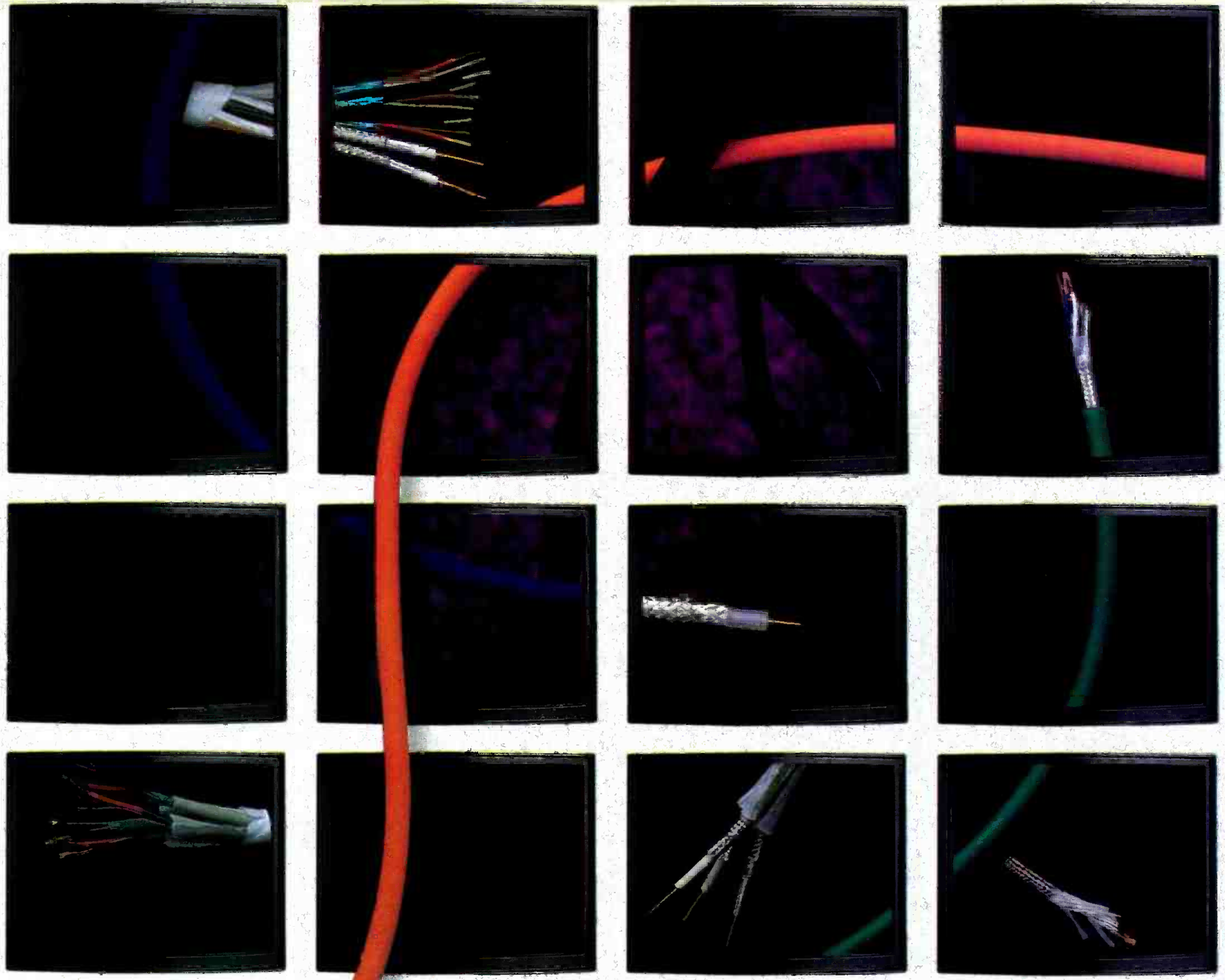
Imagine not having access to station computers and common reference materials, such as telephone lists and station wiring diagrams. (It might be worthwhile to store duplicates of important information and computer disks off-site. Remember to update them regularly.)

If you are new to a facility, enlist the help of previous engineers and subcontractors familiar with the facility. (This also will quickly increase your familiarity with the station.)

Step 2: Set priorities

Prioritize the station's systems. If the programming relies on satellite-delivery, emphasize backups for the satellite dish and receive electronics. In an all-news format, protecting incoming wire services, 2-way and microwave signals would also be paramount.

Understand that some systems cannot be practically backed-up. It is unlikely that



anyone could patch together a 12-tower directional array should six of the towers collapse in a tornado. It is feasible, however, to decide what kinds of hardware and test equipment should be on hand to patch the transmitter into one of the towers at a reduced power level. This would buy some time, and allow you to determine the next step with the information at hand.

Consider alternate program material. Backup systems are of little use if there is no programming with which to feed them. If you plan to rely on prerecorded programming, be sure it is stored with the emergency equipment. Also, include a legal station ID. You might think about preparing an emergency announcement tape as well.

When considered from a financial viewpoint, station traffic and logging functions deserve more emergency support than they often receive. These departments help generate revenue, and should therefore be included in contingency plans.

Last February, a fire roared through the building housing Washington's WGAY-FM/WWRC-AM facility. The contingency plans allowed the station to maintain programming during the four days the building was condemned. (Photo credit: Alan Felsen.)

Step 3: Make plans

Design the station backup system based on the priorities you defined in Step 2.

Resist putting off your planning sessions because funds are not immediately available. Not all contingency plans require large expenditures. It may be possible to construct backup systems with used or older equipment. Because usage is likely to be limited, state-of-the-art can take a back seat to functionality. Delaying your planning because funds might not be made available is a poor excuse. If a disaster does occur, you may have to explain why there is so much station downtime.

Contingency plans should not be overly complex. Having to carry out a complicated plan under tremendous stress is a recipe for disaster. Many external factors beyond your control will divert your focus. Work now to simplify the planning. As you plan, discuss with other broadcasters their needs. Agree to help each other whenever possible. The cost benefit of an arrangement like this is obvious.

Your emergency planning should also include developing relationships with lo-

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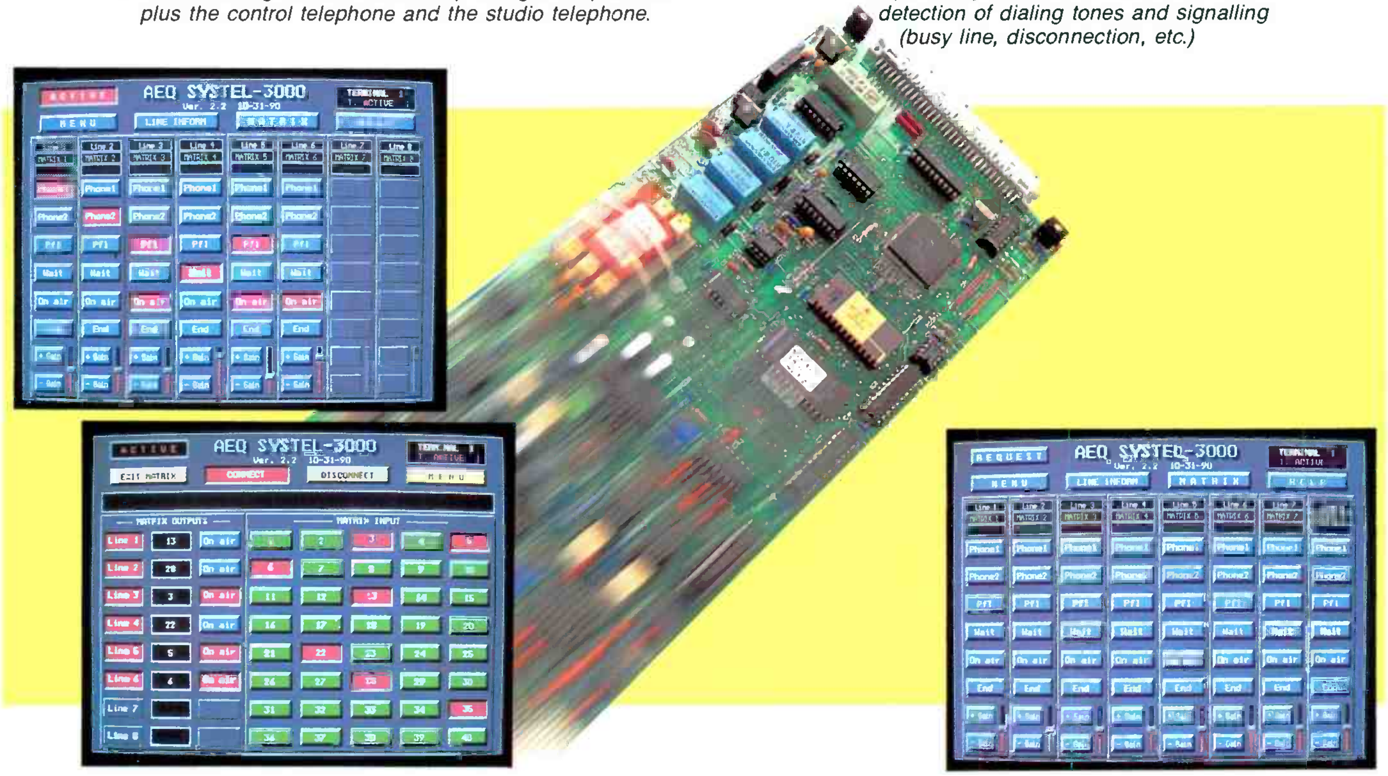

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cal emergency management agencies. These contacts can be invaluable in a time of need.

Step 4: Drill and test

Once you have created your plan and performed any installation, review the emergency procedures with the people responsible for executing them. As new employees join the company, explain to them the parts of the plan they will be asked to carry out.

If your plan takes advantage of outside facilities, stay in touch with your contact person. If the contact moves on, meet the successor and brief the person on the plan. Review the procedure for reaching the contact after hours, because that is when you will most likely need help.

Test emergency equipment regularly to assure its reliability. Verify that emergency stores are where they are supposed to be, and make sure they are in good condition. How would you feel if the station could not get back on the air because of something minor, such as a dead flashlight battery, or not having wire or a common hand tool?

The WWRC newsroom was restored to service after wiping soot off the equipment and temporarily replacing heat-cracked windows with plywood. (Photo credit: Joseph Nunemaker.)

Why plan for the worst?

Develop your plan so everyone knows what to do if the worst should occur. Conversely, realize that even the finest plans can be sidetracked by unforeseen events. However, the act of previewing potential trouble spots will go a long way toward smoothing problems out if disaster should strike. Not being prepared can ensure chaos in times of trouble. The potential for positive financial impact from good planning, and for negative impact from poor or non-existent planning, makes this topic too important to ignore.

Tower safety and disaster recovery

By Fred Moore

Careful preparation can be the key to preventing a devastating financial loss in any aspect of a broadcast facility, and towers are no exception. This preparation includes regular tower inspections to find the small problems before they become big dilemmas. Periodic maintenance is crucial to keeping any tower in top condition. A portion of this preventive maintenance program includes developing a disaster plan that details what to do if a tower is damaged or destroyed. Advance planning and routine maintenance will help ward off preventable tower failures. If the worst does happen, these steps can shorten the path to restoring your station to full operation.

Safety through maintenance

Tornadoes, ice and snow storms, hurricanes, gale winds and aircraft collisions are the most common disasters that can damage broadcast towers. You can't control nature, but you can ensure your tower is in peak condition with regular preventive maintenance. The Electron-

Moore is vice president, FWT, Fort Worth, TX.

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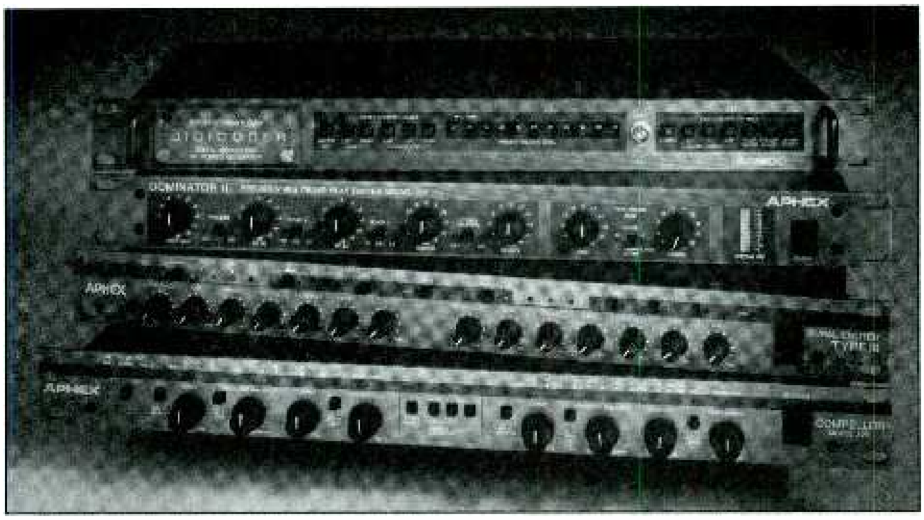

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ic Industries Association (EIA) details many inspection and maintenance specifications for towers. These help assure safety, and can extend the service life of the tower and the equipment it supports.

Every tower should have a major annual inspection. If the tower has experienced severe winds or heavy ice loads, or has been damaged in any other way, more frequent inspection may be required. Inspection and maintenance should be performed by authorized, trained personnel who are experienced in maintaining and adjusting towers. The inspector should note and record the overall condition of the tower, including structural members, finish, lighting, grounding and foundation. Any defects or problems should be photographed and referenced to their location on the tower.

- *Steel.* Structural members should not be bent or loose. Climbing facilities, platforms and catwalks should be secure. The weep holes (tiny perforations that prevent rusting by allowing condensation and precipitation to run out) should be checked and cleared of any obstructions.

- *Paint.* The tower finish should be free of rust or corrosion. The paint should be in good condition and meet FAA and International Committee on Aircraft Operations (ICAO) guidelines. The FCC is checking the condition of paint on broadcast towers, so be forewarned.

- *Guys.* Guy wire tension should be measured and recorded. Inspect all guy wire clamps and turnbuckles for rust and tightness. Verify that the turnbuckles have an anti-turn cable properly installed. Check for proper guy wire grounding. Inspect the condition of guy-damping systems and ice-sliding prevention hardware, if installed.

- *Antennas and supports.* Inspect the hardware used to secure antennas and feedlines. A poorly attached feedline or antenna could break off, and possibly damage other antennas or even the tower. Inspect coax grounding kits to ensure they are tight and rust-free. Make sure feedline and waveguide bolts are secure. Examine the feedline and antenna for corrosion, bullet holes or other damage. Remove bird or insect nests. In shoreline areas, remove accumulations of guano.

- *Marking lights.* Tower lighting is an important part of safe tower operation. Note the condition of bulbs, lenses and wiring. Change bulbs at the prescribed intervals. Attachment hardware, junction boxes and conduits should be weather-tight and secure. Make sure the tower lights turn on (and off) at the proper times, and that the strobes change intensity at the appropriate north-sky illumination levels. The operator should be able to detect tower light failure from the station control point. (This is a requirement, not an option.)

- *Foundation.* The inspection should include a thorough examination of the

ground around the guy anchors and tower base. Check the earth around these fixtures for settlements, movement, erosion, standing water, drainage and root growth that could damage the foundation. Note the overall condition of the concrete at the base of the tower. Look for cracking or splitting, chipping, honeycombing or low spots that could collect moisture and contribute to anchor bolt corrosion. Check the condition of the grout and the tightness of the nuts on the tower base. Also, inspect the tower grounding connections and the condition of the tower's lightning-protection hardware.

Although it may seem like an obvious task, detailed recording and filing of maintenance information from such inspections is rarely kept as accurate or current as it should be. Such documentation is important to future tower maintenance. It may also be your best ally if the tower ever falls and your station is sued, or the insurance company decides to challenge you on the tower's structural integrity.

Call before you add

Self-made tower disasters are often the result of owners adding additional equipment without first consulting a qualified tower engineer. Never place more antennas or feedlines on a tower than it was originally designed to support. Consult a structural tower engineer if you do need to add equipment. Often, a tower can be modified to increase its capacity. However, never attempt such work yourself, consult an expert. Overloading a tower without consulting the manufacturer not only increases the risk of tower failure, but it may also make the operator liable.

Consult the EIA and local building codes for wind- and ice-loading requirements. Load restrictions vary, depending on the tower's location. Towers on coastlines are more susceptible to gale winds, hurricanes and salt erosion. Towers in the Midwest are subject to heavy icing and other extreme conditions, so know the official wind and ice rating of your tower and be sure it meets the local requirements.

When disaster strikes

When disaster strikes a tower, first determine if any of the tower can be salvaged. Depending on the degree of damage, it may be possible to reuse some parts in restoring the tower. Most natural disasters, however, leave behind a twisted pile of steel with little chance of repair.

Once a tower has been damaged or destroyed, the operator must restore the tower and its equipment as quickly as possible. Preplanning can help.

A tornado once struck a 300-foot-guyed tower near Bloomington, IL, at 8 p.m. on a Wednesday. The tower crashed to the ground with no chance of salvage or repair. As part of a preset disaster recovery plan, the tower owner kept an emergency tower on hand at its Texas headquarters. Six hours after the tower fell, the emergency tower, along with essential equipment and antennas, was put on trucks headed for the Illinois site.

The wrecked tower supported 12 antennas when it went down. The emergency restoration tower was designed for only nine. Fortunately, engineers were able to design quick modifications so the tower would meet the 12-antenna load. After a rapid cleanup of the old tower and site, the new tower, stacked with antennas and waveguides, was fully erected by 11:30 p.m. the following Sunday. Service was completely restored Monday morning (only four days after the tower's fall), with a minimum loss of revenue.

Having access to an emergency tower as part of a disaster plan lends to rapid deployment and minimum downtime. If you don't have an extra tower available, you may be able to find a tower manufacturer that has a supply of towers for backup purposes. If a tower goes down and the manufacturer has to build the tower from scratch, it increases the broadcaster's wait to get back on the air. Downtime translates into lost dollars, and every hour counts.

Don't be surprised

The first step toward preventing a tower disaster lies in regular inspections and thorough preventive maintenance. The second step lies in developing contingency plans before disaster strikes. This advance planning will help you recover more quickly when and if disaster does strike. These two activities can help your station minimize the financial loss that often accompanies the damage or loss of a tower.

Generator maintenance

By Morgan Smith

A backup generator is only valuable if it will start when needed and carry the required load. A properly and systematically maintained standby generator will last for many years, and will start and run when it is needed. This article lists 11 tips for maintaining generators and establishing a scheduled maintenance program.

• *Information.* The primary source of information on a generator is the manufacturer's specifications and recommendations booklet. This, together with the manufacturer's local representative, will provide you with everything you need to know about your emergency power system — from oil type and viscosity, to what items should be included in a routine maintenance program.

The next important information source is a log book of maintenance activities. Make a simple generator performance check list and put it on a clipboard near the generator controls. (See Figure 1.)

Smith is a building engineering manager in Salt Lake City, with experience in broadcast properties.

• *Generator exercise.* The less an engine is run, the harder it will probably be to start. A good rule of thumb is to exercise the generator once a week. Run it long enough to warm the engine thoroughly. This will drive out moisture, prevent rusting and stop water from accumulating in the oil. Periodically observe engine temperature during operation to ensure that it is normal. Check alternator current and voltage during operation to make certain that the batteries are charging properly.

Load testing, working the generator up to its rated capacity, is an important and often neglected part of a generator run program. Load tests should be performed annually. Some facilities do them monthly. Use the tests to discover problems or to ensure that none exist. Document these tests as part of the generator maintenance program.

Some facilities have convenient loads, such as snow melt systems in driveways. A facility can also rent a dummy load. This can help accurately measure the generator's output and frequency.

When testing, check the AC waveform with an oscilloscope. Look for flattopping at full load or other forms of distortion. If problems occur, contact individuals trained in the service of load and speed regulation devices to obtain corrective maintenance. Usually, the representative can determine the cause and provide a remedy.

• *Fuel.* A generator needs an adequate supply of clean, fresh fuel. Gasoline and diesel fuel deteriorate with age (gasoline separates, diesel turns to jelly). Additives can extend diesel fuel storage life. Exercising the generator provides an opportunity to use and replenish fuel.

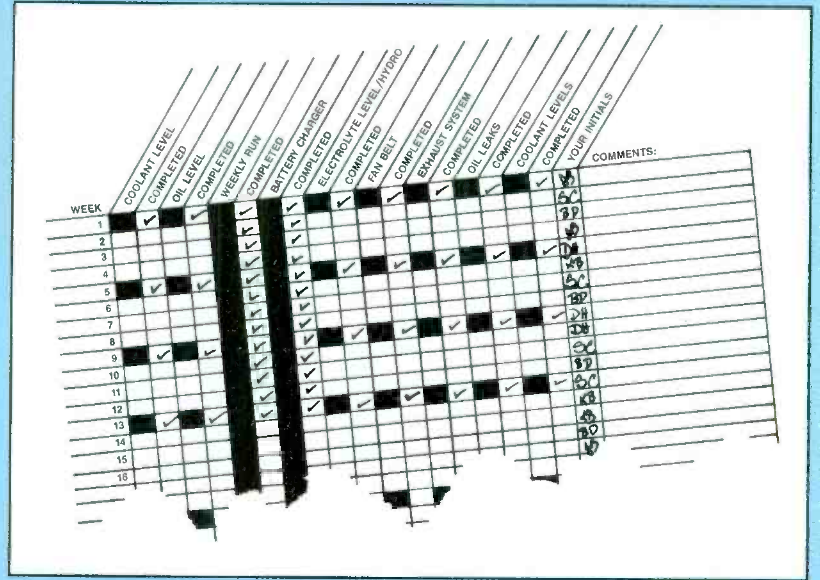

***Figure 1.** A suggested maintenance schedule.*

Document the overall age and condition of the fuel. If the fuel becomes too old, it may be necessary to pump the tank and replace it with a fresh supply. Some fuel distributors are willing to work with customers on an exchange program.

Frequently check liquid petroleum and propane tanks for leaks. These fuels are heavier than air, and leaks tend to accumulate in low spots, which can be extremely hazardous.

Local codes are becoming more specific about fuel storage. Underground storage tanks may rust through and begin to leak. Many facilities are replacing underground tanks with above-ground tanks, or are installing underground tanks that are corrosion resistant. Check local codes when considering any tank modifications.

• *Engine oil.* Change engine oil as often as necessary to keep it clean. At the minimum, follow the manufacturer's recommendations. In units that are not used often, change oil on an annual basis.

Warm the engine thoroughly before the oil change. This will suspend sludges and solids for removal with the oil.

Have an oil analysis performed annually. The cost is minimal and worth the effort. The lab report will indicate the

presence of extraneous material in the oil (babbitt from bearings, cast iron, steel, aluminum, acids, water and fuels). Use these results to determine normal or abnormal engine wear. Use year-to-year comparisons to evaluate engine condition and special maintenance considerations. Ask your generator representative to recommend a lab.

• *Filters.* Clogged air filters rob engines of power and fuel economy. Generators may have oil filters, fuel filters, air filters and, sometimes, filters over the radiator. Change oil filters with the oil, and check or change all the other filters at the same time. The oil change is also a good time to perform other maintenance items, such as checking coolant levels, servicing batteries and changing belts. Give the engine a good run afterward to check for leaks and ensure proper operation.

• *Coolant.* The more a generator runs, the more its coolant levels should be checked. Never wait more than a month. Use a good grade of antifreeze, even though the generator might be in a hot climate. Cooling water can rust engines. Antifreeze contains inhibitors that coat and protect metal surfaces. When antifreeze becomes contaminated or dirty, replace it.

• *Batteries.* A hand crank is not standard generator equipment, so generator batteries are important. Stationary backup generators usually have a battery trickle charge circuit that is powered from the building's electrical system. Its purpose is to maintain charged batteries. If the charger current is too low, the battery will run down; too high of a current will boil the battery dry, and may destroy it. The battery manufacturer can specify the correct maintenance charging current and voltage.

Consider installing a second set of batteries equipped with a separate starting solenoid or heavy duty manual alternator switch. If one set should fail, it will not take long to engage the second set to start the generator.

Battery maintenance consists of checking electrolyte levels and charge levels using a battery hydrometer. Electrolyte specific gravity relates directly to battery charge condition. Check batteries weekly. Check battery cable connections at battery terminals, solenoids and starter terminals for loose connections.

Batteries can be dangerous, and extreme care must be taken when working with them. Large batteries have high short-circuit current, so protect them from short circuits and mechanical damage. Do not short-circuit a battery or terminal connection to ground, because damage to tools, batteries or personal injury may result. Do not store batteries directly on concrete floors, because concrete affects batteries. Battery acid can burn clothes and skin, so use caution and wear protective clothing and goggles. It is good practice to have a face and eye wash station close at hand when doing battery maintenance.

• *Block heaters.* Most diesel generators need block heaters, which keep the engine block warm. Such heaters are also a good idea with gasoline or LPG-fueled generators in colder climates. Block heaters are electrical heating elements that extend into the engine water much as the elements on a home water heater. A temperature switch controls the heater output. A quick look at the engine temperature gauge when the engine is not running provides a good indication of block heater operation. Check with the manufacturer for recommended block standby temperatures. Check the heater elements and temperature switches monthly or at the manufacturer's recommended interval.

• *Exhaust systems.* Exhaust systems often extend through a building to the outside. Check pipes to ensure that they are free of combustibles and are sufficiently removed from combustible building members to prevent fires. Check exhaust systems for leaks, rust and blockage. To prevent water from running into the engine during storms, make sure that the top end-closing system works. Put these checks on a monthly list.

• *Transfer switches and wiring connections.* Check wiring connections each year. Loose connections generate heat that destroys switch gear and terminal boards. Before tightening a connection, disconnect the power source, and use a reliable volt meter to verify the circuit is not energized. If circuits cannot be turned off, an infrared scan can detect heat from loose connections. Perform infrared scans while the circuit is under high loads.

Check transfer switch contacts for severe pitting and wear. Replace worn contacts — do not file them. Check timing on automatic transfer switches. Transfers must be made "in-phase" to avoid power bumps. Out-of-phase transfers can damage equipment, blow fuses and trip circuit breakers. Your local representative can assist in solving transfer timing problems.

Exercise circuit breakers regularly. Circuit breaker contacts are self-cleaning by the wiping motions of closing and opening. Exercising breakers minimizes breaker overheating and nuisance tripping during normal loads.

Every department in the station may feel their special project deserves to be powered by the emergency power system. But emergency circuits are just that — to keep the station on the air in an emergency with only minimum capabilities. Too much extra equipment may overload the generator.

• *Cleanliness.* Keep the generator area clean. Watch for oil leaks, and repair the generators when they occur. Water pumps occasionally lose a seal and leak antifreeze, which can lead to low coolant levels and engine overheating. Replace cracked fan belts.

All these conditions are easier to spot if the generator room has a clean floor, and the generator is wiped down occasionally. A clean environment demonstrates competence and concern for safety. This can go far toward calming the concerns of fire marshals and building inspectors.

A final note about safety: generators are remotely operated mechanical devices that burn highly combustible fuels to produce electricity. This makes the generator room one of the most hazardous locations in the station. Never work on the engine without disabling the remote or automatic start system. (Test start from the remote-control point when you are done to make sure you've re-enabled the system). Work calmly and carefully on the electrical system. Never attempt to do maintenance in the generator area if you are drowsy or smoking.

New products

Audio control

By AIR

• **Model 500TV:** microphone processor for TV studios, remote locations; inputs match mic to line levels; servo-balanced output avoids ground loops in console installations; combines compression, de-essing, expansion and equalization to reduce hollow room noises and effects; protects against studio environment noises.

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FO data link

By BT&D Technologies

• **XMT1300-1.2, RCVR1201-1.2:** "Logic-to-Light" transmitter, "Light-to-Logic" receiver for high-speed fiber-optic link with logic-level interface; data rates to 1.2Gbit/s to distances of 10km; 28-pin DIP packages have 1.5×1×0.25" dimensions.

Circle (353) on Reply Card

FDDI link

By AT&T Microelectronics

• **ODL 125 series II:** light-wave data link with transmitter, receiver modules; permits communications between computers using FDDI standard at data rates in 20-125Mbit/s range; 1,300nm wavelength permits typical link spans to 3km. (ODL 70 similar, permitting high-speed and low-speed operation.)

Circle (352) on Reply Card

Digital meters

By DI International

• **DPM301, 302 series:** digital panel meters covering ±199.9mV to ±600VDC and ±1.99μA to ±1.999ADC; five models each available for voltage or current needs; 0.56-inch LED indicators; floating input; automatic zero, polarity; overrange indicators; connector, mounting bracket supplied.

Circle (356) on Reply Card

Animation equipment

By Digital Arts Film & Television

• **Motion controller:** transputer-based motion control system; portable unit operates in real time; multi-axis motion permits 360° pan, 270° tilt; hand-held remote or computer console control; operates on horizontal or vertical rail; adapters available for most 16mm, 35mm and video cameras.

Circle (357) on Reply Card

Product literature

By Pomona Electronics/ITT

• **Test accessories catalog:** 140-page publication spans a wide range of electronic testing products; test clips for microprocessors to measurement probes and probe adapters; accessories for SMT devices.

Circle (374) on Reply Card

Circle (56) on Reply Card

Circle (57) on Reply Card

Circle (58) on Reply Card

At Ampex, we engineer so you can create.

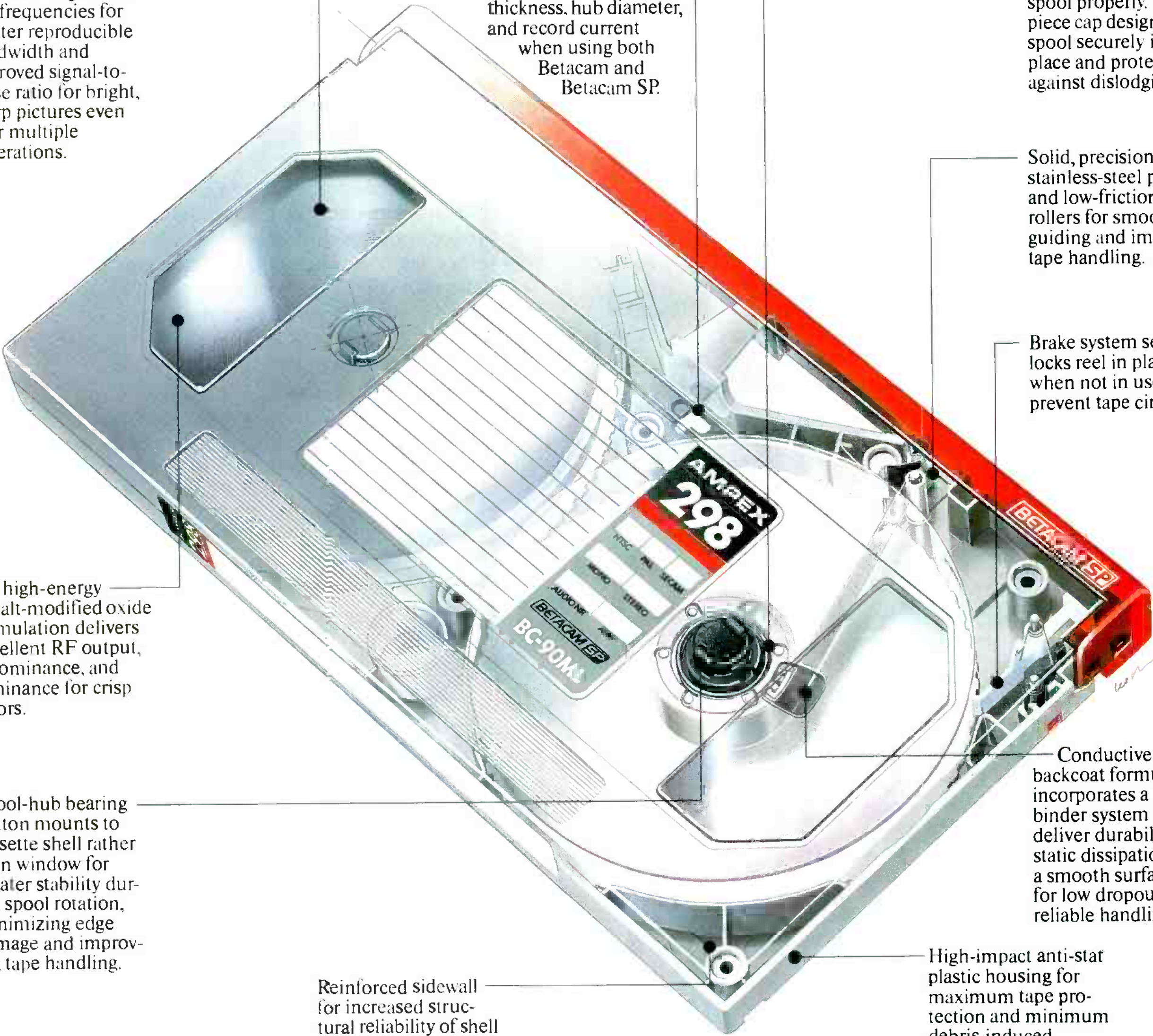
298 metal particle formulation enables the use of higher carrier frequencies for greater reproducible bandwidth and improved signal-to-noise ratio for bright, sharp pictures even after multiple generations.

Identification tabs allow machine to sense tape type and adjust for thickness, hub diameter, and record current when using both Betacam and Betacam SP.

Hold-down spring with precision-ground ends helps center spool properly. One-piece cap design holds spool securely in place and protects against dislodging.

Solid, precision stainless-steel pins and low-friction rollers for smooth guiding and improved tape handling.

Brake system securely locks reel in place when not in use to prevent tape cinching.



198 high-energy cobalt-modified oxide formulation delivers excellent RF output, chrominance, and luminance for crisp colors.

Spool-hub bearing button mounts to cassette shell rather than window for greater stability during spool rotation, minimizing edge damage and improving tape handling.

Conductive carbon backcoat formulation incorporates a tough binder system to deliver durability, static dissipation, and a smooth surface for low dropouts and reliable handling.

Reinforced sidewall for increased structural reliability of shell even under rugged handling conditions.

High-impact anti-stat plastic housing for maximum tape protection and minimum debris-induced dropouts.

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Circle (59) on Reply Card

Audio accessories
By Har-Ken Specialties
• **Product brochure:** data on various connectors and adapters designed to solve audio signal problems; series includes phono, phone and terminal to XLR connectors, as well as in-line transformers and attenuators.
Circle (358) on Reply Card

Instrument literature
By John Fluke Manufacturing Company
• **1991 product catalog:** features new event counters, timers, frequency counters, RCL meters and DMMs; 20-page catalog includes established lines as well as numerous accessories available through the Fluke dealer network.
Circle (359) on Reply Card

Product literature
By KOBOLD Instruments
• **Flow meter brochures:** product overview covers a variety of fluid measuring devices, including items usable in water-cooled transmitter systems.
Circle (360) on Reply Card

Video drive link
By Communications Specialties
• **ScreenSender:** transmitter, receiver unit permits PC TTL/digital RGB monitors to be located as far as 1,000 feet from computer; may also be used to drive multiple MDA, CGA, EGA or Hercules monitors from one computer.

Circle (354) on Reply Card

Computer accessories
By Lamp Technology
• **B9591T Eternacells:** lithium replacements for PC clock circuits; high-reliability units with 10-year shelf life hold CMOS memory of clock and PC configuration; available for almost all computer systems.
Circle (361) on Reply Card

Equipment rental/lease
By Leasametric
• **91/92 catalog:** 200-page publication provides detailed information on test equipment available on rental/lease or purchase plans; large inventory for quick service.
Circle (362) on Reply Card

Lighting control
By William McIntire Enterprises
• **ShadowMaster Plus:** flexible lighting controller in 3×4×7-inch package; three 20A dimmer circuits with U-ground connectors; dual-level with flicker generator for special effects; 16 memories include digitized events, including candles, lightning, firelight and chaser/tracer effects.
Circle (382) on Reply Card

Power protection

By Liebert

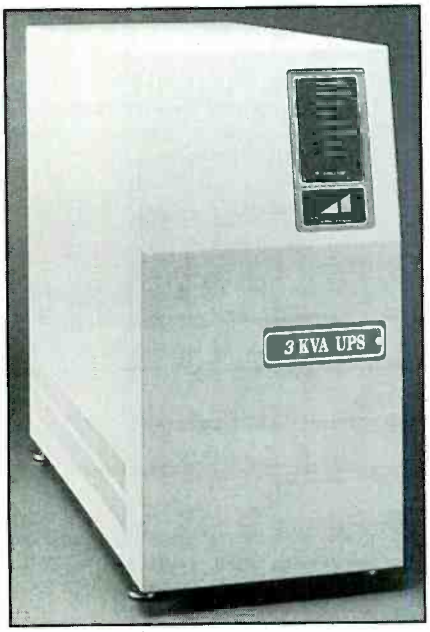
• **ActiveTracking filter:** institutes surge suppression, transient and ringing attenuation into power sources for computer and telecommunications equipment; supplements UPS and power conditioning units; 30-4,000A range in single and 3-phase models over 120-600VAC; surge current capacities to 150kVA per phase.

Circle (363) on Reply Card

Power conditioning

By LorTec Power Systems

• **Series 1000:** UPS products for isolation and protection of computer-based systems; series covers range from 3kVA to 15kVA in 120, 208 and 240VAC inputs and outputs; eliminates transients, harmonic distortion, spikes, brownouts and power losses.



Circle (364) on Reply Card

Reference chart

By Motorola/Literature Distribution

• **USA Frequency Spectrum:** color chart shows changes to allocated frequencies governed by FCC part 15; also includes coverage to 4GHz with clear indications of AM, FM broadcast bands.

Circle (370) on Reply Card

Wiring management

By NELCO Products

• **Releasable cable ties:** cable ties of Type 66 nylon for indoor or outdoor installations; quick-release design permits tie to be opened if wiring bundle must be changed; in $5^1/_2$-inch, $9^3/_4$-inch lengths for bundles to 3-inch diameter.

Circle (371) on Reply Card

Circle (62) on Reply Card

Circle (63) on Reply Card

Personal workstations

By Modgraph

• **GX-series computers:** portable units based on 80386/486 microprocessors; caching feature option; includes Ultra-VGA color monitor with 256-color 800×600 resolution; expansion slots; hard drive options to 200Mbytes; 3.5-inch, 5.25-inch HD floppy drives; co-processors, external high-resolution monitor options available.

Circle (365) on Reply Card

MIDI equipment, data

By Opcode Systems

• **MIDI Translator:** entry-level interface connects a Macintosh printer or modem port and up to three MIDI instruments; permits computer software to control creation, recording and editing of music with sequencer or synthesizer; links music to multimedia presentations.

• **Book of MIDI:** interactive HyperCard stack with information about MIDI equipment; requires Macintosh PC with hard drive and 2Meg RAM.

Circle (372) on Reply Card

Product literature

By Plitron Manufacturing

• **Toroidal transformer catalog:** 12-page brochure discusses selection guidelines for toroidal power transformers for replacement or original equipment designs; product line includes audio output and custom application devices.

Circle (373) on Reply Card

Component literature

By D.A.T.A. Business Publishing

• **D.A.T.A. Digest:** compendium and cross-reference of discrete semiconductors and ICs; high-reliability component information includes device function, MIL spec equivalents, military-to-commercial cross-reference, JEDEC/package outlines, manufacturer directory.

Circle (355) on Reply Card

Computer-to-video

By Visionetics International

• **VGALink:** combines video with VGA computer graphics displays to produce NTSC or S-VHS signals; gen-lock to external NTSC; overlay, chroma-key, fade features; supports VGA 640×480 mode.

• **DIVA:** desktop-integrated video animation system; produces multimedia productions with image capture feature; combines graphics with live video; output recordable on industrial VCRs; suitability for broadcast not stated.

Circle (380) on Reply Card

Monochrome monitors

By Wells-Gardner Electronics

• **Customiser III monitors:** P4 phosphor video monitors with 900-line resolution; five 14-inch diagonal CRTs; available with fixed horizontal frequencies between 15.7-35kHz; 10% linearity; low power consumption units.

Circle (381) on Reply Card

EMI shielding

By Spraylat

• **Wave Guard:** water-based copper paint; produces more than 60dB attenuation of EMI/RFI signals between 10kHz-1GHz; available in flat, semi-gloss and gloss finishes; significant cost savings over normal shielding materials.

Circle (377) on Reply Card

Test signal source

By Standard Research Systems

- **Model DS345:** synthesized function generator; 1μHz resolution to 30MHz; sine, triangle, ramp and square waves; synthesized modulator for phase linear/logarithmic sweeps and standard modulation.

Circle (378) on Reply Card

TV/video modules

By Tektronix

- **11T5H, 11A34V:** video trigger and video amplifier for TEK 11000 series oscilloscopes, DSA 600 signal analyzers; multistandard trigger and wideband amplifier support signal frequencies to 1,280 lines per frame for HDTV and computer graphics applications; trigger includes display clamping; 4-input device for component video systems.

Circle (379) on Reply Card

EMI measurements

By Rohde & Schwarz

- **System EZ-10:** 4-wire T-network system covering ranges from 9kHz to 150MHz in two models; designed for electromagnetic compatibility measurements; permits measurements on lines that may carry phantom-powering potential, including ISDN communications equipment.

Circle (375) on Reply Card

UPS protection

By SOLA/General Signal

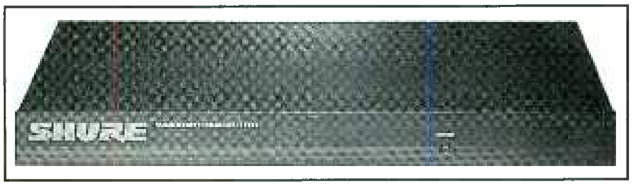
- **CPS II power supply:** uninterruptible power supply for computer-based equipment; for 50-60Hz systems in 10-100kVA power range; 5-section design with rectifier/charger, battery assembly, PWM inverter, static switch and manual bypass; 91% AC-to-AC efficiency; sealed lead acid, NiCad battery options.

Circle (376) on Reply Card

Wireless mic accessory

By Shure Brothers

- **WA 400 amplifier:** 2-input, 8-output amplified antenna distribution unit; permits up to four diversity wireless mic systems to operate from two antennas; may also be used for eight on-diversity mic systems; rack-mount unit usable with most wireless mic receivers.



Circle (393) on Reply Card

Safety standards data

By Underwriters Laboratories

- **1991 standards catalog:** guide to new and revised standards for safety; covers topics relating to the workplace as well as environmental concerns.

Circle (405) on Reply Card

Product catalog

By RITTAL Corporation

- **Innovations '90:** describes full line of racks, consoles, equipment enclosures for communications, climate control and power distribution applications.

Circle (390) on Reply Card

Circle (66) on Reply Card

Circle (67) on Reply Card

Automation control
By Utah Scientific

• **EMC:** combines Motorola 68000-series processor, custom operating system for distributed processing and intelligent machine control; interfaces with existing Utah Scientific master control switchers; capability for serial and parallel device control.

Circle (406) on Reply Card

Monitoring accessory
By Videoquip Research

• **HA-2 headphone amplifier:** 20W/ channel amplifier unit drives two headsets; volume level controls for each set of headphones; four models available; −2, −2B are desktop units with unbalanced RCA or 1/4" TRS input connectors; −2R, −2BR are rack-mount versions.

Circle (407) on Reply Card

Airflow monitoring
By Warren G-V

• **SAF series sensor:** solid-state unit detects airflow by close tolerance measurement of ambient temperatures; output drives logic circuits, alarms, relays for protection against overheating in air-cooled electronic systems.

Circle (408) on Reply Card

Power line cleanup
By Staco Energy Products

• **SPP-2000:** AC power conditioner; plug-compatible with almost any computer system or product requiring 5-15kVA single phase power; distribution panel feature can be customized with standard NEMA receptacles.

• **AC voltage controls:** 0-560VAC 3-ϕ output from 240VAC or 480VAC 3-ϕ input; output voltage selection from raise-lower push-buttons or FRC-10 series 1/4% regulation controllers.

Circle (400) on Reply Card

Audio management
By Target Technology

• **Quad-5 router:** enables destination control of four input signals to any of four outputs; permits mono-sum derivation from any two inputs; unbalanced X–Y output for CRT display to monitor stereo coherency; low distortion with signals to +26dBm.

Circle (401) on Reply Card

Electrical protection
By Sola/Unit of General Signal

• **Interact UPS line:** 3kVA and 5kVA units added to series to cover computer systems such as IBM AS/400 models, CAD/CAM workstations, file servers, telecommunications products; LAN and WAN networks; protection against power disturbances, blackouts; RS-232 port for mode control, metering, alarm messages, threshold level adjustments.

• **CPC line expansion:** additional 60Hz 3-phase systems for power ratings of 50kVA, 75kVA and 100kVA; protects against brownout, high voltages, surges, sags, spikes, noise; reduces odd-order harmonic currents produced by non-linear products by conversion to near sinusoidal currents.

Circle (396) on Reply Card

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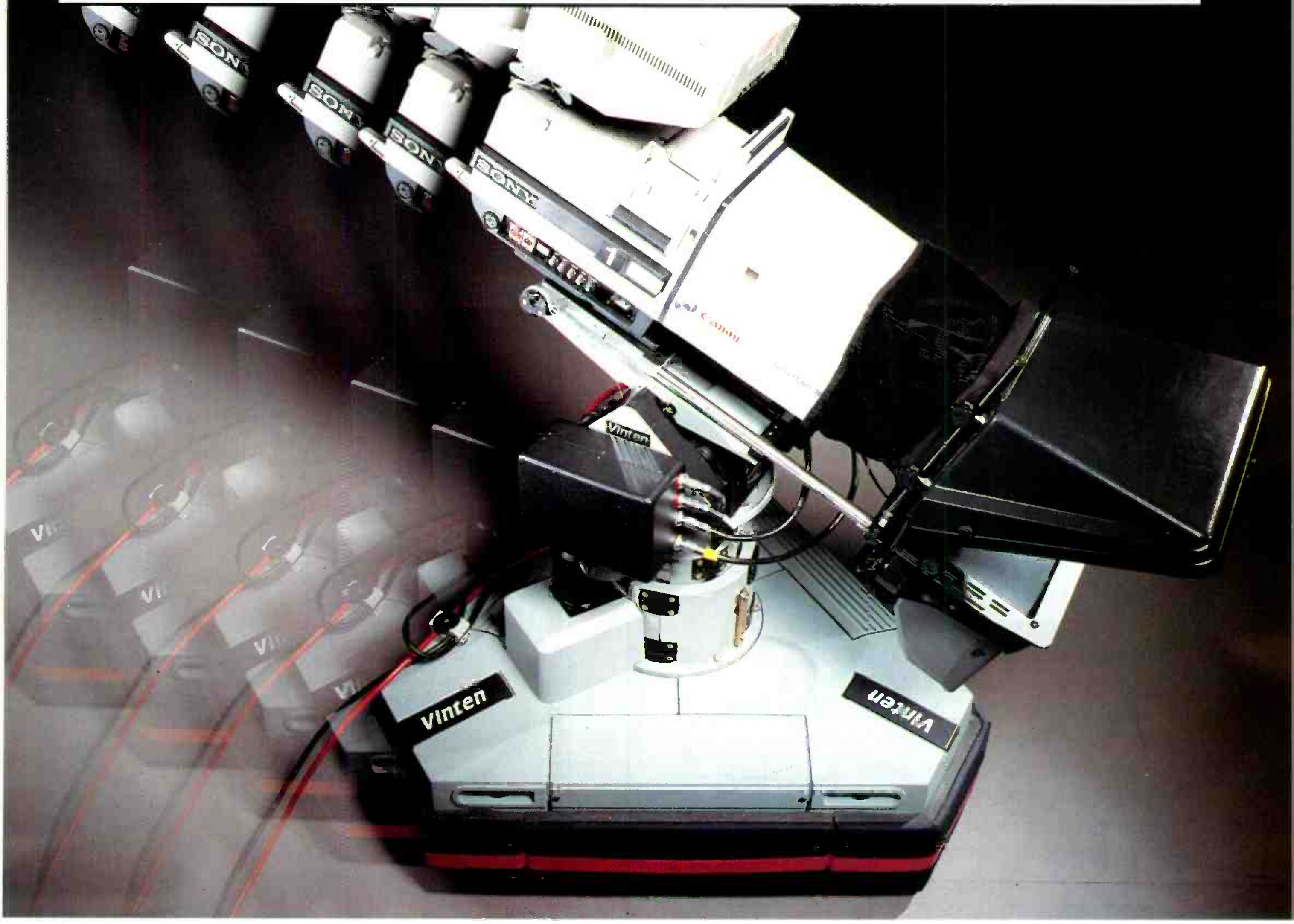

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A single MicroSwift system can consist of up to eight cameras on free navigating and static pedestals as well as on fixed heads, storing up to 500 shots each. A wide variety of control system include control panels that can also operate with a graphics tablet. And now, there's a new Touch Screen control system with its own trim panel.

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Circle (71) on Reply Card

Preview

August...

VIDEO TECHNOLOGY SPECIAL REPORT

- **Comparing the Options in Advanced TV Systems**

Engineers need to understand the basics behind some of the proposed advanced TV systems. The article looks closely at the theory and technology involved in some of the proposed systems. Understanding these systems is the first step to being able to make knowledgeable choices in advanced TV equipment. A related article will compare the various formats for HDTV audio systems.

- **Standards Conversion**

Converting between different types of video signals is neither easy nor impossible. *BE* takes a look at the processes available to convert your signal to one that your neighbor can use.

- **Connecting PC Video to NTSC**

Many broadcast stations and post houses are looking for ways to get the high-quality images from their PC onto their video recorders and broadcast chains. The process is not as simple as it might appear. Editors draw on their experiences in video graphics and PCs to lead a path to successfully moving images from the PC to professional video.

- **Digital Audio Workstations**

Recent technological improvements have brought the digital audio workstation to broadcast facilities. The article looks at some of the features found on many of the systems. It also looks behind the front panel, providing an insight into the technology used.

September...

AUDIO-VIDEO CONTROL SYSTEMS

- **Interfacing Small-Format Editing Systems**

As technology reduces the size and cost of cameras and recorders, engineers must find ways to interface the new equipment into fast-paced editing suites. The article looks at the requirements of the new small-format equipment, and how the new hardware can be interfaced in edit and post suites.

- **Routing Digital TV Signals**

Digital TV signals represent new challenges to control and routing systems. Switching and distribution of these complex waveforms requires special equipment and careful planning to avoid expensive mistakes. The article looks at some of the new technology that makes the process easier.

- **Interfacing Multiple Control Systems**

Automation comes to life as video and audio equipment rely on one of two common standards: SMPTE and MIDI. The problem is when one standard must talk to the other. The article takes the mystery out of connecting what sometimes appears to be incompatible devices.

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AA/EEO

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JIMMY SWAGGART MINISTRIES is seeking audio maintenance engineer. Facilities include Neve and Harrison consoles. Otari 24-track, Studer ¼″ machines. Must have 2-4 years experience. Contact: Ed Smith, Director TV, 8919 World Ministry Ave., Baton Rouge, LA 70821. 7-91-1t

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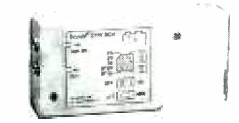

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