EMBER-DECEMBER 1985

db

THE SOUND ENGINEERING MAGAZINE

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ON THE COVER

• This month's cover features Control Room One at WGBH Radio in Boston, Ma. This control room utilizes a Neve console, Studer tape machines, and Analog and Digital Systems' ADS-910 monitors for live radio production and audio/video pre-production.

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Letters

ISRAEL IS HOME

To The Editor:

This letter is in regard to "Sound In The Holy Land" in the March/April 1985 issue of *db*. Firstly, thank you for showing Israeli studios in a favorable light, and at last bringing to the fore our thriving music industry.

Secondly, although I was thrilled to see a special profile about myself, I was horrified as to the number of misquotes and my apparent attitude towards British Studios, as suggested by the author.

My reasons for coming to live here in Israel are many, but to say, "He did not see much room for advancement in English Studios," must be the misquote of the year!

During 1981, after having progressed from "Tape Op" to "Senior Engineer,"and within the space of four years, I was looking to advance to a larger studio or company. But due to the poor economic climate of the British music industry and the shortage of work around the London studio scene at that time, I could see no immediate room for advancement.

While on holiday in Israel and visiting many studios here, I was offered such a tremendous amount of work that I decided to give it a go for a year or so.

The author's quote seems to me, and to others who have read the article, like a "kick in the teeth" to the London Studios where I trained and worked happily for 5 years, with world class equipment and some of the best musicians in the world. London has always been, and will continue to be, THE leader in recording and number one in the world. But Israel is, for now, my new home.

Having had one of our most successful years to date, with an average of 120 hours per week, we are in the process of designing a new studio complex. In the meantime, we have upgraded/added to much of our equipment, as follows:

GAL-KOL STUDIO UPDATE

Studer A810 1/4-in. master recorder
Lexicon PCM-42 digital delay
Yamaha monitor amps
Orban 53 A de-esser
Drawmer DC 221
Drawmer DS 201
EMT 140 stereo plate
Sankon CO 41 mics
Pearl drums
Kawai piano

Graeme Jackson

DOUBTFUL IMPRESSION

To The Editor: I read your Monster Cable ad in the Sept/Oct 1985 issue of *db*. I'm convinced that you are sincere in your convictions regarding this product. Yet I have been told that there is no basis in fact for the audio improvements claimed.

I have never seen an engineering article discussing technical reasons for the improvements claimed for audio transmission. Normally, technical breakthroughs are accompanied by informative engineering in the reputable audio magazines.

At this point I must conclude that any engineering expended was applied to a clever sales effort and the impressive appearance of the product.

Helmut Vles

MEDIUM DEVELOPMENT

To The Editor:

The sound engineering community has been long overdue for an article on the likes of "Have Wire Will Travel" (Sept/Oct 1985). Ian Eales is most perceptive in his description of the effects of passive components on sound quality. I'm glad to see that people in the recording world are finally beginning to acknowledge what some of the audiophile community has known for years! Maybe now more record companies will begin to produce better sounding recordings. Maybe too, they'll start noticing the distortions (yes, distortions!) introduced by the digital recording process. And perhaps then improvements will be made in digital recordings, or better yet, we'll see further development in the areas of the analog recording medium.

Ronald A. Howance

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Audio Engineering Standards

• In our last discussion on standards, we explored the more general philosophic aspects of the issue. It is now time to turn the discussion towards the impact on audio engineering, and digital audio in particular. Because the life cycle of standards often spans many decades, it will be more instructive to first consider a more classical audio standard. To illustrate the process, let us take the standard "Method for Measurement of Weighted Peak Flutter of Sound Recordings and Reproducing Equipment" which was recently published in the June 1985 issue of the *Journal of the Audio Engineering Society*. This particular standard is referred to as ANSI S4.3-1982 and AES6-1982.

Firstly, we notice that the standard is defined under an ANSI number. This stands for the American National Standards Institute, located in New York. It serves as a US umbrella organization of standards of all types. This organization is subdivided into divisions or secretariates. For most audio engineering standards relevant to our industry, the codes will be in the S4 secretariate. This particular standard on wow and flutter is further subdivided into the S4.3 subsection.

The document published refers to the Standards Committee S4 which has membership of many other organizations and institutes. The list includes: Acoustical Society of America, Audio Engineering Society, American Loudspeaker Manufacturers Associates, Consumers Union of the US, Department of Defense, Electronics Industries Association, Institute of Electrical and Electronics Engineers, National Audio-Visual Association, National Association of Broadcasters, National Association of Photographic Manufacturers, National Bureau of Standards, National Council of Acoustical Consultants, Record Industry of America, Society of Motion Picture and Television Engineers, Society of Professional Recording Studios, US Information Agency, and several individual members. Notice the extensive list of members in this S4 committee. The committee is basically an open committee which considers and reviews proposed standards.

Within each of these member organizations, there will be working technical committees which do the work of generating proposed standards. The Audio Engineering Society has several such committees. The Acoustical Society of America has an ASA Standards Secretariate. Not all of the member organizations will be formally active in each area although they have the option to be if they wish. Because of the large amount of work involved in preparing a standard, organizations will often be selective about the degree and scope of commitment to particular standards.

The reference to the AES wow and flutter standard has a reference to the fact that the AES sponsored that standard under AES6-1982. This does not mean that they did all of the work involved. This particular standard, or a very similar version, had appeared previously as a DIN document (Deutsche Industrie Norm), which is the West Germany equivalent of the ANSI group. Prior to there being any standard, one finds various technical research papers discussing the perception of wow and flutter which appeared in various technical journals. Thus, the first work in this area was at least twenty years ago. Following the public presentation of the underlying science, several manufacturers started to make measuring equipment based on the published ideas. After many years, most people were using a similar approach to making this measurement. The consensus was collected in a single definition. This became the standard. When enough countries use the same standard, a worldwide organization can then consider adopting that as a universal standard.

The legal meaning of the resulting standard is perhaps best illustrated by the qualifying paragraph in our example which appeared with that standard in the *Journal of the Audio Engineering Society*:

"An AES standard implies a consensus of those substantially concerned with its scope and provisions and is intended as a guide to aid the manufacturer, the user, and the general public. The existence of an AES standard does not in any respect preclude anyone, whether or not he or she has approved the document, from manufacturing, marketing, purchasing, or using products, processes, or procedures not conforming to the standard. This document is subject to periodic review, and users are cautioned to obtain the latest editions."

If you read this paragraph to mean that the standard has no legal force, you are correct. The best way to understand the activity of standards writing is by the analogy of defining a word in a dictionary. A wow and flutter measurement system could use any method, but if the manufacturer claims that it conforms to ANSI S4.3-1982, then

JESSE KLAPHOLZ

The Role of Audio Today

• Working in the audio field, we are a link in the chain of information communication. Our role in communicating information may be our involvement in many forms such as the spoken word in a room, a radio/TV broadcast, or a concert. Audio's role in communications is increasingly becoming more important in today's information society. Now, more than 100 years since the beginning of data communication, we stand at the threshold of a mammoth communication revolution. We have, for the first time, an economy that is based on a key resource that is not only renewable, but self-generating.

For example, around 7,000 scientific articles are written each day. Scientific and technical information now increases forty percent per year because of new, more powerful information systems and an increasing population of scientists.

That means that data will double every twenty months. Right now the volume of information is approximately seven times what it was at the beginning of this decade. But this is, *information* we are talking about—not knowledge.

We are overwhelmed by technical data because of "Information Pollution." Information technology brings order to the chaos of information pollution and therefore gives value to data that would otherwise be useless. If users, through information utilities, can locate the information they need, they will pay for it. The new electronic publishers who provide on-line data bases and communications networks, is rapidly approaching a five billion-dollar-a-year business.

There are many organizations involved in the field of communications and information technologies management. These associations have conferences, workshops, and publications within their respective specialty areas. A short list of these many organizations is included here.

Perhaps an indicator of audio's direction and involvement in information-communications are some of the activities of the well-known acousticians Bolt, Beranek, and Newman of Lowell, Mass. BBN has designed computer-networking systems for large corporations including MCI and Wang. The Wang contract was for the tune of ten million dollars to design and build a wide-area network. The system uses BBN's latest "packet switching" technology, or equipment that automatically sends bits of information most economically over leased telephone lines.

RIDE THE ELECTRONIC HIGHWAYS TO YOUR OFFICE

While we are talking about telecommunications, how about telecommuting? More and more knowledge workers—including many engineers, computer programmers, systems analysts, and technical managers—are "going to work" via phone lines instead of highways. Now, with the advent of electronic work stations, remote work is feasible for just about anyone who deals with information, and seldom has to attend meetings.

Marcia Kelly, president of Electronic Services Unlimited of New York City, estimates that fifteen to twenty percent of the population will be telecommuting by the year 2000. (ESU is a consulting organization that helps companies implement telecommuting programs.) "Every knowledge worker is a potential teleworker," says Kelly. "Over fifty percent of America's employees now work with information; in the future, the figure will jump to ninety percent." In 1984, according to an ESU estimate, approximately 450 companies had instituted telecommuting for some of their staff. Telecommuting offers a variety of financial, productivity, convenience, home life, and personal benefits:

1. No driving to work.

It's hard to beat a ten- or twenty-second commute from your kitchen to your den. Imagine: no more traffic jams, long drives, highway fumes, or crazed drivers threatening your fenders. You simply walk up to your terminal, dial the office, check for electronic messages, retrieve necessary data from central files, take a coffee-break, and do some work.

2. Savings on clothing and food. You don't have to "outdo" everybody in the office with a better suit, or pay a few dollars at the cleaners every week. "At home I don't even wear shoes," says an anonymous audio consultant. Eating at home can save $20 to $50 or more a week if you normally go out for lunch.
3. Control over work scheduling and work environment. Imagine working in your bedroom with the VCR playing your favorite video, uninterrupted by bosses telling you what to do or that drinking beer on the job is a no-no. "If I want to go skiing at 2:00 in the afternoon, I can," says a Colorado programmer.

It's "here" for only a few aspects of business, such as computer programming, word processing, financial analysis, certain engineering functions, and information processing jobs such as news reporting and technical writing. Even in these professions, the technology has a little way to go. Imminent advances that will make telecommuting more viable include the two-way integration of data with images and sound via the same phone line.

Another development that will help is the continued computerization of paper files, and the advent of local area networks within companies. Only when more "file cabinets" are electronic, and only when more than silent text can be easily transmitted, will telecommuting become a reality for a broad range of knowledge workers.

Several advances in micro-computer technology/pricing will make telecommuting even more attractive. Modems are shrinking down to one chip, and likewise, prices have rapidly fallen below $100. As an interesting aside, Consultant's Choice Inc., in Atlanta, GA, has developed a modem technology capable of using ac power lines for data transmission at speeds as high as 80,000 bits per second. They claim to have overcome the problems of line-noise with their technique. Consultant's Choice expects the modem to be a hit with utilities, homeowners, department stores, and office buildings for a variety of chores, from remote meter reading to personal-area computer networks.

TAKE CHARGE OF THAT COMPUTER OR TECHNO-STRESS MIGHT GET YOU

Cybernetics, ergonomics, and human engineering can help us adjust to computers as extensions rather than controllers of people. Even with this attention to the human interface, however, it is still apparent that too many have problems accommodating the computer implosion. Either *you* command the computer, or it will be in command and tell *you* what to do. As a result, some people suffer from a form of techno-stress due to fear of computers (techno-anxiety), identification with computers (techno-centering) or a combination of both. How do people react when a computer is introduced? A manager's early enthusiasm can cause a computer-authority syndrome, with enormous flights of what-if fantasy in which the computer is used to create piles of official-looking but useless documents, generated too rapidly and without substantial basis. Resistance to technology can also develop based on selfishness, laziness, the dislike of being disturbed, fear of exposure, or a natural distrust for outsiders (the computer).

WHERE ARE THEY NOW?

It used to be that hardware was what you bought at the local store along with the grass feed. Computer systems brought in the concept of software—that is, stuff that made the hardware run. Some observers have commented, though, that software is, in fact, quite brittle and inflexible, but that is another matter. The term "firmware" was introduced to designate software that was built into a piece of hardware—for example, a computer's built-in operating system, as opposed to the applications software that might be loaded and run on the hardware using the firmware for support. More recently, several new wares have come on the scene. Courseware, for computer-aided instruction, is one. Live-ware, as in "consumer-compatible liveware"—denoting the unreliable protoplasm-based peripherals of a computer system—is another. Liveware modules compute with a central processor, sometimes called wetware. And for something even softer than software—namely, software that is announced but never quite makes it to market—there is the new appellation "vaporware."

NEW PREDICTIONS FOR A NEW YEAR

Forecast: In 1986, computer hardware and software will continue to grow better, faster, and cheaper. But new years merit more than stating the obvious. Therefore, we proudly present our top-ten forecasts—with tongue epoxied securely to cheek—offering a peek at what 1986 may bring to those who frolic and/or toil among disk drives.

—The FBI will break up a PC bulletin board that serves as an electronic bookie, and PC net runners—those who play around on personal-computer networks—will be arrested for violating interstate gambling laws.

—A gang of septuagenarian PC users will break into a bank computer system, take the money and move to Brazil, prompting a surge of PC sales to the elderly.

—Apple Computer will drop the price of its 512K Macintosh to under $500 in response to Commodore's Amiga high-end machine with Mac-like operating system. Commodore will then abandon the Amiga and start production on a digital recording console.

—IBM will introduce a 512K, $600, lap-size computer that everybody likes.

—A Japanese company will sell a $250 machine in the United States with a database manager, word processor, spreadsheet programs, and dot-matrix printer built in. It will use Macintosh-like display and mouse. For an extra $50, a complete audio/acoustic analysis and design software package in ROM will be available. Just plug it in and go.

—A song that has lyrics referring to personal computers will be recorded and hit number one on the charts.

—In Philadelphia, clients walking into a a well known audio consultant's office will be seated at a PC and asked to answer a series of "project-profile questions" covering the entire history, objectives, and/or present symptoms. The consultant then will get a printout of the computer analysis of the profile, and the client is charged $5,000.

—The "digital glove" will be introduced. The user will slip his hand into a special glove that links into the computer, and by manipulating the hands and fingers, the user can move objects around on the screen. Full MIDI, IEEE, and RS232 capabilities will be incorporated.

—The luster of computer-aided education will disappear. Several leading universities will remove PCs from the classroom, saying it is more important for college students to practice reading and writing than program games in Pascal or Forth. Companies making education and games software will continue to byte the dust.

—A touring sound system company will devise a scheme whereby the entire set-up, alignment, and operation is completely robotically and computer controlled. The company, Rhambo Sound, will be hailed for producing the most extreme achievement in sound reinforcement for 1986. Their slogan: "Sounds so good, that it hurts!"

—P.S. Watch out for the Commodore Amiga.

ORGANIZATIONS INVOLVED IN COMMUNICATIONS AND INFORMATION

American Federation of Information Processing Societies, Inc.
1899 Preston White Drive
Reston, VA 22091
(703) 620-8900

American Society for Information Science
1010 16th St, NW
Washington, DC 20036
(202) 659-3644

Associated Information Managers
1776 E. Jefferson St/Suite 470S
Rockville, MD 20852
(301) 231-7447

Association for Educational Communications and Technology
1126 16th St, NW
Washington, DC 20036
(202) 466-4780

Association for Educational Data Systems
1201 16th St, NW
Washington, DC 20036
(202) 822-7845

Association for Information and Image Management
1100 Wayne Avenue, Suite 1100
Silver Spring, MD 20910
(301) 587-8202

Association for Systems Management
24587 Bagley Road
Cleveland, OH 44138
(216) 243-6900

Association of Data Communications Users, Inc.
PO Box 20163
Bloomington, MN 55420
(612) 881-6803

Association of Information Systems Professionals
1015 North York Road
Willow Grove, PA 19090
(215) 657-6300

Data Processing Management Association
93 Standish Road
Hillsdale, NJ 07642-1110
(201) 664-4600

Industry Information Association
316 Pennsylvania Avenue, SE, Suite 400
Washington DC 20003
(202) 544-1969

International Information Management Congress
PO Box 34404
Bethesda, MD 20817-0404
(301) 983-0604

Society for Information Management
111 E. Wacker Drive
Chicago, IL 60601
(312) 644-6610

SUNY/OCLC Network
State University Plaza
Albany, NY 12246
(518) 474-1685

Urban and Regional Information Systems Association
1340 Old Chain Bridge Road/Suite 300
McLean, VA 22101
(703) 790-1745

Videotex Industry Association
1901 N. Fort Myer Drive/Suite 200
Rosslyn, VA 22209
(703) 522-0883

World Future Society
4916 Saint Elmo Avenue
Bethesda, MD 20814
(301) 656-8274 ■

William Spurlin

John Voci

WGBH-FM: Host To A First

WGBH-FM, Boston, produces the first live direct-to-local digital audio broadcast.

WGBH Control Room One with Neve console and Studer tape machines.

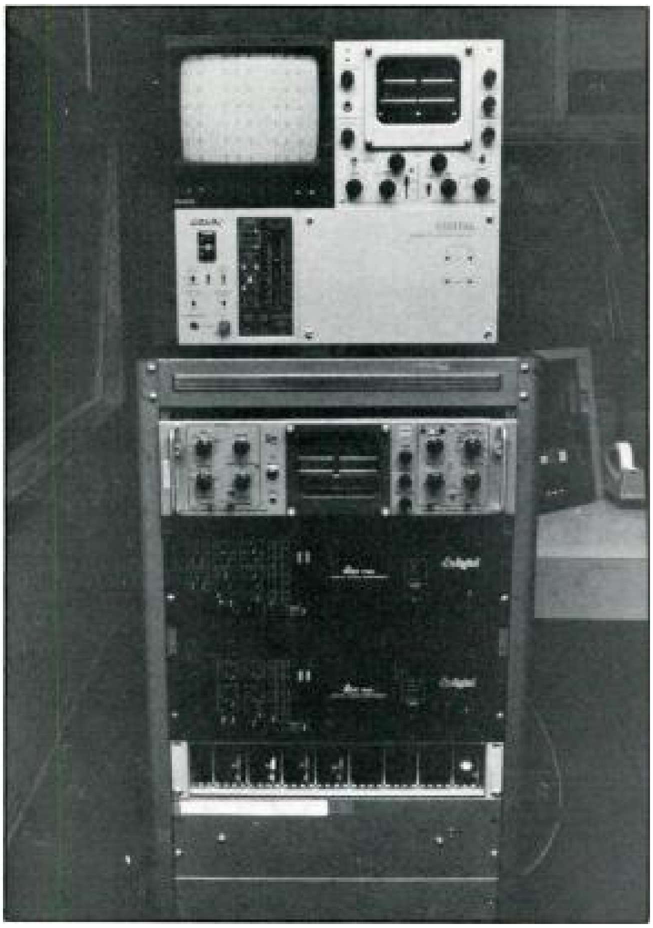
On Halloween, public radio listeners in nine cities around the country heard a live concert by Switzerland's Orchestre de la Suisse Romande in what was the nation's first live direct-to-local station digital audio broadcast. Produced by WGBH Radio, Boston, the all-Ravel concert by the Swiss orchestra was beamed digitally to public radio stations in Chicago, Los Angeles, Washington, Pittsburgh, Philadelphia, Orlando, Hartford, San Mateo and Boston, direct from Kresge Auditorium at the Massachusetts Institute of Technology.

Utilizing dbx Model 700 Digital Audio Processors, nine public broadcasters decoded and re-transmitted for local broadcast, a signal which had traveled in the digital domain from the concert hall. The 80 dB signal-to-noise ratio 105 dB dynamic range and the flat frequency response provided by the dbx 700, far surpassed the capability of conventional analog circuits ordinarily used to transmit and distribute live music programming. From the opening flute solo to the final crescendo, the Orchestre de la Suisse Romonde's performance of Maurice Ravel's "Bolero" was a spectacular demonstration of the digital medium. Listener response was overwhelmingly positive. "It was like sitting in the concert hall with the orchestra," remarked one listener.

John Voci is the operations director, and William Spurlin is the engineer in charge of maintenance and transmission at WGBH-FM.

LIVE SINCE '51

For WGBH Radio, live transmissions of concert music have been a staple of its programming since the station's



State-of-the-art digital equipment in Control Room Three. Pictured (l. to r.): Picture Monitor, wa Waveform Monitor, Sony 1630 Video Format Digital Audio Processor, Waveform Monitor, dbx Model 700 (two), and video distribution amplifiers.

inception thirty-four years ago. Since October 6, 1951, the day WGBH signed-on the air with a live broadcast of the Boston Symphony Orchestra from Symphony Hall, the station has striven to provide listeners with broadcasts of the greatest fidelity. In 1984, WGBH, which had earlier pioneered both stereo and quadraphonic broadcasts, began transmitting the Boston Symphony to its studios using a 2 gHz ENG microwave and the dbx 700. The digital link from Symphony Hall freed the station from its dependence on the telephone company's broadcast lines, a service which delivered 65 dB of signal-to-noise ratio, frequency response of 50 to 15 kHz, and persistant and annoying circuit crosstalk. When WGBH went on-line with the digital link, local audiophiles praised the "new" symphony broadcasts as the best ever.

In addition to local broadcast applications, WGBH began experimenting with the dbx 700 in the transmission chain for programs for the public radio system, WGBH sought a means to improve transatlantic transmissions of concerts from various European music festivals. In 1984, WGBH, in conjuction with the West German state radio network, conducted an international test of the dbx 700 from the COMSAT uplink in Reisting, West Germany. The test program, which left Germany cleanly, arrived in Boston with a sufficient number of errors to cause the dbx to mute. Investigation showed that processing amplifiers operated by AT&T between the Etam, West Virginia, downlink, and

John Voci (l.), WGBH Radio operations director, and William Spurlin (r.), WGBH Radio engineer in charge of maintenance and transmission, seated in Control Room One.

public television station WETA in Washington, DC, and by public television stations along the Eastern Educational Network route from Washington to Boston generated sufficient changes in the digital bit stream, particularly to the sync pulses, to be unrecognizable by the dbx 700. For future broadcasts, WGBH felt that it was necessary to both control and limit the relay points through which the signal would pass. On August 31, 1985, WGBH accomplished the first international relay using the dbx 700. Rather than use the Eastern Educational lines, which routed the signal through five relay points on the north bound path to Boston, the signal was downlinked in Etam from the Intelsat satellite, uplinked to a domestic satellite and downlinked by a portable earth station at the WGBH facilities. Austrian Radio's production of the Pittsburgh Symphony Orches-

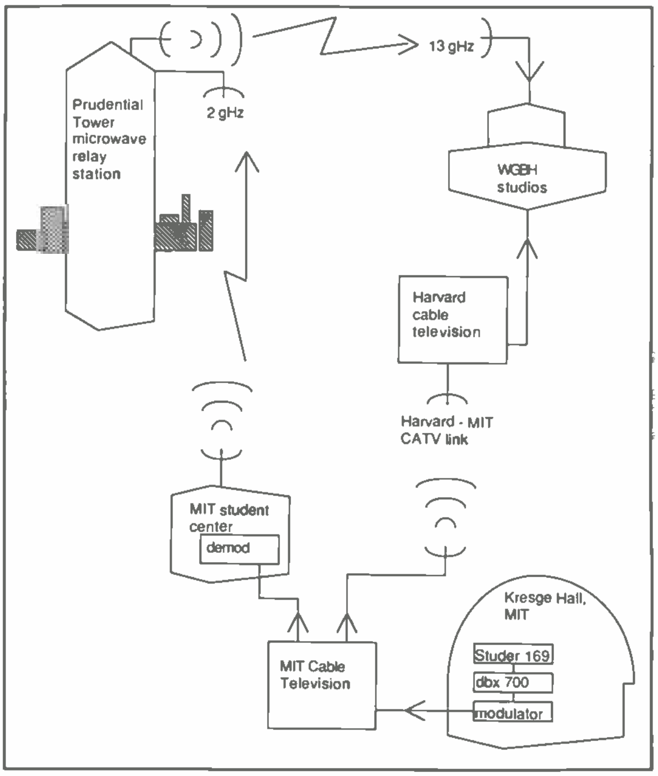


Figure 1. Signal transmission from Kresge Hall to WGBH studios.

tra's performance at the 1985 Salzburg International Music Festival arrived in Boston successfully.

As a means to enhance nationally distributed concert programs, WGBH, working in cooperation with the Newton, Massachusetts-based dbx Corporation, decided to try to deliver a digital program to other co-located radio and television stations. The halloween broadcast was to be the first point to multi-point digital relay.

INTERFERENCE

The only problem WGBH has experienced with its Boston Symphony digital link has been interference from other ENG shots. Co-channel interference due to the congestion on the 2 gHz ENG band occassionally caused the dbx 700 to mute. For the Orchestre de la Suisse Romande broadcast to provide redundancy in case of either equipment failure or microwave interference, WGBH engineers provided two digital paths from the concert hall to the WGBH studios, a distance of 1.6 miles. The primary path routed the signal on cable television lines operated by the Massachusetts Institute of Technology and Harvard University—the first use of cable television circuits for the transmission of a digital audio signal. The secondary route utilized the ENG microwave ordinarily used for Boston Symphony broadcasts.

At the concert hall, WGBH engineers fed the left and right outputs of a Studer 169 mixing console into two dbx 700 processors. The dbx 700's video outputs fed a modulator

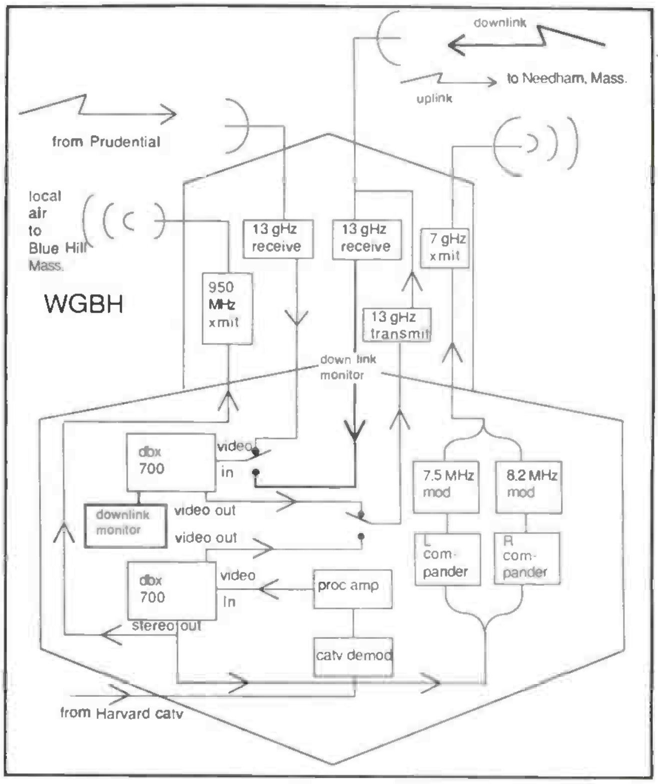
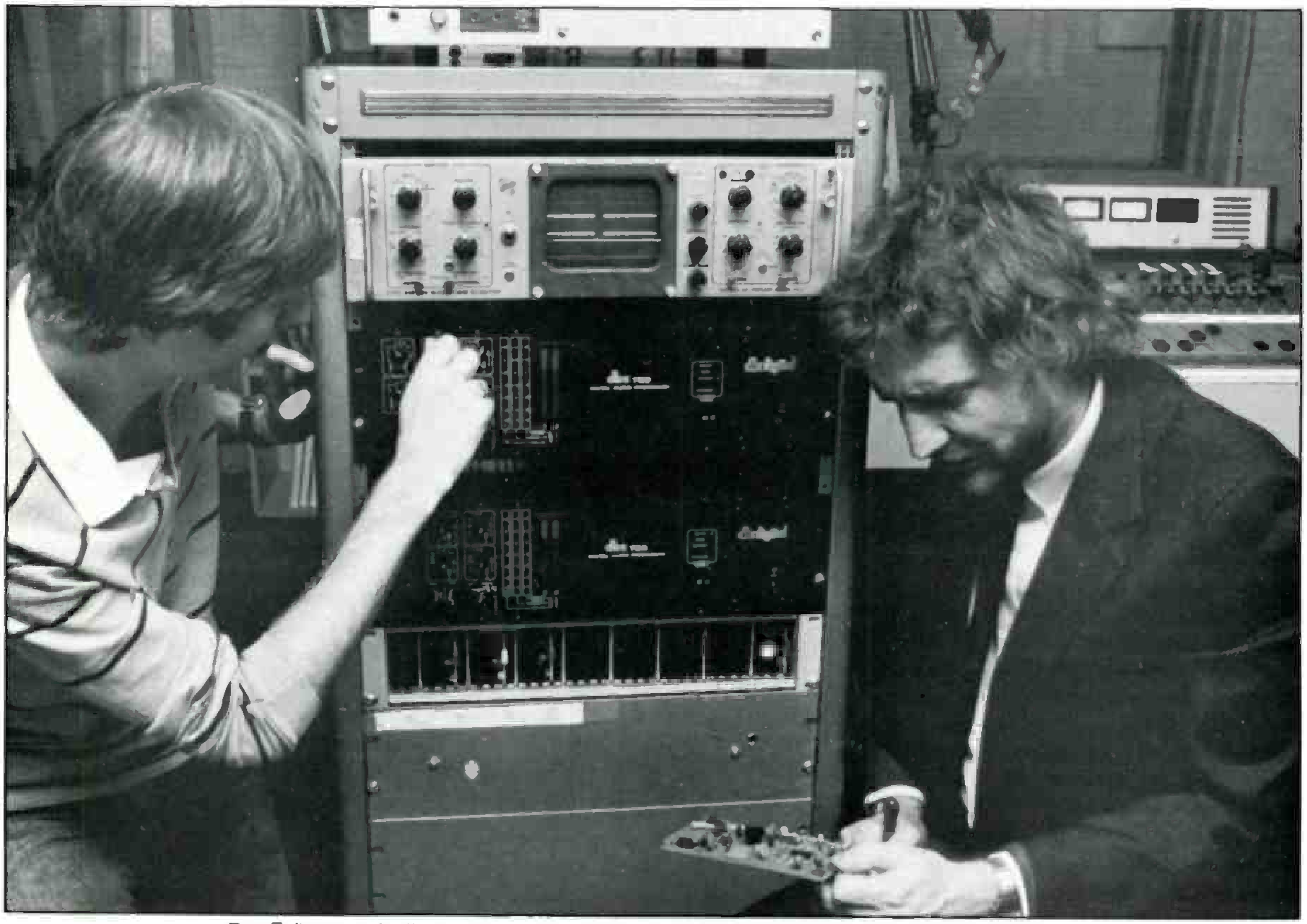


Figure 2. Uplink and downlink systems used for broadcast.



Ray Fallon, production and transmission engineer for WGBH Radio, assists Spurlin with adjustment of the station's state-of-the-art digital equipment in Control Room Three.

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For you, it's the sixth session of the day. For them, it's the biggest session of the year. So you push yourself and your board one more time. To find the perfect mix between four singers, 14 musicians, and at least as many opinions. To get all the music you heard on to the one thing they'll keep. The tape.
We know that the tape is the one constant you have to be able to count on. So we make mastering tapes of truly world-class quality. Like Scotch 226, a mix of Scotch virtuosity and the versatility to meet your many mastering needs–music, voices, effects. And Scotch 250–with the greatest dynamic range and lowest noise of any tape, it is simply the best music mastering tape in the world.
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-70
-80
-90
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1.7
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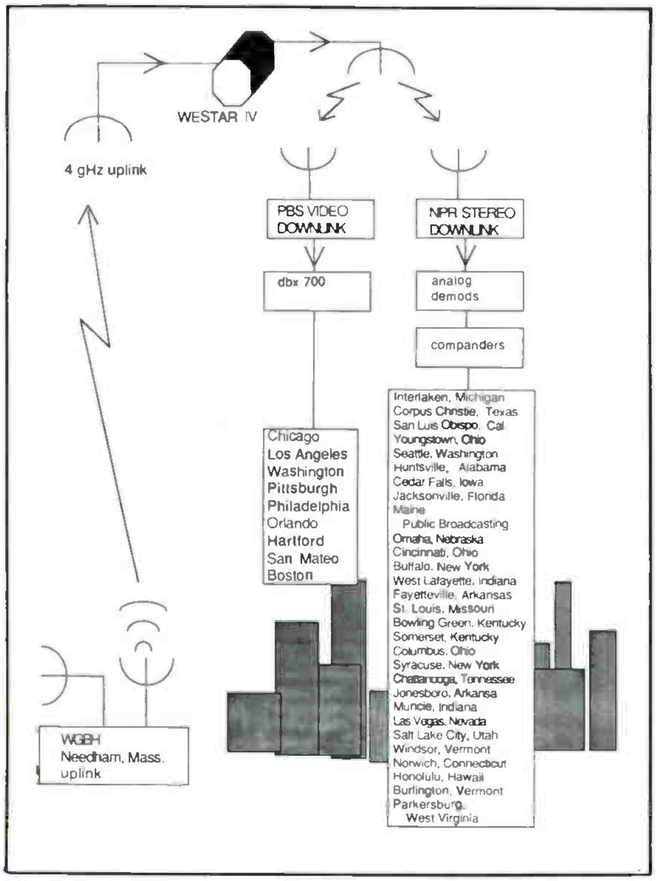

Figure 3. Signal distribution.

operated by the Massachusetts Institute of Technology cable television system. A microwave link between MIT and Harvard University fed the primary signal to the Harvard Information Transmission System, which, in turn, relayed the signal to WGBH studios. (See *Figure 1.*) A demodulator atop the roof of the MIT Student Center received the backup signal from the MIT cable system and fed it to the 2 gHz microwave. The signal was transmitted to a WGBH relay station atop the Prudential Tower in Boston, upconverted to 13 gHz and then re-transmitted to the WGBH studios.

THE LOCAL FEED

At WGBH, both primary and backup signals were fed to a dbx 700 for error correction. For the digital transmission, the video output of the primary system was fed via 13 gHz microwave to WGBH television's uplink facilities in Needham, Massachusetts, ten miles from the station's studios. (See *Figure 2.*) The video format digital audio program was uplinked on the Westar IV satellite, transponder 10D. Local stations downlinked the video signal decoded with dbx 700s and retransmitted for local broadcast.

For thirty-three stations unable to receive the video-based digital audio signal, WGBH fed the analog output of the primary system's dbx via audio subcarriers on a 7 gHz microwave to the station's audio uplink. The analog version was fed on Westar IV, transponder 2D. (See *Figure 3.*)

The Orchestre de la Suisse Romande concert introduced listeners to a new level of concert hall fidelity, not only by the enhanced quality of the digitally transmitted music, but by a sense of ambience hitherto unexperienced by radio listeners. ■

Breakfast With Bear Witme

Bruce Howze, founder of Community Light & Sound, is featured in the first of a series of breakfasts with Bear Witme.

THERE ARE ONLY A FEW large loudspeaker companies in the business that started around unique personalities: James B. Lansing a.k.a JBL; Paul Klipsch a.k.a. Klipsch; Avery Fisher a.k.a. Fisher; and so on.

Community Light & Sound also has a "unique" personality as its founder and helmsman. Yes that's right, folks—Bruce Howze. We all may know who he is, but at breakfast the other day we learned a lot more interesting things about him. We would like to share what we learned about Bruce in our first installment of "Breakfast with Bear Witme."

It's 2:00 pm and we're getting out the bagels and cream cheese for breakfast with Bruce and Bear. We're in Bruce's office which is almost *overly* neat. A rather unusual object in the corner attracts my attention—an old GGM speaker enclosure-mold converted into a turtle tank. About the office are various other memorabilia familiar to many of us who have used Community Light & Sound products before. Amongst the cigarette and cigar smoke we sip our coffee and begin reminiscing about the 60s and 70s.

Bear Witme: Tell us how CL&S got started.

Bruce Howze: I actually started the business in 1967. Started out doing custom lighting equipment, that was back

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Bear Witme is the obvious pseudonym of a well known writer/engineer who is based in the Philadelphia area.

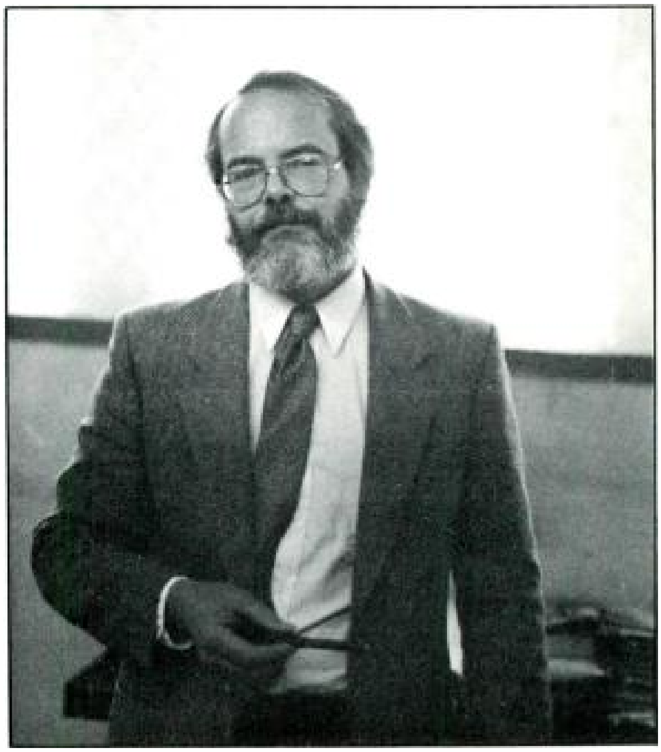

Photo courtesy of Community Light And Sound.

Bruce Howze.

in the psychedelic days. At that time the industry was in its infancy and there weren't that many products commercially available. So I got involved in doing custom lighting equipment and my first account was with the Electric Factory.

In the beginning when I was choosing a name for the company I had intended to do both lighting and sound work.The sound system installed at the Factory wasn't exactly what one would call engineered sound. The owners had a number of local corner bars, so they had their "bar jukebox guy" do the sound system. He put in the biggest jukebox system you have ever seen—big Seeburg speakers everywhere. That didn't work; he found that you couldn't put speakers all around the room and call it pa. So, he took them all off the walls and stacked them up on the stage...and that didn't work. Finally it got to the point that when anybody said they couldn't hear, the next day there would be a paging horn, or a column, or whatever he found in his basement, pointed at that spot. From those early days of the Electric Factory I got involved with the Festival group, which was a local touring sound company owned by the the Electric Factory and Dave Hadler. One of their first accounts was the Jefferson Airplane. I did some engineering work for that and also did some touring. At that time they were carrying around RCA W-folded horn bass-bins, and Altec multi-cells—it was very easy for me to see that this was not the optimum equipment to do that sort of job. The only thing that existed that made loud sounds at that time was movie theater equipment—it was never designed to be portable. You just stuck it behind the screen and that was it; nothing moved. You didn't care how heavy it was; you didn't care how big it was. When you start carrying it around you damn well start caring how heavy and big it is, and (you care that) if it's dropped it won't break. And when you start inputting 300 watts instead of 30 watts it doesn't catch fire (chuckles).

Anyway, it was easy for me to see that this was an industry in need of products, and I am just a designer and builder of things by nature. I am not a road type person; it was interesting to do the tour work but I could see that I didn't want to do that for the rest of my life. My talents and abilities were much more in the area in the development of products and manufacturing products. It was very fortunate that I had this relationship with the Festival Group so I knew what their needs were, and when products were developed you could go out immediately and test them. I could finish a horn Friday afternoon and Friday night it was playing at the Spectrum. That's great because you get immediate feedback and you can see what works and what doesn't.

BW: What was the first "product" that you made?

BH: I do remember one of the first horns that I made, the original LMF, which was a round mouth mid-range horn. It was logical that a round mouth would be stronger and so on. When we tested the LMF with the British rock group, Ten Years After, it never occurred to me that when you'd sit it down it would roll off (laughs). We set the horns down on top of the bass-bins and they rolled off and across the stage.

Fortunately that was at the sound check. So we learned that round horns had limitations and that we needed something that wouldn't roll off the stage. That was the first horn.

BW: When you first started who was Community?

BH: In the beginning it was a one-man operation working

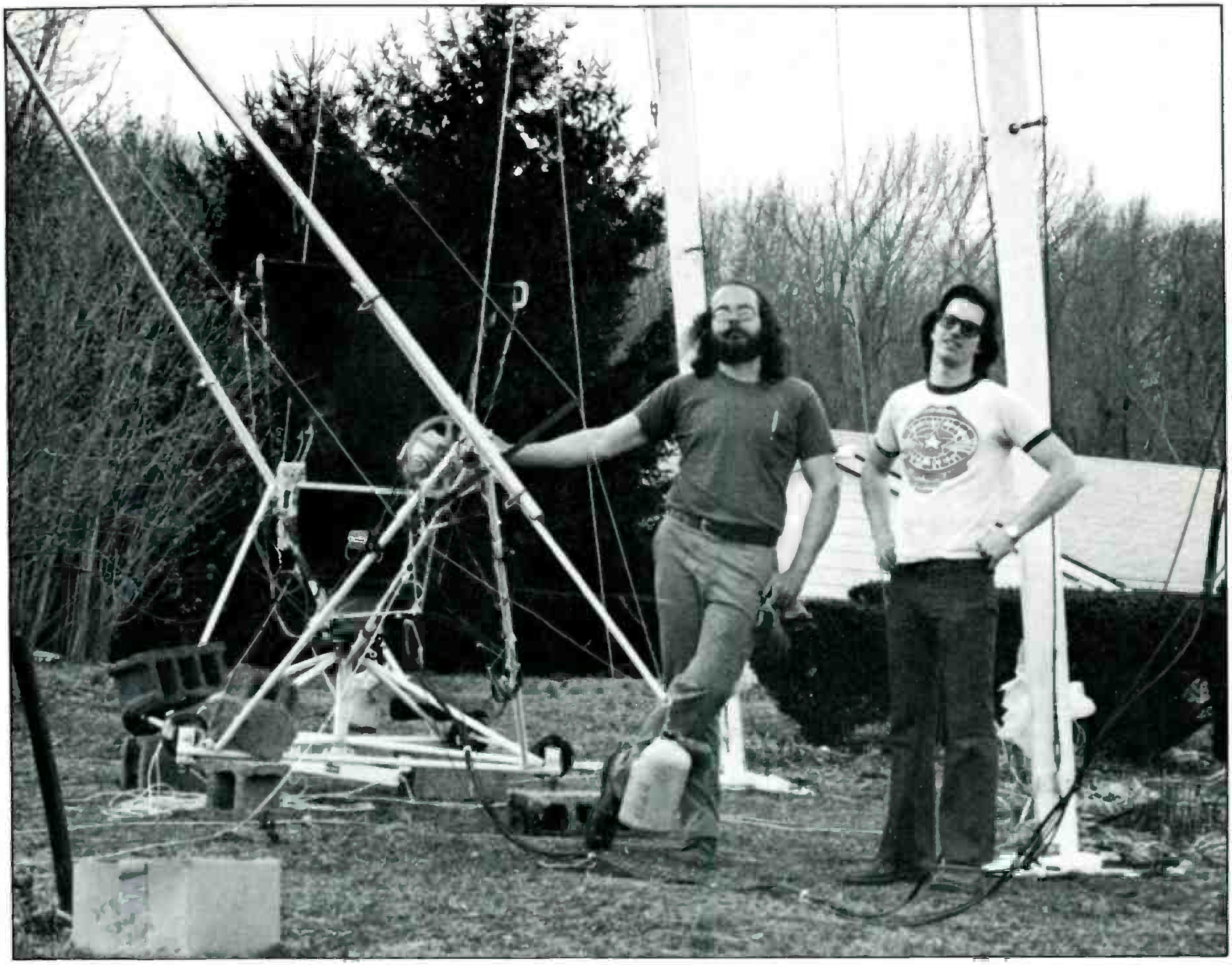

Photo courtesy of Community Light And Sound.

Bruce Howze and John Wiggins, who is now the executive vice president of CL&S, at a field test site.

out of a store front on South St. at 16th, right down the street from the Royal Theater. I lived there and also had about 1,500 square feet of work space. Tom Walter was the first person to join the company that stayed for any length of time, other than people that I would occasionally hire to do wiring and stuff like that. And he was the first real part of the company as we know it today. Tom was also involved in the Electric Factory and that's where we met. About 1969 we moved to our first real "factory building"—which is a building in which we did not live (laughs). It was up on Ridge Ave. in an old industrial property. It was an old mill built in the Civil War times and was taken over by a management company who was renting out space cheaply. We soon found out why it was so cheap; the first time it rained the water started coming through the walls.

We were doing more electronic work at that time and John Wiggins came in to work on a large recording console and help us wire it. He became more and more involved with the company and eventually became an owner of the corporation, as was Tom Walters. In about '72 or so we moved into a larger facility in South Philadelphia on Reed St.. With each move we kept getting bigger. The Ridge Ave. place was about 3,500 square feet, and the Reed St. place was about 5,000. We were growing at a fairly rapid rate. We were doing very well with our horns and fiber glass work and we picked up some fairly substantial OEM accounts. We were designing and making products for Peavey, Strom Communications, Bogen, and a few other people.

BW: This whole radical line of fiber glass loudspeakers appeared almost out of nowhere. How did that evolve?

BH: The second product was the RH-60 radial horn; next came the Leviathan, the PBL, and the NC-12, all around about the same time. We were basically looking at existing products from other manufacturers and looking at the jobs that needed to be done and seeing places where those products did not fit those jobs. For one thing we were surely pioneering the use of fiber glass. At that time horns were either sheet metal, as with Altec horns, or they were cast, as with JBL horns. It was obvious that fiber glass was a superior material to either one of those. We certainly had some negative reactions to the fiber glass. People would talk about the plastic horns—"How could a plastic horn work.... This is not strong enough," and so on and so on. Now the situation has completely changed; all the major manufacturers are using fiber glass.

BW: The names Community used in the early days were not exactly run-of-the-mill. For example, what did PBL stand for?

BH: That's a funny thing. I'm sure you're familiar with the dance posters that were up all over town in those days. They usually had some pretty creative names for those dances and one was the Psychedelic Boom a Loom. Very few people knew that story. The BLT stood for Bass Long Throw. That was when we were getting into our more sedate phase. As we added more products we had to use model numbers instead of names. Everything had to make some sort of sense.

BW: Are there any interesting stories behind the Leviathan?

BH: Well, we almost got arrested the first time we tested it in 1971. The idea was that we were trying to make a compact straight bass horn that was an extremely efficient device with good response down to 80 Hz. We took the Levi out into the parking lot which faced the Schuylkill River; across the river was the expressway. The first night we had Danny Starobin from Sweet Stavin Chain, bring over his guitar and we set up an instrument preamplifier and a DC-300A, and he was playing it quite loudly. Later we had reports that motorists on the expressway heard it very clearly. We also had reports from the neighbors via the police that didn't take too long to arrive on the scene. They didn't arrest us, but we had to stop.


Photo courtesy of Community Light And Sound.

Bruce testing an air powered siren driver.

BW: Tell us about the electronics you were producing in the beginning?

BH: We were originally doing both electronics and loudspeakers. In electronics we made some microphone mixers, and some OEM work for Guild. In the early 70s I was working on a prototype amplifier that had a power supply without a power transformer—just rectified the ac-line. Although there are other methods of doing it now, that idea is still practical and I haven't forgotten it—it's filed away perhaps for future use. We gradually got out of electronics because we were not as good at it as we were at loudspeakers.

BW: CL&S came out with a comprehensive book around 1975. Why?

BH: We often asked ourselves that, particularly when we were in the middle of doing it. It was indeed monumental. Our desire was to basically show that we made some excellent products and that our products compared very favorably to those products of our larger, and more famous, competitors. The only way to do that was to do fairly elaborate tests, and, of course, the test equipment that is available today was not available then. The best option for us was to test outdoors since we were going to test a lot of low frequency stuff. We did it all with a real-time analyzer, and very tediously transcribed all those little dots from all the slides we took (laughs). We erected a tower that lifted the speakers about thirty-four feet off the ground, and attached a servo-controlled mic-boom. It took John and me about a month to do the tests.

BW: The "white-book" was referred to for a long time by a lot of people.

BH: It made a lot of people take notice of us who had not taken notice of us before. It showed that we were real serious and our products were real serious. It did a lot of good, both in terms of sales and in recognition.

BW: Tell us what a day in the life of Bruce at Community might be like.

BH: A day in the life of Bruce at Community is like this: I start at about 10:00 am and work like hell till about 10:00 at night about six or seven days a week. That's it! (laughs). I do my designing on the drawing board at home besides that. My approach to designing products is very much integrated with the actual making of the product, so when I say that I do all my designing at home, I actually do probably half to three-quarters of it here because I do it while I'm making the products. I also do a lot of the tooling work and I certainly have a hand in the production setup. So I'm *looking* at a product. I'm not just handing somebody a drawing and saying, "Here, go make this," or "go buy this from somebody and tell me how much it costs." Usually the level of products we're making doesn't lend themselves to that.

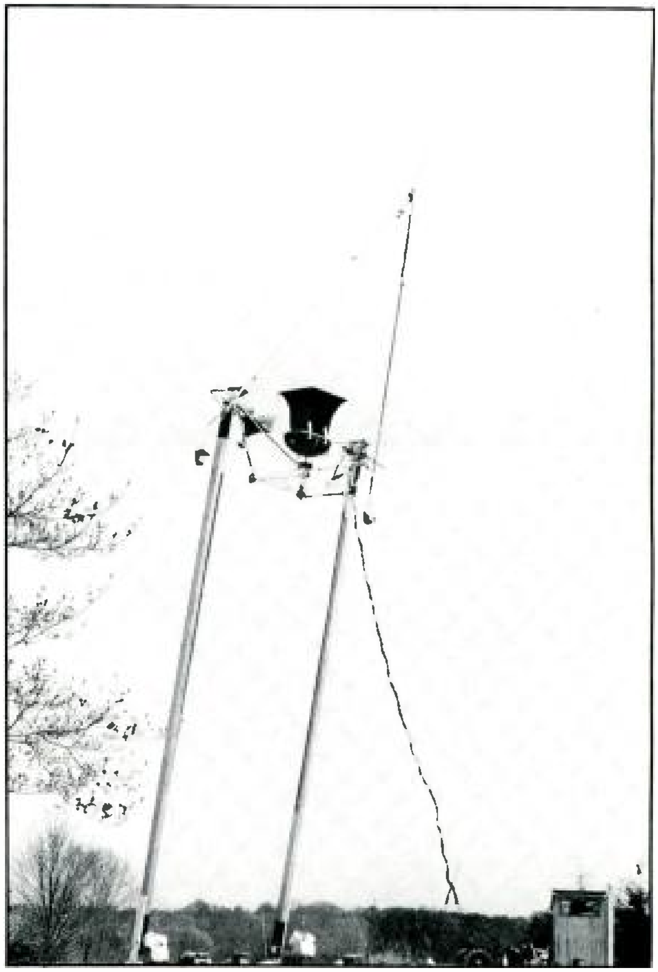
Photo courtesy of Community Light And Sound.

Horn elevated on mast for outdoor testing. Note the microphone at the very top of the mast.

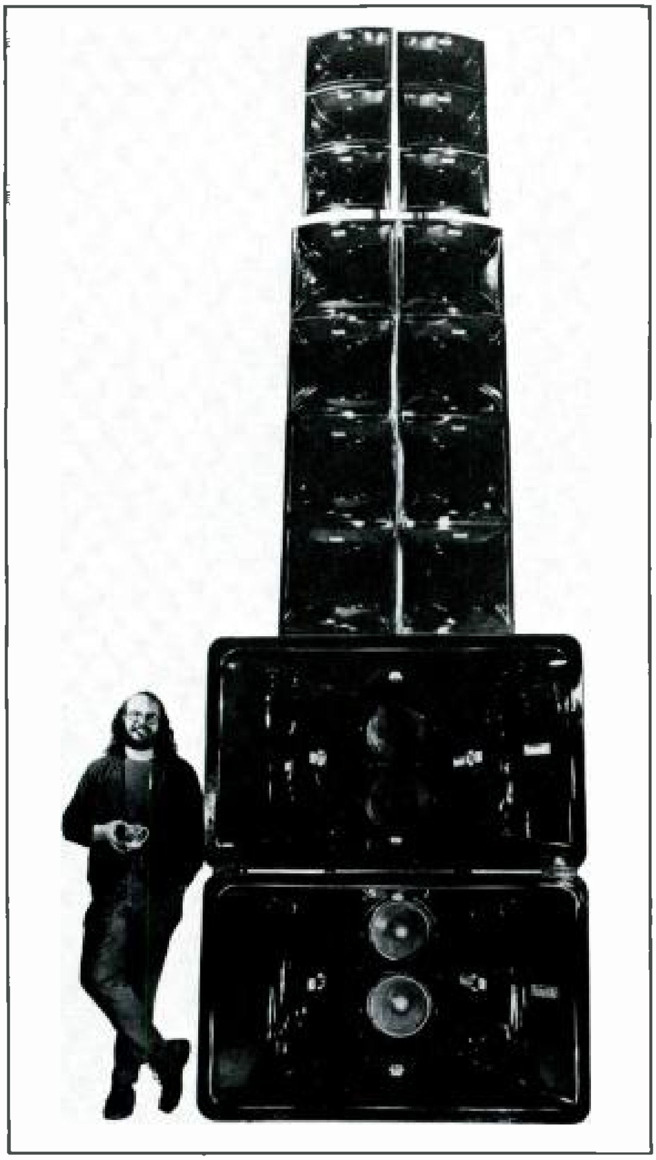

Photo courtesy of Community Light And Sound.

Bruce Howze with speaker stack. This photo was long used as an ad for CL&S.

I do the initial design of the product, all the prototype fabrication, and in that process there's a lot of design work that is going on—it's back and forth. I will very easily discover things that don't work, or I will discover things that do work but I didn't even think about. In other words as I start making the thing, as it appears before me, I see other things and I make changes. So a product may evolve so that what comes out at the end may be quite different than what was on the piece of paper in the beginning because it has gone through changes as it has gone along. And that's simply the way I design things; that's not the best way or the worst way—that's the way *I* do it.

BW: How many projects do you work on at one time?

BH: Looking at things in all phases from an initial design to a finished projec—probably a dozen. One new product design a week, some of them go straight to the circular file. We come out with somewhere between five and fifteen new products a year, some of which might be OEM so you won't hear about them. They would not appear to the public with our name on them.

BW: Can you tell us about the "nuclear power plant horn"?

BH: We got involved with Whelan because they had a need for a horn or a system that would enable their large electronic sirens, which are essentially loudspeakers, to compete in SPL with the mechanical sirens. The specifications for large siren systems around nuclear power plants were written strictly in terms of SPL; everybody considered voice warning as something nice. It wasn't in the spec, but if you could get it for free and still meet the SPL spec, that would be great. What I did for them was to design a horn that took their same sixteen drivers and produced about 10 dB more output, entirely through directional control; the horn has a horizontal pattern of 45° and a vertical of 5 °. We have conducted tests that produce intelligible sound at two and a half miles, and I own a patent on the horn. That enabled Whelan to compete directly with the mechanical sirens. In fact they have been so successful that the other manufacturers have been forced into producing electronic sirens, because specifications are being written more and more around voice/siren-warning systems. Sirens simply can't provide any information.

BW: What was the most challenging project you ever worked on?

BH: They all have their own special challenges, but probably the O-4. Of all the things we make it's probably the most dramatic product, and the most innovative.

BW: How did you come up with the idea?

BH: I had been thinking about it for a while. We always were working with mid-range, and you don't have to get too far before you realize the limitations of a cone loudspeaker. I can make a great horn for this speaker, but that's it. The speaker just doesn't have any more stuff in it. Using a cone speaker you quickly come to the end of what you can do with it—there is simply no magic to it. Obviously something else was needed, and so I had been thinking of it since '74 or '75. Over the years I experimented with different diaphragm fabrication methods and developed the one we use now in the M-4, which is one of the things that makes it work. Without it, it wouldn't.

BW: A tour of this facility shows some unusual equipment. Can you tell us about some of it?

BH: Our magnetizer was bought from RCA. It was used in their research labs. We have a ballistic galvanometer also from RCA that we use to measure magnetic flux. It works very well because it is a field-coil sort of thing that measures over a larger area than if you could stick a probe in somewhere and get a high reading. So we read the average flux in the gap. We also have a berrillium deposition machine that we experimented with during the M-4 diaphragm development. We were actually making berrilium diaphragms before any of the Japanese.

BW: Are there any projects that stand out from the others?

BH: I have enjoyed them all. There are some projects that have been more fun than others. I like to make things, it's as simple as that. Sometimes it gets to be a little tedious when you're working on something for the twentieth time, but there's always the excitement of generating a new product.

BW: In what direction is CL&S headed?

BH: We have the same direction we started from. Certainly our product line is tending more towards packaged systems, because that's what the market wants. We are spreading our scope a bit as far as price range, and we have proven that we can also produce very good low-cost/high-performance audio products. High-performance audio products is still what we do. ■

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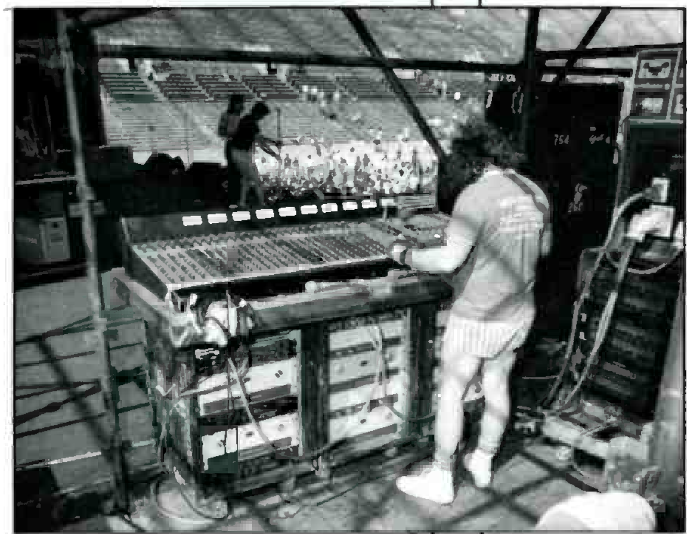

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Jesse Klapholz

Polymer Film For Transducers

Here we delve into current applications of piezoelectricity.

SINCE THE DISCOVERY of piezoelectricity by the Curie brothers in 1880, there has been continued research and development because of piezoelectricity's practical and economical applications, as well as scientific importance. Piezoelectricity, simply defined, is "pressure electricity" and is a property of certain materials that alter their shapes under the influence of an electrical field, or vice versa. Many different types of piezoelectric transducers have been manufactured over the years: for example, phonograph pickups, microphones, strain gauges, vibration pickups, underwater sound sensors, and tweeters.

The first materials that were used in piezoelectric applications were Rochelle salt, barium titanate, and quartz. However, many "synthesized" ceramic materials were subsequently developed for their improved piezoelectric activity and mechanical advantages. Today there are synthetic polymer films being manufactured that are ferroelectric, polarizable, pyroelectric, photoluminescent, photoconducting, or piezoelectric. This article focuses on the piezoelectric properties (as applied to audio and acoustics) of poly vinylidene flouride, or PVDF.

SOME BACKROUND

The real beginning for PVDF was the work in the late 60s by Kawai (1). Kawai found that PVDF could be poled to a higher level of piezoelectric activity than any other polymer material. His work was of great significance because beyond the development of PVDF's piezoelectric activity, Kawai showed the necessity for using highly oriented crystalline forms. The earliest report on an audio application of PVDF was a "planar" loudspeaker design in 1972 by Ohga (2). The first commercial PVDF products were reported by Tamura et al. at Pioneer Electronic Corp. in 1974 (3). Tamura and his co-workers at Pioneer's Acoustical Engineering Research Laboratory had succeeded in producing high quality micro-

Jesse Klapholz is a contributing editor for db Magazine and runs an audio consulting firm specializing in acoustical analysis and desing in the Philadelphia area.

High piezoelectric sensitivity in planar and thickness modes

Good fidelity with a wideband response

Low mechanical and acoustic impedance

High dielectric strength and high operating field strength, much higher than ceramics

Resistance to moisture, high humidity, and contaminants

Tough, flexible, and lightweight

Available as large area, thin film---readily cut and shaped to form complex configurations

Easily laminated to produce bimorph and multimorph elements that multiply transducer response

Not subject to breakage and loss of dipolar properties when subjected to mechanical impact.

Figure 1. General characteristics of PVDF film.

phones, phono cartridges, tweeters, and stereo headphones. Their work was a turning point in that it showed the commercial viability of PVDF in audio applications, and since its publication, has been cited in many other works. Further work concerning applications of PVDF began in 1972 at the Marconi Research Laboratories. At Marconi, a loudspeaker design was proposed that used a bimorph (multilaminates of film) PVDF configuration with alternating opposing polarities to create a bending mode in the assembly, thus extending the low-frequency response.

Extensive research in PVDF was also conducted in the Matsushita organization in Japan. At the AES Convention in 1977, Naono et al. and Rikow (4) presented the most comprehensive report to that time concerning microphone applications of PVDF. They detailed curvature mode of operation, PVDF elements with design equations and experimental models. At the same convention, Locanthi et al. (5), presented a new loudspeaker design, the model HPM-150. The HPM-150 employed a new version of a PVDF tweeter (earlier reported in ref.#3) which was then horn loaded, yet retained its omnidirectional characteristics.

CURRENT APPLICATIONS

In the communications sector, Telephonics Corporation became interested in the possibilities afforded by PVDF and has marketed a variety of low-profile, noise-cancelling boom microphones for telecommunications headsets (6), (7). Generally, the microphone element in these types of systems is located in the ear-piece assembly (making noise-cancelling impossible); however, the PVDF element can be made small enough to be located at the end of the boom and retain noise-cancelling characteristics. The simplicity of the microphone design makes it more cost-effective than dynamic or electret elements, while at the same time has the added feature of being moisture-resistant and virtually indestructible.

Countryman Associates Inc. has introduced a new lavalier microphone, the "Isomax TVH." Countryman claims that their microphone is the first to use "Active Vibration Isolation," which they further state yields 30 to 40 dB less handling noise than other lavaliers microphones (8). The "active element" is a PVDF diaphragm used as a noise-cancelling element (9). Musical instrument contact trans-

C capacitance	417pF/cm^2
Za acoustic impedance	2.7×10^6 Kg/m^2 (transverse)
Ze electrical impedance	1000ohms (for A = 100cm^2, t = 6μm, @1kHz)
Maximum Operating Temperature	80°C – 120°C
Maximum Operating Voltage	750V/mil = 30V/μm
Breakdown Voltage	2000V/mil = 100V/μm

Figure 2. Typical properties of PVDF film. (Thin film at room temperature.)

ducers can be easily made from PVDF film. Raad, a Canadian musical instrument manufacturer, currently manufactures a line of instruments including violin, viola, cello, and double bass with integral PVDF elements as contact pick-ups (10). Gibson, the legendary guitar manufacturer, recently introduced a new line of acoustic guitars with a PVDF film permanently mounted in the guitar, which functions nicely as a contact pick-up. Because this type of pick-up design uses the bending mode of the PVDF film, it is less susceptible to feedback than piezo-ceramic accelerometer type pick-ups. Furthermore, the bending-mode PVDF pick-up readily lends itself to equalization techniques of "sound hole simulation" (11).

BASIC CHARACTERISTICS OF PVDF

PVDF films are made from a semicrystalline, high molecular resin which is extruded into sheet form. When PVDF is initially extruded, the film must be stretched so the orientation of the polymer chains in the plane of the film can achieve enhanced piezoelectric activity. The stretching of the film is performed either along one face dimension (uniaxially), usually in the direction of machine travel, or in both machine and transverse directions (biaxially). The film is then coated on both sides with a conductive electrode metal, usually by vacuum deposition, to provide intimate electrical contact with the PVDF film during poling as well as during use. Often, masking is used in the vacuum deposition process to yield complex electrode patterns. The film is then polarized by an intense electric field at elevated temperature followed by cooling within the applied field. Piezoelectric activity is directionally dependent upon the stretching and polarization directions used during the manufacture of the film. *Figure 1* outlines the general characteristics of PVDF film, and *Figure 2* is a general specification of PVDF's properties. PVDF film has wide-band characteristics and low Q, unlike ceramic piezo transducers which have narrower band-widths and higher Q resonances.

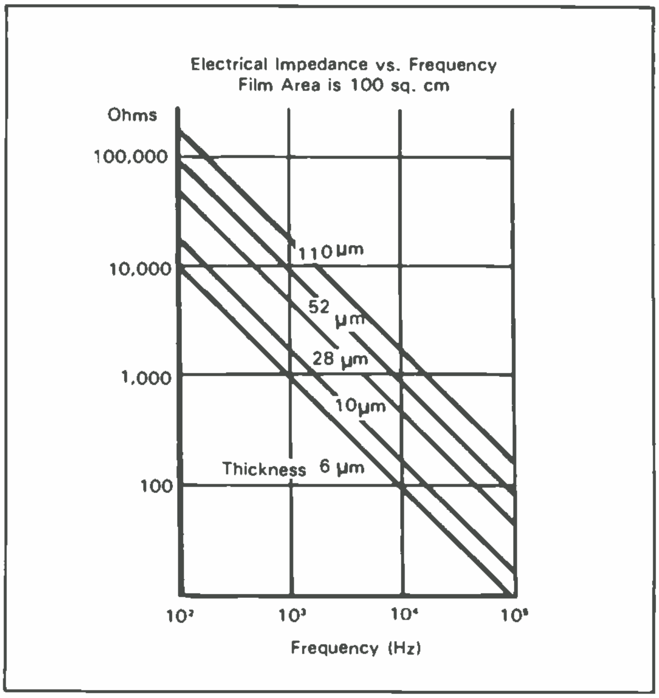

Figure 3. Impedance vs. frequency of PVDF films.

PVDF film can be used over a frequency range from DC up to the gHz region. The flat frequency response over a wide range is a consequence of the polymer's softness, which eliminates the self-ringing found in brittle materials. The basic free-air resonance of 28 um PVDF is approximately 40 MHz, and the resonance has a low Q. By varying the film thickness or using multi-laminates, the resonance frequency can be changed anywhere from the low MHz region up to the gHz region. Compared to ceramics, PVDF film has high mechanical losses (low Qm) resulting in highly damped resonance frequencies, useful if the bandwidth must be extended through resonance. The capacitance is high

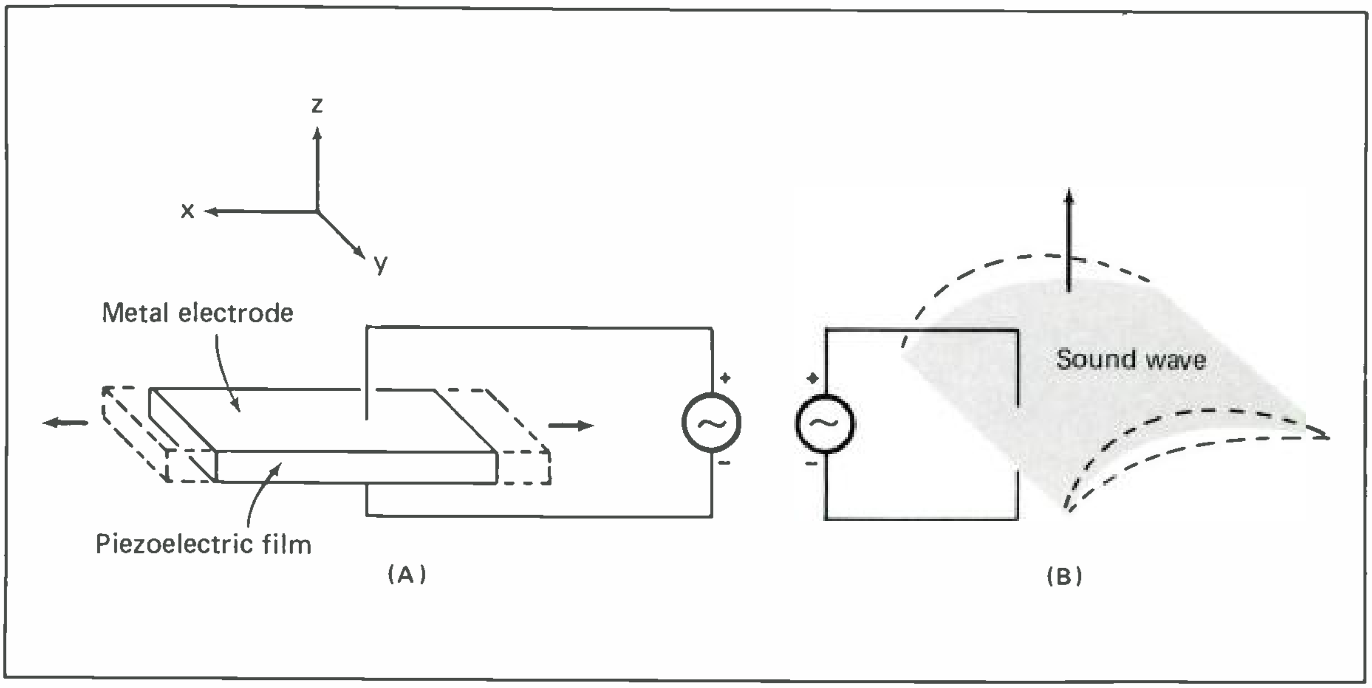

Figure 4. Diagrams showing longitudinal operating mode of piezoelectric film and how transverse vibration is converted into pulsating movement.

enough to allow for high-frequency operation when the film's output is terminated into a low impedance. The compliance of PVDF film is ten times greater than the compliance of ceramics. When machined into very thin film, PVDF can be directly attached to structures without disturbing the mechanical motion of these structures. The relationship of electrical impedance versus frequency for several film thicknesses is shown in *Figure 3*.

The efficiency of energy transfer is low compared to ceramic transducers, but PVDF film elements can be operated at much higher fields than ceramics (30V/um or 760V/mil as compared to 0.5V/um or 13V/mil). Therefore, the maximum input capability for electrical energy per unit volume is approximately thirty-six times greater than ceramic and the corresponding output of mechanical energy per unit volume is five to six times greater than ceramics.

When an a.c. field is applied to the film along the Z-axis, the film vibrates in a transverse direction (X-axis) as shown in *Figure 4A*. When the film is curved as shown in *Figure 4B*, the transverse vibration is converted into a pulsating movement. As a result the PVDF film efficiently generates radially directed sound waves. The easiest geometry transducer to fabricate is a cylinder. One can simply hold up in mid-air, a rectangular shaped piece of film, and VOILA, you've got a transducer. In holding this piece of film, as it is reproducing music, let's say, it is immediately apparent that the output rises significantly as a cylinder shape is approached, thus demonstrating what we stated above concerning the operation of the film. Now, if one could terminate the edges of the cylinder, and restrict the tendency of the film to simply expand, greater outward movement would be produced. Unfortunately, this would create other problems, namely unwanted resonances. Therefore, a suspension must be designed to eliminate resonances and maximize the output of the transducer."What about a spherical shape?" you ask. Good question—and an avenue that is currently being explored. Vacuum molding of the film can inexpensively mass produce spherically shaped diaphragms. *Figure 5* shows the forces of these two geometries.

Looking at transducer applications here, we can see that there are electromechanical and mechanoelectrical considerations. The physical mounting, housing, orientation, and termination of the film are all important factors that determine the performance of the film as well as its conversion efficiency.

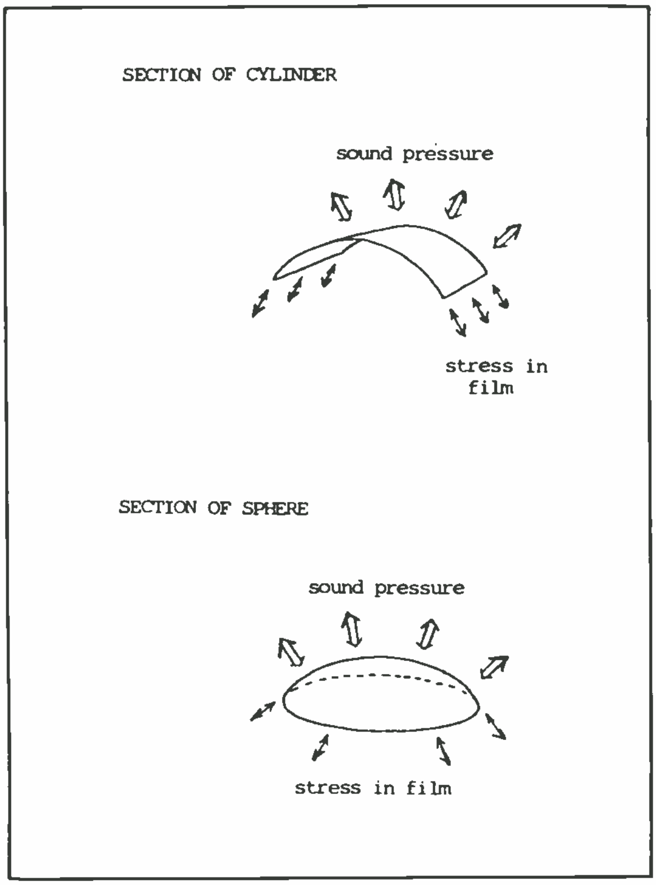

Figure 5. Diagram of forces on the two geometries.

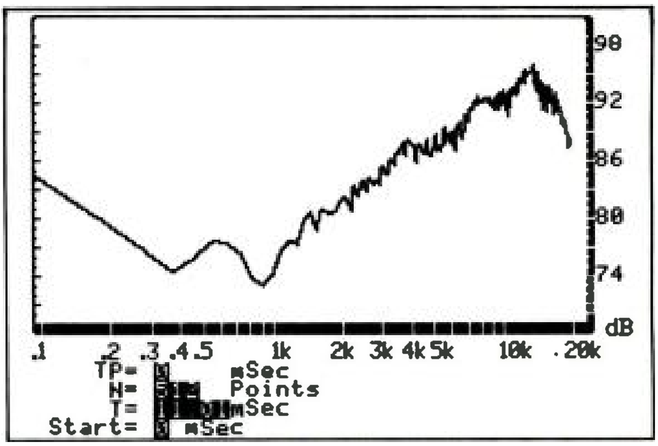

Figure 6. Tweeter response.

Also, PVDF films have a dielectric strength seventy times greater than PZT ceramics.

AUDIO TRANSDUCER APPLICATIONS

Electromechanical transducers utilize either the longitudinal or the transverse piezoelectric effects. Ultrasonic transducers usually utilize the longitudinal piezoelectric effect, whereas, flexure-mode devices and transducers for airborne sound utilize the transverse piezoelectric effect. The operating principle is such that when voltage is applied to the metal electrodes deposited on both sides of the piezoelectric film used as the diaphragm, elongation and contraction develop in the stretching direction of the film.

Equally important, in the case of electroacoustical applications, is how the final transducer will be driven electrically. Conventional power amplification is not applicable; we are not driving a coil of wire that is pulsing a piston. We are driving a piece of film that has capacitive characteristics, and as such the requirements differ slightly. The film, as was mentioned above, has a high dielectric strength; e.g., it has high voltage-handling capacity. However, it does not require the current capabilities that conventional power amplifiers are designed to produce. In other words, if one could provide the high voltage swings that the film is capable of handling, a conventional power amplifier could be virtually eliminated.

Thus, with several of the new chips currently being designed and prototyped, signal processing and the circuitry necessary to drive a PVDF film loudspeaker could inexpensively be built into, say, a one-rack-space unit.

PHYSICAL MODELS

Several PVDF transducers have been selected for demonstration and evaluation: a tweeter, vibration pick-up, and a microphone. The tweeter is constructed from 126cm(2) of 28um film, formed into a cylinder, and clamped at the top and bottom. The radiation pattern exhibited from this configuration is 360° horizontal by 120° vertical. The frequency response throughout these coverage angles is consistent and is shown in *Figure 6*. The tweeter's impulse response is shown in *Figure 7*; minimal resonance is exhibited here. A simple vibration pick-up was fabricated with a 20cm(2) piece of PVDF film. The transducer was applied to a Bodysonic low-frequency coupler (12), which in turn reproduced an impulse. For comparative purposes, a Genrad type 1560 accelerometer was also used to pick-up the output of the Bodysonic transducer. *Figures 8* through *11* show the respective impulse responses, as well as power spectrum of each vibration pick-up. The advantage of the PVDF film is its ease of attaching to any shape surface, while at the same time not contributing any mass to the structure to which it has been applied. As a last example, a simple microphone was constructed with a simple clamped suspension, with no loading or back chamber for the diaphragm, so that the film is exposed on both sides. A "difference" frequency plot between the PVDF microphone and a 1/4-inch condenser microphone of a reference loudspeaker's output is shown in *Figure 12*. This type of microphone design would work well in production testing of transducers, or any similar "high-abuse" application where durability and accuracy are both important. This type of microphone might also work well in a plane-wave-tube for compression-driver testing. Again because the film is so compliant, yet rugged, it easily fits many applications where more expensive microphones are simply not practical.

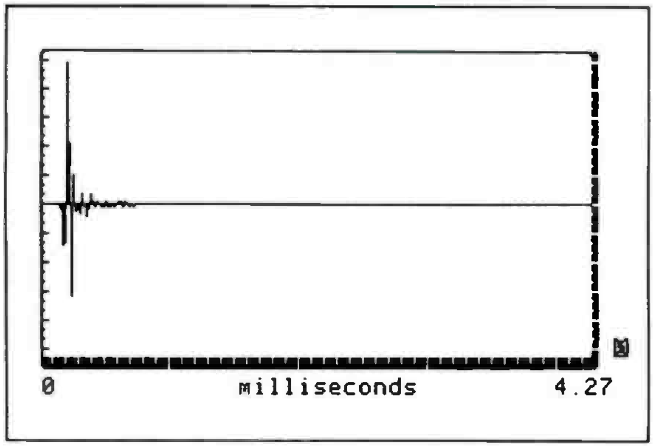

Figure 7. Tweeter impulse response.

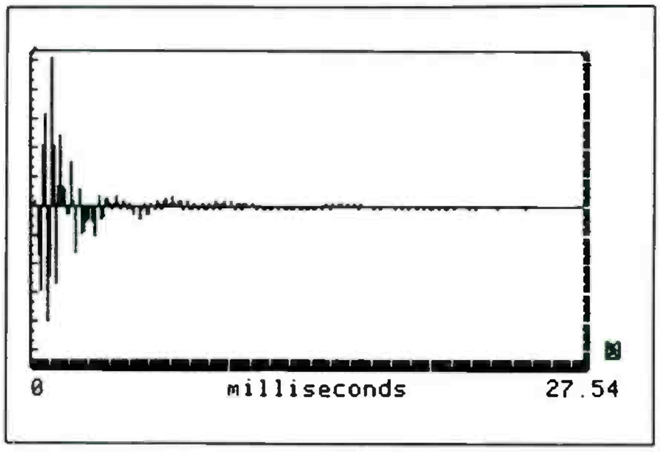

Figure 8. Genrad impulse response.

DISCUSSION

The applications discussed above are elementary in that PVDF's capabilities are limited only by the designer's imagination. Digital loudspeakers are around the corner and PVDF could certainly be applied here to great sonic, as well as economic, advantage. Ricoh Co. has been working on an AM modulated loudspeaker system (13), in which PVDF film would be appropriate. Since PVDF does not require high power, but rather high voltage, thepower amplifier necessary to drive PVDF film loudspeakers could be designed with much lower power/current requirements,

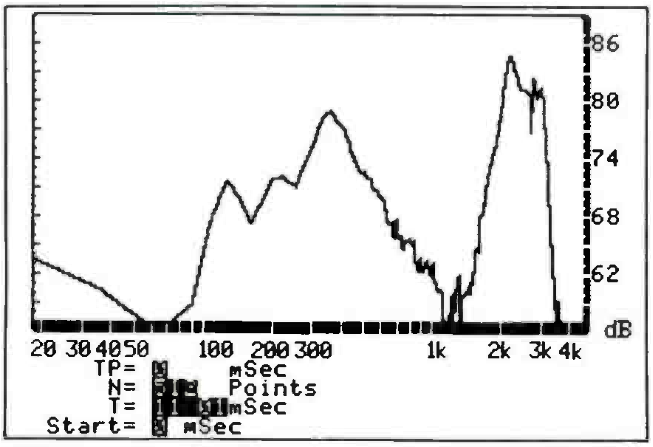

Figure 9. Genrad response.

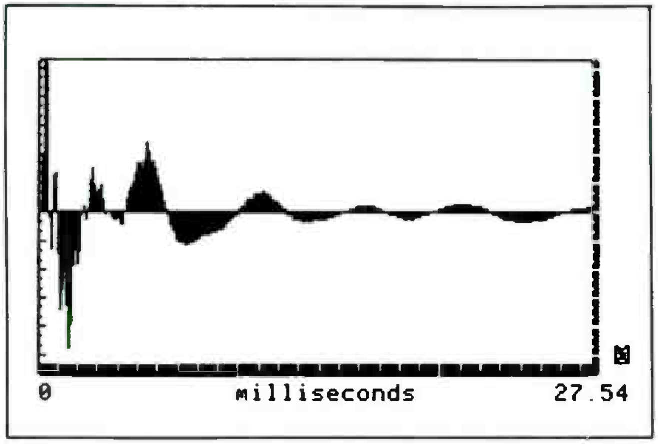

Figure 10. PVDF impulse response.

thus yielding a more efficient, compact, and economic system. Although PVDF's optical properties were not discussed here, PVDF would lend itself to new opto/acoustic/electric conversion techniques currently being developed (14). Another application currently being developed is the use of a small piece of PVDF film attached to a

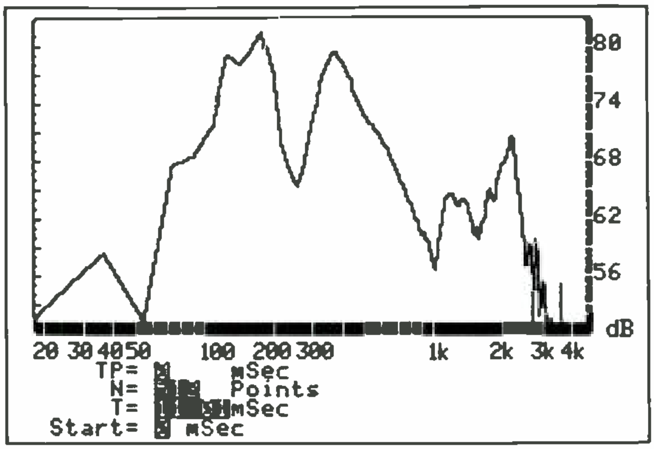

Figure 11. PVDF response.

loudspeaker cone serving as a transducer in a feedback/correction circuit (15).

Most of the research and early audio products using PVDF used film manufactured by Kureha of Japan. However, Pennwalt Corp. of Philadelphia made their line of PVDF film commercially available in 1982. They have recently instituted a proprietary technique of testing the material as part of its manufacturing process. This has eliminated the previous problem prevalent in all PVDF film manufacturing, which was a 6 dB high-frequency response variance between different parts of the film. The limiting

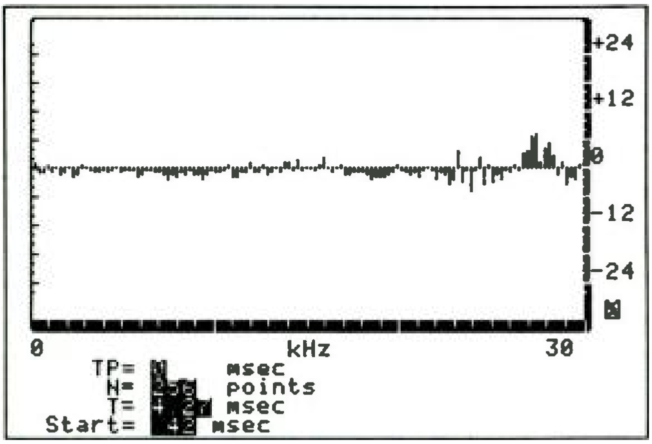

Figure 12. Difference curve.

factor in PVDF's introduction to more audio applications is the lack of widespread, dedicated, research in a "radically" new concept. However, PVDF film has been, for some time now, aggressively researched, developed, and applied in many related fields. Loudspeakers, as we know them and use them, have not changed much since their introduction in the early part of this century. It was the impetus of the motion picture industry that set the stage for many large companies to invest many dollars and years in high-level research. Companies like Bell Labs, RCA, GE, etc., no longer have such an interest in pursuing new loudspeaker technologies; their efforts are concentrated on new high-tech communications technologies. It was these companies, however, that practically wrote the whole book on loudspeaker technology.

Work is currently progressing on PVDF piezo-film applications in audio and electroacoustics, and another report will be forthcoming.

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db TEST

Len Feldman

Tandberg TCD 910 Master Cassette Recorder

General Information

TANDBERG'S TCD 910 is designed to replace both reel-to-reel and cartridge machines in a variety of professional applications. The total control flexibility afforded by the TCD-910 is based upon an 8-bit microprocessor with 32 K of EPROM memory. This cassette recorder incorporates many of the features which were formerly available only on professional reel-to-reel recorders. For example, unlike cassette decks intended for home use, this deck can be placed into the record mode even when in the play mode ("flying start recording"). Various methods are provided for introducing pauses between recorded selections, so that later these spaces can be used for various

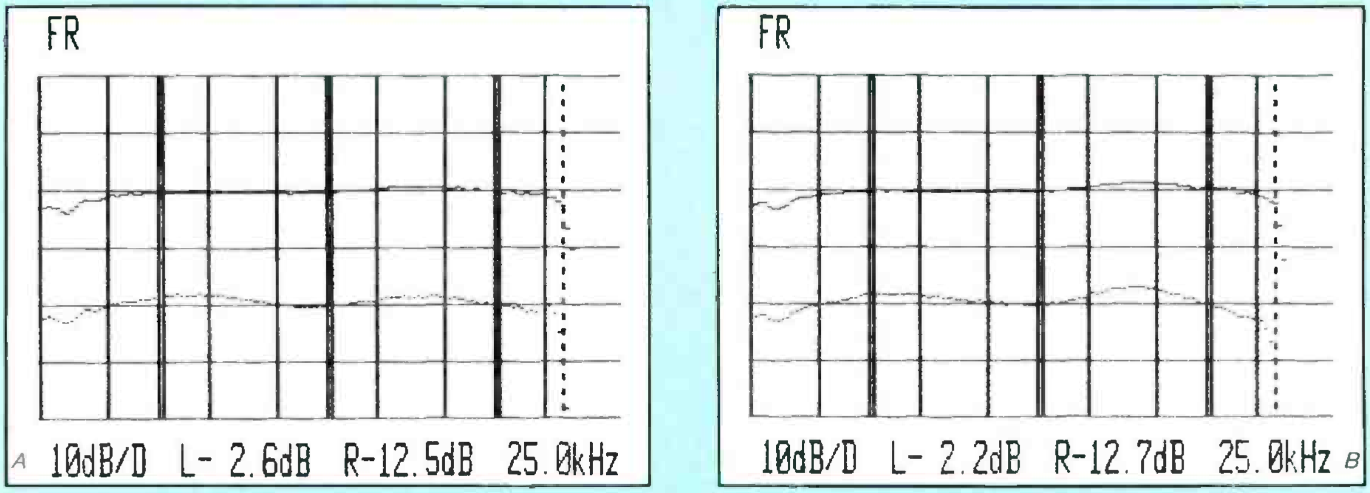
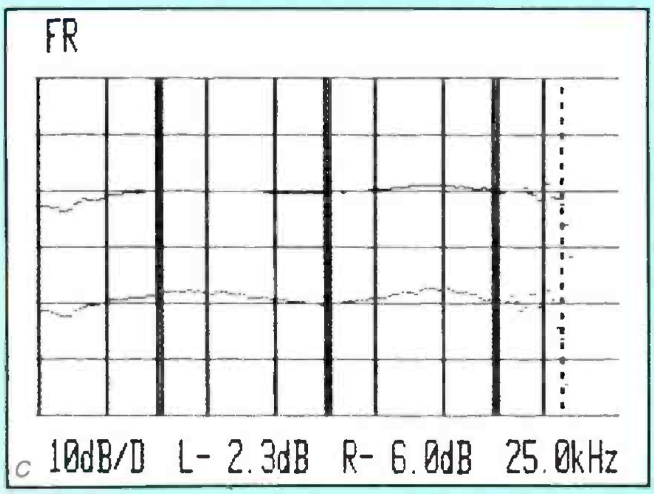


Figure 1. Frequency response curves without (upper trace) and with Dolby C for types I, II, and IV (metal) tapes. (A, B, and C.)



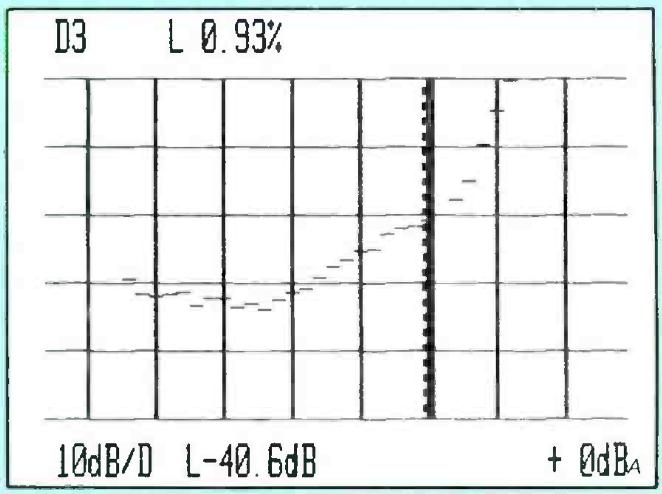
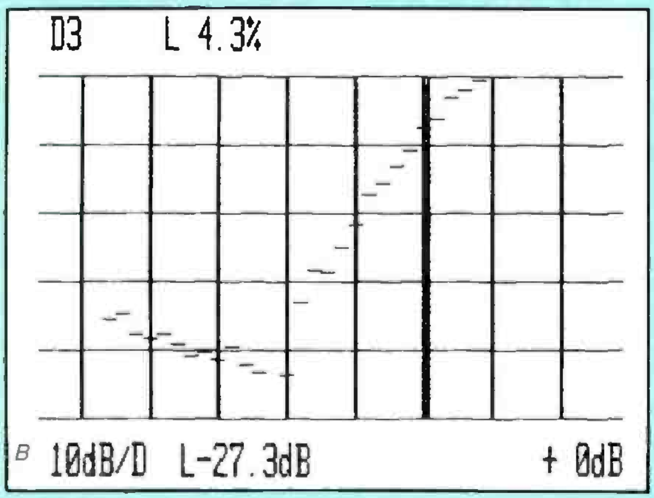
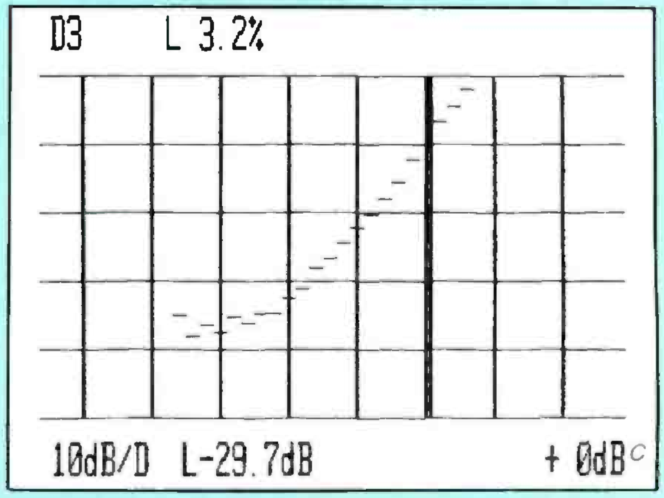


Figure 2. Third-order distortion vs. record levels for types I, II, and IV (metal) tapes. (A, B, and C.)





cueing functions during playback. A feature known as "Recap" allows you to rewind the tape slowly, all the while hearing the sound, and enabling you to resume play from any point you choose. The tape counter operates in two modes: displaying either arbitrary counter numbers or real-time indications, depending upon whether or not the system has been put through its calibration mode. If that mode has been activated, the system measures the thickness of the tape in use, and after a short period of time, it calculates the present position of the tape in terms of playing time. The counter can be used to program the deck to stop at various "addresses". An address is a programmed stopping point with a display on the counter as the reference point. Up to ten stopping points can be stored in the deck's memory —eleven points when the STOP button is pressed.

CONTROL LAYOUT

To appreciate the versatility of this cassette deck it's necessary to describe its many controls in somewhat more detail than usual. Over at the left is the multi-mode display and its reset button. This display shows actual tape time after the deck has gone through its tape calibration mode. It can also tell you what length of tape is installed in the cassette well. A three-position lever switch below the display selects tape type (Type I, II, or Metal). Surrounding this switch is a total of twelve small holes, six of which provide access to bias current adjustment of left and right recording channels for each of the three generic tape types. The other six access holes allow similar individual channel adjustment, by tape type, of recording current. More about

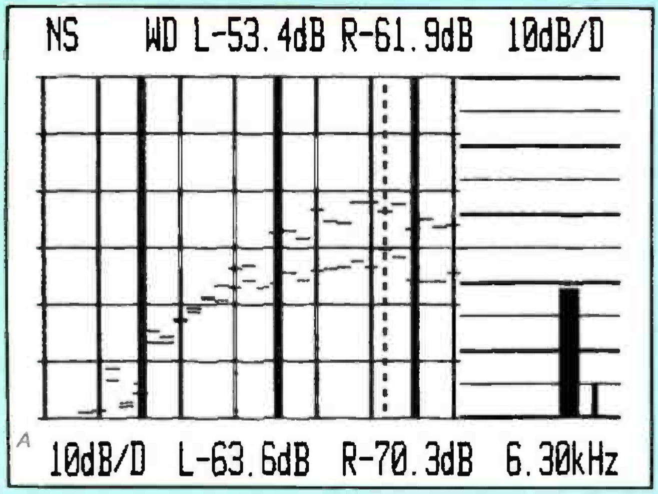
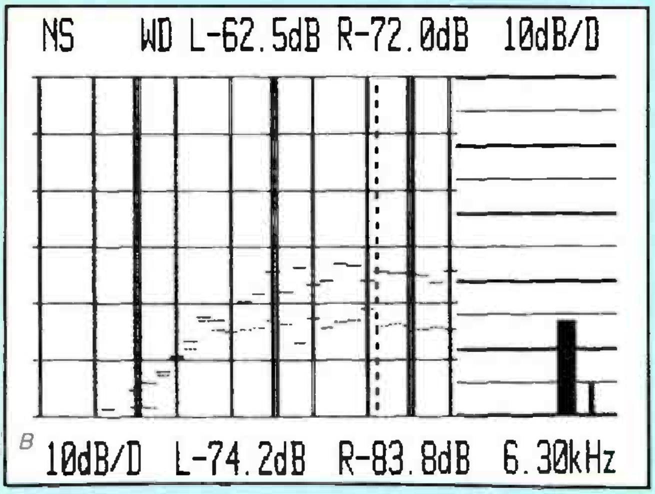
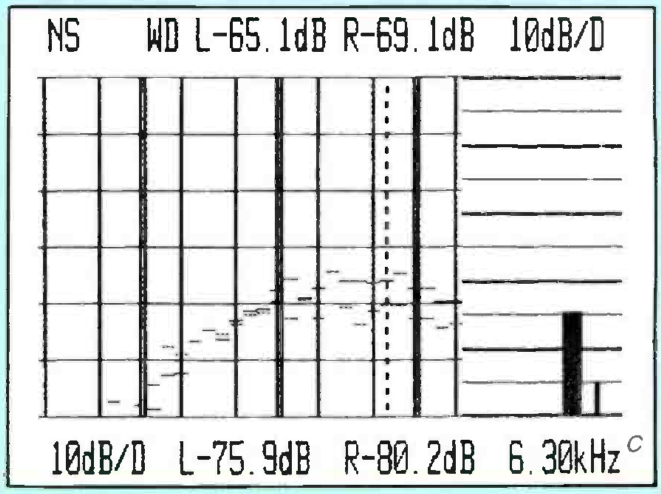


Figure 3. Signal to noise analysis without (upper plot) and with Dolby C for types I, II, and IV (metal) tapes. (A, B, and C.)





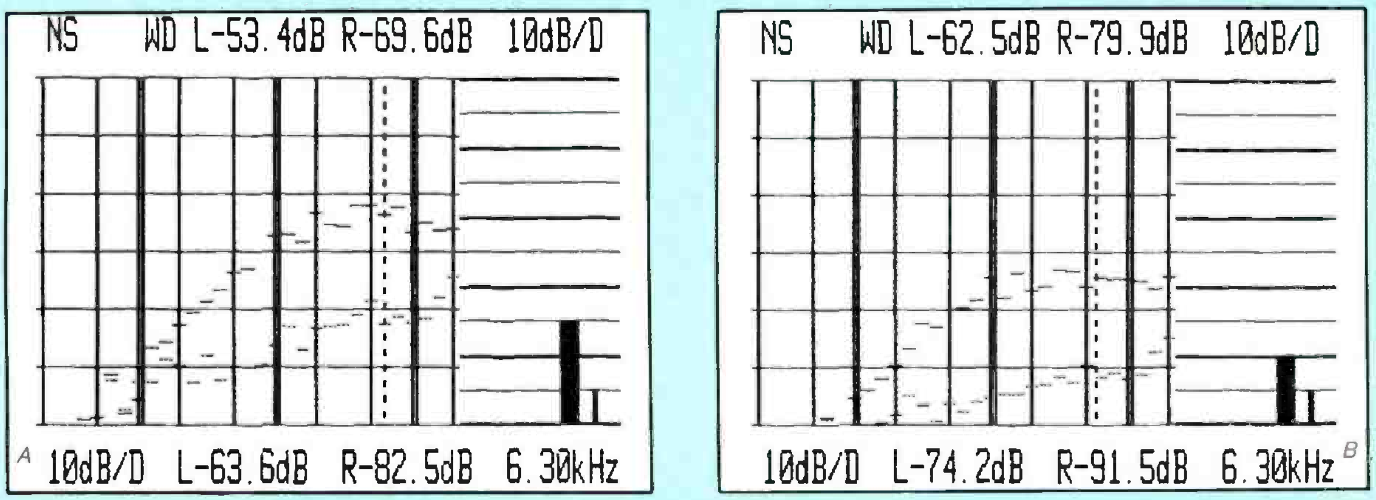
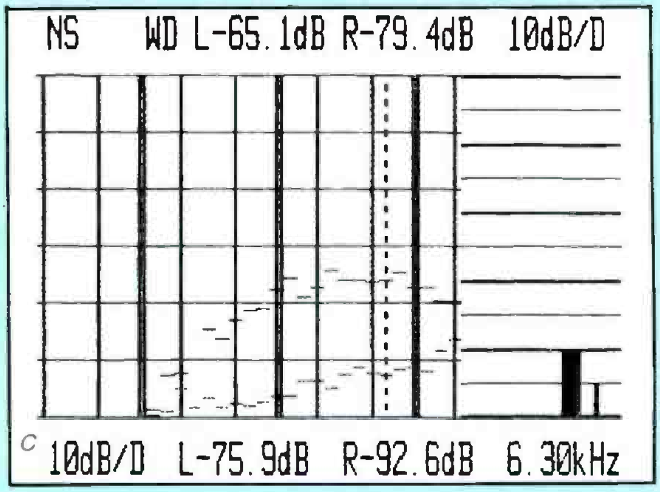


Figure 4. Signal to noise analysis without (upper plot) and with Dolby C for types I, II, and IV (metal) tapes. (A, B, and C.)



these adjustments in a moment. A continuously variable rotary record azimuth control is located directly below the tape-type switch, while below it is a headphone jack and the main power on/off switch. The cassette loading platform is positioned to the right of the display and other controls just described. Cassettes are loaded onto the takeup hubs directly; there is no swing down door. This makes for extremely consistent and accurate positioning of the cassette.

Peak reading meters are located to the right of the cassette loading area. The meters are designed to read actual recording current. Each is calibrated from above +6 dB to −20 dB. In addition to the usual 0 dB reference points, the scales of these meters also have a special marking, "M," at just above the +3 dB points. This designation is provided as the nominal reference level to be used when recording on metal tape. Metal tape, of course, can handle higher recording levels than oxide formulations and these special extra reference points let you take advantage of that. Six pushbuttons below the two meters handle the standard tape transport functions such as FAST-WINDING, PLAY, STOP, RECORD and RELEASE (of the cassette from its mounted standby position). Holding the RECORD button and depressing the STOP button briefly, automatically records a pause of four seconds, following which the deck goes into the STOP mode. Longer pauses can be recorded by keeping both the STOP and the RECORD buttons depressed for as long as you wish. Release the STOP button and the deck will go onto the RECORD mode. Release the RECORD button and the deck will go into the STOP mode. A pause can be introduced at the beginning of a recording, too, by holding in the STOP and RECORD buttons while briefly pressing the PLAY button. If the STOP button is released first, the deck will continue in the RECORD mode.

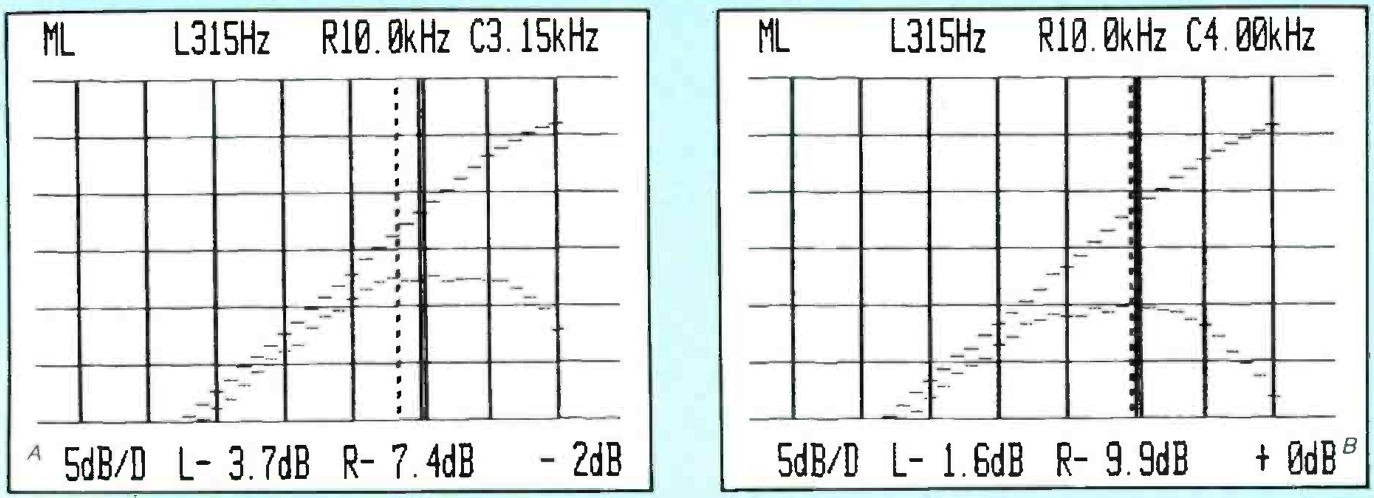
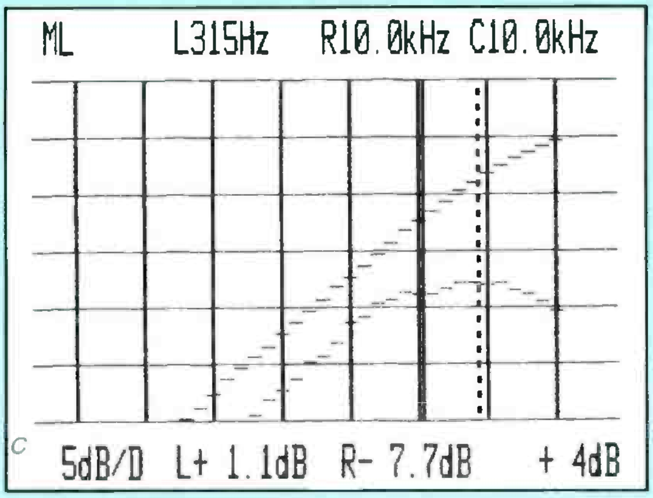


Figure 5. MOL analysis at 315 Hz and 10 kHz, without Dolby, for types I, II, and IV (metal) tapes. (A, B, and C.)



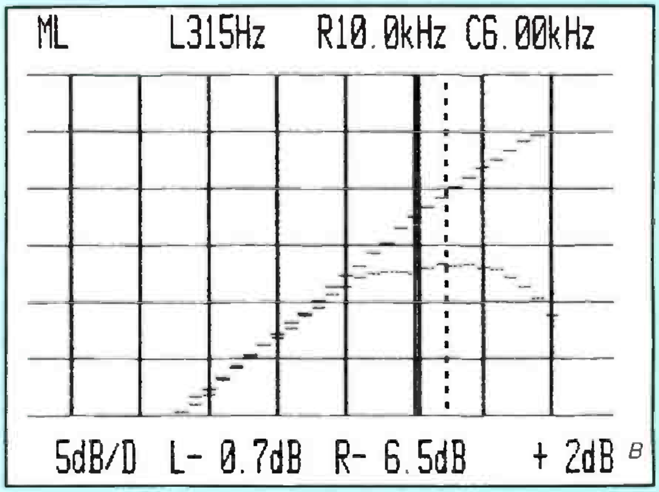
During tape playback, pressing the PLAY and STOP buttons simultaneously activates the RECAP function, allowing you to rewind the tape slowly while listening to sound output. This feature is extremely worthwhile when you are trying to cue up a precise point on the tape for "flying start" recording or for any other editing or dubbing purpose.

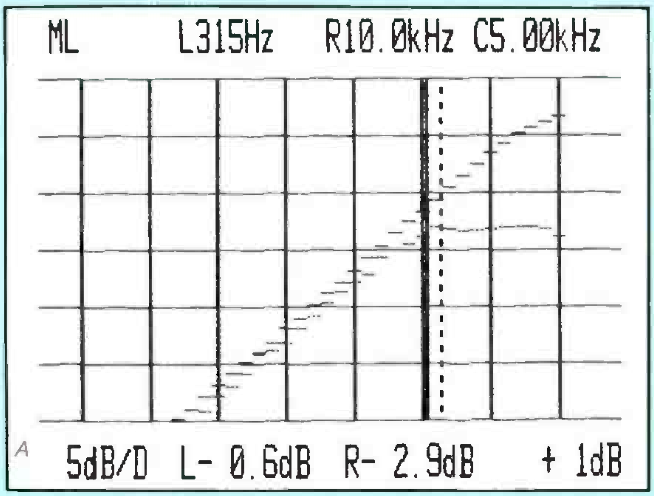
Perhaps the most unique functions of this recorder are tied to three 3-position switches located below the row of pushbuttons. If the first of these switches, the Cue Points selector is set to the No position, the display will show stored address numbers that have been programmed into the deck's memory. In this position you can search in the address register by using the FAST WIND buttons. If you press the FAST FORWARD and REWIND buttons simultaneously you will see the counter values for the address number shown in the display. If the Cue Points switch is set to GTO, the tape will wind to the address shown in the No position of the selector.

The second of these three switches is the Search selector. If set to the Cut No. setting, programs can be located using the recorded pauses. When the required program is located, the tape will stop one second in front of the beginning of the program and the deck will switch over to the STOP or PLAY mode, depending upon which mode has been programmed. The third switch, known as the Cut Selector is used to stop the tape when a pause is found or to rewind the tape to its beginning. If long pauses are encountered (greater than four seconds), the deck will mute after four seconds and will wind forward to the next program and will stop one second before the start of that program. If the pause exceeds thirty seconds, the deck will go into the STOP mode after thirty seconds. The Cue Selector function operates in the RECORD mode as well. The deck will wait approximately twenty seconds for a new program to record. If none is present at the input after that length of time, the transport will stop. Another setting of the Cut Selector switch, BOT (Beginning Of Tape) will cause the tape to rewind to the beginning when it is in the PLAY mode and a pause of at least four seconds is encountered. This function is not available in the RECORD mode.

Four small rotary knobs are arranged along the lower portion of the panel. The first of these selects Source or Tape (monitoring), and also delivers either of two test tones—15 kHz and 315 Hz—which are used along with the level meters to adjust bias and recording current for optimum recording and playback results with a given tape. The other three controls are for phone level adjustment, activation of Dolby B or C, and channel balance. Finally, a large rotary control with a concentrically mounted index or reference pointer adjusts recording levels of both channels simultaneously. The control is calibrated from +12 dB to −30 dB, with the 0 dB reference level positioned at twelve o'clock on the knob.

Left and right line inputs are connected at the rear of the





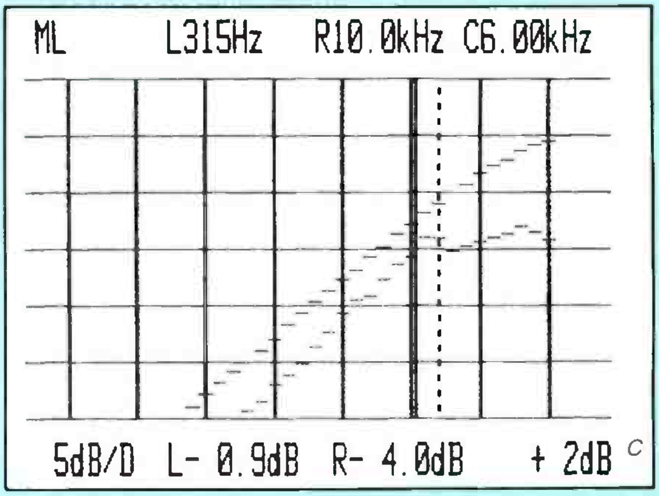


Figure 6. MOL analysis at 315 Hz and 10 kHz, with Dolby, for types I, II, and IV (metal) tapes. (A, B, and C.)

deck via a pair of female XLR connectors while outputs are available from a pair of male XLR connectors. Output levels for each channel can be calibrated so that the 250 nWb/m recording level corresponds to any dBu setting from −2 to +12 dBu. A 25-pin connector of the familiar RS-232 type is also found on the rear panel of the TD-910. A diagram showing what function can be controlled by shorting the particular pin to ground should enable you to wire up your own remote control system for this deck. Even the various LED indicators are assigned pin numbers in this connector so that they will light when the appropriate function (RECORD, PLAY, STOP, REWIND, etc.) is remotely activated.

TEST RESULTS

A summary of the most significant test measurements is shown in the VITAL STATISTICS chart at the end of this report. In optimizing the deck for the particular types of blank tape that we used in our tests we followed Tandberg's instructions, using the 15 kHz and 315 Hz test signals and adjusting bias and record current so that the 15 kHz output during monitoring was as close in level as possible to the 315 Hz output. This procedure resulted in extended frequency response with a sacrifice of headroom and higher-than-normal THD readings. It is important to understand that if we had elected to ease off on the bias a bit, we could have reduced THD considerably at some slight sacrifice of S/N and frequency response. We decided to report the test results obtained by following Tandberg's instructions, but you should certainly feel free to adjust bias and record current on this machine so as to favor those parameters that you deem most important.

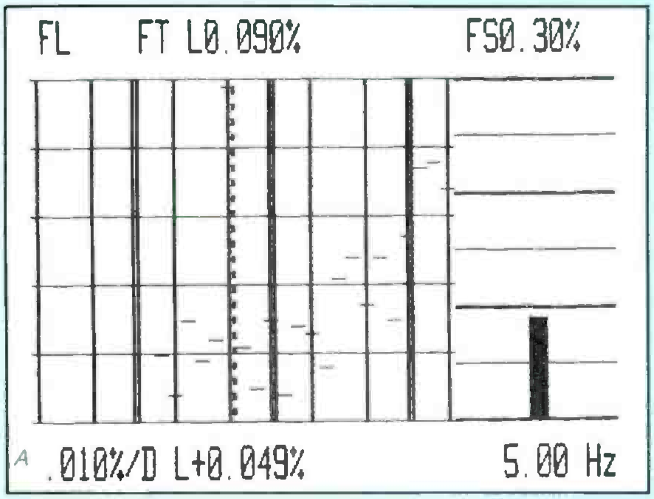
For all of our tests, we used Maxell UD XLI as our Type I (Normal) tape samples, Maxell XLIIS for Type II samples and TDK Type MA as our Metal Tape Samples. *Figures 1A, 1B,* and *1C* show RECORD/PLAY frequency response plots for these three types of tape. The upper trace in each case represents results obtained without Dolby NR, while the lower trace is the result obtained using Dolby C. Vertical scale is 10 dB per division. The dotted line electronic cursor has been set to the frequency at which attenuation for the non-Dolby response is approximately −3.0 dB.

Figures 2A, 2B, and *2C* are plots of third-order distortion as a function of recording levels. The cursor has been set in each case to read out the distortion at 0 dB reference recording level, with the actual reading in percent shown at the top of each graph, and in dB below reference level at the bottom of each graph.

A-weighted signal-to-noise analyses for the three types of tape tested on this deck are shown in *Figures 3A, 3B,* and *3C.* Results shown at the top of each graph are overall S/N figures without Dolby (listed as L) and with Dolby B (listed as R). All results shown at the top of each analysis are referred to 0 dB, rather than to the 3 percent distortion point. So, to arrive at the figures shown in our VITAL STATISTICS chart, you have to add (or subtract) the dB levels shown for 3 percent distortion. For example, in *Figure 3A* you would add 3.0 dB to the non-Dolby reading of 53.4 dB, to obtain a S/N value (referred to the 3 percent THD level for this particular tape), of 56.4 dB. The left section of each set of graphs depicts the actual distribution of the noise, in third-octave increments. The upper set of third-octave blips is for non-Dolby operation while the lower set was obtained using Dolby B.

Much of the same sort of study is represented by *Figures 4A, 4B,* and *4C,* except that this time, the comparison is between non-Dolby operation and Dolby C results.

Figures 5A, 5B, and *5C,* represent plots of input versus playback output for test signals at frequencies of 315 Hz and 10 kHz. The tests are designed to show high-frequency MOL (Maximum Output Level), for the three types of tapes when they are used on this deck. The dB figures alongside the R



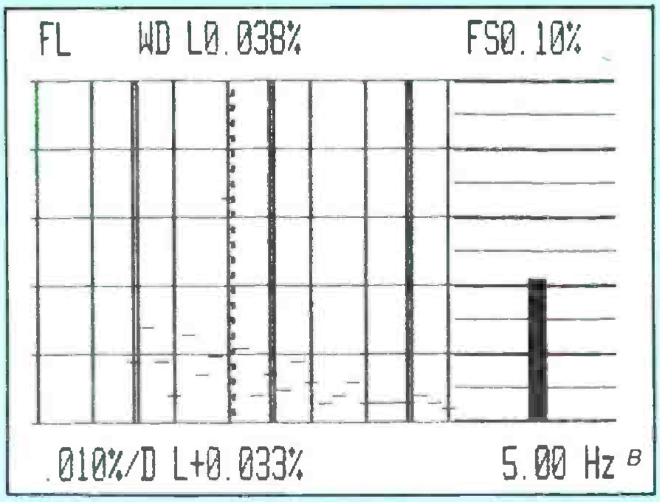


Figure 7. Wow and flutter analysis. Peak weighted study is shown in Figure 7A; WRMS study is shown in Figure 7B.

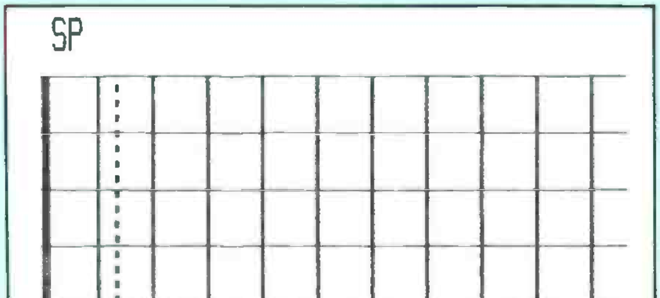


Figure 8. Speed accuracy plotted against time. Verticle scale is 0.2 percent per division.

SPECIFICATION	MFR'S CLAIM	MR&M MEASURED
Frequency Response (Hz–kHz,–3dB)		
Normal Tape (No Dolby)	20 Hz – kHz	20 Hz – 25 kHz
Normal Tape (Dolby B/C)	20 Hz – 16 kHz	20 Hz – 23 kHz
Type II Tape (No Dolby)	20 Hz – 20 kHz	20 Hz – 25 kHz
Type II Tape (Dolby B/C)	20 Hz – 18 kHz	20 Hz – 21 kHz
Metal Tape (No Dolby)	20 Hz – 23 kHz	20 Hz – kHz
Metal Tape (Dolby B/C)	20 Hz – kHz	20 Hz – 23 kHz
THD @ 0 dB (Ref. Level 250 nWb/m)(%)		
No Dolby (Normal/Typ.II/Met.)	1.8/2.8/1.5	.93/4.3/3.0
W/Dolby C (Normal/Typ.II/Met.)	N/A	.7/2.2/2.0
Level for 3% THD (dB)		
No Dolby (Normal/Typ.II/Met.)	N/A	+3.0/–1.0/0.0
W/Dolby C (Normal/Typ.II/Met.)	N/A	+4.5/+1.0/+1.0
Signal to Noise Ratio (dB)		
No Dolby (Norm./Typ.II/Met.)	See Text	56.4/61.5/65.0
W/Dolby B	64.9/71.0/69.1	
W/Dolby C	68.0/71.0/74.0	72.6/80.9/80.4

Wow and Flutter (Avg./Peak) (%) N.A./0.1 0.038/0.09
Speed Accuracy +/–0.5% –0.27%
Line Input Sensitivity (mV) See Text 200 mV adjustable
Line Output Level 1500 mV
Headphone Output Level N/A 500 mV/8-Ohms
Fast Rewind Time (Co-60) 35 Seconds 34 Seconds
Number of Motors 3 Confirmed
Number of Tape Heads 3 Confirmed
Bias Frequency N/A N/A
Power Consumption N/A 50 Watts
Dimensions (W × H × D, inches) 17-1/8 × 6-9/16 × 13-3/4 Confirmed
Weight 21.8 lbs. (9.9 kg) Confirmed
Suggested Retail Price: $ ______

notation below each graph represent the maximum output at 10 kHz obtainable with each tape, referred to a 0 dB at 250 nWb/m. These tests were repeated with Dolby C turned on, and as you can see from the results shown in *Figures 6A, 6B,* and *6C,* the high frequency MOL improved significantly when Dolby C was employed. For example, comparing the high-frequency MOL for Type I tape in *Figure 5A* (–7.4 dB), with the high-frequency MOL for the same tape when Dolby C is used (*Figure 6A*), we see an improvement of nearly 5 dB.

Peak wow and flutter for this deck measured 0.09 percent as against 0.1 percent claimed by Tandberg. (See *Figure 7A.*) Most manufacturers provide WRMS wow and flutter figures, and if the Tandberg TD-910 wow and flutter figure is measured that way, the figure obtained is only 0.038 percent (*Figure 7B*). Speed accuracy was well within claimed limits, measuring –0.262 percent as against +/–0.5 percent claimed by Tandberg. Required input and output levels appear in the VITAL STATISTICS chart, as do such other physical characteristics of the unit.

COMMENTS

The Tandberg TCD-910 is a good example of just how far cassette tape recording has come in the twenty-odd years since Philips introduced the compact cassette tape format. They intended the format, we are told, to be a convenient method for transcribing voices at business meetings and as a substitute for the then-bulky dictating machines used in offices. From those lowly beginnings we have progressed to the point where a machine such as the TD-910 can truly be used for many professional applications in which physical, or razor-blade editing is not required.

The combination of microprocessor control, excellent tape head construction and improvements in cassette tapes themselves, have made it possible for Tandberg to produce this truly elegant professional deck. If there is one criticism I can levy against Tandberg with regard to this machine it has to do with the poorly written owner's manual. With so many controls having dual (and sometimes even triple) purposes when used with other controls, a clearer explanation of how each function is to be achieved would have been helpful. As it was, we had to experiment on our own to puzzle through some of the very vague and incomplete instructions concerning programming and search modes, not to mention access to the various forms of display which the machine can deliver—once you figure out how to make it do so!

All of which does not detract one bit from the excellence of design and the superb performance of the deck itself. The professional recordist will want to experiment with bias and record-current settings to achieve somewhat lower distortion readings than we did. I strongly advise doing so; you will find that the machine has considerably greater headroom than is indicated by our own measurements performed after following Tandberg's adjustment procedures.

Construction of the TD-910 was superb; with all the ruggedness and reliability you would expect from a machine that is expected to deliver uninterrupted service for many hours per day. I'm not suggesting that this excellent cassette deck can take the place of your favorite reel-to-reel mastering deck in every instance. You will find, though, as I did, that there are times when a well-produced master cassette tape is all you need, and for those instances I can't think of many cassette decks that will do as good a job as this Tandberg TD-910 can do. ■

Jim Norman

Profile Of a Part-time Roadie

Find out firsthand about the trials and tribulations of (and shortcuts learned by) a part-time roadie.

db IS A MAGAZINE for everyone in the music and/or sound recording/reinforcement business. However, having read *db* for many years, I've seen precious few articles aimed at the small-time or part-time professional. We may not be the nations's top-dollar act, and we may not make ALL our bucks in the music or sound business, but we ARE PROFESSIONALS, make no mistake about it.

Normally, the mixer or technician with the big-buck touring group or sound contractor (usually) has the advantage of a complete technical staff, stage hands, technical support of his state-of-the-art system (yes, he's a valuable customer and good for endorsements), advance men, and sufficient time to set it up right. However, most small part-time country, pop, rock, and gospel groups, (and surprisingly enough, many full-time acts), that sustain themselves on one-nighters, have neither a roadie nor trained technician in the crew. Nor are they afforded the luxury of the time at each venue to go through a super-elaborate, highly complicated set-up—sound check, adjust/troubleshoot, re-check, tweak, re-check, etc.—procedure.

The usual situation is that the sound is mixed from the stage (house and monitor(s) on a single board or console), the mix is done by some picker or vocalist doing double-duty (VERY thick skin is required!!), and, after initial set-up and a brief sound-check (if time allows), it's "set it and forget it." Experience is a great teacher; a good ear helps. However, every trick and short-cut that can be learned should be filed away in the memory banks for, since Murphy's Laws haven't yet been repealed, the situation will repeat itself sooner or later—sooner, if you don't remember how you solved it the last time!

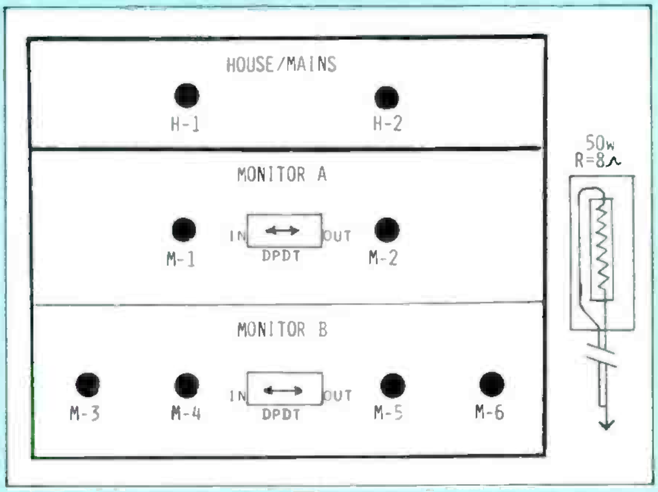


Figure 1. Diagram of speaker patchbay. (See text.)

In twenty-plus years as an untrained learn-as-you-go technician, roadie, mixer, fixer, and whipping-boy (oh yes, and bass singer) for several part-time gospel groups on the Southern gospel circuit, I've HAD to learn how to set it, fix it and run it—quick! With the entire group, and usually part of the audience, peering down my neck—all the while maintaining some semblance of decorum and stage presence; and a smile (the smile is the hardest!).

I won't bore the reader with details on my group's present sound system, but I'll tell you that the mains are large, horn-loaded, two-way cabinets with passive crossovers, and that the monitors consist of two- and three-way wedges, and several small, non-powered "personal" boxes. The rack contains two channels of 10-band graphics, a pink-noise generator, a real-time analyzer, and a Yamaha P2200. The Yamaha runs in stereo mode, with one side on the house and the other on the monitor mix (we use a single monitor mix).

Jim Norman labors in the contract sound/remote recording/production business under the trade name of Gospel Music Services, and has been a bass singer and soundman since 1963.

The board is a 12 x 2 x 1, with the usual sends, patches, eq section, bells and whistles.

The group consists of four vocals, with keyboard, steel, lead guitar, electric bass, and drums. I've found that this group will require very little variation in the mixes from show to show, so, after establishing a "good" mix in a fairly decent house (the band even said the monitors were "right" for a change), I decided to let the system set its own level. Using a hand-held sound-pressure-level meter (Radio Shack, approximately fourty dollars) and the eyes, ears and hands of a friend during the show, I found that we were playing at an average of 88-93 dB in the house, with 98-103 dB peaks, and endings—and at 93-98 dB on stage, with 103-108 dB peaks. After the show, I went into the audience with my trusty SPL meter, and had an assistant inject pink noise into an unused channel (all eq flat; no reverb), until I reached a reference level of 85 dB.

Then using the monitor send on the same channel, I repeated the procedure on stage, reaching the same 85 dB reference level. I then wrote down the entire mix, with particular notation of ALL settings of variables in the pink noise injection. At set-up for the next show, I went through my normal equalization procedure, using pink noise, a Shure ES-615 equalizer mic, and the RTA. Then I re-set my "start-up" mix, set up the pink noise levels on the unused channel EXACTLY as the reference set-up, then brought system gain up to the reference of 85 dB, USING THE MIDDLE OF THE HOUSE AS A REFERENCE POINT. I then repeated the procedure on stage, from the position of one of the vocalists. I then adjusted positioning of side-fills and other stage monitors for best band coverage and best approximation of reference noise level at all pickers' positions, and VOILA!...instant mix; instant levels.

We went through the show with no ears bleeding, and no complaints from EITHER side of the mics. Well, I did hear a few cracks out of the group, but I guess they need me to know that they're alive and listening. I've used this same basic premise to set up for about five years since, and it still works like a dream. No more of that "checking, testing, one, two, one, two, three; can you hear me in the back? Is this better?" stuff. I just tap each live mic and pluck a string on each direct instrument after set-up, to be sure everything's alive and well, then go somewhere and relax.

PHASING FOR FUN AND INTELLIGIBILITY

One of the most useful tricks I've picked up is that of proper phasing within the monitor system; more particularly, de-phasing as necessary, to fit given problem situations. As I will discuss later in more detail, there are many situations where a knotty intelligibility problem or incurably "muddy" or "thin" sound, even after equalizing the building and stage, can be solved by knowing which monitor(s) to knock out of phase from the rest of the system, AND having the guts to try it. After all, running one or more speakers out of phase is diametrically opposed to everything we've been told by those who are supposed to know; but then, what do I know? I'm an accountant. But I have learned a few tricks along the way, and I'll try to pass them along.

First, I've designed a speaker patch-bay, which incorporates two DPDT switches. Proper selection of patching and phasing (and usually some trial-and-error), can really help (nothing is absolute!), clean up some inherently bad rooms, as we shall see. *Figure 1* is a diagram of my speaker patchbay. All cabinets are nominal 8 ohms. For two mains, plug into M-1 and M-2. Impedance will be 4 ohms. For four mains, use series pairs into M-1 and M-2. Impedance will be 8 ohms. Plugs M-1, M-3, and M-4 are always in phase with the mains. Plugs M-2, M-5, and M-6 are phase reversable by the DPDT switches as to mains and plugs M-1, M-3, and M-4. In Monitor A, both plugs are parallel to eachother and to the mains. In Monitor B, the pairs are parallel to eachother, to the mains and to Monitor A. The 8 ohm, 50 watt resistor is hard-wired into the rack at any convenient location. Enough pigtail is available, terminating in a 1/4-in. phone jack to reach any plug in the bay. This gives capability to insert a dummy load into any point for impedance matching.

WHEN, WHAT, HOW

The capability to connect one or more of the monitors out of phase with the other monitors and/or the mains gives the operator more flexibility in curing problems inherent in small, highly reflective rooms with cramped stage areas and limited choices of main/monitor speaker placement arrangements. These types of rooms are common venues, along with the uncontrollable, unforseeable variables, for the part-time, semi-pro gospel quartet, country band, cabaret band, etc.—with more frequency than most would like, for many full-time touring pros. Ditto for the gym, rec-hall, and other "blimp hangar"-type rooms which come equipped with their own sets of problems. Either a closet or a cavern, never a symphony hall.

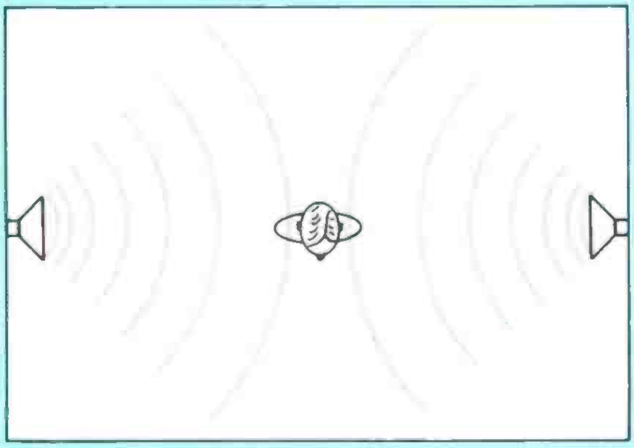


Figure 2A. Perceived low frequency SPL appears lower due to low frequency cancellation.

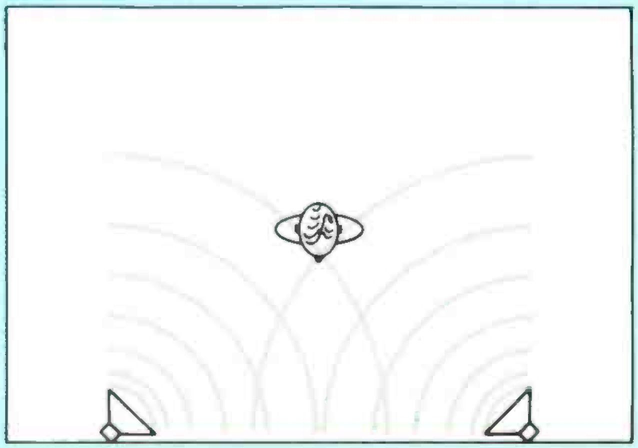


Figure 2B. Effect decreases as angle of opposition decreases.

Three common problems, their causes, and a suggested solution for each, are cited. These problems always affect the sound perceived by the performer AND the audience. Therefore, after selecting a proposed solution by phasing, ALWAYS run an A-B comparison for listeners in several typical audience listening points, and select the setting that is most pleasing to the audience, even to the detriment of the monitor sound, if necessary (remember who's paying you). If, to get a pleasing house sound, the monitor sound must be degraded beyond the point of acceptability, then a combination of slight lowering of monitor levels and judicious low frequency eq adjustments in both systems will usually achieve an acceptable compromise.

FACT #1: WHENEVER the listener is directly between two speakers [on the line between facing (opposing) speakers], and the speakers are in phase, the PERCEIVED low frequency SPL will appear lower to that listener due to low frequency cancellation of the opposed drivers. This effect decreases as the angle of opposition from the listener decreases from 180° towards 0°, and is negligible below 45°. *See Figure 2.*

FACT #2: The house speakers and monitor speakers will ALWAYS (well, most of the time, anyway), interact with each other AND the building to cancel OR reinforce the low frequency SPL perceived by the audience, the performers, or both. Sometimes the effects are opposites as between audience and performer. Some of the factors involved are:

A. Size of room and interior construction;
B. Shape of room, interior angles;
C. Distance of speaker(s) from, and angle of opposition to, opposing wall(s);

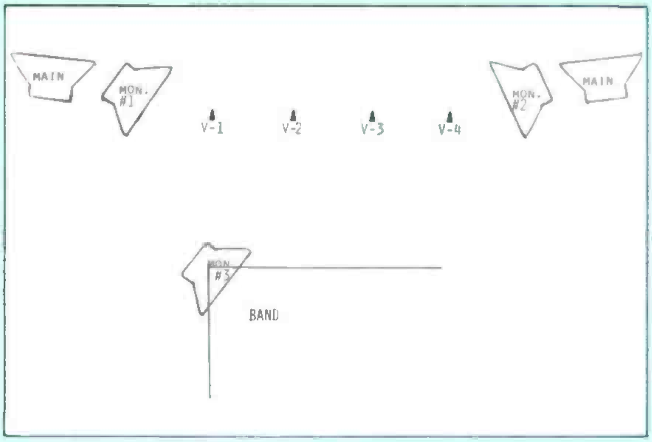


Figure 3. Illustration of possible small church or club set-up.

D. Reflectivity of front and rear walls and, to a lesser degree, side walls, floor and ceiling;
E. Resonant frequency of the room;
F. Tendency of room to retain standing waves;
G. And number of people in the room (may all your halls be full).

Unfortunately, we never play a live show in a recording studio environment, so some or all of the above come into play in every venue.

PROBLEM #1: Front monitors or side-fill monitors, or both, must be opposed (nearing 180°), due to stage layout or space limitations. The illustration (*Figure 3*) could be a small church or club. The performers shown as V-2 and V-3 will perceive low frequency cancellation. Performers V-1 and V-4 should not be bothered, nor should the band.

SOLUTION: The speaker shown as Monitor #2 should be run out-of-phase with the other monitors and the mains. Check response in-house for adverse effects as in problem #3 below.

PROBLEM #2: The performers are forced to set up BETWEEN the mains and their front monitors, (*Figure 4*). This, of course, is the least desirable of all possible set-ups, due to feedback, etc., but sometimes it is totally unavoidable. Control feedback with proper mic selection and eq. The performers shown as V-1 and V-4 will perceive low frequency cancellation; performers V-2 and V-3 may or may not, depending on variables too numerous to mention here. The band may or may not perceive low frequency cancellation; depending upon proximity too, and relative levels of monitors 1 and 2 should be run out-of-phase with the mains. If low frequency cancellation is perceived in the band area,

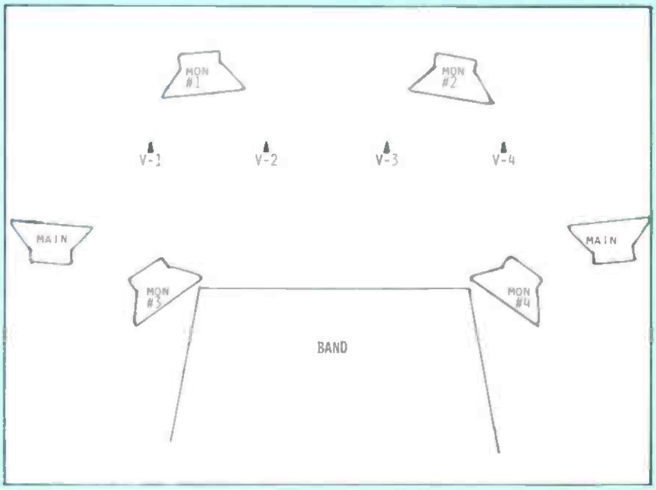


Figure 4. Undesirable set-up due to feedback.

monitors 3 and 4 should also be run out-of-phase. Check response in the house for adverse effects as in problem #3 below.

PROBLEM #3: After all set-up is finished, eq is ruler-flat, and final tweaking complete, both house and monitor systems are fired up together and, EGAD, something about the bottom end just isn't right! What has probably happened is that interaction of direct and reflected energy from mains and monitors have caused either perceived cancellation or reinforcement of low frequency energy to the house or the performers. Either is undesirable. When the effects are opposite upon house and performers, the result can be chaotic, since the knee-jerk reaction (volume/eq corrections) to correct the one will usually increase the adverse effect upon the other.

SOLUTION: When cancellation or reinforcement of low frequency energy is detected in either the house or stage area, the monitors should be un-phased either singly or in pairs, with a quick A-B comparison made in both locations after each change, until the best combination is found. The end result can be tweaked with subtle gain/eq changes to obtain the optimum result between bad and fair. (Pardon my sarcasm, but I've found VERY few venues where very-good-to-excellent results can be achieved at both ends.) If a trade-off becomes necessary, always trade in favor of the house. When you want to hear yourselves to perfection, look for it in the studio or the rehearsal room! And remember, KEEP SMILING! ■

jon gaines

CONSTRUCTION PROJECT: LED BAR GRAPH METER

Completed project

As part of a series of articles on electronic "building blocks" for audio, this article describes the construction of a 12-segment LED meter that may be built as a stand-alone project or combined with other equipment. The need to visually monitor audio signal levels has traditionally been met with VU (volume unit) meters and, more recently, segmented type meters including LED, LCD, and vacuum fluorescent displays. VU meters provide a simple and visually attractive means of monitoring the average loudness of an audio system, but have the disadvantages of being expensive, bulky, and somewhat fragile due to their delicate mechanical construction. Also, the meter's inherently slow response time makes it unsuitable for monitoring fast rise time signals, ie. the peaky transients that are an important characteristic of pop music. Further, the mechanical meter's low impedance requires an active buffer to be used to drive it so that it does not load down the line and add distortion to the audio signal.

Totally electronic meters, such as the LED meter presented here, have none of the limitations of mechanical meters. The speed of the meter's response is determined by the detector chosen to drive the display. Peak response, average response, PPM, peak-hold, and VU are some of the possibilities. Early versions of this type of LED meter used strings of LEDs driven by individual comparators and generally required lots of close tolerance components to make the meter accurate. In recent years, however, at least two semiconductor companies, National Semiconductor and EXAR, have offered ICs that consolidate most of the functions and parts of early discrete meter circuits into 18-pin DIP packages, making it easy to build LED meters that are simple, accurate, compact, and cheap.

This construction article focuses on the EXAR 2278 Bar Graph Display Generator and provides practical information on the applications of the IC. The 2278 features a built in peak detector circuit that's flat from 1 Hz to over 100 kHz and drives twelve individual LED's over a nominal display range of —20 to +8 dB. The project requires a single supply voltage of 10 to 14 volts DC, with very low current requirements. A unique series configu-

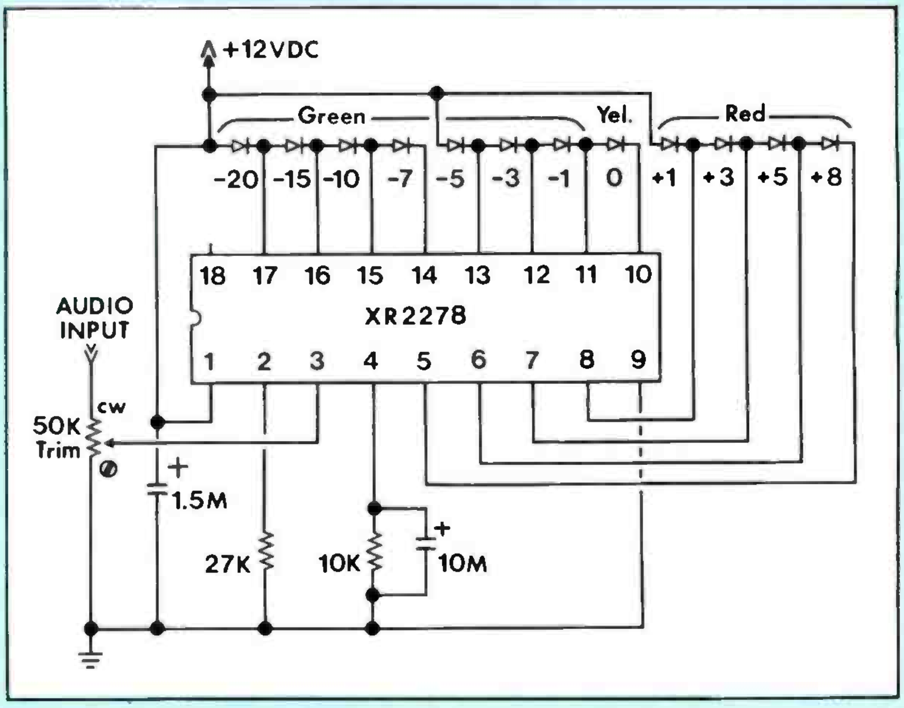


Figure 1. Schematic

ration for the LEDs keeps current consumption under 50 mA total. Constant current outputs for the LEDs insure consistent display brightness even if the power supply fluctuates, and LED brightness is even readily adjustable. Any size, shape, and color LED may be used with the IC. The high impedance buffered input allows you to hang this meter onto any audio line without fear of loading or degrading the audio source. Input level and nominal zero are adjustable via an on-board trimmer to match your system's nominal zero level. Finally, the decay characteristic of the display is easily adjusted and a peak-hold function can be incorporated just by adding a switch. And all of this with virtually no parts!

Potential applications for this project range from a single channel test meter for monitoring signal levels throughout your system to 8- or 16-channel rackmount versions that enhance the metering of your console or multitrack recorder. The unusually simple powering requirements make this a good project for first time kit builders.

The schematic diagram shown in *Figure 1* is the complete meter project based on the XR-2278. Component values shown are for typical decay characteristics and brightness. *Figure 2* gives a layout with which you can fabricate a printed circuit board, and *Figure 3* shows the component locations as viewed from the component (non-copper) side of the board. If you are using a printed circuit board, simply insert the parts as shown, paying attention to correct orientation of the LEDs, capacitors, and the integrated circuit. ICs usually have a notch of some kind at one end, indicating the "pin 1" end. Pin 1 is also indicated by a tiny circular indentation in one corner of the IC. The remaining pins are counted consecutively, counterclockwise from pin 1. This is true of all ICs in dual-in-line (DIP) cases.

Most LEDs have a "flat" on one side of their plastic lens, sometimes indicating cathode, sometimes anode, sometimes neither as is the case with the ones I chose to use. Check the literature that comes with the LEDs you use to make sure you put them in correctly.

Solder carefully with resin-core solder, avoiding the use of excess solder and heat. Be sure to use a socket for the IC. *Figure 4* is a photograph of the completed circuit board. Note that the LEDs have been bent over at a right angle and the capacitor leads are also bent so that they lay flat to the circuit board. You may not need to bend the components, depending on how you eventually want to mount the card to its panel.

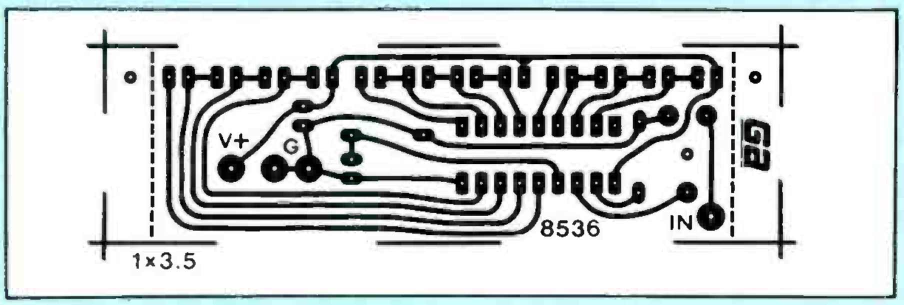


Figure 2. PC Board artwork. This view is from the copper side of the board.

POWER SUPPLY REQUIREMENT

The XR-2278 requires a single supply of 10 to 14 volts, DC (15 volts absolute maximum) at 50 mA. You can either build a dedicated 12 volt supply or purchase a standard plug-in supply of the type commonly used to convert AC for tape recorders and calculators. For example, Radio Shack sells an AC Power Adapter (Cat. No. 273-1652) which will power up to ten of these meter cards with all lights on simultaneously and even more under normal

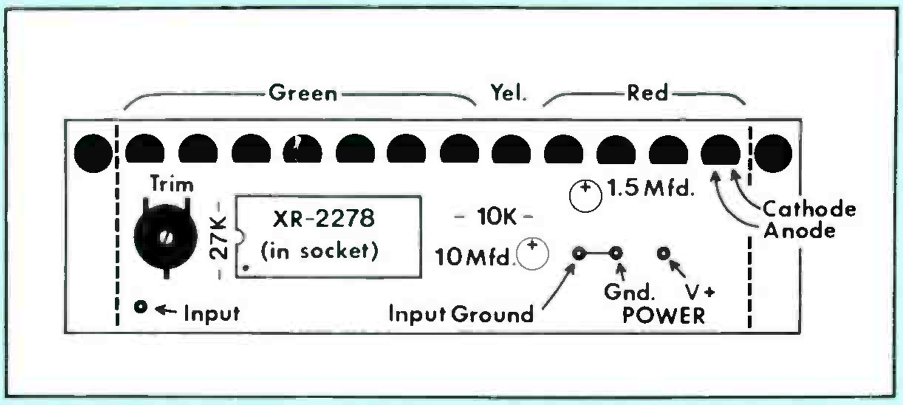


Figure 3. Component insertion guide. This view is from the component side of the board.

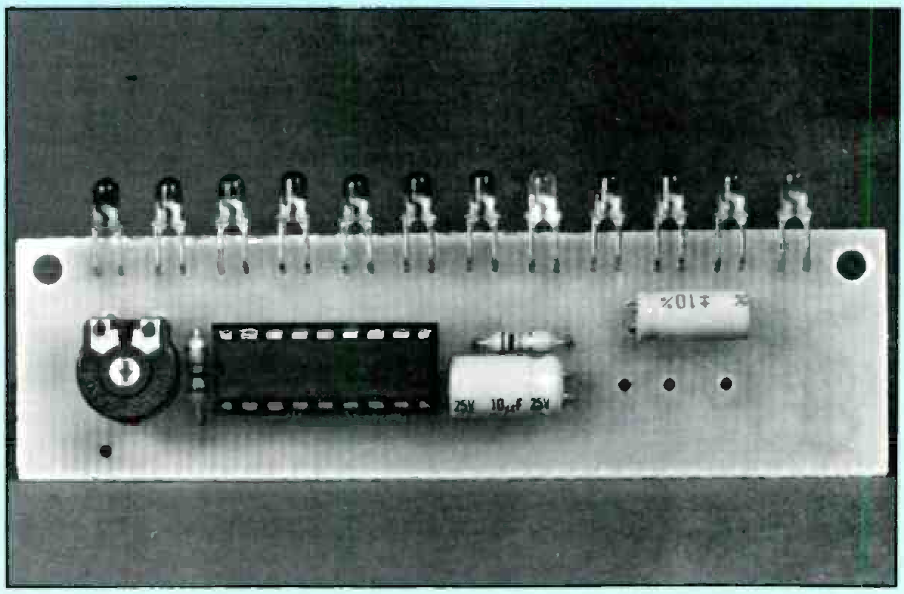


Figure 4. Completed circuit board.

program conditions. The adapter is adequately regulated for this application and sells for about $10.00. Be careful that you get the polarity of the supply correct when you hook it up to the circuit board or damage is inevitable.

If the meter is to be retrofitted into existing audio equipment, you may be able to share that unit's power supply. Most contemporary audio equipment has an internal +15V DC supply. Depending on how much "headroom" a particular supply has, you might be able to draw an additional 50 mA from it without overload. If the supply is marginal and the regulators are already running hot, consider other options. If in doubt, consult the manufacturer of the unit.

Note that the XR-2278 specifies a maximum supply voltage of +15. Therefore, if you do use a +15V power supply with this project, it would be a good idea to drop the supply a bit by inserting a couple of diodes between the supply and the PC board as shown in *Figure 5*.

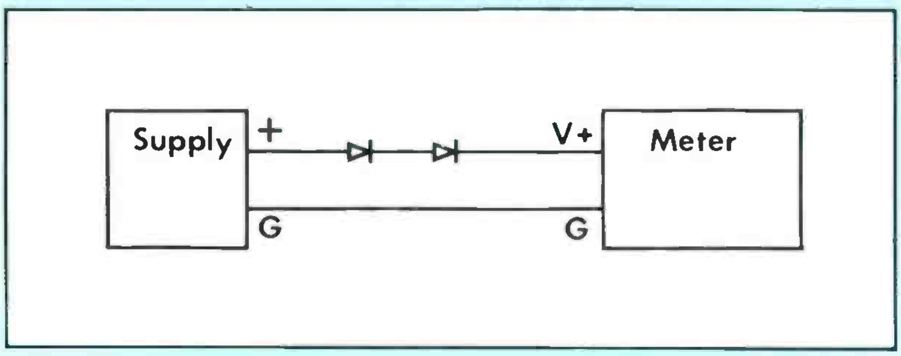


Figure 5. Dropping a +15V supply to +13.5V with two 1N4148 diodes. This step is unnecessary if operating from a 10 to 14 volt supply.

OPTIONAL MODIFICATIONS

Peak Hold: The decay time of the display is determined by the resistor/capacitor combination attached to pin 4 of the IC. By increasing either the 10K resistor value or the 10 microfarad capacitor you can slow down the decay time of the display without affecting the attack time. Stretching this concept a bit further, a peak hold function can be accomplished simply by disconnecting one leg of the 10K resistor for as long as you wish to hold the peak level. A SPST switch, either momentary or latching, is all that's required.

Brightness Adjustment: The LED intensity can be dimmed by increasing the value of the 27K resistor attached to pin 2. You may substitute a 50K or 100K potentiometer, but keep the 27K resistor in series with it.

CALIBRATION

A trim pot is provided to allow you to set the 0 dB reference level for your system. Set fully clockwise, the meter will display its zero LED with an input level of approximately −20 dBV. Turning the trimmer counterclockwise allows you to pad the input for higher level signals, ie. 0 dBV, +4, +8, or even speaker level inputs. A 0 dB input should just light the yellow LED after you've adjusted the trimmer.

MOUNTING THE METERS

The meter card layout was designed with two alternate mounting positions in mind. In the first, the LEDs extend straight out from the PC board, 3/8-in. standoffs are attached, and the card mounts parallel to the front panel of the chasis. Or, the LEDs may be bent over at a right angle as in the photo of *Figure 4*, in which case the card mounts perpendicular to the front panel. The latter position allows for denser packaging of multiple units. However, since the standoffs would now be oriented incorrectly for screw mounting, you'll need to attach the meters to the panel with a dab of epoxy.

EXTERNAL WIRING

Connect the power supply to the pads marked V+ and G. Connect the audio input to the pad marked IN and the second ground pad. It's then a good precaution to check for correct power supply operation with a voltmeter before plugging the IC into its socket. You should see +12V on pin 1 and ground at pin 9 of the XR-2278.

PARTS LIST

10K ohm resistor, 1/4 watt
27K ohm resistor, 1/4 watt
50K ohm trim pot
1.5 Mfd. electrolytic capacitor, rated at 16 volts
10 Mfd. electrolytic capacitor, rated at 16 volts
EXAR XR-2278P Bar Graph Display Generator IC
18-pin DIP socket
Green LEDs (7)
Red LEDs (4)
Yellow LED
3/16D x 3/8L inch standoffs (2), 4-40 tapped, with screws
Printed circuit board

KIT INFORMATION

Parts for this project are available from Gaines Audio, PO Box 17888, Rochester, NY, 14617, (716) 266-0780, as follows:

Complete meter kit as described, minus chassis and power supply, $19.95, plus $2.00 shipping.

Circuit board only, $7.50, postage paid.

IC only, $8.40, postage paid.

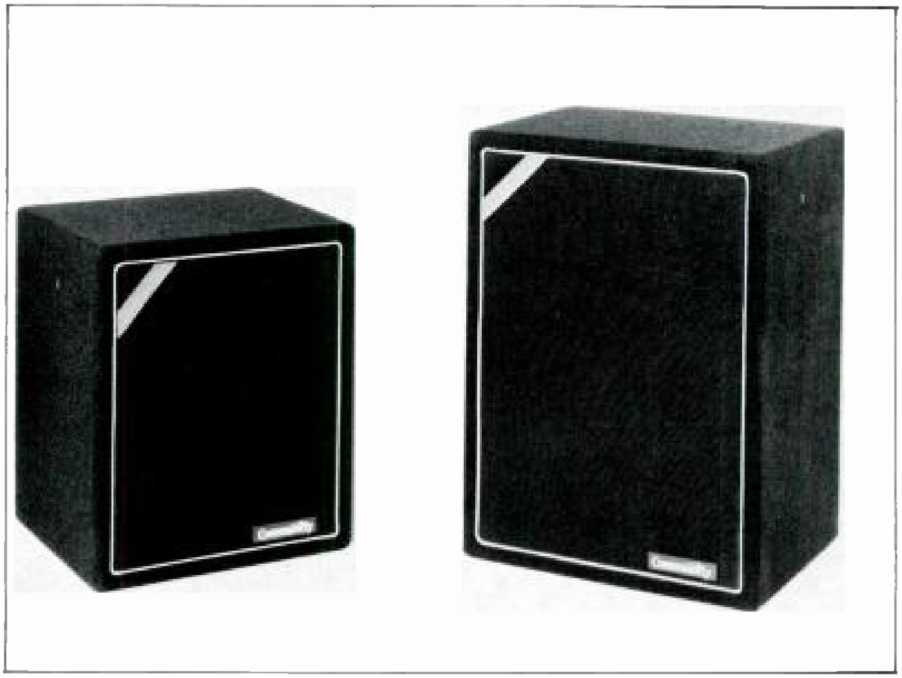
Money order, VISA, MasterCard, COD, and checks accepted. NY residents, please add 7% state tax. Thirty day return privilege for full refund if not satisfied.

SPECIFICATIONS

Range	+8 to −20 dB
Frequency response	1 Hz to 100 kHz
Input impedance	50k ohms
Input range	−20 dBV to +30 dBV
Supply voltage	+10 to +14 VDC
Supply current	50 mA maximum, 4 mA minimum
Size of PC card	1 x 3 1/2-inches
LED spacing	1/4-inch between centers

COMMUNITY'S CS25/CS35 MULTI-USE LOUDSPEAKERS

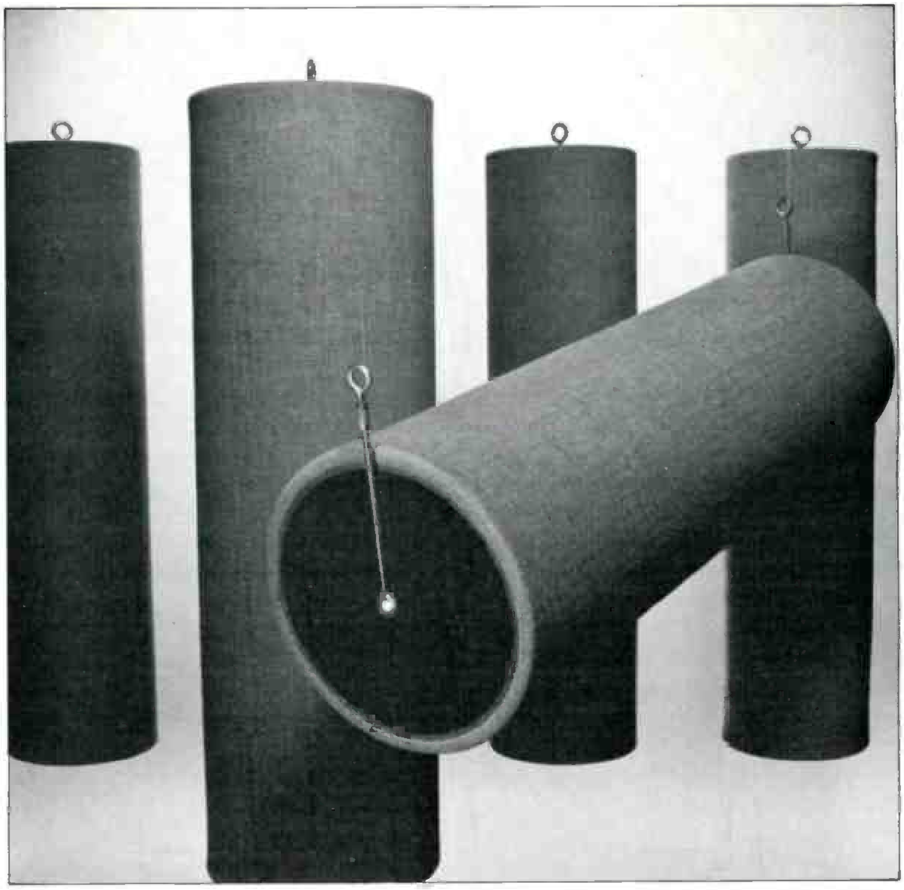
• Community Light & Sound's new series of performance/installation loudspeakers, the CS25 and CS35, provides working musicians and club/restaurant owners with high-sensitivity speakers of a reliable and cost-effective nature. Basing the new speaker's design on the company's years of in-house research and installation experience, the units are exceptionally appropriate for musicians and establishments that can't justify huge case outlays for a "wall of sound" approach, and also can't afford equipment breakdowns. The CS25 features a 12-in. driver with a single high-frequency horn, while the CS35 has a 15-in. driver with an integral high-frequency horn and ducted port assembly. A variety of Community designed features assure consistently fine highs and lows, smooth wide-angle coverage, and reliability in all situations. A specially designed filter network on the horns provides both units with exceptional crossover characteristics and high sensitivity. The CS35's 15-in. driver is cooled with a ferrofluid that increases power handling. Both loudspeakers incorporate driver protection systems that are extremely reliable—they are automatic systems that don't use fuses and allow the cabinet to always operate. Constructed of rugged but lightweight 3/4-in. fiberboard, the CS25 and CS35 cabinets are covered with a durable non-woven black carpet that simultaneously protects the speaker and anything it bumps into. A recessed pocket handle in the back of each cabinet is also the location of the input connector. Placed within the handle cavity, the input and its 1/4-in. connector are thus protected from damage if the speaker happens to be knocked over or bumped. The cabinets also feature socket-stand adaptors, so the speaker can be slipped over a stand to raise them for better sound dispersion. Standard T-nuts on the sides and top and bottom of the cabinets facilitate permanent hanging installation—optional brackets provide easy wall or ceiling mounting. The CS25 weighs 32 lbs.; the CS35, 43 lbs.



Mfr: Community Light & Sound, Inc.
Price: CS25-$239.00; CS35-$289.00

Circle 55 on Reader Service Card

ASC LOW END TUBE TRAP

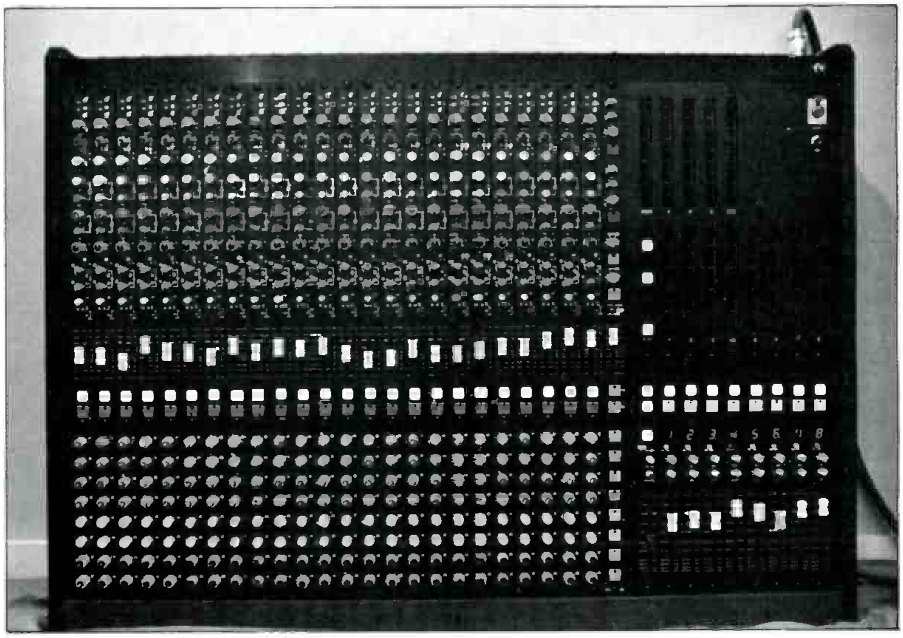


• ASC has developed the Tube Trap, a modular line of small, efficient, lightweight, portable and patented bass traps. These 3-ft. tubes can be used alone, stacked into columns, installed with wall or ceiling hangers, or used as a studio gobo system. The Tube Trap is a pressure zone trap, it's best utilized when backloaded by a wall or corner. Tri-corner installation in rectangular rooms damps all major room resonant modes effectively. Two models are in production: M-90 a voice range trap, 9-in. in diameter that provides 10 sabines per 3-ft. section absorption, and the M-45, a trap that ranges an octave below to 45 Hz, 11-in. diameter providing 15 sabines per section. Tube Trap installations clean up transient distortions due to room acoustics. Low end phase shifting in the attack of a tone burst is reduced by trapping the first, strong reflection. Room resonance modes that color the tone burst decay, muddy the sound, and cause a low end build-up, are controllable by Tube Trap placement. The result is a faster and more detailed low end acoustic room. All Tube Trap products have a "midrange crossover diffusor panel" section that keeps the sound field balanced even when listening in the nearfield of the absorption unit. Rotation of the tube adjusts mid-range brightness. Lower frequency units are also available upon order. Typically, the M-20 operates thru 20 Hz. ASC has built Tube Traps to 5 Hz. Modular half round Tube Traps that provide mid-bass wall treatment are also available.

Mfr.: Acoustic Sciences Corp.
Price: About $12.00/sabine.

Circle 58 on Reader Service Card

MEYER SOUND LABS STAGE MONITOR CONSOLE



• Meyer Sound Labs is distributing a very high quality, mid-sized monitor console. The console is a combined effort of Meyer and its Japanese distributor, Acoustic Technical Laboratory (ATL), and will be available in limited quantities for those interested in high fidelity stage sound. Available configurations are 24 x 12 and 32 x 12. All twelve outputs have large segmented LED metering that may be switched to VU or peak reading. Any of the twelve mixes may be reassigned in any order to the 8 main outputs via a fast electronic matrix assignment system. Each transformerless input channel has switchable phantom power, a high pass filter, and calibrated 4-band true complementary eq. There is also a switching system to allow a master fader to control the input send to the matrix. Insert points and direct outs are furnished for each input. Monitoring solo points are at input, summing, and output stages and peak indicators at each of these stages allows distortion free operation. Talkback can be assigned to individual outputs for improved musician-mixer communications. Two auxillary inputs can be used to route effects to any output.

Mfr.: Meyer Sound Labs
Price: $40,000 for the 24 x 12; $48,000 for the 32 x 12

Circle 60 on Reader Service Card

EVENTIDE REVERB AND EFFECTS SOFTWARE

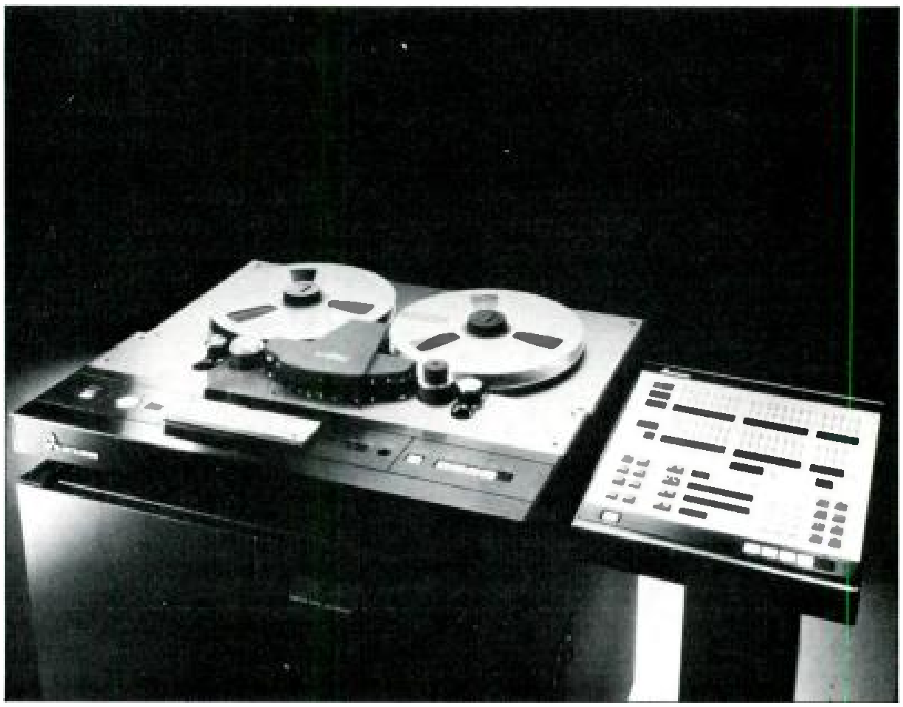
• Eventide has enhanced its SP2016 Effects Processor/Reverb with four new reverberation programs, a vocoder program and an automatic panner program. The RMX Simulation Plus programs provide accurate simulations of two of the most popular AMS RMX 16 reverb programs: Reverse Reverb and Non-Linear Reverb. But unlike the single channel AMS unit, the Eventide SP2016 gives the user two independent channels simultaneously. As a counterpart to these special-effect-decay reverbs, a "Natural" Reverb, featuring natural decay ambience is also being introduced and a Gated Reverb program has also been added. All of these new reverb programs have been added at no extra charge to the standard Generation II software package, and are available as free software enhancements to present SP2016 owners. Each SP2016 now comes with a total of ten reverberation programs and twelve effects programs. The newly-released Automatic Panner program provides delay panning as well as amplitude panning functions. User-adjustable parameters make possible a wide variety of crossfade and panning effects. This program is available as an option on new SP2016s as well as units in the field.

The new Channel Vocoder ROM is also available as an option. Another Eventide exclusive, this program highlights the versatility of the SP2016. With the Channel Vocoder ROM, the SP2016 can function as a full 18-band, professional quality vocoder, with performance that compares favorably with the best high-end single purpose vocoders.

Mfr: Eventide Inc.
Price: $6,895.00

Circle 57 on Reader Service Card

MITSUBISHI 32-CHANNEL DIGITAL AUDIO RECORDER



• The recording format of the Mitsubishi Pro Audio Group X-850 32-Channel Digital Audio Recorder is in full compliance with the recently agreed PD Pro Digital format standard between Mitsubishi, AEG, and Otari. The X-850 is fully compatible with the digital recording format of the predecessor model X-800, providing a total of 45 tracks on 1-in. tape There are 2 analog cue tracks, 2 digital auxiliary tracks and 1 time code track, in addition to the 40 tracks used to provide the 32 channels of digital audio. The X-850 is capable of cut-and-splice editing and then overdubbing over the mechanical splice. Other unique features include the RS-422/RS-232 serial interface to other recording or synchronizing systems, as well as the ability to accept sync inputs on 9.6 or 8 kHz, 50, 60, or 59.94 Hz and composite video. A new attractive appearance design for the machine, as well as the remote control system, makes the X-850 stand out as a true high technology studio system. It can accept up to 14-in. reels of tape providing more than one hour recording capability on a reel. By utilizing the digital output ports, master multitrack tapes can be digitally dubbed for copying purposes without any generation loss whatsoever. The X-850 features a high quality error correction system. Up to 8 tracks could be totally lost due to tape dropouts, head damage, or even clogged heads, before there would be a noticeable loss in audio quality. And all channels remain available to record and play-back, even punch-in and punch-out. All 32 channels are available all of the time, due to the power of the Mitsubishi error correction scheme.

Mfr: Mitsubishi Pro Audio Group
Price: $154,000

Circle 59 on Reader Service Card

DOLBY XP SERIES MULTITRACK

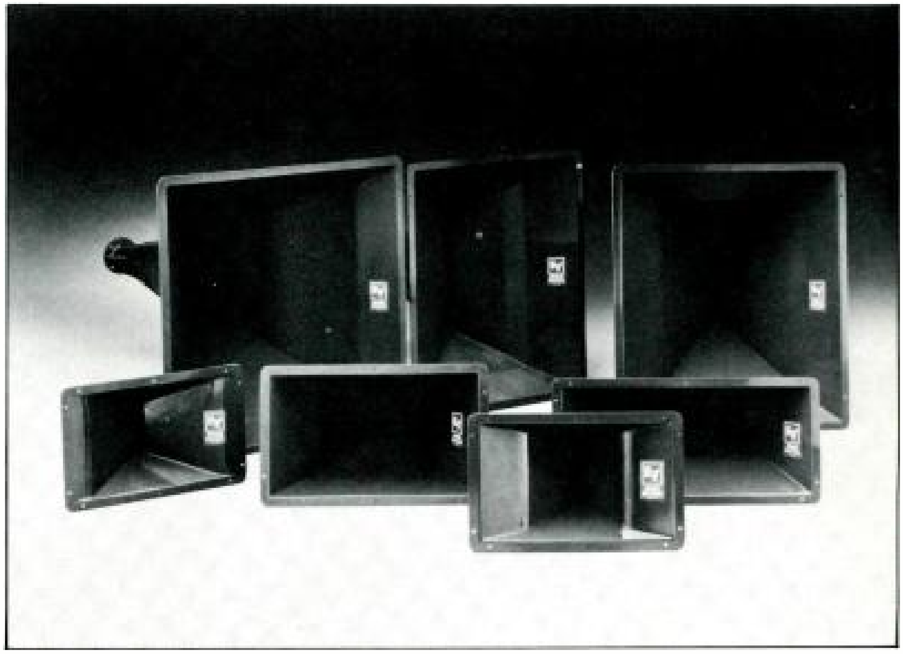
• Dolby Laboratories' new multitrack noise reduction package, the XP Series, contains up to 24 channels of Dolby A-type noise reduction, a 4-band, low-level signal processor offering 10 dB signal-to-noise ratio, and crosstalk and print-through improvement. The XP Series is identical in performance to the SP Series multitrack noise reduction unit introduced by Dolby Laboratories in 1981. Among the XP's cost-effective engineering changes are a new power supply (Model PS3), and dedicated noise reduction circuit design, rather than carrier card/plug-in assemblies. The XP also features detented calibration trim controls, discrete FET noise reduction control switching, and individual channel hard-wire bypass.

Mfr: Dolby Labs

Price: $14,950 for 24-tracks

Circle 52 on Reader Service Card

ELECTRO-VOICE CONSTANT-DIRECTIVITY HORNS



• EV's TransPlanar HP high-frequency horns offer the uniform beamwidth control with flared and conical surfaces. The series' three "large" HP horns maintain rated beamwidth control down to 500 Hz in both the vertical and horizontal planes. The four "small" HP horns maintain horizontal beamwidth control to 500 Hz and, with smaller, more convenient vertical dimensions, exhibit vertical control to 1,500 Hz. The HP horns have mouth sizes only as large as necessary to maintain the rated coverage angles down to the low-frequency limits described above. This permits both compact cluster design and exceptional directivity control. Cast into the metal throat section of every HP horn, angled waveguides restore full coverage patterns above 10,000 Hz, correcting the problem of high-frequency beaming, or narrowing of coverage angle, found in other 2-inch-throat constant-directivity designs. Constructed of non-resonant, fiberglass reinforced composite, all HP designs feature a smooth transition from radial to planar surfaces. This unique joining of curved and straight-sided geometries results in unusually smooth frequency response, optimal driver loading, uniform coverage to 20,000 Hz and extremely convenient mounting. The throat section terminates in a flanged, 2-inch exit with a four-bolt mounting pattern, the "standard" format for many applications. All horns are directly compatible with EV's new DH1 and DH2 2-inch exit drivers. The 2-inch, die-cast metal throat is captured in the horn's fiberglass sidewall during the manufacturing process, giving the HP horn and throat assembly the strength of unitized construction. Encapsulating the throat in this manner eliminates the cost and inconvenience of bolt-on throat sections. They have a flat, predrilled mounting flange which simplifies installation in both cabinets and component clusters. All large format HP model numbers describe the horn's nominal coverage pattern: HP4020 (40 ° H x 20 ° V), HP6040 (60 ° H x 40 ° V), and HP9040 (90 ° H x 40 ° V). The model numbers of small format horns abbreviate coverage-angle information: HP420 (40 ° H x 20 ° V), HP640 (60 ° H x 40 ° V), HP940 (90 ° H x 40 ° V), and HP1240 (120 ° H x 40 ° V).

Mfr: Electro-Voice, Inc.

Prices: HP4020-$645.00;
HP6040-$430.00;
HP9040-$430.00;
HP420-$310.00;
HP640-$187.00;
HP940-$180.00;
HP1240-$187.00.

Circle 53 on Reader Service Card

• **West Oak Recorders** of Westlake Village, California, has become the first commercial studio in the United States to use the **Sony 1630 digital processor**. Rented from CMS Digital, Inc., the 1630 was utilized in a project featuring a seventy-five piece orchestra with Sarah Vaughan singing translations of poetry of Pope John Paul II. The album will be released on Jazzletter Records.

• **F. Davis Merrey, Jr.** has been appointed president of **Altec Lansing Corporation** of Oklahoma City, Oklahoma, a leading manufacturer of sound products for commercial and professional sound systems. Merrey holds a B.S.E.E. degree from the Virginia Military Institute and an M.B.A. degree from Golden Gate University. He has worked as manufacturing manager for Transaction Systems, and was later manager of product development for Litton Dental Products. He joined Electro-Voice in 1976 as manager of one of their plants, and later was made general manager of Electro-Voice's EV-TAPCO facility in Seattle. In 1981 he became Electro-Voice's vice president of operations...**Gary H. Rilling** has been named vice president of marketing, and a corporate officer of Altec Lansing Corporation. In his new post, Rilling will be in charge of distribution and promotion of Altec Lansing sound products world-wide. He studied industrial electronics for two years at the Pennsylvania Institute of Technology, and, in addition, studied electronic theory part-time for four years in a technical school in Philadelphia. He worked for a number of sound contractors as a salesman, technician, designer, and administrator. He joined Altec Corp. in 1974 as district manager, and was later promoted to regional manager. He was brought to company headquarters and made national sales manager in 1979, and then was made vice president of Commercial Sales (USA) in 1981. Rilling is also a member of the Audio Engineering Society.

• A recent addition to Miami-based **Pat Appleson Studios, Inc.,** is a new custom-designed 8-track stereo mixing console installed in Appleson's Studio A. Manufactured by Pacific Recorders & Engineering Corporation of Carlsbad, California, to president Pat Appleson's rigid specifications, this custom **ABX-34** model is the largest configuration produced to date, according to Pacific president Jack Williams. The console's mic input circuitry and mix buss circuitry incorporate the latest Dean Jensen design for transformerless operation and very low noise floor with fast rise time. It is regarded as one of the smoothest and cleanest boards ever built.

As part of the expansion process, Appleson Studios acquired an additional 2,000 square feet to accommodate the construction of another studio as well as additional personnel, most of whom are already on staff.

There has been an ongoing equipment replacement program since mid-1984. All 2-track decks have been replaced with new Otari 5050B-IIs, with two more decks scheduled for replacement this year. Four Harmon-Karden CD491 cassette decks were also added. These decks feature Dolby HX professional head room extension kits that give excellent high frequency performance without the need for any type of decoding on playback. Other additions include an Aphex Systems Aural Exciter-Type B and the renowned Aphex Compellor Leveling System. The Patch Bay has been totally replaced with ADC's Ultra-Patch. Other improvements include an updated monitoring system throughout the plant using Hafler P-500 VMOSFET Power Amplifiers in BI and Tri-amp configurations. Since quality assurance level of masters not originating in-house has always been a problem, Appleson Studios has acquired a Pacific Recorders & Engineering Stereo Display Instrument. It is a combination of a stereo monitor oscilloscope and a phase compatibility meter. Also, a Dorrough Electronics Model 40-A Loudness Monitor is used to check for uniform loudness from source to source.

Other additions include: stereo monitoring in the announce booth, stereo cue on all channels, interlocked digital timers for engineer and talent, plus a sundry of tweaks and enhancements. Custom-built illuminated remote panels offer complete machine control from the mix position.

According to president Appleson, an often overlooked source of trouble is the a.c./mains supply. To insure line protection from spikes and brown-outs a new dedicated cable was run from the FP & L (Florida Power & Light), utility box to a main tech distribution point. There the current hits a dynamic power conditioner and isolation transformer in addition to general electric metal oxide varistors and on to the studio where all equipment plug strips have more MOVs and the faster acting Panasonic ZNRs. Calibrated line/load meters are used at the tech power distributution point and a duplicated set sits on the console overbridge in the control room.

For both audio and video, Appleson Studios' "concept through completion" service is designed to provide clients with everything they need to produce successful material.

• **Acoustic Spaces Corporation** has made Woodstock, New York, its east coast base. Offices have been established in the recently completed **Dreamland Recording Studios** which was designed and constructed by Acoustic Spaces and KDP Engineering.

• In response to increased sales in the area, the magnetic tape division of **Agfa-Gevaert, Inc.,** Teterboro, New Jersey, has announced that **Scott Kaplan** will join their Los Angeles staff as technical sales representative. Kaplan will provide technical and sales support for all Agfa tape products in Los Angeles, Santa Barbara and Hawaii. Prior to joining Agfa-Gevaert, Mr. Kaplan was assistant sales director at Studio Film & Tape. He received his B.A. degree from S.U.N.Y. at Binghamton, New York...**Donald E. Rushin** was named marketing operations and international director and **Joseph L. Leon** was named sales director, both new positions, in 3M's magnetic audio/video products division for its broadcasting and recording and commercial and educational markets. Rushin will direct the marketing organization for the division's BR/C&E markets with ongoing responsibilities for international marketing in the professional business. Leon will be responsible for the division's professional sales force. Rushin most recently was international director for the division with twenty-six years in the company; Leon, who joined the company in 1959, was director of professional markets.

• **Tom Irby**, a professional audio industry veteran has been appointed to the position of operations manager of **Valley People**. Irby has been involved with the audio industry in various capacities since 1972. In that year, he helped coordinate the development of a SMPTE time code synchronizer while with 3M Company. Later, Irby, was promoted to southeastern sales manager for 3M's Mincom Tape Recorder Division, where he remained until 1974. At that time, he became a sales representative with Studio Supply Company. Eventually, Irby purchased Studio Supply, where during his ownership he designed studios and supplied equipment packages for noted performers and producers such as, Willie Nelson, Larry Butler, Mickey Gilley and Chips Moman. Earlier this year, Studio Supply was sold to John Alderson. As operations manager at Valley People, Inc., Mr. Irby will be responsible for overseeing all aspects of the company's daily activities. Ray Updike, sales manager of Valley People, Inc., has announced the appointment of the following representatives for the company's product line: Darmstedter Associates, Electro-Acoustic Marketing, Wilson Audio Sales, Bencsik Associates, Dobbs Stanford Corporation, YoreCo, RM Associates, and Radon and Associates. Updike stated that the appointment of these firms was necessary to reinforce the rapid sales growth experienced in the recording, broadcast, communications, sound contracting, and sound reinforcement marketplace. He further stated that as Valley People continues to introduce new product lines and increase its prominence in the marketplace, "We will be adding additional representatives and distribution outlets to better serve the industries in which we are involved."

. . . & Happenings

BEARSVILLE STUDIOS INSTALLS SOLID STATE LOGIC CONSOLE

Steve Bramberg, manager of Bearsville Studios, is proud to announce the installation of a new **Solid State Logic 48-channel 6056E** (with total recall), into their Studio B complex. The new desk will be in place by mid-November and is to be linked with two Studer A800 Mark III multitrack machines by means of an Adams Smith 2600 locking system. The new 48-track studio, (designed by George Augspurger), will be unique in that many of the design modifications and custom features of this console were suggested by producer/mixer **Bob Clearmountain.** In addition, Bearsville has also added a full range of outboard equipment, supplied by Martin Audio, N.Y.C. (i.e., the Yamaha REV-1), to compliment their new state-of-the-art mixing facility.

In addition to Studio B, Bearsville's very large Studio A (which is 90' x 40' with a 40' ceiling), has been extremely active as a result of their recent installation of the Neve 8088. This studio also contains two Studer A800 Mark IIIs, which are locked by a Studer TLS 4000 system.

SOUND IDEAS FINISHES UP

Sound Ideas, under new management and ownership, has put the finishing touches on its three rooms. Studio A has a Neve 8108 32-input console with Necam automation and a Studer A-80 MKIII 24-track tape machine. The room has four isolation booths and is ideal for tracks and overdubs of all kinds. Studio B has recently acquired an SSL 6000E 56 mainframe console with Total Recall. It has a Studer A-80 MK IV 24-track tape machine. The room has two isolation booths, and is excellent for both tracks and mixing. Studio C is an overdub and mix room with a Harrison 36/24 console and an Ampex MM-1200. All rooms have UREI 813 monitors. Four-track, 1/2-in. and 1/4-in. 2-track machines are also available. A full complement of outboard gear, as well as Dolby A and dbx noise reduction, are on hand. Also on the premises are several synthesizers, two Yamaha 7-ft. grand pianos, two drum kits, a Hammond B3 organ, and assorted amplifiers.

In addition, Sound Ideas has added eight more modules to their SSL6000 E in studio B completing its 56 mainframe. They've also acquired several new pieces of outboard gear, including the Roland SRV 2000 digital reverb.

ROLAND ADVANCES IN MUSIC EDUCATION

RolandCorp US demonstrates its firm commitment to music education through a generous donation of microcomputer-based music equipment to nearly 500 school districts throughout the United States. Totaling more than a half-million dollars worth of equipment, the CMU-800 CompuMusic Systems work in conjunction with Apple II computers and are ideal for use in existing music education programs.

RolandCorp US is the exclusive US distributor of Roland electronic musical instruments, music software, signal processors, sound reinforcement equipment, BOSS products, and Roland DG computer peripherals.

AGFA-GEVAERT BREAKS GROUND

Groundbreaking ceremonies for a new 43,000 square foot marketing/training and regional distribution center, being constructed for **Agfa-Gevaert, Inc.**, recently took place in Irving, Texas. The new center will replace the present facility in Las Colinas, which has been in operation for the past ten years.

The building is being developed by SMC Properties, headquartered in Beverly Hills, California, and the general contractor is Baggette Construction of Decatur, Alabama. The facility should be ready for occupancy in December, 1985.

MIDI IMPACT AT NEW YORK'S MEDIASOUND STUDIOS

Michael Hektoen, president of Mediasound, Inc., a New York-based recording studio, announced the opening of its new MIDI IMPACT room this week. The room provides the client with the following features: 64-track MIDI/SMPTE sequencing on IBM, Macintosh and Commodore computer music systems; Synclavier music system with SMPTE interface capabilities; displays and printouts of finished tracks as well as keyboard improvisations instantly; enormous custom sample sound and percussion library; 3-dimension computer graphic representation of sampled waveforms with total editing capabilities; ability to synchronize all percussion and synthesizer parts with analog multitracks, film or videotape; fast synthesizer sound editing and program recall through computer-assisted interfacing; 32-voice digitally-controlled analog synthesis system; array of algorithm and phase distortion digital synthesizers; interfacing capabilities for MIDI guitars.

Mediasound's in-house programmers, Frank Doyle and Mark Kovach have produced, recorded, and programmed with The Fixx, Evelyn "Champagne" King, Rock Steady Crew, Moto Sano, CBS, Island, Inner City and Virgin Records. They have worked on radio, television, and industrial film promotions, as well as electronic music productions for cable television.

DUPONT/N.V. PHILIPS

The **Du Pont Company** and **N.V. Philips** of the Netherlands announced they will form a joint venture which is expected to become the world's preeminent supplier of optical discs. Du Pont and Philips will combine all their existing compact disc and high-density information storage disc operations in a new venture. The venture will manufacture and supply discs for the worldwide audio, video, and data markets. "From its inception in early 1986, the new company will be the world's largest supplier of CDs for prerecorded music," Gerrit Jeelof, vice president of Philips, said in London. "It also will be the world's most diversified optical disc operation, supplying optical discs for every audio, video, and data market," he said. In New York, Du Pont Vice Chairman Edgar S. Woolard, Jr., said the undertaking will be the largest non-enery joint venture in Du Pont's history and underscores Du Pont's intent to enlarge its participation in electronics. Mr. Woolard said, "The new company will have unmatched technological, manufacturing, and marketing strengths. Philips will bring to the venture the world's largest compact disc manufacturing operations and its reputation as one of the world's largest electronics companies. Du Pont will contribute its high-density disc operations, its recognized expertise in electronic materials and manufacturing, and its US market presence in data processing." Initial assets of the enterprise will include the world's largest CD manufacturing facility in Hanover, West Germany; and existing "LaserVision" video disc and data operation at Blackburn, England; an existing high-density optical disc development facility near Wilmington, Del.; a development facility at Eindhoven, the Netherlands; and access to all optical disc research of both Du Pont and Philips.

To meet the rapidly growing world demand for compact discs in the audio entertainment market, the venture will initially nearly double capacity at the Hanover plant to approximately fifty million CDs annually. The partners also have taken an option on a 160,000-square-foot building and 95 acres of land in Kings Mountain, N.C., near Charlotte, which will be refurbished as a CD-audio manufacturing plant. This facility is expected to begin operations in 1986, with capacity about equal to the expanded Hanover facility.

By 1990, CDs are expected to capture 40 to 50 percent of the recorded audio entertainment market. Between 1986 and 1990, the venture plans to bring total CD-audio capacity to more than 200 million discs annually by adding facilities worldwide. In addition to CDs for recorded audio, the new company also will manufacture and sell compact discs and high-density optical information storage discs for the data storage and retrieval markets, respectively, and video discs for the entertainment, educational, and instructional markets. The compact discs include the so-called CD-ROM and CD-PROM, which can be programmed once. CD-ROMs, designed for use with personal and home computers, are expected to have their greatest demand in the business and instructional markets. The high-density information storage disc is expected to serve a variety of markets such as banking, petroleum, insurance, industrial, military, and education, where large volumes of information are stored. Beyond these write-once discs, the venture plans to sell erasable discs in the future, drawing upon the current research programs of Philips and Du Pont. Philips is doing development work based on both magneto-optic and phase-change technologies. Du Pont is conducting research on alternative erasable technologies based on organic material. From this broad activity, the venture will commercialize those technologies which best fit the needs of the marketplace. The venture plans to start construction in 1986 of production facilities in the US to meet expected demand for CD-ROM, CD-PROM, and high-density discs for the data markets.

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FOR SALE: WESTREX 3DIIH, $4,400; Haeco SC-2, $6,000; Haeco SC-1, $1,700; Grampian D, $325; Westrex 2B, $400. Haeco cutterheads new, other cutter-heads rebuilt and within specifications. New solid state amplifiers available. **International Cutterhead, 194 Kings Court, Teaneck, NJ 07666. (201) 833-4421.**

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Gentle on the Wind. The A820 handles your valuable tapes with kid gloves. The DC capstan motor starts, following a defined ramp, only after the pinch roller is engaged. A closed loop servo system monitors tape tension and reel inertia to provide optimum acceleration and braking. Both tape tension and tape winding speeds are user programmable.

The Wheel Thing. The A820's dual thumbwheel shuttle/edit control makes tape-cut editing a breeze. One wheel fast winds tape in either direction at increasing speeds while the other precisely positions tape for the edit.

Also Noteworthy. The A820 incorporates Studer's new generation of phase compensated audio electronics, available with either transformer or active balanced inputs and outputs. In sound quality, the A820 takes a quantum leap ahead of recorders made just a few years ago. Options for the A820 include a center-track SMPTE time code channel and test generator.

The Payback. The "hardware" in the Studer A820 is made to give you dependable service for years to come. That's the Studer tradition. Plus, with its advanced software, the A820 also does more different jobs, gets more jobs done in less time, and produces sonically superior results. That's the bottom line in any upgrading program.

For more information on the new A820 Analog Master Recorder, please contact: Studer Revox America, 1425 Elm Hill Pike, Nashville, TN 37210; (615) 254-5651.

STUDER REVOX

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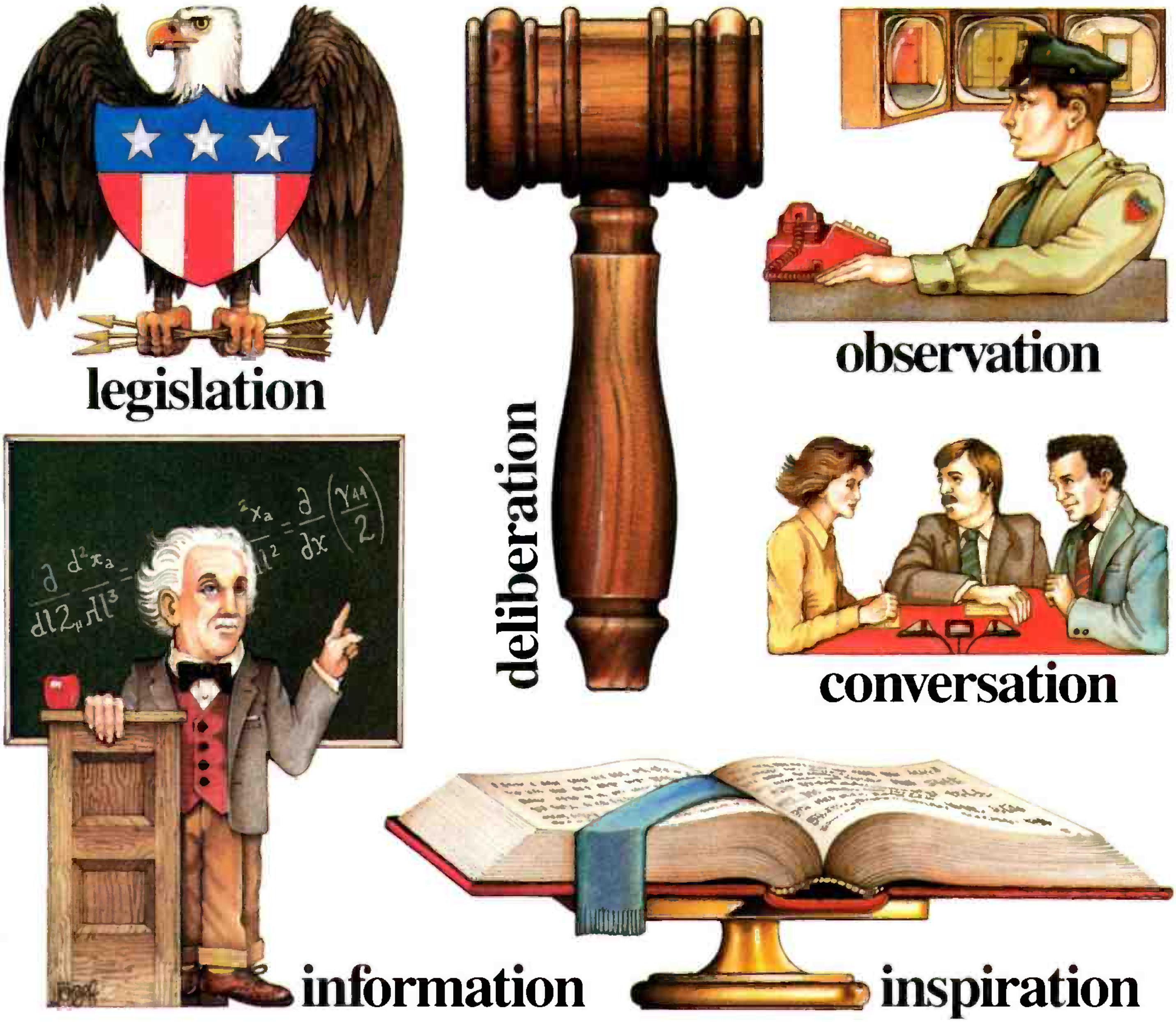

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