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THE SOUND ENGINEERING MAGAZINE

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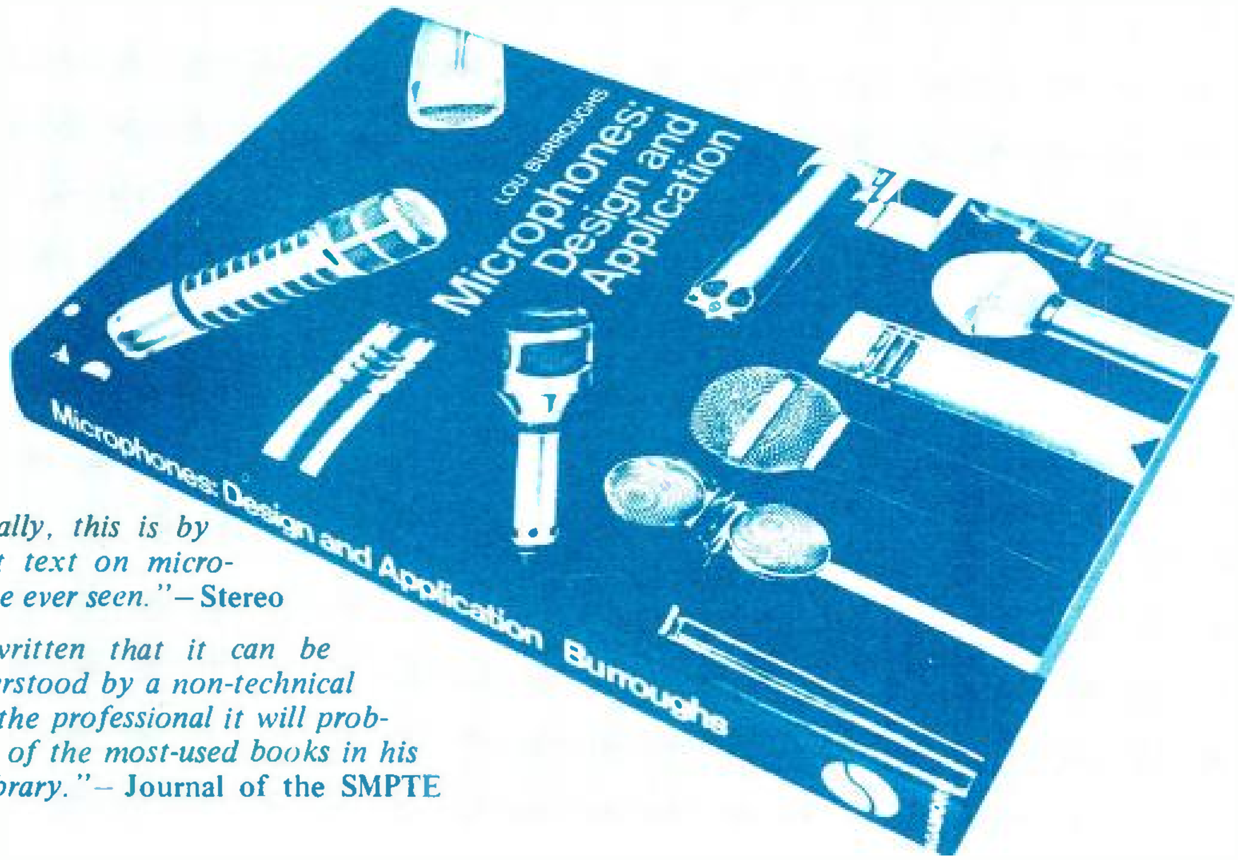

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And the rave reviews go on and on. "At last...a decent book on microphones," said David Lane Josephson in *Audio*. "Excellent chapters on various aspects of microphones, which are discussed in great detail," said Werner Freitag in *The Journal of the AES*.

They're applauding Microphones: Design and Application, by Lou Burroughs, who has written this practical, non-theoretical reference manual for everyone involved in the application of microphones for tv, motion pictures, recording and sound reinforcement.

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Coming Next Month

• In April, we'll take a look at what's new in test equipment, and testing procedures. Wayne Jones will get things underway with a survey of recent developments in audio tests and measurements. We'll also have a close look at those X-Y plotters briefly mentioned in this month's Time-Alignment feature. And a report on some new methods of amplifier evaluation, including 3-D testing.

• In case you thought that PCM is the only way to go digital, we'll have something to say about Delta Modulation. And speaking of digital, we might even have a quick look at a new digital tape recorder or two.

• And, what about some more audio problem solving, using programmable calculators and computers? And, a status report on—finally!—f.m. quad. All this, and more, in the April issue of **db—the Sound Engineering Magazine.**

About The Cover

• The SACMEX Centro de Grabacion—studio A. which is nearly one hundred feet in diameter. John Woram's article on this new Mexico City complex begins on page 37.

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THE SOUND ENGINEERING MAGAZINE

MARCH 1979 VOLUME 13, NUMBER 3

db is listed in **Current Contents: Engineering and Technology**

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db Calendar

MARCH

20-22 Hilton Inn. Salt Lake City. Utah. For registration forms or information on either seminar. contact: SYN-AUD-CON. P.O. Box 1134. Tustin. CA 92680. (714) 838-2288.

25-28 **National Association of Broadcasters (NAB) Convention,** Dallas Convention Center. Dallas. Texas. For more information contact: Dallas Convention & Visitor's Bureau. Dallas Chamber of Commerce. 1507 Pacific Avenue. Dallas. Tex. 75021 (214) 651-1020.

APRIL

2-5 **First Annual Architectural Acoustics Exposition and Seminar.** Hyatt Regency O'Hara. Chicago, Ill. Contact: Wayne V. Montone, Executive Director, 464 Armour Circle. N.E.. Atlanta. Georgia 30324.

10-12 **Synergetic Audio Concepts Sound Engineering Seminar:** Sheraton Harbor Island. San Diego. CA. For registration forms or information. contact: SYN-AUD-CON. P.O. Box 1134. Tustin. CA 92680. (714) 838-2288.

23-26 **Audio-Visual '79.** Wembley Conference Centre. London. Contact: British Information Services. 845 Third Avenue. New York. NY 10022. (212) 752-8400.

MAY

12 **1979 Midwest Acoustics Conference. Topic:** Digital Technology: Impact on Recorded Sound. Norris Center. Northwestern University. Contact: William R. Bevan. Shure Bros.. Inc., 222 Hartrey Ave.. Evanston. Illinois 60204. (312) 866-2364.

15-18 **63rd AES Convention (Los Angeles),** Los Angeles Hilton. California; Chairman will be Martin Polon, Director, Audio Visual, U.C.L.A., C.A.S.O., Rice Hall 130, 405 Hilgard, Los Angeles, Calif. 90024. (213) 825-8981.

22-24 **Synergetic Audio Concepts Sound Engineering Seminar:** Sheraton-Universal Hotel, No. Hollywood. CA. SYN-AUD-CON, P.O. Box 1134, Tustin. CA 92680. (714) 838-2288.

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db Letters

To The Editor:

If issue must be taken with Mr. Dunn's letter, I feel it is over his rather subjective statement about making music "sound better." The classic communications model, in simplified form, is communicator, transmission path, and receiver. Mr. Dunn would seemingly change this to communicator, interpreter, and receiver. Rather like the t.v. network censor, he apparently feels that he, rather than the artist or listener, should have the final say over the message content. If the artist deemed a 60 dB separation necessary between the drum and the triangle, I might well disagree personally, but professionally I realize that, as a broadcaster, my function is to let the artist stand the test of the listener to the very best of my technical ability.

Please don't read into this, agreement with your "Unknown Engineer" —I am forced to disagree with his view point just as strongly. To best serve the interest of both artist and listener requires an understanding of the limitations of both the medium and the receivers the listener is forced to use. Perhaps it is ultimately impossible to serve both perfectly, but the various devices the engineers supposedly foist on us are designed to make closer approximations of this goal possible.

Commenting on the Chris Edwards letter, I would like to note that although the ARB indicates greater numbers for radio in general, it does seem to be the case that a smaller percentage of the available population is listening—and for shorter periods of time. If it is true that the "punch out" factors in radio, such as poor commercials, stop sets with no real content, and perhaps the lack of control over the play list, are offset by the drawbacks of playing your own music, dead "air" between cuts, and of course the purchase and care of the material; then perhaps the decline represents listeners who are tired of the program directors' interpretation of what "sounds good." Objectively, doesn't it seem somewhat odd for a program director to decide that Stevie Wonder needs tempo enhancement, Roberta Flack needs more punch, Neil Diamond more compression, and that Fleetwood Mac's latest cut isn't bright enough? This, in the most general case, results from someone whose musical background consists of listening to 45s for 10 years at a peanut-whistle station.

Marshall P. Brown
Chief Engineer
KTKT, Tucson, AZ.

To The Editor:

I started last fall to answer your "anonymous chief" (**db**, September 1978) and now I can no longer hold back my two cents worth. As chief of a thousand watt class-IV, I am always looking for ways to better compete with my more powerful neighbors. Only recently have I seen the way—Limiters! Every limiter you see advertised offers you at least three dB over anything you are now doing, witness this quote from a 1940 brochure for my Western Electric 1126 type limiter:

"The 1126A . . . will give a 5 db (sic) increase in average signal level. . . ." *

Now, all I need in order that I may equal my 50 kilowatt friends is approximately 16.989700004 dB more signal power. So, if I place about five or six of these new limiters head to tail—I shall be there!

More seriously, my job as engineer is to give the management and program people the kind of signal that they believe will do the job they want done. I should, however, do my best to educate them on the physical limits of the art and the expected results and trade-offs of anything that I may do to our signal.

While I believe that the people who make the records know what

Preliminary Information, WESTERN ELECTRIC 1126A AMPLIFIER,* Western Electric Company sheet WECO-T-1810, 1-L-40-4C. This delightful device takes less than half a meter (19¼ inches) of rack space and uses but 12 tubes. But, the specifications and claims could have come off the pages of this month's **db. I rather like the old western stuff and am always willing to add either the equipment or the literature to my personal collection.

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sound they want and have the people and equipment that will get that sound on vinyl for them, others may not. Part of the problem, of course, is the philosophy of the programmer. Is he working for a high "cume" or for retention? If he wants a high cume, then the louder the better—even at the expense of fidelity. But, the distortion inherent in high levels of compression and clipping will tire people so that they cannot stay tuned in all day. But, if you believe the listeners to be a fickle lot, constantly twisting their dial, this is no problem—attracting their attention is. So, my job is to deliver to them what they want while explaining the consequences and options as best I can.

F.M. presents a special problem with its 75 microsecond preemphasis. When Paul Dunn (db, Letters, December, 1978) asked "where is it written in stone that a tap on a cymbal or triangle cannot be recorded at a high enough level so that it can be heard easily . . ." I was ready to tell him §73.317(a)2 of the Rules and Regulations of the Federal Communications Commission:

> "Preemphasis shall be employed in accordance with the impedance-frequency characteristic of a series inductance-resistance network having a time constant of 75 microseconds."

All of which is a six-bit way of saying that a high at ten kilohertz must be 10 to 14 dB down (if it is not when you start, the limiter will do it for you) and even a not so high of five kilohertz will be between 5 and 8 dB down. Fletcher-Munson does not help either. Listen and watch the modulation monitor while playing the "Theme from Shaft"—you won't hear much, but oh! will that needle bounce. Yet, I have seen "rock jocks" who would crank the high end all the way up when carting music and then complain that the station was not loud enough. There just isn't any way around that high end problem (except maybe Dolby).

And as for multi-band limiters that are now making a big noise—that is just what the ultimate multi-band limiter would give you. If we build a limiter that will do a good job of keeping every frequency at the full peak, nothing will come out of it except "white noise." High compression will make you sound "louder" when close-in, but when you get to the fringe area the noise and distortion (and when you clip—either "hard" or "soft"—you add distortion: sounds that were not in the original input) will add to the natural poor signal effects to make you sound "more distant" than you would otherwise—i.e., a net loss of coverage!

All of this is only a long-winded way of saying "there ain't no free lunch." As an engineer, I can only work to deliver the sound that my masters want (within the physical limits of the art and part 73, FCC R&R) while questioning and teaching.

LEE S. PARR
Chief Engineer
WWIN, Baltimore, MD

TO THE EDITOR:

After 5 years as a broadcast engineer, I felt I had to move to the recording studio. Paul Dunn's letter in the December, 1978 **db**, provides the reason for this.

Why, I'll bet Mr. Dunn aligns his tape machines as often as once every six months—whether they need it or not!

PHIL MENDELSON
Los Angeles, CA

TO THE EDITOR:

It seems that "A Chief Engineer" has opened a Pandora's box. At any rate, I feel I must also add my little say. First, let me say that I have been a broadcast engineer for more than twenty years. I have had a fling at production director, program director, news director, technical director, chief engineer (experience in a.m., f.m., and f.m. stereo), and t.v. programming and engineering. I also have a Bachelor degree in Radio and Television production. So much for "qualifications."

Dear Chris Edwards:

After reading your letter (**db**, December, 1978) I feel that, to use your own words, "I can do nothing but sadly shake my head. Your attitude is so typical." But is it really? The thing which provokes me is the attitudes presented by both "Mr. Engineer" and yourself. Having been on "both sides of the fence" I can see validity in both positions. I can certainly recall, as a "deejay", having to cut some of my favorite recordings short so I could get in all those blankety-blank commercials those blankety-blank salesmen sold. But, as you said, they paid all the bills, including my salary (which, by the way, wasn't terribly great at that time). What really is the crux of the situation is summed up in your final words. Our goals are the same—MONEY! The difference lies in the loads we wish to take to get there, and in the attitudes of the individuals. Management is dedicated to making the greatest net profit with the least effort. Programming is dedicated to producing the greatest creativity. Sales people are dedicated to the art of selling an intangible. Engineers and technicians are dedicated to the quality of the product from a technical standpoint. And therein lies the battleground.

You have ears, you say. Most assuredly you do. But they are not the ears of a technical person. The engineer, hearing ANY distortion, wants to eliminate it because it irritates him, and, to a technically oriented listener, it reflects on the ability of the engineer(s). What you say about the technician not having an interest in anything but volts, current, deviation, and the tech logs doesn't make much sense —after all, THAT'S HIS JOB, and SHOULD BE his primary interest. If he weren't interested in those technical things, what kind of a product would the sales staff have to sell? The serious broadcast engineer has always decried the unrealistic dynamic range the recordists put on their discs. But don't put all the blame on the recordists. Sometime the artist is to blame, because he expects the "idiot recording equipment" to do things beyond its technical limitations. Whatever the reason, however, the poor broadcast engineer has to try to compromise his stations equipment, which likewise is technically incapable of encompassing such a wide dynamic range. But, until a specific standard is agreed upon, and all recording company people interpret the standard the same way, and make their recordings in strict adherence to that standard, we all have to compromise. We not only compromise technically, but also, at least to some degree, our principles. WE KNOW that what we hear is not as good as our equipment could make it. WE KNOW that we must process audio to some degree in order to meet regulatory requirements. That doesn't necessarily mean that we like it.

Believe me, we technicians DO give credit for common sense when we see it regularly displayed. But too often, by all concerned, when we say "compromise" we are really saying, "You give me my way, or I'm gonna be mad at you!" Compromise means that we each give a little and end up somewhere between the two extremes. Engineers are not always right—but neither is anyone else. After all, there is no ONE RIGHT WAY to do the job in broadcasting. There are many RIGHT ways, and the astute broadcaster, whatever division he works in, is always open to suggestions. I can say, unequivocally, that some of my best technical ideas came from a seed of thought planted by a very non-technical station manager. On several occasions, when asked why a certain thing had been done that way, in an effort to arrive at the reason. I really

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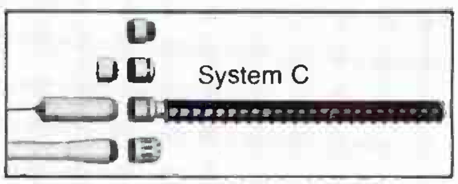

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didn't know. It was just because I'd "always done it that way." (And HIS way often proved better.)

Perhaps the single idea that has always caused me a great deal of displeasure is that all arguments are based from extremes. All right; we need to process audio, not only to meet regulations, but also to give us an overall clear sound. Probably nothing will lose listeners any faster than levels that are all over the place from the infield to the outfield. The next thing that will drive listeners away is an "un-clean" sound. Many listeners really do not understand what is meant by distortion—by definition —but they know that certain stations "just don't sound good." These are not necessarily the biggest problems. nor are they the only problems. But they are a part of the total. Let me conclude by saying that compromise carries the connotation that all parties are communicating. thoroughly, and that each division, understanding the others' problems, is willing to sacrifice a little to get the BEST QUALITY PRODUCT possible on the air. Then, and only then, can we say that a "real compromise" has been reached.

E. NEIL PIKE
Broadcast Engineer
Fairview Heights, IL

TO THE EDITOR:

Having received the December issue of **db,** I noticed, with both amusement and disbelief, the continuing argument between radio engineering and programming types. This round seems to have started by an anonymous working radio engineer who, as I recall reading the original letter, was annoyed by the way the broadcast industry, in general, proceeds to degrade the relatively good technical quality of commercially recorded discs into garbage.

This poor fellow, and there is no doubt as to the reason which he wishes his identity to remain a mystery, is answered by three letters in the above cited issue. The first respondent manages a station in California, and appears to be basically a proponent of "Anon", as I will call our poor fellow. As I read Mr. Erickson, of KRJB, he has reached basically the same conclusions that EZ Communications has with regard to over-processing of audio, although his format and locale are vastly different from those our five stations operate within.

The second respondent is of the breed that is titled "program director/ chief engineer," and has several valid points about record companies not adhering to NAB or RIAA level specifications. Mr. Dunn of WDBF has not been taught, it appears, either through theory or demonstration, that music has a rather wide dynamic range, and a great deal of gain-riding usually destroys this both amazing, and very psychoacoustically satisfying sensation.

The remaining respondent is a programmer, and I am pleased to see that he does realize that our poor "Anon", in markets small and large, usually is one of the least appreciated and loved persons on the payroll. But Mr. Edwards, our programmer, seems to have little concept of the relationship between technical quality, listeners, Arbitron numbers and programming.

A look at the ARB books for the country's top markets will show that each programming type usually takes a given share of the audience. This share varies little from market-to-market, when averaged over 6 to 8 book periods, unless a particular market has a disproportional demographic or ethnic weighting. Take a look at the major rated stations in each programming type. Note that I said *stations* rather than the #1 guy in each case. If you've got a 5.7 share in Boston, it doesn't matter if you're not #1, because you are going to get some business, both national and local agency, because of those numbers. In large markets, as well as small, you can only have one #1 station. Sure,

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you should strive to be better—better in programming (whatever programming), in technical, in promotion, and, most importantly, in sales—but there can be only one #1 guy.

A closer look and a visit to the market will usually show that the major rated stations in each format probably have good technical quality, as well as good programming, promotions, and yes, sales.

A look further at the numbers will show you that a heavily processed station will either tend to have very young or very old stats. These demographics will probably tend to favor males over females. These conclusions are based on some pretty valid psychoacoustics. Most intelligent radio buys (and there are many that are not intelligent, too!) are trying to reach the 25 to 49 demographic, and a great many products are reaching out to women, who often do the buying. This alone should shock a great number of station managers and owners into reality!

Further, have you ever looked at all of those superb receivers that your listeners are carrying out of hi-fi stores and discount houses alike by the gross? You really should, because the number of households with really good audio equipment is rising meteorically. It really doesn't matter if the equipment, when the all-thumbs listener finishes installing it, isn't d.c. thru light, it is 100 times better than what he had before, and he is becoming far more critical of what he is listening to. He is also becoming OLDER (surprise, Mr. P.D.)! The average age of the marketplace is getting older, because primarily I suppose, of the post W.W. II baby boom.

As you get older, you see, you don't want to be annoyed—and compression is annoying!

All of this is why all broadcasters should take a careful look at themselves technically, and become aware that the consumer, the listener is becoming more critical. Aside from programming, promotions, and sales, which are *not* my field, the technical side of the station requires improvement also. And improvement will eventually give you listeners and sales. The station owner or manager who has an under-paid (and probably under-intelligent) engineer, who is working on antiques, with a 75¢ per week parts allowance is behind the times by five years, even in small markets!

Obviously, many will disagree with my philosophy, numbers, and my engineering persuasion, but I think a careful analysis of the facts will show essential agreement with my premise.

THOMAS L. MANN
Vice President/Engineering
EZ Communications

TO THE EDITOR:

Once, there was an amazing device called radio. It was first noticed back in the twenties and thirties. The idea was that you could sit at home and listen to things happening dozens or even hundreds of miles away. The early efforts didn't sound like much, but it wasn't too long before a person could hear a man speaking to him from the other side of town . . . and it sounded just like that man was really speaking from the big box in the living room. There was music, too . . . an orchestra, a soloist, maybe even a one-man band . . . all crammed into that big box in the living room.

Then an even more amazing thing happened. The box got smaller! There was a box in the bedroom, one in the kitchen, and even one in the car. Of course, the voices in the little boxes didn't sound quite as real as the ones in the big box but then what-the-heck, at least you could take the entertainment with you.

As time went on, the boxes got even smaller and lighter, and everybody had two or three of these "a.m. radios" that they could put in their pockets. Everything was just great until people started getting interested in a new kind of radio called "f.m.".

The first f.m. radios weren't nearly as neat and compact as the a.m. radios were. The f.m. sets were big monsters that sat in the living room and sounded "funny". Some people liked them better. They noticed that the voices sounded really natural—like the person doing the radio show was really sitting inside the box. The songs on the f.m. radios sounded just like the original recordings so you didn't have to spend a lot of money on plastic records that got scratched and noisy.

In the sixties, a new twist was added. Something called "stereo" was developed so you could have an "f.m. stereo receiver" and *TWO* big boxes for the music and voices to come out of. Having two boxes instead of just one seemed to add a sense of spaciousness that one box didn't have.

Still, some people were unhappy because you couldn't carry the f.m. radios around like you could with the a.m. radios. So, some smart engineers figured out how to put an a.m. and an f.m. radio into a single box that was just as small as the little a.m. sets. They even made combination sets for cars and later figured out how to stuff

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a tape recorder in with the two radios. Of course, the more things you crammed into the box, the less realistic the sound became—but then nobody seemed to mind except for a few "hi-fi fanatics."

By the seventies, the number of "radio stations" that provided the sounds for the radios had become so numerous that they filled up all the open spots on the radio dials. The a.m. stations and the f.m. stations were also fighting with each other to get listeners for their particular programs. The a.m. stations had the advantage of having had a head start and could also "broadcast" farther, but the f.m. stations had stereo and all of the hi-fi fanatics.

The a.m.ers and f.m.ers also fought among themselves. Somebody discovered that if you ran your audio through something called a compressor, your station could sound louder than the other stations and maybe somebody might hear your signal a little better. Of course, doing this really made the music coming out of the box sound weird but since nobody seemed to mind, what-the-heck!

The a.m.ers all bought these compressors which they called "processors" and had battles to see who could sound the loudest and most different. A few a.m. stations got worried about the better sound of f.m. and went the opposite direction, trying to make their own sound more realistic. They were fighting a losing battle, though. The little a.m. radios couldn't sound anything like the big f.m. radios and nobody made big a.m. radios anymore.

The f.m. stations seemed to be winning the fight for a while, but then the f.m.ers did something really bizarre. They figured the a.m.ers were onto something and so they also bought processors and started having loudness wars, too. This made the hi-fi fanatics really mad. They quit listening to the radios and started buying more records.

Meanwhile, the a.m.ers were still jealous of f.m.'s stereo and demanded that the government let them have stereo, too. After a lot of arguing. they were allowed to have stereo but their version wasn't as good and really limited the a.m. transmission range.

It didn't matter, though. Everybody wanted to have a radio station of some sort, so the government decided to make more spots on the radio dials. They did this by sqeezing the stations closer together. They also dumped in lots of new stations on those few spots where only a few stations had operated before.

This worked out so well on a.m. that the government decided to do the same thing on f.m. Pretty soon, there were twice as many stations as before and everyone was happy. Well—almost everyone. The people who liked to listen to the same station as they drove long distances couldn't do that anymore, so they all bought tapes and tape players without radios. The last few hi-fi fanatics were also pretty upset until somebody invented a record called a "video disc" which didn't get scratched and made it possible to have FOUR speakers instead of just two. Most everybody else found a new toy called "home-video" which had better sound than the radios and you could also watch pictures or play games.

Oh, there are still plenty of radio stations around—except they're a lot different than before. There isn't much profit in radio now, so most of the operations have ony one or two people working for them. The equipment is all automatic and the sound isn't very good. Still, that doesn't matter much—not very many people are listening anymore.

John E. Shepler
Chief Engineer
WROK/WZOK, Rockford. IL

PATRICK S. FINNEGAN

db Broadcast Sound

Audio From the RPU Receiver

• The use of small transmitter/receiver systems for remote broadcasts is almost as old as broadcasting itself, and the use is expanding today. As with most electronic systems, the remote pickup systems have their share of operational and electronic problems. The transmitting units are usually operated by non-technical people, so the engineer often needs to supply these people with suggestions on positioning and placement of the transmitting antennas for best results. Aside from this, many factors at the receiving end can work to produce noisy and poor quality audio out of the system for a particular broadcast occasion. This month we will look at the receiving end of the RPU system and some of the problems that can develop.

ADEQUATE SIGNAL

No receiver can perform very well unless there is adequate signal at its antenna input terminals. Unless the r.f. signal strength is of sufficient amount to drive the limiters into full operation, there will be noise in the recovered audio output—noise in the form of amplitude modulation of the f.m. signal. However, if the signal strength is strong enough to operate the limiters, then the noise will be clipped off and eliminated. Should the signal be weak, then the noise will not be limited and will be demodulated right along with the audio and appear in the audio output.

Assuming the transmitter is functioning properly and radiating the correct amount of r.f. signal—positioning of its antenna, objects within the signal path, as well as distance, all contribute to degrading the strength of the radiated signal as it appears at the receiver. And since most of the units operate on VHF or UHF frequencies, propagation factors and the path are very critical. Theoretically, operation at these frequencies in line-of-sight. Therefore, large buildings or structures, directly between the transmitter and receiving antennas, can act as a shield and produce shadow areas (weak signal areas) on the opposite side of the structure. So can metal buildings right next to the transmitter antenna. Proper positioning of the transmitter can often work wonders in signal strength at the receiver, and can often make the difference between a non-usable signal or a good signal. When a weak signal condition occurs, advise the announcer or News person to move the transmitter unit a few feet (or perhaps across the parking lot, across the street or the other side of the building, etc.). Non-technical people will often take small handheld units inside a building and expect to transmit across town to the receiver. Sometimes this works, but other times it's a complete failure. Much depends upon the structure of the building, how much steel it has and so forth. In any such instance, and in other projected broadcast sites, check them out beforehand.

ANTENNAS AND LEAD-IN

A good antenna system at the receiver is important. Even a strong signal can be reduced to a very weak signal at the receiver terminals by a poor antenna system. A high antenna and coaxial down-lead is recommended. Since most transmitting antennas will only be a few feet from the ground, 50 or 100 ft. additional height at the receiving antenna will make a considerable improvement in the distance the transmitter can operate from the receiver. But there is a limit to height, and a point is soon reached where the loss in the down-lead is more than what was gained by the additional height. Very high antennas are also subject to "skip" signal interference from distant stations.

An improvement in antenna system can be accomplished with a high-gain directional antenna. The gain of the antenna increases the signal strength of the incoming signal, and the directivity reduces interference from other

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signals. To be effective, the directional antenna must be pointed in the direction of the transmitter. If the antenna location is central to the area to be worked, then a motor driven positioner is necessary to orient the antenna towards the transmitter position.

Antennas and their coaxial downleads, rotators and their control cables, are constantly exposed to the elements and lightning. These need inspection from time to time—especially any coaxial connectors. Moisture may enter the connector and cause corrosion, and lightning surges may burn the inner connector. In both cases there will be high resistance, and moisture itself can cause a shorted condition. All these will reduce the signal strength at the receiver.

MONITORING SIGNAL STRENGTH

Since correct r.f. signal strength into the receiver is so important to noise-free audio output, this should be monitored in the set-up for broadcast at a particular location. A listening test of the reproduced audio output is sometimes sufficient to indicate adequate signal, but a better test is to measure the current of the 1st limiter. Some receivers have a built in meter to do this, or at least a test jack for alignment. If yours has no meter, add a d.c. microammeter for this purpose. Be sure, however, to add enough series resistance to keep the meter on scale, and to provide isolation for the limiter circuit.

When setting up a broadcast from a particular location, observe the reading on the limiter meter while orienting the antenna. The increase in signal strength from the antenna positioning will be noted as a rise or drop in limiter current. Orient for maximum current. If the signal is strong, there won't be a peak since the limiter will "flatten out" the signal as it goes into full limiting. When positioning, the antenna should be left at a point between where the signal begins to flatten out and where it begins to drop again. Should a large remote project be coming up, such as broadcasts from many voting places all over the County on election day, then each of these positions should be checked out ahead of broadcast day. Test out the transmitter at each location, orient the antenna for best results, and make a listing of the antenna direction positions and the limiter readings from each site. Leave this for the operators on election day. The operator can then preset the antenna direction before each broadcast, as well as noting if the limiter still reads the same.

INTERNAL

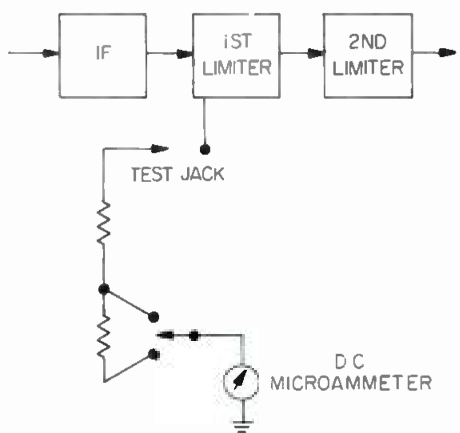
The receiver is basically a crystal-contolled, double-superhetrodyne unit that contains a few more circuits or components than a standard receiver. One is the bandwidth filter at the head-end of the i.f. stages or r.f. section. Its purpose is to eliminate adjacent channel signals from interfering with the desired signal. The r.f. and i.f. stages in the receiver must be tuned so that the resting frequency of the transmitter carrier is right down the center of this bandwidth filter. Should the transmitter be far off-frequency, or if the receiver crystals have drifted, or if someone had "twiddled" the tuning of the i.f. cans, the signal will not be directly in the center of the bandpass. This will limit the carrier deviation and sidebands on the "short" side and produce poor quality audio at the output. Under ordinary circumstances these transformers and stages are stable and require no tuning. The crystals will age and cause a long-term drift in tuning of the oscillator and should be returned. The transmitter frequency should be checked regularly with a frequency counter and reset to correct the frequency if necessary.

A good point at which to monitor whether the carrier is down the center, is at the test point of the "zero" tuning of the discriminator. With the transmitter signal being received, the meter should indicate zero. But if the indication is far off to the plus or minus side, something is off-frequency (assuming the other receiver tuning has not been disturbed). If the transmitter is known to be right on frequency, then touch-up the receiver oscillator frequency tuning to reset that reading to zero.

SQUELCH

The receiver contains a squelch cir-

Figure 1. A good indication of signal strength is the 1st limiter current. If the receiver doesn't have metering, add it.



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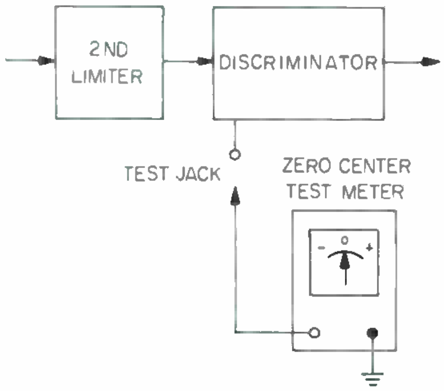

Figure 2. Measure, at the discriminator test point, to determine if the carrier is coming down the center of the bandpass.

cuit that is intended to mute the speaker output when no signal is being received. Without a squelch, the noise would roar out the speaker (or into the station's audio channels). Proper adjustment of the squelch is important to the reception of signals and can effect the audio output. This is essentially an operational problem (assuming no circuit fault). With no received signal, the squelch control should be adjusted to just mute the speaker. A problem can arise when high antennas are used and there are "skip" or other interfering signals. These will cause the squelch to "break open" intermittently, so the operator may readjust the control to shut these off. With the unit over-squelched, a weaker signal from your own transmitter may not be able to break it open and the call is missed. Although the squelch may be opened up on a stronger signal, it may still be at the point where the audio is chopped or sounds as if it is breaking up. Readjustment of the control will quickly clear up the problem.

INTERFACE

The audio from the RPU is intended for broadcast over the station's normal facilities, so interface to the station's audio system is important to quality. Some receivers are already equipped with a balanced 600 ohm output and VU meter, but the industrial types are intended for speaker operation only. This type requires more careful interfacing to the station's audio system.

A receiver that contains only a low impedance, unbalanced speaker output can best be interfaced to the system with a speaker-to-line transformer. This will not only effect the impedance match to the 600-ohm system, but will isolate the unbalanced from the balanced. A loss pad should also be added so that the internal control of the receiver can be operated high enough for speaker operation—thus keeping the audio to the system within correct proportions. Regardless of the type of receiver being used, the external audio wiring to the station's system should be shielded, and normal care taken with the shields. Carelessness here can result in noise, hum and crosstalk problems.

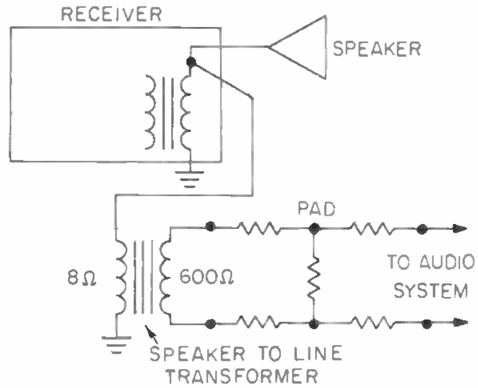

Figure 3. Use a transformer and pad to interface the receiver with the audio system, if the receiver only has a speaker output.

Some receivers contain a switchable audio filter. This can be used, as the case requires, to improve a noisy signal. A listening test on the particular occasion will indicate the best setting for the equalizer.

ALIGNMENT

Proper alignment of the receiver requires good quality, specialized test equipment and knowledge on the part of the engineer. Unless there have been major circuit failures, it is best to leave the alignment alone. One mistake is to think the engineer can "touch-up" all the r.f., i.f., and discriminator tunings by listening to the audio output. Most of these adjustments are very critical, and it is very easy to quickly get the entire unit so out of alignment that a major realignment is necessary.

RECAP

The use of remote pick-up systems is expanding. Many non-technical people are operating these systems, and need instruction on proper positioning and location of the transmitting antenna. Signal propagation at the high frequencies and antenna systems have a considerable effect on the signal reaching the receiver. Good quality requires the transmitter be right on-frequency, and the receiver tuned so that the carrier comes right down the center of the bandpass. Proper squelch operation, as well as the limiter and audio interface affect the noise and quality. Receiver alignment is very critical and requires special test equipment and knowledge. Tweaking up the alignment while listening to the audio output can put the whole system out of commission. ■

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And that's not all. Technics MKII Series adds quartz accuracy to whatever pitch variation you desire. In exact 0.1% increments. At the touch of a button. And instantaneously displayed by the front-panel LEDs.

And to take advantage of all that accuracy, Technics has a low-mass S-shaped universal tonearm that's so accurate, friction is down to 7 mg. (vertical and horizontal).

Technics MKII Series. Compare specifications. Compare quartz. And you'll realize there's really no comparison.

MOTOR: Brushless DC motor, quartz-controlled phase-locked servo circuit. SPEED: 33⅓ and 45 RPM. STARTING TORQUE: 1.5 kg · cm. BUILDUP TIME: 0.7 seconds (= 90° rotation) to 33⅓ RPM. SPEED DRIFT: Within ±0.002%. WOW & FLUTTER: 0.025% WRMS. RUMBLE: −78 dB. PITCH VARIATION: ±9.9%.

Technics MKII Series. A rare combination of audio technology. A new standard of audio excellence.

Technics
Professional Series

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NORMAN S. CROWHURST

The Role of Computerization

● In 1970 the Oregon State System of Higher Education's Teaching Research division retained me as an Assistant Professor, because of my expertise in engineering, an expertise they expected me to apply to education, in the areas of Research, Development, Diffusion and Evaluation (to quote words from the title of the project for which I was hired).

In the course of this project, tremendous use was made of the state's computer system, which was continuously "on line." We collected a great deal of "data," which the computer programmer fed into the computer and "instructed it," to get the same data out, in rearranged form. To me, what came out was just that—a rearrangement of the same data. It told me nothing that was not fairly obvious from the data we fed in.

But many of the others, including the program director, had become computer worshippers, mentally endowing that device with superman powers. They were positive that by its "skill" in rearranging facts, it could find some relationship, information, or something, that was not obvious from the "raw data." How did it do this supposedly miraculous feat?

The program director, or somebody else, programmed it to seek such a relationship. The data were based on questions that had been asked of many educational workers engaged on projects all over the country. If the questions were biased, then the answers were similarly biased, to agree or disagree with the bias of the question. Could the computer take that into account?

Not unless it was programmed to do so. But since the program director and others who programmed the computer, were unaware of their own bias, how could they program the computer to become aware of it? What they regarded as "facts" must be accepted as facts by the computer. The computer is no more than a digital device with a memory, that does precisely as it is told.

But its programmers persisted—and all over the world today, they still persist—in endowing the computer with magical powers. Presto, the computer has found some information we could never have deduced without its fantastic help. Really, all the computer can do, is put together such information much *faster* than is humanly possible.

HUMAN ERROR

Much is made of a computer's faultlessness—it cannot make a mistake. On modern electronic musical instruments, a computer-type memory can instantly "memorize" a musical phase and keep repeating it until that program is cancelled and another is given. If the musician who thus programs it, hits a wrong key in the first run-through of the sequence, the computer will keep repeating that wrong key, every time it repeats the whole sequence. It has no means of knowing what key the musician should have hit.

So it repeats its programming faultlessly—even to copying mistakes. How do you suppose a computer makes the "mistake" of crediting someone's bank account with $1 million instead of $10? Could be the operator leaned on the zero, for a little too long! But once it's in the computer, because of its "faultlessness" it can be very difficult to correct. The computer does not know it has to subtract $9,999.990! Where would it get that figure from, since it had no means of knowing the operator made a mistake in instructing it, in the first place?

Computers are a natural for engineers. Where a few decades ago, any engineer worth his salt had a sliderule sticking out of his pocket, now he will have a pocket calculator, with many computer functions on it. If he wants sin 30.35°, he selects the "degree" function, hits 3-0-.-3-5, verifies that his readout displays 30.35, then hits the "sin" button and, within a fraction of a second the appropriate sine is displayed.

The computer did not read it from tables, as the engineer before the day of the sliderule would have done. Nor did it use an analog device like a sliderule, to estimate the value of the sine. The computer could read such a device no more accuately than yesterday's engineer could. No, it calculated that particular value from first principles, just as the original tables were calculated—only very much faster.

The engineer who used tables might have been using tables with a misprint. It did happen. The engineer who used a slide rule could have mis-read his slide rule, making an incorrect interpolation. Now the computer makes it fool-proof, doesn't it? Suppose the angle for which he wanted the sine was 30.35, and the buttons he hit represented 33.5—how would the computer know about that? It would accurately give him sin 33.5°, no questions asked. And he would use that figure, believing it to be sin 30.05°.

The computer will not prevent him from making that mistake. So how can he guard against it? Let us take a simpler example, that more readers could follow. Suppose you want to multiply 3,396 by 4,507. You can hit the proper buttons, and get 15,305,772. But how would you know if you had hit a wrong button? Maybe you would not. You want a way to check the result.

The computer did not make a mistake, but you might have. So how can you check for that possibility? 3,396 is 3,400 −4. And 4,507 is 4,500 + 7. If you remember your algebra, 3,396 × 4,507 should be the same as 3,400 × 4,500 plus 7 × 3,400, minus 4 × 4,500, minus 4 × 7. So one way to check would be to put the 15,305,772 figure in the memory, and perform that somewhat longer calculation. 3,400 × 4,500 is 15,300,000. 7 × 3,400 is 23,800, added to 15,300,000 makes 15,323,800. 4 × 4,500 is 18,000, subtracted from the previous result is 15,305,800. Finally 4 × 7 is 28 which, subtracted from the previous result is 15,305,772. You pull the previous figure out of the memory and find it agrees: you did not make a mistake.

MASTER/SLAVE RELATIONSHIP

But what does being able to do that require? You must understand what you are doing, and what you are having the computer do. You are the master and the computer is your slave. The attitude that the computer can make no mistakes, leading its operator to accept whatever answer it displays as "gospel," is a dangerous one.

Supposing, by mistake, the operator feeds in 3,936 × 4,507, instead of the previous figure, the computer will read out 17,739,552. That must be the answer: the computer said so. Correct, it is the product of 3,936 × 4,507. It is *not* the product of 3,396 × 4,507. The computer did not make a mistake: you did!

How does all this apply to our more general subject? Human beings can think. Previous human thought developed the algebraic formula $(a + b)(c - d) = ac + bc - ad - bd$, which is what you used for checking.

If you learned the principles of calculation, you would know this, not so much from memory, as from first principles. You might remember it, to save having to memorize first principles, but you would know it, one way or the other. You would be satisfied it is true.

Departure from that started a long while ago. Back around the turn of the century, multiplication tables (now called multiplication facts, in an attempt to avoid the obnoxious tables) were taught by rote. Only persistent individuals, who questioned why, realized there is a reason that 7 times 9 makes 63. Others accepted it as unaccountable fact. And if their memory went faulty, that was too bad. They had no basis for checking back.

Advanced educators realized that education must teach people how to learn, how to think, how to communicate effectively, how to study and develop knowledge, and so forth. They suggested reorganization of curriculum to bring this about. Really, there was nothing wrong with the previous curriculum, but with the way it was taught.

So we had the "new math," and a lot more new curriculum material, aimed at correcting all this. But what has it really done? It has merely substituted a far more meaningless set of "data" for assimilation into kids' heads. The rote manner of its learning has not changed. If anything, it has become worse, because there is no rhyme or reason, by which parents can come to the kids' rescue, when teachers do not know what they are doing.

PROGRAMMED RESPONDERS

Kids have become, with each generation, more a form of human programmed responders. They acquire the habit—if they get good grades that is—of being able to give the answers this particular teacher wants. In another area, social studies, one teacher may believe in individualism; so the students respond in a way to please that teacher. Next year, for social studies, they have a teacher who believes in collectivism; so they learn how to respond to that kind of teacher.

In the real world, there are places for individual thought, and places where collective reinforcement is necessary. Both have their proper place. We all live in the same world—we cannot ignore that fact. But this phoney programmed response bit has generated a whole population of people incapable of individual thought.

Some have asked if it will ever be possible for a computer to "think." If you ask a person that, analyzing his answer will be interesting. If the person you ask has lost the power to think for himself, he will tell you that it is only a matter of time before computers are as "smart" as humans. But if the person you ask can still think, as an individual, he will know that a machine can never have true intelligence. It can only do what its programmers have given it power to do.

On this point, many a time, in a political or religious conversation, I have sought to find out what a particular person believes. The fact that he is a Republican or Democrat, or that he belongs to a specific religious denomination, does not mean he believes that group's tenets, down to the last comma. Or it should not. But what do I find?

Very often, if he is a programmed responder, he will seek to find out the answers I want, just as he did in school, and has been doing ever since. Perhaps, from the way I ask a question he will think I am a conservative, politically. So he answers the line he has learned pleases conservatives. But I am not satisfied. I wonder how well he has thought that comment through, and ask a further question.

Now, because it appears that I am questioning the typical "conservative response," he thinks he must have made a mistake—I must be a liberal. So he tries that response. He does not realize that his successive responses contradict one another—he is just trying to say what he thinks I want to hear. And what I really want to hear, is what he himself thinks.

But he does not think. He just responds. So he finds this "conversation" difficult—he cannot "please" me. Eventually, he will erupt with something like, "Well, what do you expect me to say?" You would be surprised how many people like that there are around today.

And what are they? They are little better than well-programmed computers. They listen to you, using a recognition training—what kind of answer do you want? Then they respond, as conditioned, for that kind of question. They have no opinions or thoughts, of their own.

If you try to discuss the relative merits of conservative *vs.* liberal philosophy they will compare the two, as entities. This has these good points, those bad points, and so on. They see only that choice—select the better of the two. They cannot think from first principles, to develop what is right, and then see how each philosophy deviates from truth. Oh no, that requires the ability to think for oneself. And neither a computer nor a programmed human being has that power. ■

New Products & Services

MICROPHONE STANDS

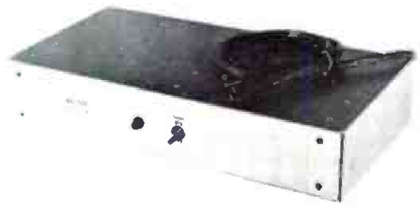
• The Porta-Series Models PSC and PSC-3 microphone stands feature grooved cam construction, machined into the tube assembly, for positive-locking and non-rotational brake action. Tempered springs within the tripod assembly assure increased stability. The vertical tube assemblies, on the stands, are chromed-plated with all-metal "grip action" clutches and equipped with standard ⅝-inch-27 male thread terminations. The Model PSC-3, designed for use by seated performers, extends from 26 inches to a 66-inch height—telescoping to 22 inches for transportation. The Model PSC extends from 35 inches to 63 inches, and folds down to 32 inches.
Mfr: Atlas Sound
Circle 56 on Reader Service Card

TAPE MACHINE

• A new mastering deck, with optional dbx, the 35-2 includes full logic (with motion sensing), up-front bias and EQ controls, four high-density permaflux heads for 2-track record/playback and 4-track play, pitch control, cueing and editing, and punch-in recording facility. The 35-2 has an overall frequency response of 40 Hz to 22 kHz @ 15 in./sec., a S/N ratio of 100 dB (with dbx), wow and flutter of 0.03 per cent @ 15 in/sec, and an overall harmonic distortion of 0.6 per cent at normal operating level.
Mfr: TEAC Corporation of America
Price: $1,900
Circle 58 on Reader Service Card

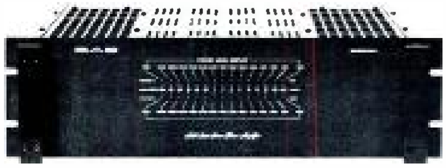
REVERB SYSTEM

• A mono reverb system for line level operation, the Model 1155 utilizes a dual 14-inch reverb spring assembly, with a bandwidth of 50 Hz to 6 kHz. The input signal is fed to an amplitude limiter and constant current generator. A model 522 power supply provides the bipolar 24V to operate the system.
Mfr: Opamp Labs Inc.
Price: $275.00
Circle 57 on Reader Service Card



Circle 43 on Reader Service Card

POWER AMPLIFIER



● Complementary circuits ensure maximum power output in the Model 2300—delivering 150 watts per channel minimum rms, into 8 ohms, from 20 Hz to 20 kHz, with no more than 0.05 per cent thd. The model 2300 utilizes full thermal protection (in case of high temperature conditions), signal relays to protect speakers from low frequency information which might damage them and electronic protection for transient overloads and short circuits.
Mfr: Scientific Audio Electronics, Inc.
Price: $700.00
Circle 50 on Reader Service Card

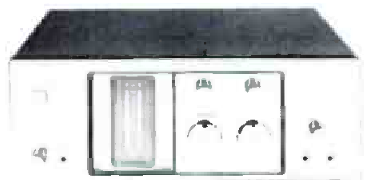
REVERBERATION MEASUREMENT

● The Model 232A Reverberation Timer computes and digitally displays room decay time within each of 19 frequency bands, from 63 Hz to 12.5 kHz. The unit features automatic level detection, two noise averaging filters, zero-crossing circuitry for external inputs, internal pink noise generator, recorder output, AKG phantom-powering, and calibrated send and receive controls.
Mfr: Acoustilog, Inc.
Price: $795.00
Circle 51 on Reader Service Card

DELAY LINE/FLANGER

● Providing a wide range of unusual effects, as well as regular delays, the Series 440 Delay Line/Flanger combines VCO time based processing with straight delays from 0.5 msec up to 160 msec. A dual output allows stereo effects and generates stereo synthesis from a mono input. The regeneration and internal mixing controls permit variations in depth and dynamics. The balanced input will accept signals as high as +18 dBm, and the outputs are equipped with discrete line drivers capable of +18 dBm into 600 ohms.
Mfr: Loft Modular Devices, Inc.
Price: $800.00
Circle 52 on Reader Service Card

DIGITAL AUDIO PROCESSOR



● Using pulse code modulation (PCM) technology, the PCM-1 encodes a live sound source and converts it into a dense group of variable pulses. Signals from the PCM-1 can be recorded on a videocassette recorder (Betamax, U-Matic or other NTSC-standard models). The unit records 13-bits of information on each channel (equivalent to 1.4 million bits of data per second) with a response range from 2 Hz to 20 kHz, and a dynamic range of 85 dB. Wow and flutter, tape hiss and speed deviation are virtually eliminated from the recording process.

Mfr: Sony Industries
Price: $4400.00
Circle 53 on Reader Service Card

REMOTE CONTROL

● Offering volume control, fade control, and volume mute, the 3BX-R is a remote control unit for the 3BX dynamic range expander. The remote unit, in addition to duplicating the 3BX main controls, offers a release time control and a high frequency transition level control, with upward and downward expansion displayed on six led's—two for each frequency band. The 3BX-R is powered by the 3BX, and comes complete with a 25 foot cable terminating into a 12-pin socket on the 3BX's rear panel.

Mfr: dbx, Inc.
Price: $149.00
Circle 55 on Reader Service Card

CORDLESS MICROPHONE

● Incorporating a special dynamic expansion circuit, the System 22E cordless RF microphone offers a dynamic range of 95 dB. This broadcast quality cordless microphone system consists of a crystal-controlled VHF-FM pocket transmitter and matching receiver. Due to the logarithmic compressor circuitry in the transmitter section of the system, over modulation is prevented.

Mfr: HM Electronics, Inc.
Price: $1,945.00
Circle 54 on Reader Service Card

Editorial

Brussels, Belgium—The Audio Engineering Society is holding its annual European convention here this month. And what better excuse do we need to dedicate this issue of **db** to the subject of international audio? As you read this, we'll be in Brussels (strictly business, of course), finding out about all that's new in professional audio. As for the rest of the world, we've managed to squeeze at least a few corners of it between this month's covers.

We start off with a journey of a trifle more than 22,000 miles—not around the world, but away from it. Specifically, into geo-synchronous orbit, where we discover **An Improved Audio Pipeline.** It's Western Union's WESTAR satellite, which gives our National Public Radio network a better means of distributing its programs around the country, free from terrestrial limitations. It's not quite international audio, but that shouldn't be too far away now, with just another satellite or two. What's a geo-whatchamacallit? Wayne Hetrich explains it all in this feature story.

A little more down to earth in our **Report From Mexico City,** where we find one of the most advanced recording facilities in the western hemisphere. After reading this report, you shouldn't be at all surprised if this *Centro de Grabacion* becomes well-known in international audio circles.

Although New York is home, it's also part of the international scene, and our coverage wouldn't really be complete without a **Report From New York,** which tells us a little bit about what's going on at Harry Hirsch's Soundmixers.

In Europe, the *Centre Georges Pompidou* has attracted international attention, and so we asked John Borwick if he wouldn't mind dropping in at the Institute for Research and Coordination—Acoustics/Music (IRCAM), which is a part of the *centre*. Borwick didn't deed much persuading, as he cheerfully admits in his **Report From Paris.**

And in Germany, Jeff Nieckau flew into West Berlin, from where he filed his **Convention Report: The 11th Tonmeistertagung.** It's an interesting contrast to the conventions of the Audio Engineering Society. which are perhaps more familiar to most **db** readers.

We conclude our international survey with a **Report from Japan,** in which Sierra Audio president Kent Duncan brings us up-to-date on his recent studio construction experiences in Tokyo. As Duncan has discovered, it's a long way from beautiful downtown Burbank, but it's a trip he heartily recommends without hesitation to any serious audio pro.

We return from our 'round-the-world survey just in time for another look at time, or rather, at **Time-Aligned Loudspeaker Systems.** Speaking of the Audio Engineering Society—as we were at the beginning of this page—an AES paper by Ed Long on time-alignment techniques marked the beginning of the development of the UREI 813 Time-Aligned Studio Monitor System. Here, author Dean Austin describes the development of this system, with special emphasis on the significance of time coherence.

Although we'll be back next month with more on the subject of digital audio, it's not too early to remind you now of the next Midwest Acoustics Conference. The subject is **Digital Technology: Impact on Recorded Sound**. The conference will be held at Northwestern University (Evanston, Illinois) on Saturday, 12 May. For more information, contact William R. Bevan, Shure Brothers, Inc., 222 Hartrey Avenue, Evanston, Illinois 60204. (312) 866-2364.

J.M.W.

Have VOCAL STRESSER WILL TRAVEL!

......says Tony Visconti*

"As a successful record producer, I am continually travelling to studios all over the world, recording such people as **David Bowie, Thin Lizzy and Mary Hopkin.** *I have to deal with a wide variety of equipment in various studio settings; so in order to ensur that I have the best Compressor-Limiter equipment to hand, I invariably pack a Vocal-Stresser in my suitcase*

In my opinion, **Audio & Design** *make the finest range of auxilliary processors available and their equipment offers the producer/engineer ultra flexibility in the creation of good music*

* Managing Director of **Good Earth Productions** and freelance producer of many hit records by illustrious pop stars

The **Audio & Design Vocal Stresser** uniquely combines a multi-ratio compressor with overall peak limiter and low level noise expander/gate with a parametric type equaliser in one package.

The combination provides most facilities necessary for improving and processing programme material whilst retaining maximum operational flexibility

The equaliser is simple to operate and can be switched *before* **(pre)**, or *after* **(post)**, the compressor-limiter, as well as into the *side-chain* **(s.c)** of the compressor section for frequency modulation effects and changing spectral energy balance.

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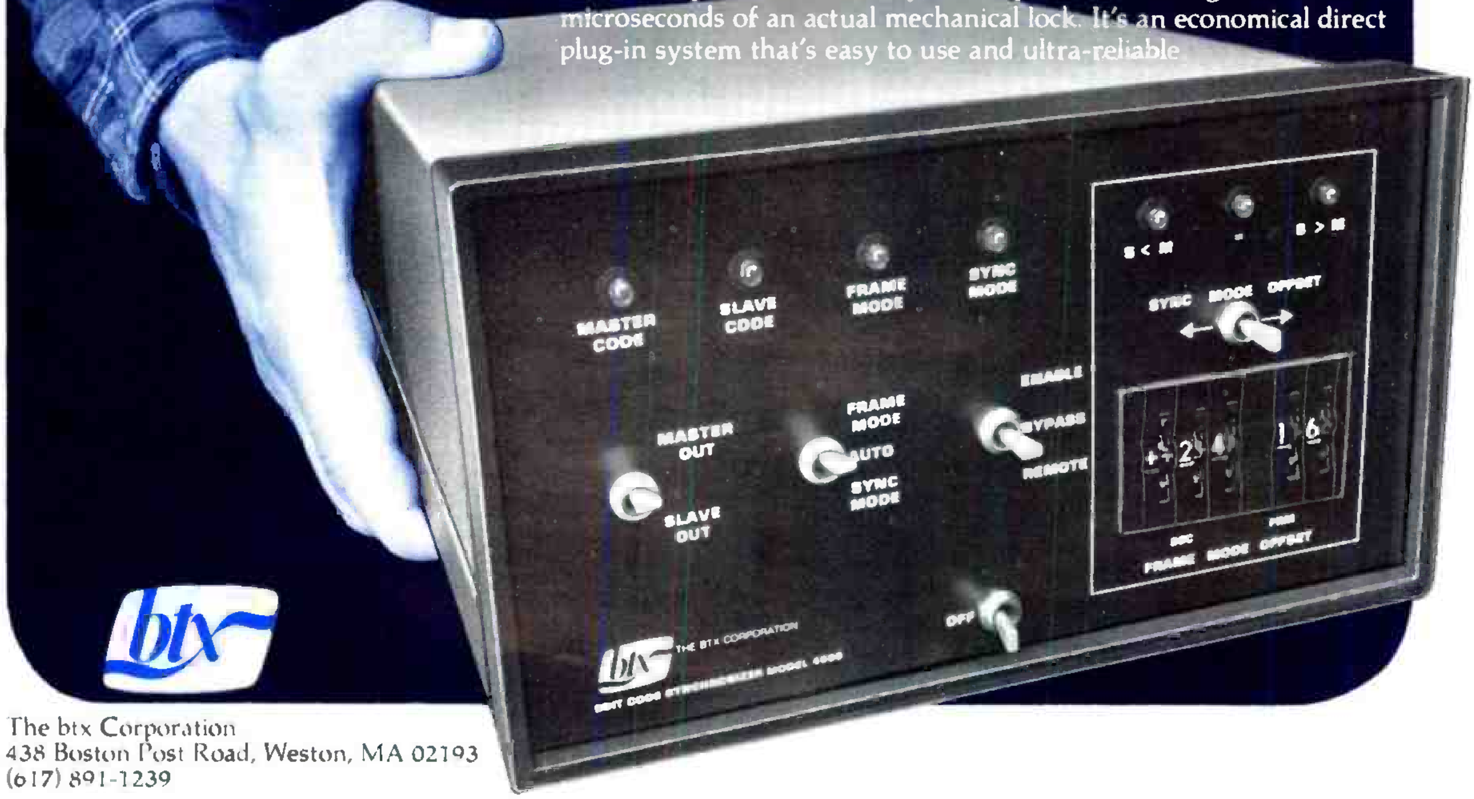
btx presents the 30-track audio recorder

The btx 4500 SMPTE interlocking system lets you operate any two multi-track recorders in tandem for 14, 22, 30 or 46-track capability. Using standard SMPTE time code written on one track of each machine, any two recorders may be synchronized, including video to audio. You can even mix makes, formats, speeds, and numbers of tracks, with or without servo controlled capstan drive. The btx 4500 is a micro-processor-based system capable of tracking within 50 microseconds of an actual mechanical lock. It's an economical direct plug-in system that's easy to use and ultra-reliable.



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WAYNE L. HETRICH

An Improved Audio Pipeline—NPR's Satellite System

Prompted by the insufficient bandwidth of conventional terrestial interconnection systems, NCR is shifting to satellite technology for high-quality program distribution.

NATIONAL PUBLIC RADIO (NPR) is a private, non-profit membership corporation established in 1970 specifically to provide a national program service for the nation's public radio stations.

Since its beginning, NPR has considered plans for shifting its interconnection service for program distribution from a terrestrial system to one based on communications satellite technology.

NPR's present service is severely limited by the lack of terrestrial circuits of sufficient bandwidth to permit high quality monophonic and stereophonic programs to be distributed nationwide. Most public radio stations operate in the f.m. band and have high-fidelity capability to provide stereo programming. To match this, NPR must presently distribute such materials on prerecorded tape, which is sent through the U.S. mail at a library rate to each station. A three to four week period must be allowed for this delivery.

BASIC OBJECTIVES

To meet the basic objectives of NPR, a satellite interconnection system must provide for the simultaneous national transmission of two or more programs of up to four channels each. The material will be distributed in real time with 15 kHz fidelity in mono, stereo, and quadraphonic transmissions, with provision for maximum local flexibility in selection and scheduling. The system will also provide for transmission of locally-produced programs, both for national, regional, or other less-than-national uses.

Initially, four 15 kHz channels will be transmitted through the system simultaneously, and each station will be able to receive all four simultaneously. Eventually, up to 12 simultaneous channels may be used. The choice of a single-channel-per-carrier system gives public radio the most flexibility in how it combines channels to make up stereo or quadraphonic program formats. Multiple circuits and their configuration in stereophonic and quadraphonic modes will make it possible for NPR to make a significant breakthrough in the breadth of its program service to the American public.

RECEIVER SUB-SYSTEM

Each licensee will be provided with one receiver sub-system, consisting of a down-converter, four demodulators and associated audio expander units (FIGURE 1). The down-converter may be tuned to one of two wide-band-width transponder frequencies, and each demodulator unit may be independently tuned to one of 12 narrow-bandwidth frequencies within the selected transponder. The system concept allows for eventual expansion from the basic four program channels to twelve 15 kHz program channels. The system concept also includes two low-power, narrow-band "Coordination Channels," to be used for teletype, message services, and remote control functions.

Earth terminal receiving antennas (FIGURE 2) are 4.57 meters in diameter and have the capability for manual re-pointing to another satellite in the event of a major or catastrophic failure of the prime satellite.

Artist's eyeview of satellite in orbit.

Wayne L. Hetrich is senior engineer, research & development for National Public Radio, Washington, D.C.

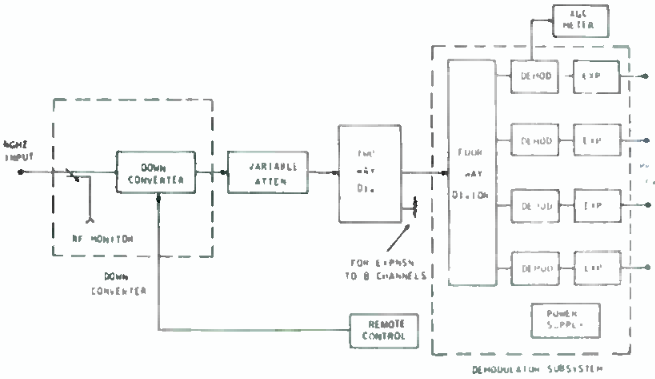


Figure 1. The receiver sub-system.

The design also accommodates 15 uplink transmitter sites. Each of these sites will have the capability of transmitting two independent 15 kHz program channels simultaneously to the satellite. The basic system is designed to allow for expansion of the uplink system as the public radio satellite system grows.

Initially, about 190 radio stations will be served by the satellite interconnection system, which is expected to grow to over 300 stations in the next ten years by adding from ten to twelve stations each year. NPR will effect a coordinated transition from the terrestrial interconnection to a satellite interconnection system. The work is to be done in segments.

THE MAIN ORIGINATION TERMINAL AND NPR TECHNICAL CENTER

The Main Origination Ground Terminal System (hereafter referred to as the M. O. Terminal) has been constructed for the distribution of Public Broadcasting Service television programming, and will be expanded for National Public Radio programming. The M. O. Terminal site is at Bren Mar, Virginia—approximately 10 miles southwest of Washington, D.C.

An M. O. Terminal Terrestrial Interconnection Link is to be provided from the NPR Technical Center on M Street, N.W., in Washington, D.C. to the M. O. Terminal site. There will also be return loops from the M. O. Terminal back to the NPR Technical Center, for monitoring and other program uses. Initially, two spare link channels will be provided between NPR and the M. O. Terminal, for a total of six duplex program circuits.

The M. O. Terminal will have the capacity to transmit at least four television programs simultaneously and also be capable of handling public radio's need in another transponder on the same satellite.

In the future, the M. O. Terminal and M. O. Terminal Terrestrial Link will be expected to handle up to 12 primary single-channel-per-carrier program signals along with spare facilities for the public radio system.

This M. O. Terminal will be fully equipped with back-up (redundant) systems to achieve high reliability, and consists of two 11-meter antennas, both of which will be able to receive as well as transmit. The Public Television System provides four low-noise amplifiers (LNAs), five transmitters, five receivers, and a terrestrial microwave link to the PBS Technical Center.

The overall design of the M. O. Terminal takes expansion capability into account, so that public radio satellite interconnection system equipment needs can be accommodated in a cost-effective and timely manner.

AUDIO AND LINK PERFORMANCE DESIGN GOALS

The audio performance goal for the NPR network is to deliver a high quality signal with at least a 67 dB signal-to-noise ratio. The system will make extensive use of a new compander which produces about 29 dB subjective processing improvement, thus greatly reducing the performance requirements for the transmission channel itself. It is expected that the subjective signal-to-noise ratio obtained at NPR stations will be in excess of 70 dB.

GEO-SYNCHRONOUS ORBITS

When an object is placed in what is termed a geo-synchronous orbit around the earth, the centrifugal force acting on the rotating object is exactly balanced by the pull of earth's gravity. Thus the orbit is maintained, and the object neither flies out into space, nor falls to earth. From these facts, it can be reasoned that there must be a particular orbital distance at which gravity and centrifugal forces are in balance when the object makes just one revolution in a twenty-four hour period. This distance is about 22,300 miles above the earth's surface, and is known as a geo-synchronous or, more commonly, a synchronous orbit. By placing the satellite in a synchronous orbit in the plane containing the equator (FIGURE 3), the satellite remains fixed in space with respect to the earth, since both are rotating at the same rate. Thus, the satellite appears to hang over a fixed geographic point, and can serve as a platform for a microwave repeater.

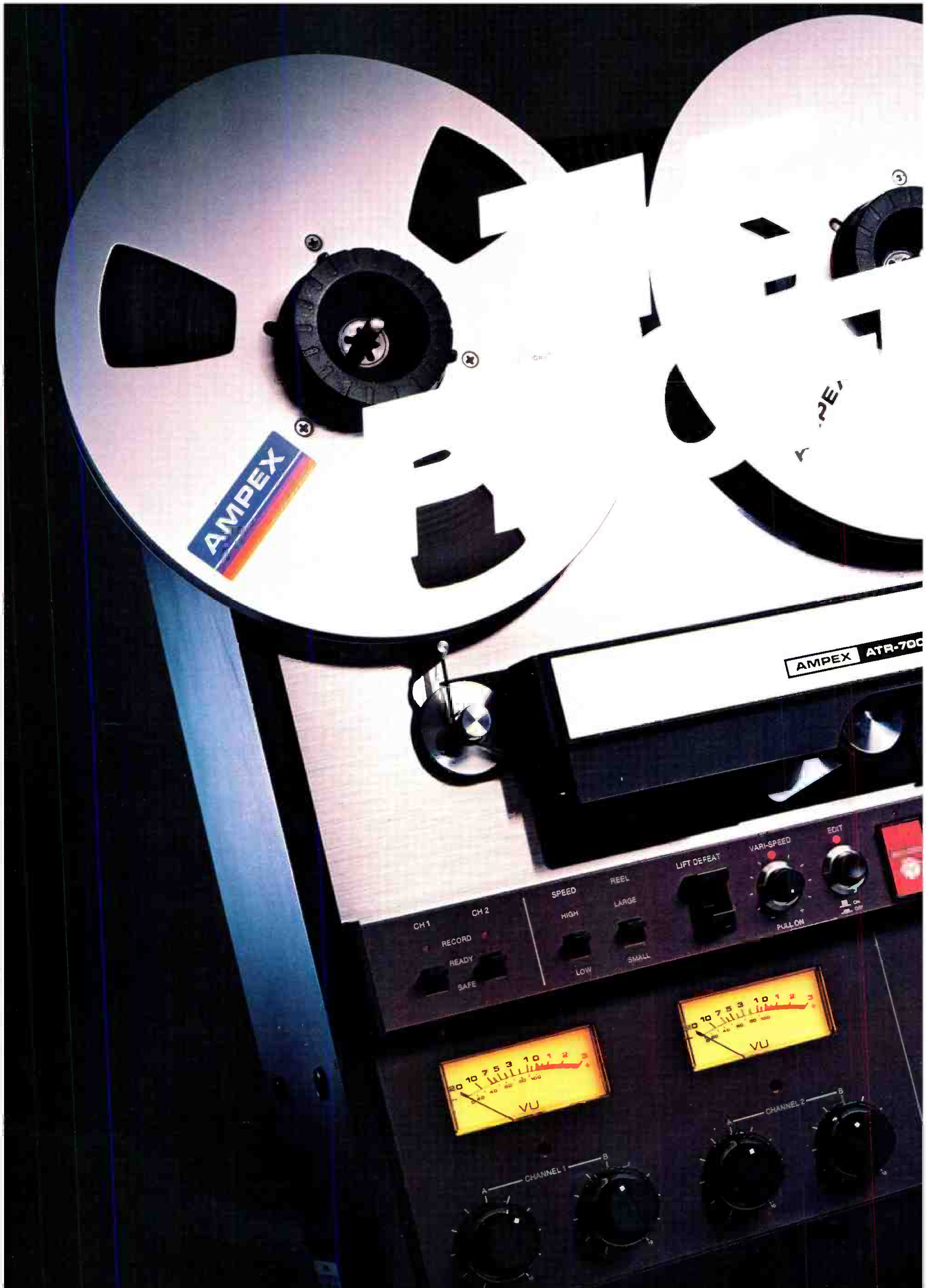
The satellite system itself is really just a long distance microwave transmission link which serves to connect the program origination source to all NPR receiving terminals. We accomplish this through a repeater (called a transponder, and carried on a synchronous satellite).

SATELLITE REPEATER COMPONENTS

The basic parts of a satellite repeater are a transmitting antenna, receiver, frequency translator, power amplifiers, and receiving antenna. These are shown in FIGURE 4. Signals are transmitted "uplink" from the ground-originating terminal on a carrier frequency in the 5925 to 6425 MHz range (or, about 6 GHz). These signals are received by the satellite and translated downward in frequency by 2225 MHz, amplified, and transmitted "downlink" in the 3700 to 4200 MHz (about 4 GHz) frequency range by the satellite. Ordinary frequency modulation is used to impress the program material on the microwave carriers.

In order to maximize power and bandwidth capacity, the NPR satellites (WESTAR I and II, owned by Western Union Telegraph Co.) are divided into twelve "transponders," each having a nominal bandwidth of 40 MHz Each transponder is actually a separate transmitter (trav-

Figure 2. The earth terminal receiving antenna.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CH 2
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SPEED
HIGH
LOW
REEL
LARGE
SMALL
LIFT DEFEAT
VARI-SPEED
PULL ON
EDIT
VU
VU
CHANNEL 1
CHANNEL 2


SIONAL
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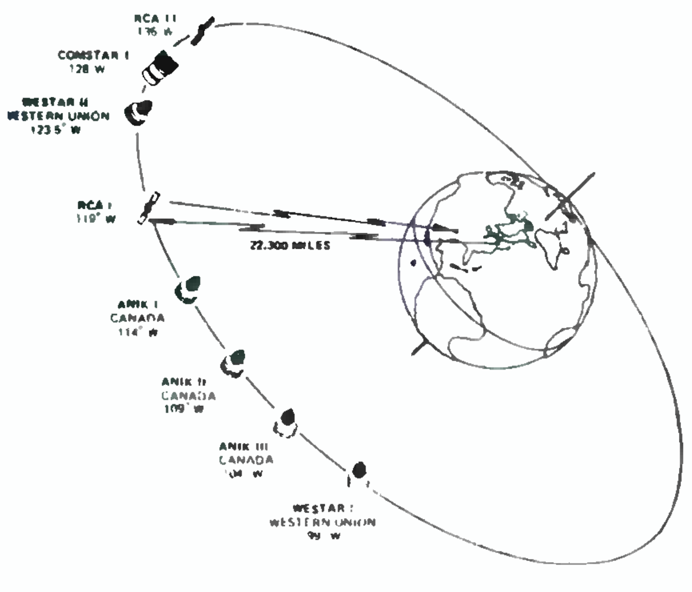


Figure 3. Satellites in geo-synchronous orbit.

eling-wave tube amplifier). Outputs from all twelve transponders are summed by combining filters to feed the satellite's transmitting antenna. The output from the satellite receiver is divided into nominal 40 MHz bands by other filters, each feeding the appropriate transponder. The transponders have been designed so that the downlink center frequencies and transponder numbers are on 40 MHz "steps."

Each of these channels is independent of the others and may be operated single-carrier, as for tv service, or multiple-carrier as for NPR service.

GROUND TERMINAL COMPONENTS

The basic parts of the ground terminal which transmits the program channel to the satellite at a typical NPR uplink site are: an f.m. compressor/modulator, frequency up-converter, power amplifier, and antenna, as seen in FIGURE 5. The additional components in the figure will be described later.

Program audio material, after audio compression, is applied to the f.m. modulator, which generates a carrier in the vicinity of 70 MHz, and frequency modulates this carrier with the compressed audio program material.

The peak deviation of the f.m. carrier, corresponding to a peak test tone level of +18 dBm, is 75 kHz. This is the same peak carrier deviation as used by the f.m. broadcast industry. Since the highest modulating frequency is 15 kHz, the modulation index is 5 which again, is the same as the f.m. broadcast industry.

INDIVIDUAL CARRIERS

The individual carriers each correspond to program channels which may be operated in either mono, stereophonic, or quadraphonic configurations. The NPR system will initally be equipped for four carriers with each R/T uplink capable of transmitting two, and capable of being expanded. Thus, the initial system will be capable of transmitting four separate mono programs, two stereophonic programs, or one quadraphonic program.

Each of the four carriers will be generated on a separate frequency in the range from 63 to 77 MHz by the f.m. modulator for a particular program channel (mono). In the case of stereo, "left" will be on one carrier and "right" on another. Note that this is a departure from the standard f.m. stereo transmission system used by broadcasters which uses a composite signal carrying *both* left and right audio signals on one r.f. channel.

The combined group of f.m. carriers in the 63 to 77 MHz range from a passive power combiner is fed to the frequency up-converter, which translates each carrier by a fixed frequency amount to the desired uplink frequency. Thus, if there are, for example, four NPR carriers at 65, 69, 72 and 74 MHz at the input to the up-converter, each of these might be translated upward by 5995 MHz. This translation would result in the four carriers appearing in transponder number 4 on the downlink.

The NPR carriers at the output of the up-converter are each at a power level of about one one-hundredth of a milliwatt. Power amplification is provided by the power amplifier system to increase the individual carrier power to a level suitable for uplink transmission to the satellite. With the high-gain directive antennas used, a level of the order to 6.5 to 45 watts-per-carrier is required at the antenna, depending on the antenna size and gain. At remote transmit stations using 4.5-meter dishes, about 45 watts is required while at the 11-meter M.O.T. antenna, only about 6.5 watts is required.

ANTENNA REQUIREMENTS

A parabolic reflector antenna is used to provide the power gain and directivity needed to establish the uplink. The same antenna structure may be used for both transmit and receive by use of specialized isolation techniques which separate the transmit and receive signals at the antenna feed.

Gain of the antenna at transmit frequencies is approximately 46 dB, depending on the type of antenna used. With 45 watts-per-carrier delivered to this antenna, the effective isotropically radiated power (EIRP) is 62.5 dBW, which is sufficient to accomplish the uplink transmission with negligible degradation to the signal quality.

The NPR receive antenna is a 4.5-meter diameter parabolic reflector structure. Gain at 4.0 GHz is approximately 43.5 dB.

Such high-gain antennae, operated in conjunction with a low-noise amplifier, provide a sufficiently high received carrier power-to-system noise power ratio to achieve a high signal-to-noise ratio.

Incoming signals in the frequency range from 3700 to 4200 MHz are fed from the antenna receive output port to the low-noise amplifier. The LNA has about 60 dB gain, which raises the level of the received signals sufficiently for proper operation of the down-converter and f.m. demodulator.

LOW-NOISE AMPLIFIER

The low-noise amplifier establishes the sensitivity of the receive system. Since it has relatively high gain, noise contributions from "down stream" portions of the receiver system (e.g. down-converter and f.m. demodulator), are insignificant. One particular type of LNA has a noise figure of 0.75 dB which is 55 Kelvin. Only in the low carrier level areas must a lower-noise temperature (and more expensive) LNA be provided. It has been determined to be cost-effective to select 150, 125, 95 or 55 Kelvin units for each NPR receive terminal, as indicated by conditions at the individual location.

The down-converter contains input band-pass filters which select the 40 MHz portion of the 500 MHz wide LNA output spectrum corresponding to the desired transponder containing the NPR carriers.

These carriers are down-converted or mixed down to

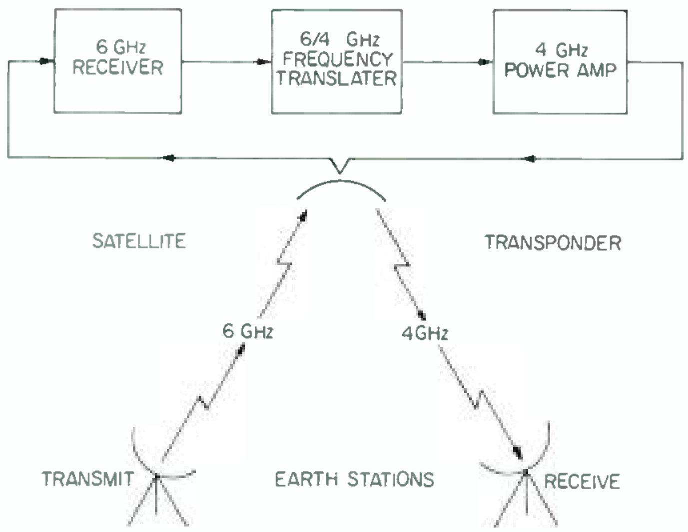


Figure 4. Block diagram of the satellite transponder.

recover the original frequencies generated by the f.m. modulator at the transmit site.

Output from the down-converter, consisting of the group of NPR carriers translated down to the nominal 70 MHz, is fed to a power divider. The output ports of the power divider each feed an f.m. demodulator, whose function is to select the desired carrier from the group and to demodulate that carrier to recover the program channel audio modulation.

Since the program audio was compressed prior to being applied to the f.m. modulator input, the demodulated audio recovered from the f.m. demodulator will be expanded. Each f.m. demodulator is followed by an audio expander to restore the dynamic range of the original program material. Output of the audio expanders feeds the NPR studio program equipment for local broadcast.

As we said earlier, transmit capability will be provided for certain public radio satellite ground terminals. These terminals will be capable of transmitting two carriers in either a mono configuration (two separate program channels) or in a stereo configuration (one stereophonic program carried in the two channels).

Thus, the public radio satellite ground terminals which are equipped with the R/T option will be capable of feeding programs into the satellite interconnect system for regional or national distribution.

In normal operation, the wide dynamic range audio program is compressed to improve signal-to-noise ratio, and applied to the f.m. modulator, which generates a frequency-modulated carrier in the 62 to 78 MHz frequency range. By means of front-panel selector switches on the compressor/modulator units, any one of twelve predetermined channels in this frequency range may be selected for transmission in accordance with the NPR frequency plan.

Output from the compressor/modulator to channel 1, is fed to the up-converter for that channel. In the up-converter, this frequency-modulated carrier is translated upward to the frequency in the 5925 to 6425 MHz transmit band corresponding to the satellite transponder in use. The translation frequency shift in the up-converter may be switched so that the program channel carrier feeds either the primary or the backup satellite transponder.

Now, the output of the channel 1 up-converter is fed to the power amplifier unit, which delivers approximately 100 watts at the nominal 6000 MHz frequency to the hybrid power combiner, as seen in FIGURE 5.

Operation of channel 2 is similar to channel 1. The equipment in both channels is identical; thus, separate carriers—each modulated with a program channel—are fed to the hybrid power combiner.

In this type of hybrid combiner, half of the power presented to either input port appears at the output port. Such a combiner has the advantage of providing a rather high isolation between the two input ports, thereby permitting completely independent operation of the two transmit channels. It is also a very simple and reliable device. When both channels are in use, two carriers on separate frequencies (each at a power level of approximately 100 watts), are fed to the two input ports of the hybrid combiner. Approximately half of the input power for each carrier appears at the output port of the hybrid, and is fed by way of the transmit feed structure, to the antenna.

ORTHOGONAL POLARIZATION

By means of orthogonal (i.e. 90 degree separation) polarization of transmit and receive signals, a high degree of isolation (at least 30 dB) is achieved between the transmit input port of the antenna and its receive output port. Hence, only a small fraction of transmit power appears at the receive port of the antenna.

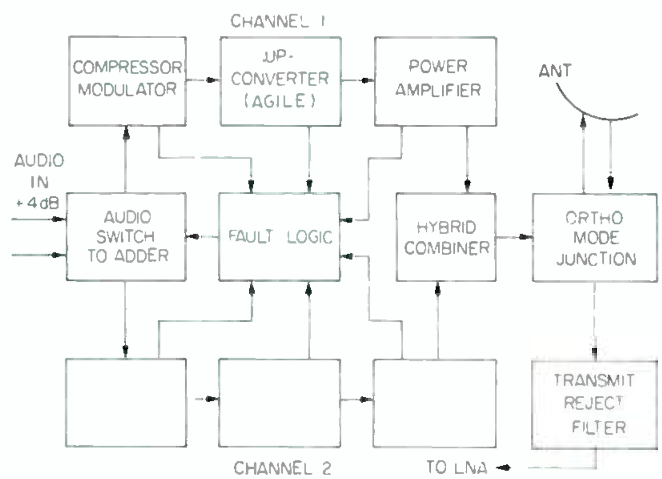
The transmit signal (one or two carriers) is fed via the transmit feed to the antenna which has approximately 46 dB power gain. At least 40 watts (+ 16 dBW) per carrier is delivered to the antenna feed. The antenna power gain results in an EIRP of at least 62 dBW, which is adequate to provide high quality uplink transmission with reasonable operating margins.

Although the polarization technique described provides a relatively high degree of isolation between the transmit input and receive output ports, it is necessary to provide additional attenuation to the transmit signal appearing at the receive port to prevent overload of the LNA. Thus, the transmit equipment chain includes a transmit reject filter (also seen in FIGURE 5). This filter has at least 50 dB rejection to frequencies in the transmit band and only about 0.1 dB insertion loss to signals in the receive band. The combined polarization isolation and attenuation to transmit frequencies of the filter provide 70 dB attenuation to the level of the transmit signal appearing at the LNA input. Thus, operation of the uplink transmit system does not affect or degrade the simultaneous operation of the receive system.

SATELLITE "FOOTPRINT"

The level of receive carrier power from the satellite will vary with geographic location because the gain of

Figure 5. Block diagram of the transmitter sub-system.



the satellite transmit antenna, like any other directive antenna, varies with the angle of the main beam axis. As a result of satellite transmit antenna gain variation, the EIRP varies across the United States. It is convenient to show EIRP at various locations by means of contours of constant EIRP, overlaid on a map of the United States. These contours are sometimes referred to as the satellite "footprint." For the WESTAR satellites used by NPR, the regions of maximum satellite EIRP are in the central U.S. In other areas, the satellite EIRP may be several dB less, with Alaska about 5 dB down.

For a constant signal-to-noise ratio, a less-sensitive (or more noisy) receiver may be used in high EIRP regions. Conversely, in low EIRP regions, a more sensitive (less noisy) receiver is required.

Saturated EIRP represents the maximum total power which the transponder can produce. The power available to each individual NPR carrier is considerably less than saturated power, since the transponder power capacity must be shared among up to twelve NPR carriers in addition to other carriers within the same transponder. In multi-carrier operation, transponder power must be "backed-off" to avoid unacceptably high levels of intermodulation in the satellite. In other words, the per-carrier power in twelve-carrier operation is somewhat less than one-twelfth of the saturated power level. Finally, an allowance of 1.5 dB is made for transponder end-of-life degradation. Thus, the per-carrier power in the NPR system is the saturated EIRP reduced by 10.8 dB for twelve-carrier power sharing, plus a suitable intermodulation backoff of 6.7 dB, necessitated by multi-carrier operation, plus about 1.5 dB for end-of-transponder-life degradation.

A typical value for pre-carrier power is about 16.5 dBW, or 44.7 watts.

The so-called path loss or spreading loss depends on the distance, or slant range, from the satellite to the receiving terminal.

Thus, a typical total propagation loss used in the calculations is the sum of the path loss and a 1.1 dB allowance for miscellaneous losses. A typical value, using this example, is 196.3 dB.

Carrier power, C, received by the NPR terminal is determined by transponder per-carrier power, propagation loss, and receive antenna gain.

In a computer program, the worst case carrier level is calculated and used in subsequent calculations. This conservative method of calculation is used to ensure that performance requirements are met with reasonable operating margin for either transponder in either satellite.

Thus, the lowest value or "worst case" carrier level using the typical values of these quantities as stated before is:

$$C - 14\text{ dBW} - 196.3\text{ dB} + 43.5\text{ dB} = -138.8\text{ dBW}$$

In any f.m. system, the signal-to-noise ratio is the sum of carrier-to-noise ratio and the f.m. improvement factor which is a function of receiver noise bandwidth, highest modulation frequency, and peak f.m. deviation of the carrier. For the NPR system in which these parameters are 200 kHz, 15 kHz and 75 kHz respectively, it can be shown from basic f.m. theory that the improvement factor is approximately 27.0 dB. Thus, the system audio signal-to-noise ratio is just 27.0 dB better than the r.f. carrier-to-noise ratio in the 200 kHz bandwidth.

It is a requirement in the NPR system that the signal-to-noise ratio of the bearer channel (not including compander improvement) be at least 41.0 dB. Thus, the required carrier-to-noise ratio (CNR) of the system must be at least 41.0-27.0, or 14.0 dB.

Antenna noise temperature is partially determined by the amount of noise power radiated by the "hot" earth entering the antenna through sidelobe responses. Therefore, the antenna noise temperature depends on the elevation angle of the antenna. Since the temperature of the earth's surface is of the order of 300 degrees Kelvin, a significant amount of noise power can enter the antenna sidelobes. At a typical elevation angle of 40 degrees, antenna noise temperature for the NPR antenna will be approximately 22 Kelvin. Using a computer program, antenna noise temperature is determined for the elevation angle corresponding to the worst case situation.

Table 1.

NPR SATELLITE CHANNEL AUDIO PERFORMANCE LIMITS

• Frequency Response (50 Hz to 15 kHz)	+0.5 to −1.0 dB
• Insertion Gain	±0.25 dB
• Insertion Gain Variations	
Short-time period (1 min.)	±0.1 dB
Long-time period (1 hr.)	±0.5 dB
• Random noise (ref to +18 dBm @ 1 kHz)	
Overall system (with compander)	≥70 dB
Bearer channel (without compander)	≥41 dB
• Crosstalk	≥76 dB
• Interchannel gain difference	±0.25 dB
• Interchannel phase difference	≤10°
• Impulse noise (C/N @ 14 dB) For peak pulse levels exceeding −52 dB ref to peak test tone level and pulse durations not exceeding one millisecond	1 pps
• Total harmonic distortion @ 18 dBm, 50 Hz to 15 kHz	≤1%
• Intermodulation distortion	≤0.5%
• Signal-to-periodic-noise ratio of overall system (ref 18 dBm maximum signal @ 1 kHz)	≥65 dB

NOISE CONTRIBUTING FACTORS

Although generally rather small, noise contributions "downstream" in the receiver system from the LNA must be considered. Such contributions include down-converter and demodulator noise, and noise contribution resulting from coaxial cable and power divider losses. Since the LNA has relatively high gain (about 60 dB) they are typically less than 1 Kelvin in most stations, but could total as much as 30 Kelvin.

Such downstream receiver noise contributions are included in the miscellaneous contribution margin.

The overall audio performance limits of the NPR satellite system are shown in Table 1.

When we consider all the design and performance requirements in terms of expansion capabilities, channel flexibility and remote input program transmission from regional centers, the new NPR satellite program distribution system opens a whole new era of expanded broadcast services to the listening public. NPR is proud to be a part of it. ■

JOHN WORAM

Report From Mexico City
El Centro De Grabacion

The SACMEX Centro de Grabacion—a recording complex of unique design.

"El Centro de Grabacion."

MOST OF US have heard of ASCAP and BMI, or, the American Society of Composers, Authors and Publishers, and, Broadcast Music, Inc. Together with Sesac, they form the "big three" of music licensing organizations.

But, what about "SACMEX"? This is the Mexican equivalent of the big three: in other words, the *Sociedad de Autores y Compositores de Mexico, s.a.* During the 40's, SACMEX composers attracted world-wide attention. Hit songs such as *Besame Mucho, Frenise, Perfidia* were recognized everywhere, even if their country of origin didn't get much attention.

But as the years rolled by, music of Mexico all but vanished from the international scene. The Mexican economy was growing, and local record companies turned their attention inward, towards meeting the needs of their own national market. With the rock revolution, the international stature of Mexican music deteriorated even further.

Studio A control room.

Studio A—almost 100 feet in diameter.

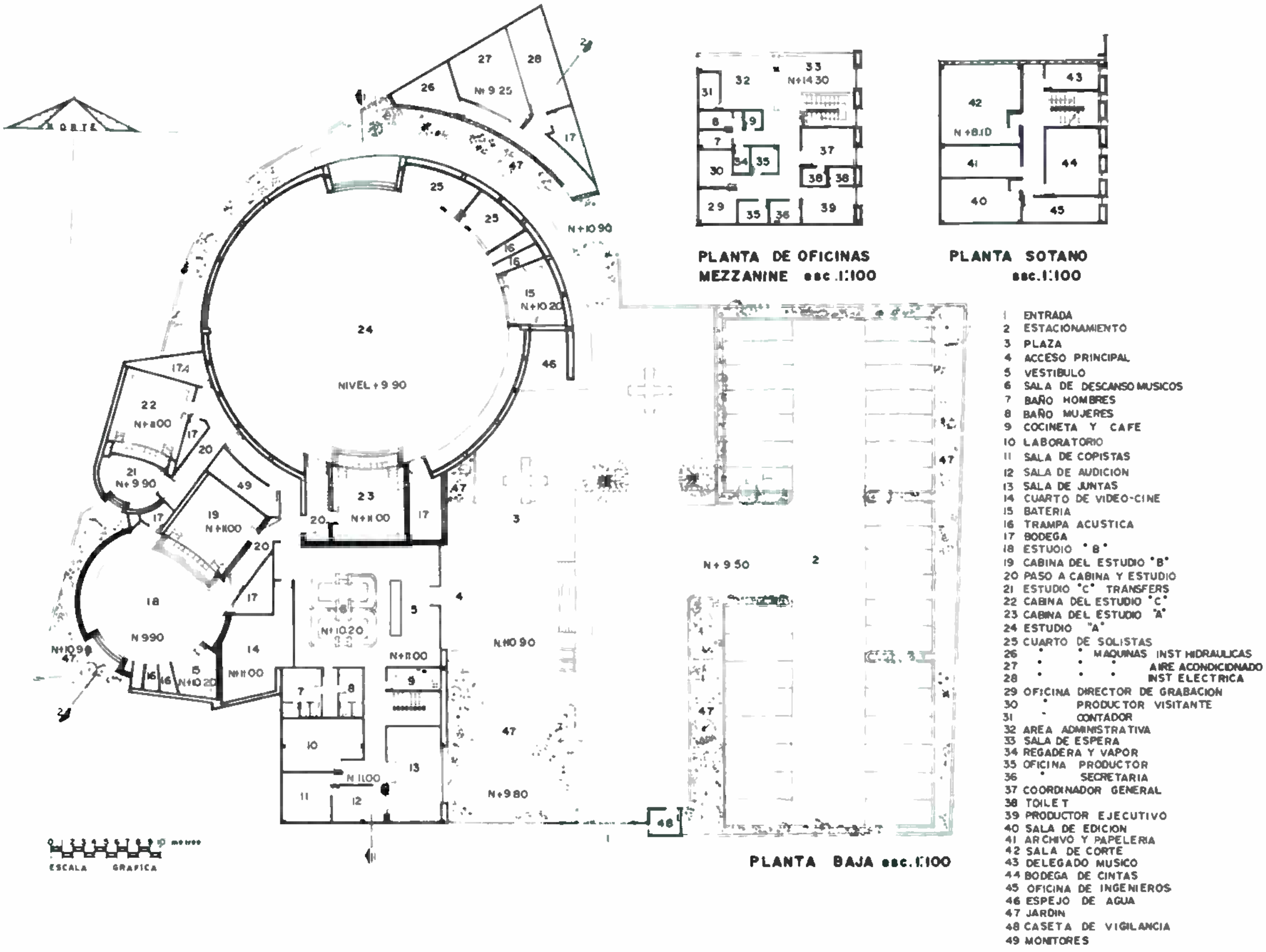
The Mexican branches of the big international record labels became interested in supplying Mexico with north-of-the-border rock, and were too busy to promote local talent abroad. The talent was there in abundance, but it just wasn't getting international exposure.

Mexican recording studios did not keep up with the state-of-the-art, and the future for the many talented artists and composers of Mexico did not look promising. The situation did not escape the notice of SACMEX, for one of the Society's obligations is to try to help develop and advance the standards of Mexican music.

After many discussions, the SACMEX board of directors came to the inescapable conclusion that they must take dramatic action, if the cause of Mexican music was to be advanced. An ambitious cultural center was planned, and is now under construction. It is safe to say that by the time it is completed, it will be difficult—if not impossible—to find its equal anywhere else. Eventually, the cultural center will contain six movie houses, three theaters, a school of music and technology, publishing houses, archives for printed music and records, and a large commercial center. And of course, a complete recording facility.

At the Harrison board.

The lounge/lobby area.



Floor plan for the SACMEX "Centro de Grabacion."

The financial planning for such a massive undertaking was developed by *Senor* Gilberto Navarro, a sub-director of SACMEX, with the enthusiastic support of the local financial community. The three theaters will be built in collaboration with the Mexican government and the *Filharmonica de las Americas.*

EL CENTRO DE GRABACION

Of immediate interest to **db** readers is the SACMEX *Centro de Grabacion*, or, recording center. The center is already functioning, and by anyone's standards, it's an eye-opener, and a controversial one at that. It consists of three recording studios, the largest of which (Studio A) is almost 100 feet in diameter. Diameter?

Yes, each of the studios is *round!* The idea came from Jose Antonio Zavala, who is *gerente de produccion* at SACMEX. Jose Antonio is part of a musical family of twelve brothers and sisters, who have recorded all over the world. During the past 25 years, he's been in and out of many recording studios, of all sizes and shapes—but never a round one. But, why *not* round, he thought? Late one night, he phoned his friend Prospero Sandoval—now the *director technico* for SACMEX—to propose the idea; which was met with a long period of silence. Eventually, Jose Antonio recalls, he shouted, "Hey Prospero, did you fall asleep?" to which Prospero retorted, "SHH! I'm thinking it over!" And then, "OK, let's do it."

The Mexican architect Manuel Rocha was called in, and eventually the studio plan found its way to paper. As the planning progressed, Ron Newdoll of Accurate Sound was asked to help specify and acquire the hardware package. Ron recalls his initial surprise at the idea of round recording studios. He found that most American and European engineers were conditioned to dismissing the idea as musically unsound. But Jose Antonio insists that the round studio should be able to satisfy *all* points of view, from the technical to the musical. As a musical director, he feels the shape allows him to spread out a large orchestra, and still maintain good visual communication with each section. Prospero Sandoval points out that the latest acoustical materials and techniques free the designer from the constraints of the traditional rectangular concert-hall approach to studio design.

In the smaller studios, the acoustic treatment lends itself quite well to multi-track recording, especially for rock groups. Separation is excellent, and groups can work out their own seating arrangement, without having to cope with the vagaries of nearby wall surfaces. The controversy really starts when dealing with the recording of larger orchestral ensembles in Studio A.

Throughout the world, there are concert halls whose acoustics range from so-so to spectacular. With few exceptions, the spectacular ones were built before the turn-of-the-century. In fact, in a recent Audio Engineering Society preprint, Professor J. Robert Ashley notes that, ". . . it would seem that we lost the recipe for a superb concert hall about the time of World War I." His paper is "A Preliminary Evaluation of 360 Degree Concert Halls." (AES preprint 1420). In the paper, Ashley asks, "Is a 360 degree hall a good recording studio?" and concludes that the answer is "No." Of course, Ashley is speaking of concert-hall type recording, not rock.

Some recent classical recordings from the 360-degree Berlin Philharmonic Hall may seem to support Ashley's position. Although the recording is remarkably clear, the traditional "depth" of sound has all but vanished. A recording of the Beethoven 9th Symphony may be a prime example. The dynamic range is spectacular, as is the clarity of the score. But this massive ensemble sounds as though it was recorded in a shoe box, with no room to breath. In fact, it sounds like no room at all. Some will love it for its clarity—others will regret its lack of spaciousness.

And that brings us back to SACMEX's Studio A; a round room built as a recording studio, not a concert hall. How does it stand up to Professor Ashley's question?

ROUND IS BETTER?

Several months ago, free-lance engineer Carson Taylor recorded the *Filharmonica de las Americas* in Mexico City's *Palacio de Belles Artes.* Prior to his recent retirement. Taylor was chief engineer for Angel Records, and senior mixer for Capitol/EMI.

Rehearsals for the recording sessions were held in SACMEX's Studio A, and in addition Taylor did some experimental sessions there. With 90 musicians in the orchestra, and a chorus of 75, the studio did not seem crowded. The circular shape provides ample room for movement, without dealing with the problems of corners. Taylor found that the possible negative effects of acoustic foci have been well-handled in the ceilings and walls, and no real problems exist from these sources. Isolation from outside noise is excellent. It was only during one particularly violent storm that an occasional thunder-clap could just be heard.

ROOM FOR IMPROVEMENT

Taylor feels that there are still some (lack of) ambience problems to be worked out. However, the floor is presently covered with a very thick, luxurious carpet, which absorbs reflected sounds. This is just right for the smaller studios, for recording multi-track. But, Studio A is too big for anything but a large orchestra, and could benefit from some controlled reflected sounds. Therefore, he has suggested that the carpet be replaced with a hardwood floor, to enhance the ambience of the studio. With the technical excellence found here, Taylor believes that in time, SACMEX's Studio A should become one of the finest recording facilities for large orchestral work in the western hemisphere. It should also be noted here that the studio was not necessarily planned for the European concert repertoire. It was designed to suit the needs of the Mexican musical scene, and the SACMEX officers are pleased with the sounds they hear in Studio A. However, experiments are being conducted, and it would not be surprising to find a hardwood floor, with roll-up carpeting in the future.

The smaller rooms have already attracted the attention of some well-known visitors. Doug Henning, Rod Stewart and Don Costa have all recorded there recently, and Polydor—as well as other major labels—are planning future sessions.

CONTROL ROOM EQUIPMENT

Visiting engineers and producers should feel right at home in the SACMEX control rooms. All three are identical in size, configuration and equipment complement. And that includes Harrison boards, 3M M-79 24-track recorders, dbx noise reduction, UREI compressors and graphic equalizers, Eventide flangers and digital delay lines. Four live reverberation chambers have been built in the basement, and these are supplemented by AKG spring reverberation systems.

All the Electro-Voice Sentry III monitors are bi-amped, using Crown amplifiers and White crossover networks. And, each control room is designed and equipped for quadraphonic mixdowns.

In specifying the equipment package, Ron Newdoll at Accurate Sound had to give careful consideration to reliability and ease of servicing. Although Mexico City is not that far away, getting equipment shipped across the border can be a tedious and time-consuming task. Long delays are the order of the day, and so a minor replacement part can easily become a major logistical problem.

FUTURE PLANS

So far, things have been working smoothly, with very little downtime. In fact, ambitious expansion plans are underway, and these will include a complete film transfer facility, as well as a tape-to-disc transfer system. Newdoll and Prospero Sandoval have spent a lot of time "A/B-ing" cutting lathes and are pretty close to making a final decision.

Newdoll reports that future planning will also include the installation of an automation system—as yet unspecified. Actually, SACMEX has already experimented with at least one automation system, which wound up being scrapped. According to Newdoll, it was impossible to get the necessary technical support and documentation from the manufacturer.

SACMEX'S IMPACT

The presence of SACMEX's *Centro de Grabacion* is being felt throughout the Mexican recording industry. Jose Antonio Zavala notes that shortly after the inauguration of the studio, many older studios began updating their facilities. In fact, Accurate Sound has established a Mexico City office to more efficiently meet the requirements of both Mexico and its neighbors to the south. Jose Antonio is delighted with the flurry of local activity. He is quick to point out that SACMEX is certainly not trying to compete with record labels. Its first duty is to promote the cause of Mexican music, and he feels that by making advanced recording technology readily available, SACMEX composers will enhance their competitive position in the international recording world. SACMEX has opposed a recent plan by the Mexican Musicians Union that called for a law that would require radio stations to play Mexican music 90 percent of the time. SACMEX favors free competition, and feels that with a facility such as the *Centro de Grabacion* readily available to local artists, the sound of Mexico will once again be heard—not only in Mexico, but around the world.

For that matter, recording artists from around the world should be heard more in Mexico. With a recording facility such as SACMEX offers, it should be hard to keep them away. ■

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JOHN WORAM

Report From New York
Soundmixers Recording Studios

Soundmixers—from drawing board to reality; a tale with a happy ending.

NEW YORK CITY, January, 1977—Harry Hirsch is sitting in a down-at-the-heels office in New York's ancient Brill Building at 1619 Broadway; that once-famous music industry mecca with the polished brass doors on the outside and now, an air of genteel decay inside. He's describing a pie-in-the-sky million dollar recording studio that he wants to build, right here in Fun City.

Harry's fanciful tale has a familiar ring to it. We've all heard the same story again and again. "The money people are getting together next week, and we should be underway in just a little while longer."

A DIFFERENT TUNE

That's usually the end of the story. The deal falls through (it's always someone else's fault), and the story dies, only to be resurrected some time later in another locale. But this time, there's a difference, and a refreshing one at that. For while Harry is talking, a crew of workmen are gutting the second floor of the Brill Building, making way for what promises to be an exciting new studio complex —and right here in the heart of Fun City!

It turns out that Harry and I went to RCA Institutes at about the same time, although neither of us will admit to graduating as long ago as the late 50's (*very* late 50's, Harry adds). While your author sought fame and fortune in the quality control department of a large record company, Harry sought his with Charlie Leighton at JAC Recording. At the time, JAC was a very suceessful mono, and later stereo, studio across the street from the Plaza Hotel. After cutting his recording teeth there (doing Pepsi, Coke and Newport jingles), he left to help form Media Sound, moving one block south and a few blocks west, into a former church.

Although Media seemed to be doing well, Harry split with his partners in late 1974, and spent the next few years thinking about building yet another new studio. Along the way, he met John Storyk of Sugarloaf View, and was impressed with his conceptions of layout and "ambience". And then there was Robert Wolsch, one of the big apple's more inventive photographers, and an expert with color and lighting. (For a first-hand look at Wolsch's work, see our November and December 1978 covers—Ed.)

And now, some twenty years after that graduation, we sat together surveying the floor plans for what would soon become known as Soundmixers—the largest independent recording studio complex ever built in New York City. The plans revealed three separate recording studios with identical control rooms. These are hexagonal in shape, and occupy about 425 square feet each. A fourth studio will have a slightly smaller control room.

A MILLION DOLLAR EFFORT

Altogether, Harry anticipated that the lease-hold improvements on the building would come to about $500,000 (not counting Excedrin, of course). Another $500k should take care of the equipment package, which was to be coordinated by Ham' Brosius at Audiotechniques. Harry decided to go with Audiotechniques after making the decision for MCI consoles and tape recorders. As a leading MCI dealer, Brosius was the logical choice for the "everything else" as well.

And what swayed Soundmixers to MCI? According to Harry, it was a combination of features, "track record" and industry acceptance. With artists, engineers and producers expected from all over the world, it was going to be important to have a console that would be immediately accessible to the most number of people. Chances are, if a client has any sort of studio experience, he's already familiar with MCI.

Accordingly, Soundmixers ordered three MCI 542 consoles for Studios A, B and C. (For the time being, a smaller

The lobby at Soundmixers, ready and waiting.

Harry Hirsch surveys one of his studios, not quite ready for a session.

Of course, you recognize the MCI console. If not, there's a better view in one of the other photos.

custom-built board was slated for D.) The MCI boards were re-configured to place eight VCA sub-mixers at the center of the console. This was done to make mixdown sessions a little easier for engineers who don't have two dozen fingers. Since these studios were to be equipped with 24-track MCI recorders, the engineer could sit dead-center and in most cases, do his mixing on the eight sub-mix groups.

As for noise reduction, Harry would have both Dolby and dbx available. At the moment, a "house tape" had not yet been chosen, but testing was under way.

About 75 microphones were on order. These included the usual complement of AKG, Electro-Voice, Neumann, Sennheiser and Shures, plus some RCA 44s and 77s. As for amplifiers, no decision had been made yet, although the field was narrowed down to about five contenders.

FILM SCORES

Harry hadn't forgotten his ad agency background either, and was also looking forward to getting more involved with film scoring. Accordingly, plans were being made for a Philips tele-cine projector, which would look into a color tv camera. On every frame of the color picture, a BTX SMPTE Time Code Generator would display an addressed frame, either in real-time or in footage. In the control room, the client and engineer would watch over a 19-inch color monitor, while out in the studio, the musicians would be able to follow the action on a 9′ by 15′ screen. With 2-foot high time-code characters displayed on the screen, there would be little chance for missing a cue.

Speaking of cues, there would be four audio cue feeds to the studio—typically, two mono, one stereo and one for click-track-plus-program. In addition, an intercom with private phone would be available at the conductor's podium.

The lounge area, with "coffee and . . ." available.

Soundmixers Studio A control room.

SIGNAL PROCESSING DEVICES

Complementing the MCI consoles and machines, there would be an ample supply of UREI 529 one-third octave equalizers, 1176LN and LA-3A compressors, 964 digital metronome, Eventide Harmonizers, delay lines and Flangers and a Marshall Time Modulator. Reverberation would be handled by EMT plates, and AKG BX-10s and BX-20s.

Soundmixers hadn't forgotten musical instruments either. A Steinway B had been ordered for Studio A, and Yamaha C7s were going into the other rooms. Also on order were Musser vibes and a xylophone, three complete drum sets, two Hammond organs with Leslie speakers, tympani and an assortment of electronic keyboards.

About those drums, Harry did not plan to enclose them in any sort of "sealed container," as seen so often these days. A drummer himself, he wanted the drummer to be able to see and feel the rest of the group, and with Storyk's soffit system, there should be no problems.

IN OPERATION

January, 1978—It's now about two years since our first meeting. Soundmixers opened its doors on September 15, 1977, and has been busy ever since. Its "advertising attitude" has attracted a steady stream of ad agency clients such as J. Walter Thompson, Doyle, Dane & Bernbach, and Young & Rubicam. In between commercials, the studio has earned RIAA gold for Meatloaf's single, "Two Out of Three Ain't Bad," and Kenny Loggins' "Nightwatch" album. In addition, albums recorded at Soundmixers are now available on A&M, Arista, Atlantic, Casablanca, Columbia, Elektra, Polydor, Private Stock, RCA and United Artists. Artists include Sylvia Syms, Melba Moore, the Fania All-Stars, John McLaughlin, Laura Nyro and Woody Herman. In the works are projects by Baby Grand, the Average White Band, the BeeGees, and Peter Frampton.

Also, Harry has a new office—with a rug, yet!

Two years ago, Harry was confident that Soundmixers would attract a lot of major recording work back to New York. He was right, but he underestimated a bit. On the drawing board, Soundmixers was conceived as a multimedia house, with work split between film, advertising and label sessions. Now, two years later, it turns out that almost 80 percent of the bookings are record dates.

Accordingly, Tom Hidley has been called in to make some modifications to the three main control rooms; finetuning them to better meet the needs of today's market. Basically, the work involves a re-design of the acoustic trapping, and larger flush-mount surfaces surrounding the speakers.

Studio D, under construction.

And that led us into a discussion of what makes up the "perfect" control room. Harry feels that the basic obligation of an acoustical consultant is to design a control room that is flat. Once you start adding 6 dB of room equalization, you're compensating for a room that was not well-designed in the first place—and probably generating a lot of IM distortion as well.

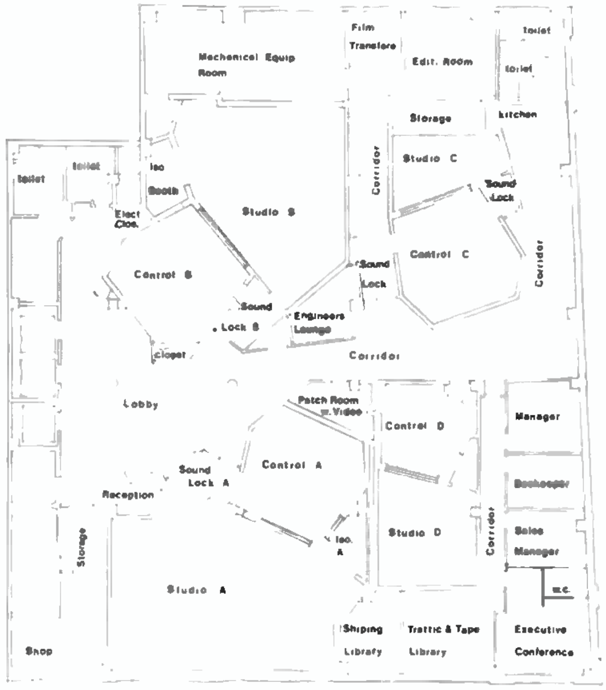
With a good room, the producer shouldn't receive any great shocks when he plays his tapes at home. Of course, it's going to sound different—that's inevitable. But, provided his home system is in good shape, he should be able to form some sort of frame of reference between home and what he heard in the control room during the session.

No changes are planned in the studios themselves. The isolation seems to be working out well, and all-at-once sessions with strings and brass are no problem at all.

CLOSE-MIKING AND MULTI-TRACK

And that brought us 'round to a few words on the subject of recording styles. Harry recalled the time—not really that long ago—when some producers (especially in film) regarded the multi-track recorder as a "safety." During the session, it was the stereo (and sometimes, mono) tape on which all the balancing decisions were made. Those days seem to be gone, perhaps forever. Now, people demand close-miking for the control it gives them—regardless of

A drummer's-eye view of Studio A.



Floor plan for Soundmixers Recording Studios.

whether they need it or not. Musicians are becoming conditioned to multi-track, and many of them don't know what they're missing. Of course, Soundmixers can provide all the control that anyone needs, but—now and then—it would be nice to see (and hear!) a real stereo sound again.

Speaking of close-miking, and of microphones in general, the AKG-414 seems on its way to becoming the all-time favorite at Soundmixers. Electro-Voice RE-20s are in demand on drums, and other popular models include the Sony C-37, Sennheiser 421 and AKG 451.

By now, Soundmixers has completed its tape evaluation program, and decided on Ampex 456 "Grand Master" as the house tape. For engineers who prefer something else, they keep a variety of other formulations available too. Noise reduction seems to be running about 60/40 in favor of Dolby, with some producers opting for 30 in/sec with no noise reduction.

New signal processing devices include the Audio & Design F760X Complex-Limiter, which has become a great favorite. The studio hasn't got enough of them yet, so fights over who gets to use it next aren't uncommon.

A LOOK AHEAD

What about the future? It looks as though Soundmixers is going to be ready for it. Last year the studio took delivery of BTX's new micro-processor-based SMPTE Time Code Package. (For more about this, see "SMPTE Time Code Comes to Audio," in the November, 1978 issue of **db**—Ed.) The BTX system was originally intended for record and film dates, but Harry finds that more and more commercial accounts are beginning to use it too. Early in 1978, Studios B and C were converted to MCI's automation system as well.

And, Soundmixers isn't ignoring digital technology either. A Lexicon digital reverberation system is on order, and Harry has made arrangements to evaluate Sony's new digital tape recorder system that was seen at last November's Audio Engineering Society convention.

So, the chances are that if its new and innovative, you'll find it at Soundmixers—right here in Fun City! ■

JOHN BORWICK

Report From Paris

IRCAM—A Unique Center for Research

In attempt to integrate music and technology, IRCAM brings together musicians, audio engineers, and scientific research workers.

"JOHN, go see IRCAM." The voice of Larry Zide telephoning from Long Island to my English countryside retreat sounded as if he meant it. Needing very little persuasion to visit Paris at any time, I hopped on a plane and soon my taxi-driver was careening through the streets—simultaneously waving both hands in the air. "IRCAM, *c'est unique*: it is the first time I took anyone there!"

The shock of parting with 70 Francs for the taxi ride was followed by a visual shock. I had been put down at what looked like a cross between a devasted bomb site and a hurriedly-erected travelling circus. Now the effect on me cannot be fully appreciated by U.S.A. readers, for whom the knocking down of characterful buildings and the putting up of characterless ones is commonplace in all the big cities. But in Paris?

By a coincidence, London and Paris are both going through the traumatic experience of tearing down not single buildings, but entire areas formerly occupied by busy market centres. In London, it is the famous Covent Garden Fruit and Vegetable Market (scene of Eliza Doolittle's meeting with Professor Higgins in "My Fair Lady") and in Paris it is the equally colourful market complex just North of the *Louvre* and *Notre Dame* called *Les Halles*. Back in 1969, a master plan for establishing a new centre for contemporary art in the heart of Paris was formulated by President Georges Pompidou. *Le Centre Georges Pompidou* was eventually inaugurated in December, 1976.

The main structure is of revolutionary design and seems to consist primarily of glass and steel with large interior spaces left free for future needs and experimental short-term installations. The building houses three major departments, a museum of modern art, an industrial design centre and a public library with enormous space for changing displays and audiovisual "happenings." The pity is that so many of the old market buildings have been knocked down (and, sentiment aside, it must be admitted that they had to go). Therefore the new Centre adjoins a wasteland occupied (when I arrived) by lads playing football and ugly tents and "shapes" housing sundry manifestations of modern art.

Of course anyone who has enjoyed the special delights of eating in a French restaurant will know that this Gallic race cares much less for external appearances than for the excellence of what is served within. So I still felt fairly buoyed up as I dodged the flying footballs and came to a sign for "IRCAM," with an arrow pointing downwards!

Then I understood all. I had been told by architect friends that IRCAM had achieved an incredibly low interior noise level approaching 0 dBA—which compares more than just favourably with the 20 dBA of a good recording studio. Obviously, the only way to exclude noise so effectively in the heart of Paris was to build the whole thing under ground. Pausing briefly to look at the 30-foot high air conditioning towers, which are the only parts of IRCAM above ground level, I went downstairs—to find a veritable Aladdin's Cave of audio riches, and a very friendly welcome.

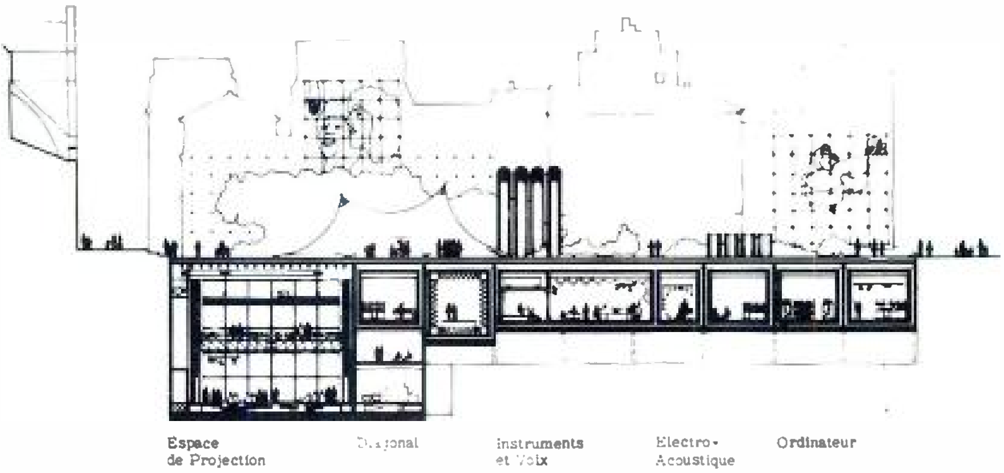
IRCAM—AN EXCITING ADVENTURE

IRCAM—the Institute for Research and Coordination Acoustics/Music is an autonomous section of the *Centre Georges Pompidou*. Its director is Pierre Boulez, a composer and conductor with vast interest and experience in electronic and experimental music of all kinds. The main objective is to bring musicians, audio engineers and scientific research workers together to work on problems of musical composition, performance and reproduction that can no longer be solved by individuals. As Boulez puts it, "the musician must assimilate a certain scientific knowledge . . . the scientist must understand the direction contemporary music has taken, and orient his imagination along these lines. In this way we hope to forge a kind of common language that scarcely exists at present."

IRCAM has departments; Instruments and Voice (directed by Vinko Globokar), Electroacoustics (Luciano Berio, the composer), Computers (Jean-Claude Risset), Research and Training (Michel Decoust), and Coordination (Gerald Bennett). Boulez has chosen a truly international team to head up these departments: two French, two American, one English, one Italian and one Yugoslav. I met all sorts of nationalities on my tour, and my friendly guide himself, the chief sound engineer Benjamin Bernfeld, is Rumanian and a very active member of the AES European Region organizing committee.

We began in the large Concert Hall (called in French *Espace de Projection*) which has overall measurements of

John Borwick is audio editor of the British publication "The Gramophone."



A sectional view of the underground IRCAM at the "Centre Georges Pompidou." The large area to the left is the "Espace de Projection."

90 x 85 x 46 feet high and can seat up to 400 people. It has a degree of flexibility and versatility that I have seen nowhere else. A better description than Concert Hall would be Experimental Acoustic Enclosure. I have already referred to its 0 dBA noise level, but even more striking are the arrangements for controlling reverberation time. I have visited studios with "variable acoustics" in Germany, and even used some during my time at the BBC, but the amount of variation has usually been minimal. So much so that the balance engineers have tended to leave the reversible panels and dummy ceilings at some half-way setting and get the degree of dryness or ambience they want by selective microphone placement and judicious use of reverberation plates or chambers.

But the IRCAM Concert Hall can be given any reverberation time from 0.8 to 4.5 seconds, by remote control from the monitor room. All wall and ceiling surfaces comprise rotatable prismatic panels, each eight feet long and with absorbing, reflecting or diffusing faces. There are 516 of these "periactes," with a very complex system of motors to turn them, in groups of three, to any one of seven positions. At present, the settings are manually selected—and the change of acoustic was very effectively demonstrated on speech and music when Pierre Boulez presented the Inaugural Concert—but there is an ambitious plan to program one of IRCAM's several computers so that any required reverberation time/frequency curve can be set up automatically.

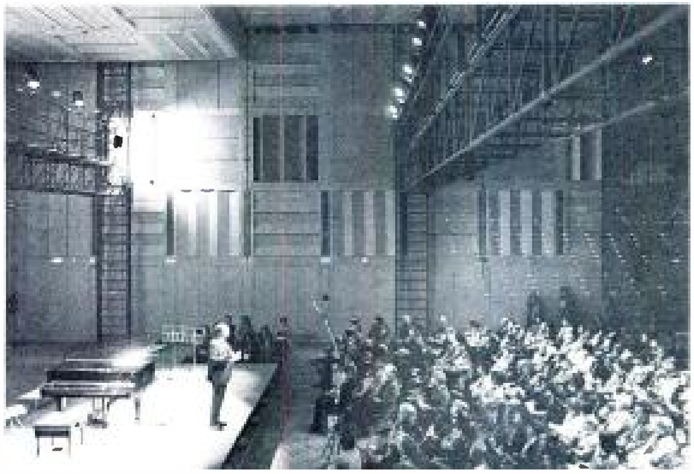
In addition, the three sections of the ceiling can be lowered to reduce the effective height to only about 15 feet. A complicated system of cranes, gantries and podiums can set up any desired relationship of musical performers, loudspeaker arrays and audience—not just the traditional raised platform for musicians and flat or raked audience area.

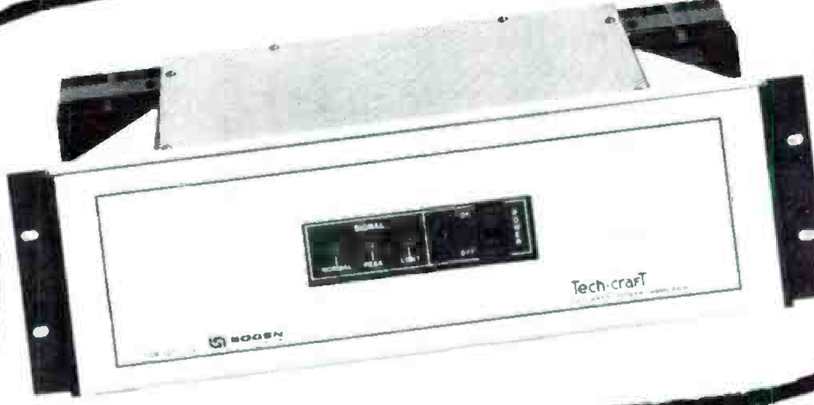
The monitor room has a 32 x 16 Neve mixing desk and large JBL monitor speakers. However, because of the enormous viewing window and multiple racks of control gear, Ben Bernfeld told me that they were not keen to make recording and mixdowns in this room. It was more than adequate for control of sound reinforcement, television producers and crews etc., but the main recording control room was just being completed next door. On entering, I could tell immediately that this would be a more ideal acoustic environment for making auditory judgements.

Reverberation time had been brought down evenly to about 0.4 second at all frequencies, using a wall and ceiling treatment made up of a metal honeycomb grid over a cloth-covered absorbent. Bernfeld had a feeling that the room was now too dead, and that they might want to increase T to about 0.6 second. This is comparatively easy to do by adding well-damped reflecting panels which simply bolt in front of the metal grid—a form of treatment I saw later in the three other studios and their control rooms. A large Neve console occupied the centre of the control room, with JBL monitors again, a 16-track Studer A80 recorder and several smaller Studer and Ampex machines. A 16-channel Dolby A noise reduction rack was fitted, with a set of Telcom boards available as substitutes. The rival dbx system was used in the Concert Hall monitor room. As well as conventional three-pin connectors everywhere, there was a very flexible 63-line ring system in all studio areas, allowing comprehensive linking of audio sources and destinations. Remote control/autocue of the 16-track A80 was also possible from one studio to another.

The largest studio was big enough for 20 musicians or

Pierre Boulez at a press conference in "Espace de Projection."





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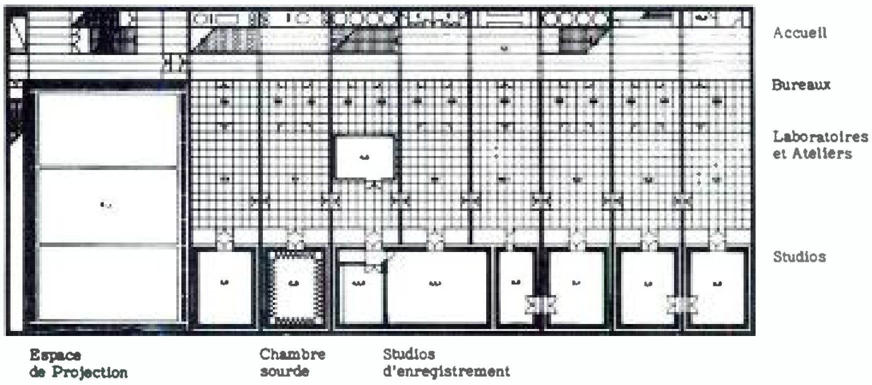


The IRCAM floor plan.

so, while the others were set aside specifically for electro-acoustic or computer-assisted music-making and research. All studios are well separated from routine noises, since they are built along one side of the underground space. The reception areas and corridor are along the other side, with access to the studios only by walking first through the associated offices and then through their laboratories.

RESEARCH IN PROGRESS

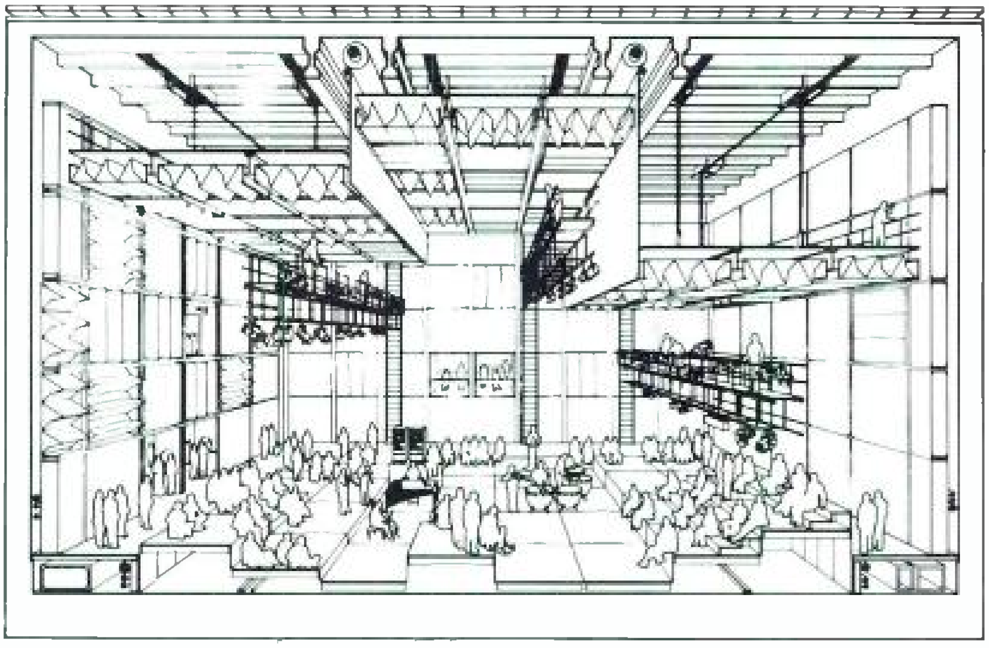
A fair number of the research projects currently in progress do seem esoteric (way out) by normal sound studio standards. For instance, I could hear a flute being played so badly that it was all breath and squeak. But this, I was told, was a part of a study of the multiphonic or "dirty" sounds that can be obtained from most instruments by departing from the normal fingering or blowing techniques. Composers are now interested in exploiting these previously unwanted sounds, particularly from the woodwind. On examining the flute I had heard, I found it had been "prepared" by sticking pieces of plasticene on some of the keys. A recorded catalogue of multiphonic sounds was being prepared and played to listening panels for putting into rank order of various criteria. The non-harmonic spectra of bells and other percussion instruments are being studied as an aid to construction and, of course, electronic synthesis.

Three aspects of digital sound synthesis are being studied. First, synthesis on general-purpose computers follows the Music V program developed at Bell Laboratories by Max Mathews and others. (He is now scientific consultant at IRCAM.) The system has 192,000 words of core memory, about 30 million words of disc storage and can be used by up to 16 people at a time. Second, synthesis on specific devices aims to give composers control of sounds in real time. The system used at IRCAM was developed by G. di Giugno and provides 256 oscillators, 256 amplitude modulators, computer-controlled frequency, amplitude and phase, and up to four wave-shapes of 4,000 words resolution can be used simultaneously. Third, psycho-acoustic studies to help composers plan their synthesis work has centered on a group of computer programs called *ESQUISSES*. This can greatly speed up the progress from a musical sound in the composer's imagination to its optimum realization technically.

Another research project—to me more interesting—was the development of specialized transducers (contact-microphones) of very high quality. The Concert Hall gives new possibilities for studies into room acoustics. Victor Peutz, the acoustical consultant for the IRCAM building, has prepared measurement programs. One of these uses a new algorithm for determination of reverberation time. It calculates the decay and relative strengths of the three most important components of the decay curve in any required number of frequency bands. The apparent contradictions

The recording control room for the IRCAM concern hall.

Giuseppe di Giugno, designer of IRCAM's digital synthesizer, seated at the Neve console. Note the acoustic wall treatment.



An artist's conception of the "Espace de Projection."

in the Sabine and Eyring formulae should soon be resolved. The computer will be used increasingly for the modelling and simulation of enclosures and, where possible, the Concert Hall will be used not just for research measurements but also for live concerts to check ideas against audience reaction.

PUBLIC LIAISON

Besides a continuing programme of concerts and lecture/demonstrations, IRCAM and the entire Pompidou Centre have a commitment to involve the general public as much as possible. Technical seminars are promised for 1979-80 and a series of 6 to 8-week courses for composers is already announced. The latter have starting dates in 1979 of February 5, July 2 and November 5. Entry will be highly selective, as only four to six participants per course are envisaged, and a tough 6-day week working schedule has been prepared. Course fee is 500 NF.

I enjoyed my visit. I regard IRCAM as a unique and ambitious step towards the integration of music and technology (something I have been modestly concerned with during the past seven years coordinating and teaching on the *Tonmeister* degree course at the University of Surrey). As well as bringing composers and researchers together, IRCAM has recruited experts from many different countries—which must be a good thing—and the friendly welcome I received proves that they are putting one of their *raisons d'êtres* into practice: "IRCAM is a creative laboratory, and also a place open to the public who will be welcome to form their own judgement and to participate."

The following *Rapports IRCAM* have been published, and are available by mail order at 20 French Francs each. The reports are printed in English and/or *French*, as noted below.

1. Research at IRCAM in 1977 (English and French).
2. *Unite electronique destinee a la transformation du son en temps reel, programmable et controlable par l'instrumentiste.—Rene Causse, et al.*
3. Computer Facilities for Music at IRCAM, as of October, 1977—John Gardner et al.
4. A One-Card 64 Channel Digital Synthesizer—G. di Giugno and Hal Alles.
5. Real-Time Synthesizer Control—Max Mathews and G. Bennett.
6. The Use of the Linear Prediction of Speech in Computer Music Application—James A. Moorer.
7. *Musica, Programme de Codage de la Musique—Giovanni di Poli.*
8. Musical Acoustics—Jean-Claude Risset.
9. The Development of Digital Techniques: A turning point for Electronic Music?—Jean-Claude Risset.
10. *Paradoxes de Hauteur des Sons—Jean-Claude Risset.*
11. *Hauteur et Timbre des Sons—Jean-Claude Risset.*
12. Low Dimensional Control of Musical Timbre—D. Wessel.
13. Perception of Timbral Analogies—D. Ehresman and D. Wessel.
14. Conceptual Structures for the Representation of Musical Material—D. Wessel and John Grey.
15. Computer-Aided Model of Stereophonic Systems—Benjamin Bernfeld and Bennet Smith.
16. A Digitally Programmable Filter—Maurice Rozenberg.
17. About This Reverberation Business—James Moorer.
18. A Composer's Notes on the Development and Implementation of Software for a Digital Synthesizer—Neil B. Rolnick.

The reports can be obtained from:

IRCAM—Relations Exterieures
31 rue Saint-Merri
F-75004 Paris, FRANCE

According to IRCAM, no mail order can be accepted for less than five reports. A cassette of musical examples is also available for 20 French Francs.

IRCAM has also been described in an Audio Engineering Society preprint (no. 1310), entitled, "The Variable Acoustics of the *Espace de Projection* of IRCAM (Paris)." ■

JEFF NIECKAU

Convention Report:
The 11th Tonmeistertagung

A look at some of the discoveries and impressions of the audio wares displayed at the meeting of the Tonmeisters.

Let's admit it: all of us enjoy the excitement of discovery, while strolling through exhibitions of audio and electronics equipment. That was one reason for my going to Berlin West last November. The other was the **11th Tonmeistertagung.** (Meeting of *Tonmeisters*). Both meeting and exhibition were organized by the *Verband Deutscher Tonmeister e. V.* (the German *tonmeister* association).

The venue was the *Sender Freies Berlin* (*SFB*), West Berlin's Free Radio station on *Masurenallee*. Meetings and equipment demonstrations were held in the studios of the *Haus des Rundfunks;* the historical Broadcasting House, which had housed the radio organization of Germany's first Republic before Hitler. Later on, (1933 to 1945), it was the *Reichs-Rundfunk;* the Nazi broadcasting chain under the supervision of big-mouth Joseph Goebbels. As a punishment for that, there followed more than ten years under Soviet control as *Berliner Rundfunk* (despite the fact that the place was located in the then-British Sector). Finally, in 1950 *Haus des Rundfunks* was given back to the present owners.

For me, it was a pleasure to see the place again, because it had housed me twice: in 1940 as a young practitioner for a studio course, and again in 1952, as a sound engineer—an assignment I terminated myself, in order to avoid renewing my acquaintance with Siberia, which I already knew well enough from P.O.W. times.

The meeting of *Tonmeisters* included a good old German-styled big opening with neck-tie atmosphere and speeches of all kinds from the top brass in the spacious studio 1. Throughout the meeting there was a hard-working atmosphere with lectures neatly divided into presentations by members and/or associates of the VDT organization, and by representatives of manufacturers and/or distributors of equipment. On the whole, it was an almost-European event. Lecturers came from Austria, Belgium, Denmark, England, Germany, Norway, Poland and Switzerland. They were from broadcasting organizations, technical institutes, music academies and from Germany's *Institut Für Radioteknik* (*IRT*)—the technical and standard-watching institution, financed by all broadcasting organizations of the Federal Republic of Germany. Themes covered a large spectrum of engineering, technology, music-engineering relations, and automation. The topics presented by the manufacturer/distributor group did not differ much, except if refering to new products. Most of the lecturers in this commercial group were Germans or Austrians.

THE EXHIBITION

A far broader scope was offered at the exhibition, which was located inside the large new TV Center of *Sender Freies Berlin.* It has not come to my attention whether anybody got lost, or died from hunger, while touring never-ending corridors in search of the various exhibitions or equipment demonstrations, lectures and—last but not least—the cafeteria. Nevertheless, the way in which a total of only 55 exhibitors were spread all over the place was a unique achievement. The most accessible part of the exhibition was the large entrance hall of the SFB TV Center, where nine top suppliers to German broadcasting (e.g. AKG, K+H, Neumann, Schoeps, Siemens) were found. Surprisingly, Philips was also right here; the largest of all, despite its not being a top supplier in Germany, at least not up to now.

But the large Philips stand was for good reason. because Philips came up with two remarkable things. First, a custom-made multi-track automated recording and mixdown console with everything one could dream of. Second, the new "LDC StoreMix System," a development out of

Jeff Nieckau is manager of commercial studio systems at Bauer Fernmeldebau und Elektronik KG, Mainz, Germany.

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the practice of the Polygram studios, in collaboration with Philips Laboratories. It's a vca-based mixdown system with storage on tracks between the recording tracks of a multi-track tape, meaning it does not steal any of your tracks. This of course is achieved by two additional data heads, mounted left and right from the erase head of a normal multi-track tape machine. The read-before-write method makes it possible to store a full mixdown program including all update possibilities on only one data track. No data erase head is required, since the write function erases all previously stored data information. On 16 recording track machines, this system offers up to 12 data tracks, meaning that up to 12 different mix programs may be stored, and that three more data tracks are available for autolocator time codes or sync impulses.

To find AEG-Telefunken, top supplier of German broadcasting, one had to move on to another place, where only six exhibitors were concentrated. I found a very modest stand: Telefunken did not play it that big, but then why should they, as business goes on and on. However, top engineers were on hand, demonstrating the "telcom c4 Compander System", which has found acceptance over here in both broadcasting and commercial studios. EMT—another top supplier—was at the same location and had on show a cleverly-selected sampling of its products (the world-famous reverb plates, electronic reverb, digital delay, studio-type high quality record players like the famous newer EMT 950, limiters, compressors etc.). Also seen here were Studer products and Harrison consoles. Spendor monitor speakers from England were shown and demonstrated, after have found several customers amongst the broadcasting organizations. The most impressive stand in this area was by the German 3M Company, featuring the M-79 multi-track recorder together with the many other professional products of this U.S. firm.

Everybody who was not yet aware what the German word *verboten* means had a fair chance to learn it here at SFB, because a lot of things were *verboten*, although the staff of SFB was generally helpful and friendly. Most amazing was the *Rauchen verboten* (No Smoking) in TV Rehearsal Studio A, where not less than 40 exhibitors were concentrated. Could this be the reason why only a very few of these exhibitors belonged to the major league of German broadcasting suppliers? The most impressive show of consoles and other studio gear was by MCI's German distributor R. Barth KG. from Hamburg, who manufacture their own studio products as well.

WAS IST EIN TONMEISTER

Tonmeister is a degree, and a position, invented and introduced by German broadcasting organizations. Initially the *Tonmeister* was supposed to be a musically-trained recording supervisor, with enough technical knowledge of the recording process to be responsible for the final sound and to maintain good relations between recording engineer and conductor. Though there are *Tonmeister* positions at every German broadcasting house and corresponding numbers of *Tonmeisters* at work, only a small number of commercial recording studios have adopted this system. Therefore, the majority of the members of VDT are employees of radio and tv in Germany, and consequently the event in Berlin was first of all a matter of broadcasting. While noting the reactions of prospective buyers to the exhibited products, I decided to pay particular attention to the judgements of the *Tonmeisters* from German radio and t.v. With its nine federal radio & tv centers. the Voice of Germany with radio and tv activities, the *Deutschlandfunk* with radio-only, and ZDF, the mighty Second German TV Network, the market is incredibly big.

Reactions of the *Tonmeisters* in view of the beautifully-designed and very workable commercial equipment (e.g. at the MCI stand) were quite mixed. One could see that especially the younger ones would definitely like to work with such equipment, but lack the courage of introducing it at their studios, knowing that IRT regulations (up to now) could be a serious barrier against the entrance of any commercial console into the holy field of German broadcasting. The older ones, still the top brass in the field, object to "too many knobs" and "too-close-together" faders and knobs. This of course refers to the standard width of 40 mm (1.575 inches) of all modules introduced and admitted over here. Naturally, the old-timers would insist in full compliance with all relevant IRT regulations before ever proposing such a console to their planning divisions.

Over here, service engineers of broadcasting organizations are not easy to convince towards introduction of commercial consoles, since all their spare parts and service facilities are for the standard type of closed-case modules, as introduced at its time by Telefunken, with the technical blessings of IRT. Most consoles, amplifier racks, and even the amplifier parts of some tape recorder models. are assembled from this type of module, which is available for all conceivable functions and applications, down to a simple talkback knob. Meanwhile, Telefunken and at least seven other German manufacturers are producing and selling such modules (e.g. EAB, ENB, Danner, Lawo, Neumann, Siemens, TAB). All stick to the same standard sizes, carry the same locking system and comply with all corresponding IRT regulations. Standardization goes right down to exactly the same pins and pin connections. No doubt, to successfully run against this wall of "panzer" module-techniques will take its time, if it is possible at all.

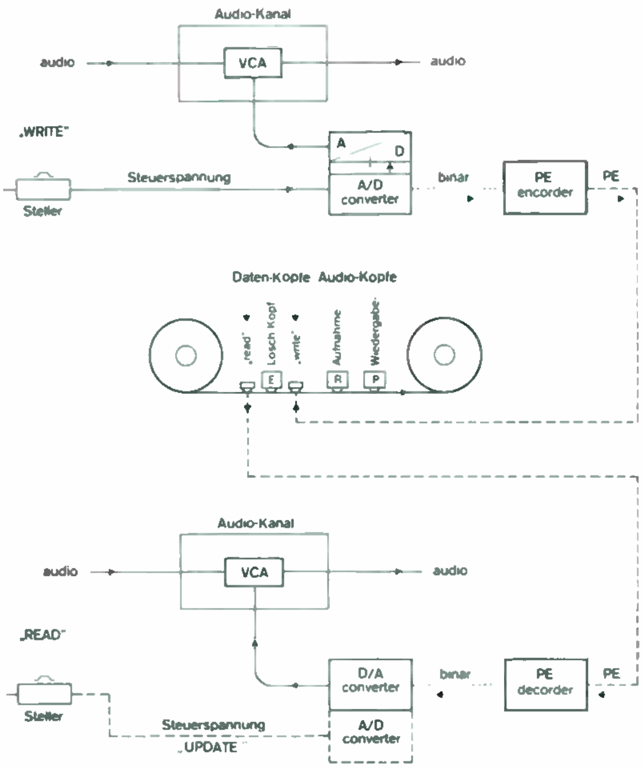
Planning engineers of the broadcasting organizations say that there is very little multi-track production work in the broadcasting houses. so that requirements can be covered within the existing module system. As for live broadcasting studios, they frankly confess that they feel much safer with the present easily-interchangeable modular system, so there would not seem to be much chance for commercial hardware to be introduced.

Well, does this mean a German Requiem for everybody who does not manufacture the one-and-only IRT-blessed "Panzer" module?

I do not think so. First, it is obvious that any equipment designed this way is many times more expensive than even the most sophisticated commercial console or equipment rack. Most broadcasting networks are no longer prepared to put inexhaustible funds in the hands of their engineers, since they do not have them. Second. the harsh technical principles of IRT regulations were broken through many times before—partly in situations where there was no other way, partly by engineers who wanted to get things going and had no interest in scientific achievements. And in truth, even IRT must not be considered as a bunch of close-to-impossible-sticking idealists. First, the technical authorities of IRT throughout the years have contributed decidingly to the high standard of German broadcasting engineering in both audio and video. Second, German radio and tv are public service companies and hence. bound to technical standards and security rules. In addition, any breakdown over here is regarded to be not just a technical accident, but an impossible failure of engineering. Third, IRT already has made some changes within its own framework and will continue to do so: anyway, a younger generation will come along with a broader look at things. closer to the practical reality of today's engineering.

A tremendous change has already taken place in the field of video at German tv stations. Whoever would have dared to mention firms like Grass Valley a couple of years ago? This person would have risked, if not his job, at least his reputation as a serious engineer. Today, just look around—control rooms are full of all kinds of formerly-

LDC StoreMix-System



System Diagramm

banned commercial equipment. It is just that the field of audio is so much older than video and therefore concessions to reality are more limited than in countries with commercial broadcasting systems.

No doubt, time works. Recently I was rather astonished when after many years of absence, I entered a broadcasting studio over here to do a couple of programs. I had carefully dubbed in all my recorded material to 7.5 in/sec. single or dual track stereo, knowing that for long years this was the minimum acceptable quality standard in German radio. Now, when entering the control room, I found an absolutely non-IRT-standard cassette (yes, cassette) recorder connected to the "Panzer" module standard console. And—oh, the shock!—they even dared to ask me whether I had any recorded material to be dubbed from cassette. You see, it's happening.

Going back to the exhibition, it is obvious that in view of the present "official situation", the smaller exhibited consoles—mostly British firms like Soundcraft, Allen & Heath, etc.—were received with not much more than a smile (at least officially). I found an apparently more-clever approach to the introduction of commercial consoles to German radio and tv at the stand of BFE KG., the largest independent German studio and system builders, in big business with all German broadcasting organizations and now dealing with commercial recording studios as well.

BFE displayed a modest-looking console built by its British partners "ZOOT HORN Audio Systems". This kind of console was first accepted only for high quality sound reinforcement systems. Later, they found acceptance as broadcast consoles and talks are now going on about a couple of studio consoles of the same system design. A closer look at the system and the modules reveals why: the width of modules is at least 40 mm, and all knobs and switches of lined up neatly and are easily visible. The knobs are shaped in such a way that even in a dimmed tv control room or van one can find out the exact position of each knob by a touch. Possibly another reason for acceptance was the high-quality Penny & Giles faders and —last but not least—the fact that all technical data are close enough to all corresponding IRT regulations.

All together, the experience to be drawn in my opinion is that playing it big will probably not smash the existing wall of "Panzer" module-techniques. The thing is not to start from the top but, from scratch, which means building up confidence in commercial gear little-by-little. However, do not expect any big deal on this potential market without making some concessions to what is used over here in size and layout. Finally, technical data are to be as close as possible to IRT regulations. For example, all line inputs and outputs, as well as all insert points, of + 6 dB level and balanced throughout. This makes it obvious that first chances are for upper-class equipment. As an example, the before-mentioned BFE KG. features the ECLIPSE C by Sphere in the class of larger consoles.

Digital delays and other gear for electronic effects were another item on which the *Tonmeisters* focused their interest. R. Barth KG. had the Ursa Major Space Station and BFE the DeltaLab DL-1 Digital Delay Module on show. Audio & Design Recording and Klark-Teknik were also well-represented. The surprise of this exhibition was presented by a hitherto-unknown manufacturer with the name of Ibemco, demonstrating a "Spectral Dynamic Processor (SDP)." According to the manufacturer, this is intended to be the break-through to "higher fidelity." With our present consoles we are in control of levels, equalizations and effects, which the inventors of SDP consider to be insufficient. What they say and what we all know is that musical instruments, when being played from pianissimo to fortissimo, do not only change loudness but timbre too: therefore they are of the opinion that the engineer should have a tool to control this effect.

As far back as 1929 a professor from Berlin (E. Schumann) published investigations on this subject, calling it "The Physics of Timbres". In 1975, H. P. Mertens published a book on the same subject. The title of the book is "Schumann's Laws Of Timbres And Its Signification for Transmission Of Speech And Music" (*"Die Schumannschen Klangfarbengesetz und ihre Bedeutung für die Ubertragung von Sprache und Musik,"* Verlag E. Boschinsky, Frankfurt am Main). For good reasons the book is a must for everybody in sound and music.

The SDP builds on Schumann's Laws and Mertens' investigations of correct dynamics corresponding to the natural changes of timbres. SDP therefore is supposed to enable the engineer to change the timbre of musical instruments within the range that such changes would take place at the instrument itself when being played in various dynamic ranges. This opens up the possibility to change timbres of a single instrument within an orchestra in a natural way, instead of using the heavy-handed equalization we are using now. Needless to say, none of the presently-existing equalizers would be able to comply with the requirements according to Schumann's Laws: only analog-digital-analog conversion can do that and it pretends to do it in the SDP. If not the way to real "higher fidelity", it is at least an idea to think about, and further developments should be watched. The demonstration at the Ibemco stand—by headphone only (speakers *verboten*—was not enough to decide whether the console of the future is unthinkable without SDP or anything like this.

All together, sniffing around in Berlin was interesting and opened some up-to-date views on the sound scene and the market over here, which hopefully I was able to reproduce well enough for the readers of **db.** ■

KENT R. DUNCAN

Report From Japan

CBS/Sony Recording Studios in Tokyo

Displaying a deep appreciation of music, and expertise in high-technology production, Japan is a natural environment for recording studio expansion.

TOKYO IS PERHAPS the only major city remaining in the world today which, in terms of recording studios, is still under-built. Japan has been rather late in following the trend towards the independent studio operation. In fact, until quite recently, all records were still being recorded in a handful of label-owned recording studios. In Tokyo, recording studio activity is hampered by the additional burden of very expensive, and hard-to-find, building space. Yet Tokyo is still *the* recording center in Asia, with only very small facilities in Taiwan, Korea, the Phillipines, Singapore, Malaysia and New Zealand. (Australia has an industry all its own which—interestingly enough—is patterned much more on the U.S. than the U.K., despite Commonwealth ties and the fact that EMI has the largest studio in Australia.)

At present, studio rates in Tokyo approach U.S. $300 per hour, which certainly bodes well for the recording studio construction business. In fact, Tokyo (and Australia) have mushrooming independent recording facilities, and while neither country has a really independent disc mastering studio yet, there are several in the planning stages.

Japan remains a world of contrasts, especially to this Western observer. A deep appreciation of music has been an important part of the rich culture for over four thousand years. The patience, politeness and attentiveness which the Japanese possess when dealing with music is, without question, the most admirable I have seen. With this background, and their expertise in high-technology production, one would imagine the Japanese would be producing their own professional recording hardware, especially considering their well-known involvement at the consumer level. But this is hardly the case. While there is one company, Denon, which makes two and four track recorders, all the other recording equipment comes from abroad. Denon equipment is found mostly in broadcast applications. However, MCI is now making a good headway with the broadcasters of Japan, just as it has done in U.S. radio.

TECHNICAL EXCELLENCE

With the exception of Germany and Switzerland, Japan is a more technically-oriented market than the rest of the world. Sales are conducted with less "hype," and minimal talk about what records were recorded where. The emphasis is solely on technical excellence, and the average Los Angeles promotion man would find it hard to cope with the endless questions on technology. When asked why a particular piece of equipment is preferred, the Japanese recording engineer seems to have *all* the answers, as to features, performance and service availability. In tape recorders, the "big three" seem to be MCI, Studer and Lyrec. There are also some 3M machines in use, and of course Ampex is very big in the video industry—second only to Sony.

STATE-OF-THE-ART FACILITY

On a personal note, Asia has always intrigued me. It's a special world all its own, and I must admit that—next to home—it is my "second love." To me, it seemed only natural to want to become involved doing business in this part of the world. So, when the people at CBS/Sony approached Tom Hidley and I, we were more than ready, and quite excited at the challenge they provided us: to build the first fully-automated, 24-track, state-of-the-art facility in Tokyo! The studio was to be a showcase for the Orient, and of course, a delightfully grand entry for Sierra into the Asian market.

From the first meeting with CBS to final contract signing

Kent R. Duncan is president of the Sierra Audio Corporation. Burbank, California.

was a very quick two months, punctuated by CBS's Koji Hazama making several trips to Europe to see equipment suppliers, confer with Tom Hidley, and finalize design details.

Due to concrete and framing problems, construction started three months late, but was still finished on time and within budget, thanks to the diligence and enthusiasm of our Japanese colleagues. Construction was handled in collaboration with a large independent Tokyo construction company, which has offices arounds the world and generally builds high-rise offices. For the project, we supplied six foremen, and forty-four Hidley TM-3 monitors, which were completely installed and interfaced by our crew.

The studio complex is quite unique for this part of the world, in that it offers such a large and complete facility. Twelve music rooms are completely self-contained, with beautiful decor which includes garden areas, view offices, a full restaurant (seating fifty-five) and lounge areas—all coordinated in as beautiful a fashion as any project we have been involved in. Features which we have not had an opportunity to include in past designs were specified by the Sony people, as they wanted their studios to be completely state-of-the-art.

SUPERIOR CRAFTMANSHIP

The interface work performed by the Japanese crew was completely unique. The craftsmanship on patch bays. connectors, a mic. matrix system to assign inputs, was superb. In every detail, the workmanship was absolutely amazing, and construction of the studios (after shell construction was finished) was completed in approximately eight weeks. Massive numbers of crewmen, working seven days a week, two shifts per day, descended on the job, in order to meet the deadline.

The studios vary acoustically by design from room to room. The large studio, which can accommodate ninety musicians, was designed for large groups up to symphonic orchestra size, but is intended to capitalize on music of the area. This studio was *not* designed for high-SPL electric bands, unlike two of the medium-sized studios, which cater to rock-and-roll.

LAYOUT

Five studios and an automated mix room are on one level. On a second level reside the editing rooms, mastering rooms and overdub/mix rooms. Five levels, including a basement parking area complete the facility, which is in the Shi-ochi district of Tokyo. The first recording session was recorded by Tom Hidley and myself, in September, 1978. We were immediately followed by David Rubinson, producing Herbie Hancock for CBS. (For more on David Rubinson, see "Automating The Automatt" in the November issue of **db**—Ed.)

Without question, recording studio expansion is on the rise in Tokyo, although the under-built condition will continue for some time. Certainly, there is a big demand for foreign talent and ideas. I have often thought that a good rhythm section from L.A., Atlanta or New York could go to Tokyo, record three sessions a day, and stay as long as they liked. Likewise, American and English technicians could certainly find their way to Tokyo and succeed if they were musically knowledgeable.

Multi-track recording in Japan is mostly 24-track. While I saw over a dozen 16-track machines, all of them have 24 on order or in their near-future expansion plans. I did not see any 32- or 48-track recorders. While on the one hand, 24-track might seem to be unnecessary now that MCI and Telefunken have introduced 32-track machines, there is still a distinct advantage to locking two 24-track machines. The advantage is not just more tracks, but the technique allows the engineer to lay down rhythm and instrumental overdubs on the first tape, and then sub-mix a cue track to the second machine, storing the original tape until it is time for mixdown. In the meantime, the thousand or so passes made on the second tape for lead and background vocals, strings and horns, need not cause deterioration of the transients on the basic rhythm tracks. When it comes to mixdown, locking the machines together is a simple affair, and can be done by many techniques, with a quality that is certainly worth the effort. There is also some safety when travelling, should the second tape become lost or damaged.

With all of these tracks, the question of the need for noise reduction comes up. Is it really necessary? With the better tapes, I think not.

After observing the growth of the Japanese recording industry at first hand, I've been asked what effect this growth will have on those U.S. studios that Japanese artists so readily patronize. Very little. While local Japanese artists can now stay at home and record, many have become accustomed to American innovation in recording techniques. The premier recording artists who have been coming to the U.S. to record will continue to do so, for the same reason that other major artists now readily traverse the globe to seek out their favorite studio or engineer.

CONTINUED GROWTH

As I mentioned earlier, people are still waiting in line for recording time in Japan, and I feel that with the additional facilities now becoming available, the Japanese music industry will continue to grow, topping all expectations and projections. In addition, American artists are beginning to plan more Japan tours, and their need for quality studios there should increase.

In recognition of the growing Japanese recording scene. Sierra's future planning calls for a full-time Tokyo office to handle Japanese and Asian business. For the past two years, we have been operating with one full-time employee, but the growth and close ties between the Japanese music business and L.A. musicians is one that will require Sierra to open a local office (and give me an even-better excuse for more visits to my "second love").

In the meantime, Japanese acts continue to travel throughout the United States in great numbers. By best count, some sixty major albums were recorded for the Japanese market, at American recording studios in Nashville, Los Angeles and Honolulu. Indeed, simply because of the high cost of the Japanese recording studios, two of the studios that we are building in the Philippines anticipate that fifty percent of their business will come from Japan. As noted before, recording time in Japan may cost $300 an hour. In Manilla, it's about $30 an hour. But you'll have to bring your own musicians.

OTHER STUDIOS

Presently, Sierra Audio is involved in another project in Tokyo which encompasses two studios. Being built for Kitty Music, an independent record label, the facility is a self-contained rock-and-roll studio located within Tokyo proper. We're also discussing the possibility of a resort-atmosphere studio which will require a two-hour ride from Tokyo on Japan's amazing bullet train. This should provide a fascinating contrast with other vacation atmosphere studios, such as the Hidley-designed Caribou (Colorado), Manor (England), Chateau (France), Mountain (Switzerland), LeGab (Mexico), Kendun (Hawaii) and ABBA (Sweden).

Japan is a cultural and musical experience that the serious music business professional cannot afford to ignore. A taste of the devotion, respect and appreciation of music that one finds in Japan, and in all of Asia, is a 'high" I would wish for all! ■

DEAN AUSTIN

Time-Aligned Loudspeaker Systems

The development of a new studio monitor system, based on Time-Align techniques.

WHEN ONE THINKS ABOUT BEGINNING a discussion of studio monitor systems, several good reasons for not doing so immediately come to mind. Foremost is the highly-subjective nature of evaluating loudspeaker system performance. Where else in the field of electro-acoustics are there such diverse—often contradictory—opinions on a subject? The very vocabulary is tenuous, at best. For example, where do you find universally-acceptable definitions of "punch," "roundness," "tight," or "transparency"? For that matter, what is the definition of "definition," as applied to monitor system performance?

Now, add to the semantic problem the often-overwhelming effect of the environment on the system. Into whose control room do we take all monitor systems for evaluation and comparison? Certainly, no one wants to listen to them in an anechoic or highly-reverberant environment; but just what do you pick in between these extremes? Also, what is a "standard signal source," acceptable to everyone?

Finally, we have to try to eliminate the witchcraft and misconceptions which abound in speaker system lore, and concentrate on solid, scientific, measureable parameters—if possible. In thinking of solid, scientific analyses, the works of A. N. Thiele and Richard H. Small stand out, almost as lone beacons in recent years, in the area of fundamental, scientific advancement of the state-of-the-art.

Now that I have acknowledged the hazards of the discussion, I nevertheless approach the subject with great enthusiasm; and especially that part of the subject dealing with the Time-Alignment system developed in conjunction with the UREI 813 Studio Monitor System. (Time-Align and its derivatives are trademarks of, and licensed by, E. M. Long Associates. The use of this terminology is licensed to UREI for use in connection with its products, and the trade-marked nature of references to Time-Alignment is hereafter implied.)

In this article, an attempt will be made to individually examine the various parameters which influenced the development of the 813 system, and show how each contributed to meeting the design objectives. This is easier said than done, because of the complex inter-relationship of the parameters. The design objectives may be categorized as follows:

1. Flat frequency response (40 Hz to 16 kHz).
2. High efficiency.
3. Correct damping.
4. Low distortion.
5. Time-Alignment of driver components.

Dean Austin is assistant to the president at UREI—United Recording Electronics Industries, Sun Valley, California.

The 813 system has been designed around the Altec 604 series loudspeaker. This "duplex" loudspeaker was selected by UREI president Bill Putnam because of its wide general acceptance as a studio monitor. His decision was an easy one to make: Why fight success?

Measurement of acoustic output, with a constant signal voltage applied to each section of a 604-E, showed a difference of several dB in the average efficiencies of the low- and high-frequency sections. Later, similar differences were measured on the 604-8G version of the speaker. For this reason, coupled with the relatively low cone compliance which results in a fall-off in response below 100 Hz, an additional low-frequency driver was incorporated in even the earliest experiments. Several systems were built, using various stock 15-inch low-frequency drivers. Some of these systems were installed in United/Western's studios in Hollywood, where they were well-accepted by clients. In use since 1976, they were replaced by the newer time-aligned system early last year.

Since these early systems displayed good efficiency and uniform response across the operating bandwidth of the low-frequency drivers, attention was now turned to the crossover region, and to the task of smoothing out the high-end response.

APPLYING TIME-ALIGN TECHNIQUES

In May, 1976, Ed Long of E. M. Long Associates presented his AES paper, "A Time-Align Technique for Loudspeaker System Design" (AES preprint 1131). The paper outlined his measurement techniques and findings, with respect to the time-alignment of multi-driver loudspeaker systems. Long's work impressed Bill Putnam, who invited him to collaborate with UREI in the development of a high-performance studio monitor system, that would take advantage of the Time-Align technique. Thus, the UREI 813 Time-Aligned Studio Monitor System began to take shape.

E. M. Long Associates is an independent acoustics design firm, located in Oakland, California. They have a free-field measuring facility, and a very-elaborate, computer-controlled acoustics laboratory. Soon after they were brought into the project, a series of new design parameters was proposed. These included the final volume and dimensions for a sub-woofer/tuned-port system. The criterion for the bass response remained that same subjective "tight" sound of the earlier monitors, but with even less distortion. The Time-Aligned network was designed around the 604-8G speaker, with a second network developed for the earlier 604-E speaker (UREI model 824).

DEFINING TIME-ALIGNMENT

It might be worthwhile at this point to give a definition of Time-Align. This is a real-time design method, utilizing proprietary instrumentation, which requires that the phase (time) relationships between the fundamental and the overtones of a complex, *transient*, acoustical signal presented to the listener will accurately match the electrical signal presented to the input terminals of the monitor system.

The realization of the Time-Align network for the model 813 required a combination of two major functions: the time delay necessary to align the acoustical positions of the low- and high-frequency transducer sections of the 604-8G, and the smooth blending of the acoustical output of these transducers.

Bringing the high- and low-frequency transducer sections into time-alignment significantly improves the smoothness of the amplitude *vs.* frequency response through the crossover region. This is because the filters are controlling the high-pass and low-pass functions of the acoustical output of the transducer sections at the same instant in time. This eliminates those additions and cancellations which will occur in the acoustical blending of staggered transducer outputs, where the resultant offset causes these outputs to be out-of-step in time and therefore, to have incorrect phase relationships. Of course, the filters used to divide the signal between the transducer sections also have their own time-delay characteristics, and this is taken into account. The actual final Time-Align network must be a synergy between amplitude control and time delay characteristics. This results in proper amplitude *vs.* time, and amplitude *vs.* frequency characteristics of the acoustical output of the 604-8G.

The UREI 813 Time-Aligned Studio Monitor System. Note that the sectoral horn has been replaced by a single-section horn.

POLARITY REVERSALS

Over the years, many users of duplex loudspeakers have inadvertently reversed the polarity of the high- or low-frequency driver, and have not been aware of the fact from listening to the speaker. There have even been some users who felt that some duplex systems sounded better when connected in reverse polarity to that recommended by the manufacturer. However, with the Time-Align crossover network, there can be no ambiguity about the correct polarity. With the proper connections to the transducer sections of the 604 drivers, the free-field response is flat within ±2 dB, throughout the frequency range from 800 Hz to 3,000 Hz. An accidental polarity reversal will create very audible dips in the frequency response.

It was determined that the performance through the crossover region could be further improved with a more-gentle slope over the crossover transition, which improved delay characteristics. This required lengthening the high frequency horn to lower its cut-off frequency.

Once that bridge was crossed, a whole new look was taken at the horn. Measured acoustic responses, using stock 604 sectoral horns, had shown narrow peaks and valleys in the response above 3 kHz, with amplitudes in excess of ±5 dB from the nominal level. Since the cross-sectional area of the sections in the mouth of a sectoral horn becomes an appreciable fraction of a wavelength at the higher audio frequencies, the obstacle presented to the sound wavefront passing through the horn creates reflections which affect the relative amplitude of the output at various frequencies.

DISPERSION VERSUS SMOOTHNESS

The reason for using the sectoral horn is, of course, to provide improved horizontal and vertical dispersion of the sound wave for a given size of horn mouth. But in a recording studio, the listeners are usually located close to the on-axis orientation of the monitor speakers, so it was decided to sacrifice some horizontal and vertical dispersion for the sake of smoothness of amplitude response. Therefore, a single exponential horn of the required length and mouth area was designed. It is about 0.9 inches longer than the sectoral horn that it replaces. The actual reduction in dispersion is approximately 10 percent along each axis.

Although there are unit-to-unit variations in high-frequency drivers, the results of having tested several hundred production 813 systems incorporating the new horn (UREI model 800H) and 838 crossover indicate that the response can be expected to measure within a ±2 dB envelope from 3 kHz to 16 kHz and beyond.

As progress on the crossover network and enclosure for the system progressed, it became obvious that no stock auxiliary low-frequency driver would meet the requirements of a very low cone resonance (the goal was to have the 3 dB-down point at 40 Hz or less, in a quarter-space mounting), and a smooth response up to the crossover frequency (in order to bring the system efficiency up to that of the 604's high-frequency section). Accordingly, Ed Long determined the design parameters for the auxiliary low-frequency driver, and designed a custom unit (UREI model 800W).

DESIGNING THE ENCLOSURE

The enclosure for the system presented some contradictory requirements. In experiments with earlier monitors, using stock low-frequency drivers, it had been established that a closed box, containing close to 40 percent volume of high- and low-density fiberglass acoustic damping material, would provide the damping desired with only a dB or so loss in efficiency. Special high-density material was obtained, which has exceptional absorption at low frequencies. The damping being sought was that which would give instruments such as the kick drum the "tight" sound that was mentioned earlier.

Translating "tight" into a test procedure resulted in exciting the system with a 50 Hz tone burst (4 cycles on and 8 cycles off) and verifying that the acoustic response decayed into the ambient noise in less than one-half cycle, after the gate was closed on the burst.

The dimensions of the new enclosure were obtained by applying a modification of conventional ported-system theory. The ratio of 1.25:1 was used in choosing each dimension, to avoid having any one acoustic path in the

The Altec 604-8G duplex speaker. Note the 2 x 3 sectoral horn in the center.

enclosure be a whole-number multiple of another. Very slight adjustments were made to accommodate standard structural material sizes. High-density, one-inch-thick particle board is used, with approximately 32 feet of 2-inch by 3-inch interior bracing to stiffen the cabinet.

It would have been much more convenient for many potential customers if the box could have been perhaps 30 percent smaller, but the physics just don't work out that way. Since the 813 went into production, research is continuing with smaller closed and vented boxes, possibly in conjunction with electronic equalization, but no smaller version—which will meet the performance requirements—is yet ready for the market.

In a completely sealed box, output distortion increases substantially (to more than 15 percent in tests of the 813 monitor without a port) when the system is driven hard enough to produce the high sound levels currently used in many control rooms (in excess of 110 dB SPL at a distance of one meter from the enclosure). The distortion is greatest at low frequencies because the amplitude of travel of the cones is inversely proportional to the frequency of the drive signal. At the outset of the 813 design, it had been determined that a ported enclosure could be designed which would retain the "tight" quality of closed box systems. The addition of a properly-tuned port (tuned to approximately 23 Hz) reduced the distortion by a factor of four, as compared to the closed box. A special sub-sonic acoustic filter is placed in the port opening, to limit excursions of the woofers at low frequencies. The combination of port and sub-sonic filter is called a "pressure-control aperture."

Design of network filters with the required electrical response and time delay characteristics was tricky, because of the inter-relationship of network impedance in the design formulas. Achieving the required delay tends to push the input impedance of the network down at the lower frequencies. It is for this reason that the 813 installation instructions recommend that the 4-ohm tap be used on amplifiers with selectable output impedance, even though the nominal impedance of the 813 system is 8 ohms across the audio frequency spectrum.

TEST PROCEDURES

A few details concerning the instrumentation and test set-up may be of interest. (Naturally, other UREI products were readily available.) A UREI X-Y plotter with sweep oscillator and warble generator were initially used for electrical frequency response measurements. Previous tests have demonstrated that acoustic measurements using a warbled sine-wave signal source give excellent correlation to measurements made using band-limited pink noise, or a Sonipulse (UREI model 100) send signal. An X-Y plotter system is thus an odds-on favorite, for convenience in making such measurements.

For acoustic response, the AKG C-451 microphone is more than adequate. During the early development of the 813, a Sonipulse was modified to provide a preamplified output from the C-451 supplied with the system. This output was then fed to the X-Y plotter.

Due to the fact that no anechoic chamber was readily available at UREI on a day-to-day basis, an ingenious substitute was devised by Bill Putnam. A conventional construction scaffold frame, 7 meters high, was set up with a crossbar-suspended pulley at the top. A test platform was suspended from a cable, which ran through the overhead pulley to a winch on the ground, which was used to raise and lower the test platform with the system under test. The test microphone was attached to the platform suspension system, and fixed at a distance of one meter above the face of the test cabinet. With this system, the test cabinet is suspended, facing upwards, approximately 5 meters above the ground. This eliminates the effect of reflections from other surfaces over the frequency band being measured, and provides a measurement of free-space response that is accurate within 0.5 dB down to 40 Hz. Comparison of measurements of a specific monitor system in the free-space set-up with those obtained in typical, well-designed studio control rooms provide the corrections to be applied to convert the free-space measurements to those of a quarter-space. Here, it must be emphasized that the 813 system provides a flat (within a 5 dB envelope) response down to 40 Hz *only* when mounted in a quarter-space configuration. This requires recessing the cabinet into a wall surface of the control room, and either mounting it close to the ceiling, or providing a baffle at the top edge of the cabinet to obtain the low-frequency loading necessary for such extended frequency response.

One last note regarding test procedures. There has been some criticism of square-wave testing of loudspeaker system response. Due to the fact that a square wave is composed of a fundamental sine wave plus odd harmonics, looking at the square-wave response of a system—especially with fundamental frequencies in the octave below the crossover frequency—gives an indication of the time coherency (or lack of it) in a loudspeaker system. And this brings us to the kernel of the nut. Just what effect does Time-Alignment, in and of itself, have on the overall monitor system performance?

At this point, we don't have the definitive answer to the question. And, it is important to remember that *all* the innovations which are incorporated in the design of a system (loudspeaker or otherwise) contribute to the total effect, and it is the combination of these that one responds to when listening. There have been a number of direct comparisons of time-aligned systems with other monitors in various acoustic environments, and the subjective reactions have been diverse, although always gratifying. One frequently-heard comment is that it is "less tiring" on those long, late-night sessions. (This may help support the theory that prolonged listening to phase distortion creates listener

The retrofit single-section exponential horn assembly.

fatigue—Ed.) Some users feel that individual instruments are better defined and easier to pick out of the combined sound. At UREI, we are convinced that Time-Alignment contributes to all of these attributes. It has also been our experience that, the longer someone listens to time-aligned monitors, the more perceptive he becomes of their unique characteristics.

EVALUATING TIME-ALIGNMENT

We did set out to determine for ourselves what Time-Alignment contributes, but that is easier said than done. It is not difficult to add or subtract delays of almost any magnitude to the excitation of either the high- or low-frequency components of the 813 system, but doing so mis-aligns the effects of the filters in the amplitude *vs.* frequency domain. Since these were carefully brought into alignment with the network design, peturbations in amplitude response will occur.

Tests were run in which the drive to the high- and low-frequency sections of the 604-8G was deliberately mis-aligned by the measured physical offset of the two sections (approximately 400 micro-seconds). The time delays were inserted so as to have minimal effect on amplitude response. Tapes, recorded live, were played in a studio control room over a single monitor system. The music contained percussion, vocal and string passages. The delay was switched in and out silently as many as ten times during any passage, and listeners were able to follow the changes and accurately identify the condition at any point in the sequence. Three types of subjective observations were noted when the system was properly time-aligned:

1. The voices were more "intimate," "natural" or "live-sounding."
2. The strings were more "transparent" or "well-defined."
3. The percussion sounds were "tight" or "better-rounded."

Somewhere within their range, virtually all musical instruments, including the voice, contain fundamentals below the crossover frequency of most loudspeaker systems. along with harmonics which fall above the crossover point. Therefore, it should be obvious that non-time-corrected crossover systems are—in some way—going to change the character of such sounds when they are reproduced. The burning question is, how much of what characteristics does our hearing detect? An analogy which may or may not apply has to do with the resolution capability of the eye. When a picture (being projected) is out-of-focus, the eye detects the condition, but really does not have much sensitivity as to the degree of out-of-focus condition, until an in-focus condition is approached. At this point, the normal eye has great powers to resolve the focus until it is very close to perfect. If the analogy holds, perhaps the ear may have resolution capabilities for time offsets that are very near to an exact time-aligned condition, and these may be considerably greater than has been presumed.

A statement which remains locked in my memory came in a conversation with Richard Small and Don Keele, both renowned loudspeaker system designers. at a seminar which was held in 1977. Dick said, as best memory can recall, that he was reluctant to discount any hypothesis regarding the human ear's ability to discern very slight differences in the character of sounds. "For example, think about the fact that, if you are quite well-acquainted with a person's voice, you will recognize that voice over the telephone, in a car, a noisy room, a reverberant church, or even in an almost-anechoic environment." In other words, our auditory system somehow sorts through all kinds of interference, to detect characteristics in sounds that may be composed of very small differences, from one source to another. Much research remains to be done to define the ear's sensitivity to such variations in amplitude. frequency content, phase, or what-have-you.

THE PERCEPTION OF TIME DELAY

What other evidence, pro or con, do we have on the subject? The bibliography of Mr. Long's AES paper is replete with references to the evaluation of time-delay effects in multi-element reproduction systems. Perhaps most-often quoted are the findings of John Hilliard. who states that delays of 10 milliseconds are definitely objectionable. However, more recent experiments have indicated that most people can determine delay effects on the order of 500 microseconds, with some individuals being able to perceive changes of two-thirds that amount. These experiments were conducted by Blauert and Laws, and reported in a paper published in the Journal of the Acoustical Society of America, entitled "Group Delay Distortions in Electro-acoustical Systems" (May 1978). (For a little more about the Blauert and Laws Criteria, see John Eargle's article on Studio Monitor Systems in last month's issue of **db**—Ed.)

Those who have reported on the subject have approached the task in a variety of ways, with an equal variety of instrumentation, and the conclusions do not always agree when subjective evaluations are presented. As mentioned earlier, one conclusion we can reaffirm as a result of work on the 813 project is that a variety of closely-interrelated physical phenomena affect the listener's perception of material reproduced by a loudspeaker system. It is, and will continue to be, a challenge to evaluate the effect of these interrelated phenomena on an individual basis. I guess this just reaffirms that, the more we learn about some things, the more we realize how little we really know.

All electrical (crossover) networks inherently contribute some delay to the signal applied to the input terminals. but in general, few attempts have previously been made to incorporate just the right amount of delay to align the two or more acoustic wave fronts so that they are in precise time-coincidence. Some networks have even introduced delays which increase the mis-alignment of the drivers. Those of us who were involved in the development of the 813 system are unanimously agreed that unique characteristics are contributed to a music reproduction system by bringing all elements of the system into Time-Alignment. The more erperience listeners have with time-coherent systems, the easier it is to perceive (and appreciate) these unique characteristics. The challenge is before all of us to prove or disprove those suppositions and opinions that have not yet yielded to hard scientific analysis. There will always be room for divergent personal opinion on the subjective aspects of sound reproduction. And that might even include the subject of Time-Alignment. ■

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db People/Places/Happenings

• Assuming worldwide responsibility for sales of Scully Professional Audio Recorders and Dictaphone Voice Communications Recording Systems, **Frank Santucci** was named international sales manager, for **Scully Recording Instruments.** Mr. Santucci formerly held the positions of marketing manager for **Orban Associates** and senior product manager at **Ampex Corporation.**

• Joining **Ferrofluidics Corporation,** Burlington, Massachusetts, as audio products manager, **Lou Melillo** is responsible for applications engineering and marketing the use of ferrofluids in loudspeakers. Previously, Mr. Melillo was chief engineer of audio systems at **Becker Electronics** in New York, designing loudspeaker systems.

• In a major realignment of his activities world-wide, **Tom Hidley** has moved his base of operation to the island of Kauai in the Hawaiian chain, while operating **Hidley Design Services** out of a Honolulu office. His exclusive representatives will continue to be **Kent Duncan** of **Sierra Audio** and **David Hawkins** of **Scenic Sounds,** London.

• Appointed president of **Studer ReVox America, Inc., Bruno Hochstrasser** will be responsible for all USA activities of both the Studer line of professional tape recorders, and the ReVox brand of audiophile components. A ten-year veteran of Studer, Mr. Hochstrasser will be based in Nashville, and assisted in sales and engineering support activities by **Tom Jenny** and **Heinz Schiess.**

• Joining **Warner Cable Corp.,** from **CBS, John A. Lack** has been appointed to the newly created post of executive vice president, programming and marketing. At CBS, Mr. Lack served as vice president of the CBS Radio division and general manager of WCBS Radio, in New York.

• Promoted to senior vice president, merchandising and advertising for **Radio Shack, Bernard S. Appel** will assume the additional responsibilities of supervising merchandising for Radio Shack's overseas operations, and for overseeing the firm's advertising departments. Mr. Appel was previously vice president, merchandising for Radio Shack.

• Named vice president of the audio division of **Sony Industries, Frank Leonardi** replaces **Gus Ishida,** who returned to Tokyo to assume a position with the International Division of Sony Corporation. Mr. Leonardi, based in New York, joined Sony's audio division last year as marketing manager for hi-fi products.

• **Elliot Schwartz** has been named director of sales for **KLH Research and Development Corporation.** Mr. Schwartz was formerly national sales manager for **Bose Corporation** and director of sales for **Teledyne Acoustic Research.**

• In the areas of marketing development and corporate communications, **Altec Lansing** has appointed three vice presidents and one director. **Robert T. Davis,** formerly director of systems/applications engineering, was named vice president of professional market development. Promoted from director of product development, **Irwin Zucker** was named vice president of consumer market development. Moving from his position of corporate director of industrial relations, **Chris Christianson** has been appointed vice president of industrial relations. Named director of marketing communications was **Curtis Pickelle,** previously manager of that department.

• **Julian J. Trivers,** president of **Trans World Promotional Services, Inc.,** and director of advertising and sales promotion world-wide for both **Stanton Magnetics** and **Pickering and Co.,** has passed away. Prior to his association with Stanton Magnetics and Pickering and Co., Mr. Trivers operated **J. N. Trivers & Co., Inc.** as a merchandising marketing and sales promotion consultant.

In memory of the late Mr. Trivers, the **National Radio Broadcasters Association** Board of Directors has voted to establish the **Julian J. Trivers NRBA Internship Program.** The internship program will be funded by the NRBA on a continuing basis.

• Assuming responsibility for the customer service and technical training departments, **Donald E. Mereen** has been named marketing director, broadcast & professional audio products group at **Telex Communications, Inc.** Mr. Mereen joined Telex in 1974 and has held positions in sales and new product development.

• **Jim Drummond,** of Atlanta, Georgia, has been appointed the new southeast sales engineer for **IGM**—broadcast equipment and automation products. Mr. Drummond was formerly manager of the southeast region for **Videomax.**

• **Soundesigns,** previously owned by **Mediasound, Inc.,** has been sold. In a reorganization, **The Audio Group, Inc.** was formed, providing a unique full service facility—selling new equipment, as well as creating a new idea for audio brokerage. The Audio Group, Inc., is located at 1780 Broadway, Suite 800, New York, NY 10019. (212) 765-7790.

• The **Community Recording Services (CRS)** is an association of individuals who make their careers in the music and recording industries. The Society offers its members access to their co-operative, state-of-the-art, multi-track recording studio, in addition to providing for the exchange of educational, technical and artistic ideas and concepts through its seminars and conferences. Membership is open to anyone actively involved in the music and recording industry, and details can be obtained by writing: **Community Recording Services, P.O. Box 4672, Vancouver, B.C. V6B 4A1 Canada.**

• Named senior editor of **Popular Electronics, Hal Rogers** joins the magazine from **High Fidelity** where he was associate audio/video editor. Previous to that, Mr. Rodgers was science editor at **Funk & Wagnalls Company.**

• **John Phelan** has been appointed to the newly created position of general manager for **Filmways Audio Services.** Mr. Phelan previously served as manager, professional sound products at **Shure Brothers.**

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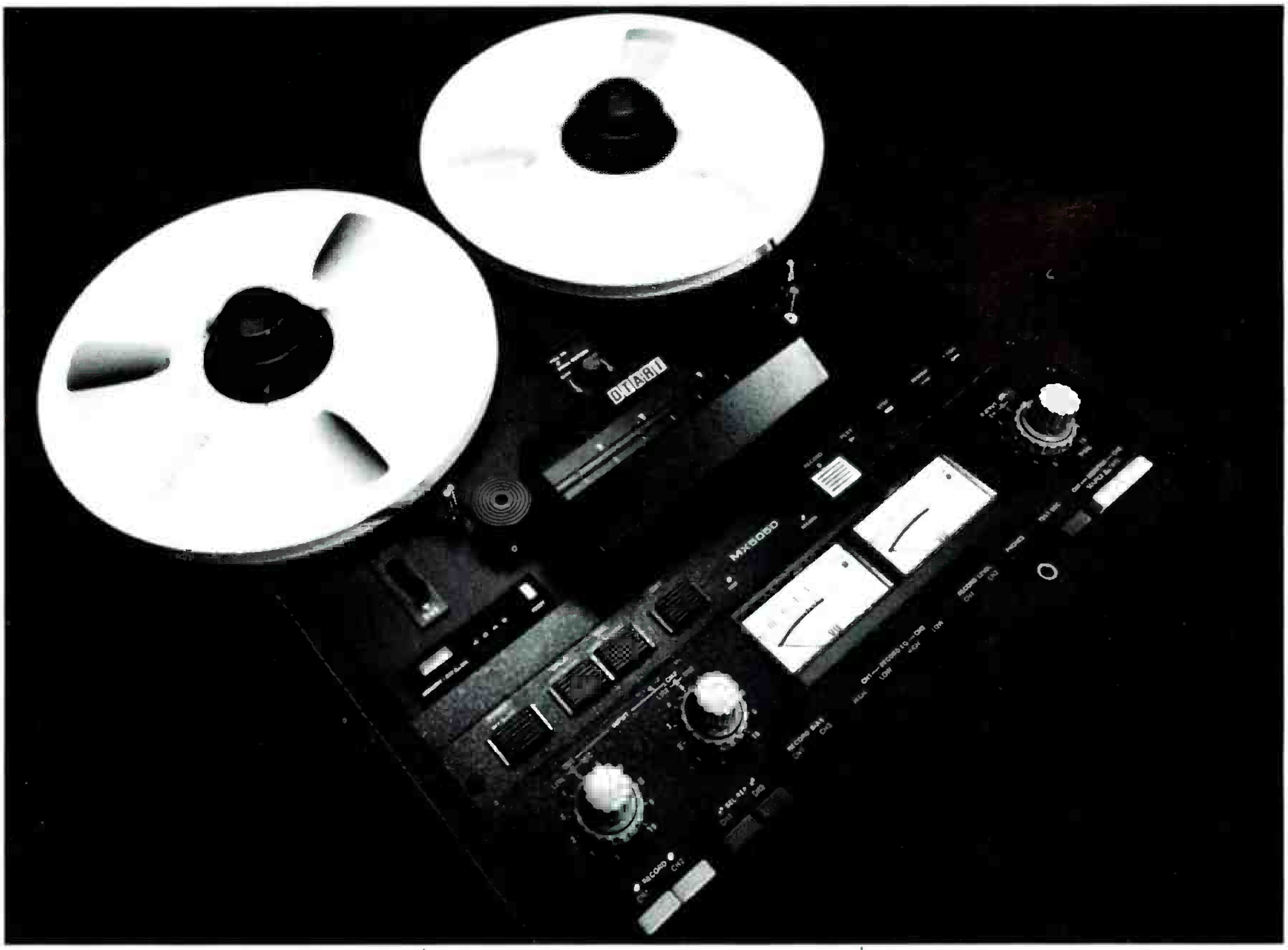

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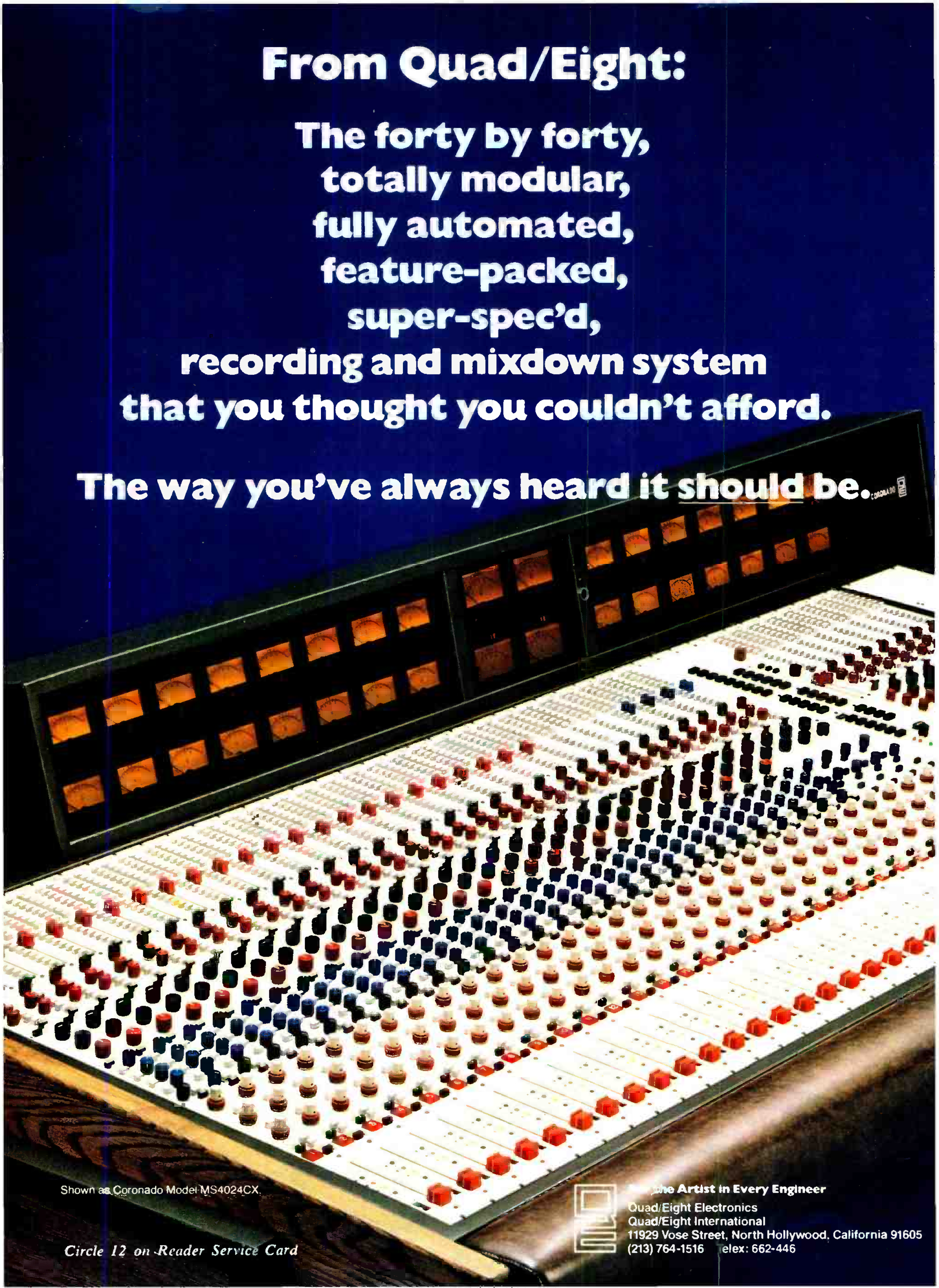
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