THE SOUND ENGINEERING MAGAZINE

FEBRUARY 1981
$1.95

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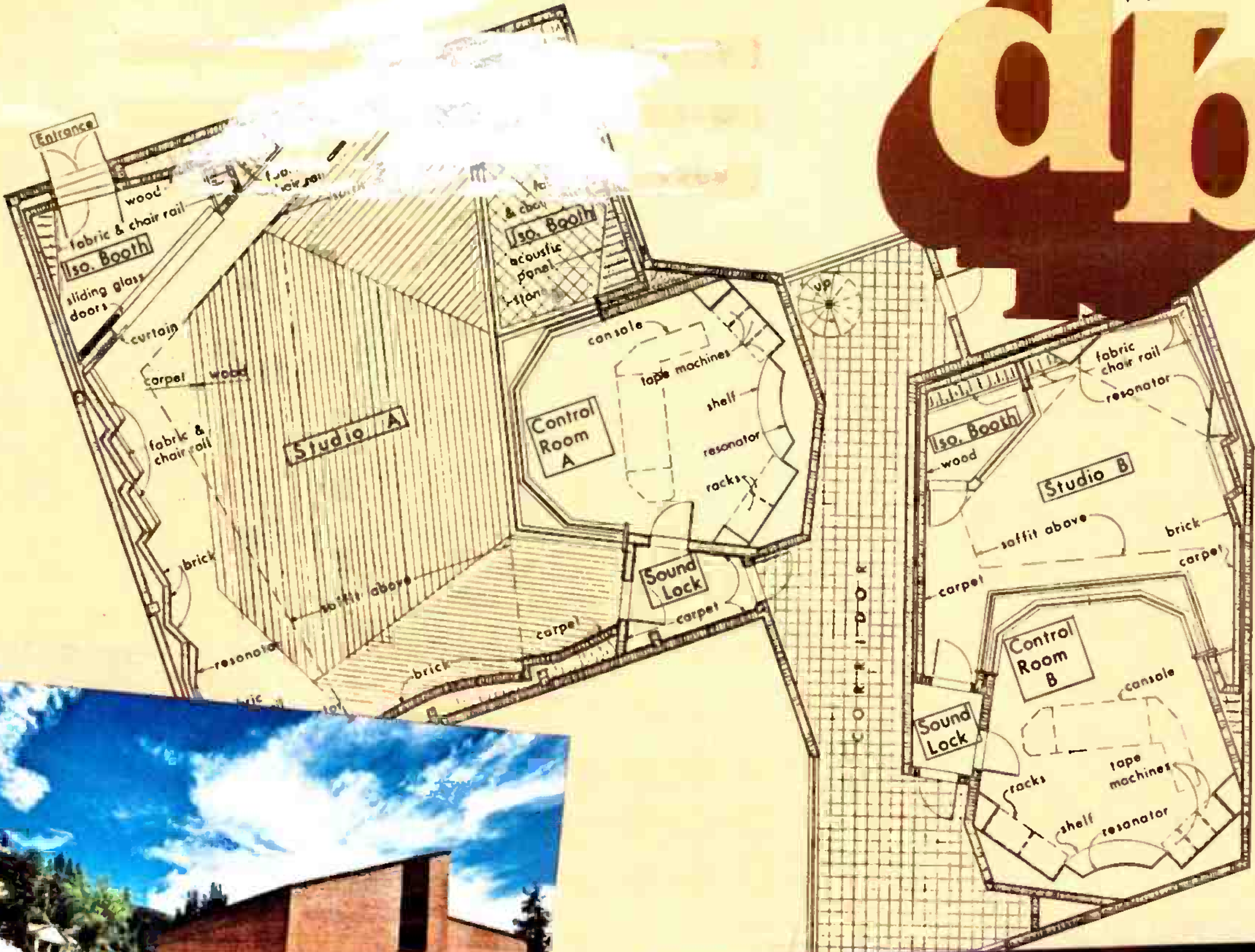

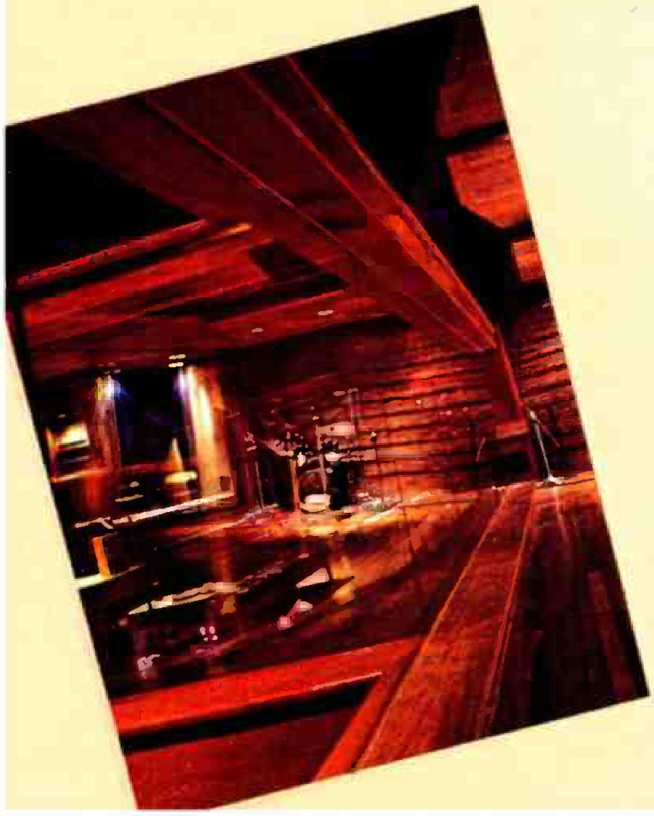

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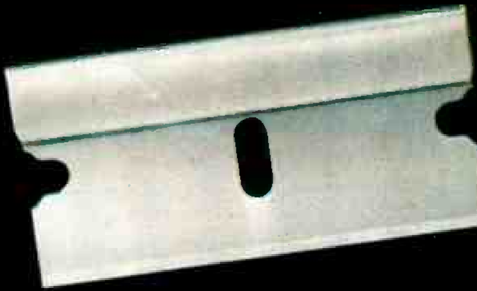

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ABOUT THE COVER—

• This month's cover takes us to South America, for several views of *Fonovision Internacional* in Bogota, Colombia. The studio complex was designed — and photographed — by architect John Storyk. For more details, see Howard Sherman's feature story in this issue.

THE SOUND ENGINEERING MAGAZINE

FEBRUARY 1981 VOLUME 15, NUMBER 2

EDITORIAL	24
FONOVISION INTERNACIONAL— COLOMBIA'S FIRST WORLD CLASS STUDIO Howard Sherman	26
THE DESIGN OF THE HITOKUCHI-ZAKA RECORDING STUDIOS Yasushi Ono	30
RCA MEXICANA STUDIOS John Woram	34
THE AMPEX SSL INTERFACE Normand L. Major	38
A STUDIO FOR RESEARCH Tom Lubin	41
SCENE FROM EUROPE: APRS DIGITAL GET-TOGETHER John Borwick	46
THE BOSE® 802 PROFESSIONAL LOUDSPEAKER Roy Komack and Brian Moriarty	48
LETTERS	6
DIGITAL AUDIO Barry Blesser	10
SOUND WITH IMAGES N. I. Weinstock	12
THEORY AND PRACTICE Norman H. Crowhurst	18
NEW PRODUCTS AND SERVICES	21
NEW LITERATURE	27
CLASSIFIED	53
PEOPLE, PLACES, HAPPENINGS	56

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db Letters

FILM-VERSUS-TAPE

TO THE EDITOR:

It is encouraging to see your magazine giving increasing coverage to the film and video fields of audio production. However, Mr. Weinstock's article "Film vs. Video" (Sept. 1980) continues to promote misinformation about film sound processes and equipment.

He describes optical sound as, "This capacitance system has severely limited dynamic range (about 500 to 14,000 Hz)." The physics of current solar cell technology is hardly capacitive, and moreover, the low frequency response of optical recording extends to DC, limited only by filtering in the record or playback electronics.

As to the "horrors" of magnetic film stock, a modern film recorder enjoys excellent frequency response, low wow and flutter, lower print-through, and increased dynamic range over standard tape formats. Ask anyone using a tape backup for live 35 mm. recording as to which folds up first.

Mr. Weinstock would have us believe that film technology has hit its peak while video's horizon is limitless. Might I suggest that the film medium is limited only to the imagination of the industry itself, whereas video, locked into the 525 line format and a maze of government bureaucracy, is quickly using up its technological headroom.

ROBERT M. BUDD
Post-Production Consultant
Kagel Sound Company

Mr. Weinstock replies:

Mr. Budd has properly corrected me on his second point—indeed, I am horrified at my own sloppy writing, which gives the impression that mag stock recordings are inferior to tape with standard backing. However, there is a "horror" involved in the use of magnetic film stock, and this pertains also to Mr. Budd's other two points as well.

While (point 1) current solar cell technology surely produces a very high quality recording, the available technology is not widely in use now, nor is it likely to be for at least a few years, if at all. For the time being, capacitance optical sound is the standard optical sound. If one is to produce a film with optical track for any sort of distribution, that track must use the old system.

Index of Advertisers

AKG 27
Ampex .... Cover IV
Auditronics 16, 17
Banff Center 43
Bose 9
Dolby 21
Elar Publishing Co. 15
Electro-Voice 23
Inovonics 14
JBL 13
Keith Monks 40
Lexicon 35
Mitsubishi .... Cover II
Neal Ferrograph 19
Neptune 20
Orban 37
Otari 31
Polyline 18
Pro Audio Seattle 10
Quad-Eight 25
R. K. Morrison Illust. Mats 6
Sescom 6
Shure Brothers 7
SME 12
Standard Tape Lab 43
Swintek 18
Telex Turner 8
The Mike Shop 14
3M Magnetic Tape .... Cover III
White Instruments 22
Yamaha 29

Coming Next Month

- In March, we present the first installment of our two-part series on Music in the Studio. Part I is on Electronic Music, which means anything from vocoders to synthesizers to metric music.
- Coming next month, in **db—The Sound Engineering Magazine.**

As to the part-legislated, part-technologically-dictated pictoral inferiority of television and video to film, I wholeheartedly agree. And, as I said recently in another magazine, all of the microprocessor-controlled advances contributing to the rapid improvement and greater availability of video can also be deployed with motion picture film. The problem with film is its lack of adaptability to what will surely be the distribution systems of the future, and its escalating expense (relative to video) due to mechchanical processes and non-recyclable materials. As a filmmaker, I too prefer the possibilities of that medium. As a writer, writing for other sound recordists, it's my obligation to state clearly just where most of our bread is buttered. At the same time, as quality must be championed, it must be recognized that film has an excess of quality that is in the process of being traded off for video's convenience, relative permanence, and adaptability to new uses.

To my mind, Godard says it succinctly in his current, Every Man For Himself: *"Cinema et video... Cain et Abel."*

db Calendar

MARCH

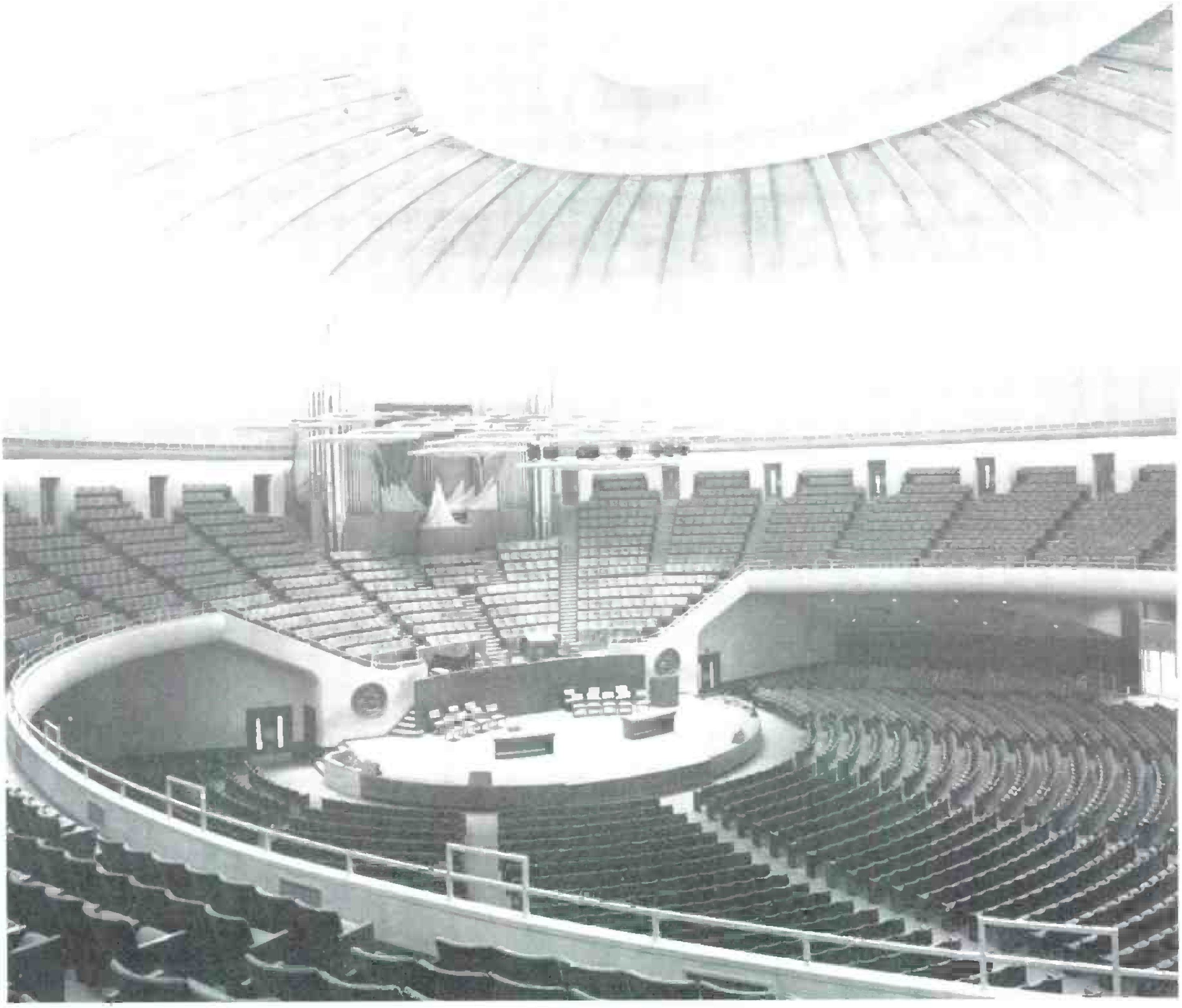
6-8 **Video and Lighting Workshop.** Pick Congress Hotel, Chicago, IL. For more information contact: Bill Ludwig, Victor Duncan, Inc., 2659 Fondren, Dallas, TX 75206. Tel: (214) 369-1165.

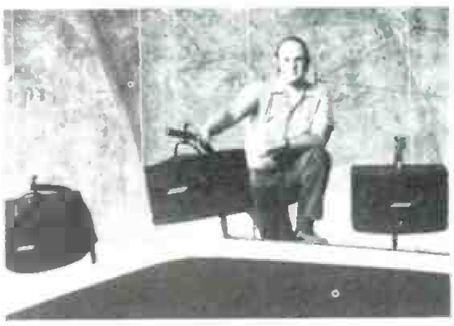
17-20 **AES 68th Convention.** Congress Center, Hamburg, Germany. For more information contact: Audio Engineering Society, Inc., 60 E. 42nd St., Rm. 2520, New York, NY 10165. Tel: (212) 661-8528.

APRIL

6-9 **National Noise and Vibration Control Conference and Exhibition.** Hyatt Regency O'Hare, Chicago, IL. For more information contact: NOISEXPO, 27101 East Oviatt Rd., Bay Village, OH 44140. Tel: (216) 835-0101.

25 **1981 Midwest Acoustics Conference.** Hermann Hall, Illinois Institute of Technology, Chicago, IL.





BARRY BLESSER

Perception of Quantization Error

• Up to this point in our presentation, we have discussed two different types of quantization error: inherent theoretical error as a result of a finite number of bits, and component defects which move the quantization levels from their theoretical values. However, we have not said much about the perception of these errors. This is a difficult topic, but it is an important one.

Since we are talking about a non-continuous, non-linear transfer curve which relates the analog voltage and the digital word, one would think that the manifestations of these errors is equivalent to distortion. While this is sometimes the case, a noise model is usually more accurate. In this article, we will examine the errors in terms of an equivalent additive noise and, in terms of distortion. Both of these terms will be treated as if they were analog errors. In other words, we will use these words for perceptual descriptions as well as for mathematical descriptions. When we say quantization noise, we mean that the error sounds like the same additive white noise which would be heard from an analog pre-amplifier. And when we say distortion, we will mean the kind of error which might come from a poorly designed output stage with cross-over distortion. The distinction between them is important since one is independent of the signal and the other produces harmonics which are related to the signal.

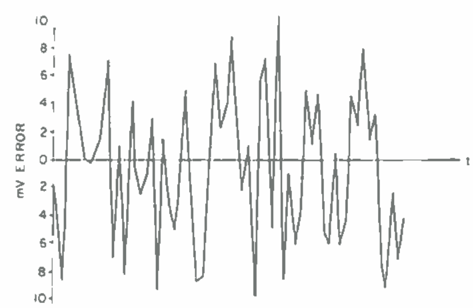
QUANTIZATION NOISE

For high level complex audio signals we will show that the quantization errors of both types are random. To do so, let us first consider a specific example using a 10 bit DAC in an A/D encoder having quantization levels at -30 mV, -10 mV, +10 mV, +30 mV, +50 mV, etc. The full range is from -10.14 volts to +10.24 volts. For our signal, we will take the summation of three sine waves at 1 kHz, 1.4 kHz, and 2.2 kHz, each with an amplitude of 1 volt. The sampling rate will be at 40 kHz. The first column is the time at which the summation happened (scaled by 25 μ sec.), the second column is the true input, the third column, the digital representation with a perfect converter, the 4th column is the quantization error.

Time	Input	Digital	Quantization Error
0	0	0.0	0.0
1	.0126	.02	+.0074
2	.0252	.02	-.0052
3	.0378	.04	+.0022
4	.0504	.06	+.0096
5	.0630	.06	.0030
6	.0757	.08	+.0043
7	.0883	.08	-.0083
8	.1008	.10	-.0008
9	.1134	.12	+.0066

Notice that the quantization error shows no pattern from sample to sample. The value is, of course, limited to ±10 mv, since we defined this converter to be accurate. The ±10 mV corresponds to the ±0.5 LSB error of a perfect converter. The input signal clearly has a pattern, since it is smoothly increasing and band-limited. The reason that an input with a pattern does not generate an error with a pattern can be understood as follows.

Figure 1. Quantization error for 10[sin(ωt) + sin(1.4ωt + sin(2.2ωt)] in an A/D encoder having as LSB = 20 mV sampled at 5 kHz, f = 1 kHz. Error is first 65 samples from t = 0.



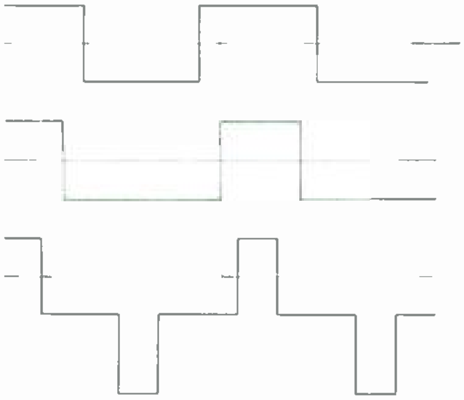


Figure 2. Digital representation of a 15 mV peak sinewave in a digital system having an LSB of 20 mV. These three examples are for different values of DC, which shift the location of the sinewave's 0 volts relative to the location of the quantization levels.

The input pattern allows us to predict the next sample very accurately. However, very small inaccuracies in predicting the next sample mean that the value relative to a quantization level is not predictable. If one looks at an oscilloscope one can almost predict the follow waveform exactly over a 25μ sec span; however, to predict it to within a milli-volt is not possible. Hence, the quantization error becomes randomized for most audio signals. To further demonstrate this fact, the actual error signal is plotted in FIGURE 1. This error is the difference between the digital representation of the complex audio in a similar example, and the actual output. Not only does it look random, it *is* random. If one were to listen to it acoustically, it would sound like the hissing of white noise. It has equal energy at all frequencies. The only difference between it and real analog noise is that the range of values is more limited for a constant RMS value. All values of the noise are equally likely between the + and − peaks. Analog noise is gaussian distributed. This means that its peaks are much higher than its RMS value.

We could demonstrate that all amplitudes are equally probable by simply keeping a record of the numbers which appear in column 4 of the table. To do so we might consider 20 possible error values in steps of 1 mV. We ask the questions: what percentage of errors are between −10 mV and −9 mV (bin 1)?, what percentage of errors are between −9 mV and −8 mV (bin 2)?, etc. The result is called a histogram or distribution of frequencies.

Before you get too comfortable with the friendly notion of quantization noise, notice that the previous discussion was always qualified with the phrase, high level complex audio signals.

QUANTIZATION DISTORTION

We are now ready to consider the other extreme case using a very low level simple sinewave. Suppose we have a sinewave with a 15 mV peak amplitude and a DC value of 10 mV. This signal only spans one quantization level. For positive values of the sinewave, the signal is between 10 mV and 35 mV, corresponding to the quantization region of 20 mV. For negative values of the sinewave, the signal is between 10 mV and −5 mV, corresponding to the quantization region of 0 mV. The digital result is a square wave! Representing a sinewave by a squarewave is rather poor in terms of harmonic distortion. Our signal is very badly distorted. The quantization error is now best represented by the hard clipping of a diode limiter. This error is not so friendly.

There are other consequences of this kind of distortion. Let us assume that the DC value were to slowly change. The resulting output would not be a squarewave but a rectangular wave with a change in the duty cycle. A further change in the DC would result in another shaped waveform. FIGURE 2 shows several possible digital outputs from the encoding of the 15 mV sinewave for different values of the DC component. Instead of DC changes, we might have a very low frequency component which acted as if it were a slowly changing DC. In this case, our sinewave's distortion would change its character as it was moved through the quantization levels.

We can furthermore consider the case where the sinewave had been only 5 mV peak. Here there would be some values of the DC which resulted in the sinewave being between two quantization levels. Since it never crossed a quantization level, there would be no output. Suddenly, all the signal would stop. A shift in our DC would then result in the signal reappearing.

Needless to say, this kind of distortion sounds very ugly and it has no analog counterpart; it is a uniquely digital distortion. Those of us who work with digital audio systems call this kind of distortion *granulation* noise, since it sounds like playing a record made of sandpaper. The difference between the nice white noise which we discussed previously and this ugly granulation noise comes from the difference in the input signal. In the former we assumed high level complex audio, in the latter we assumed a low level simple sinewave.

Because the granulation noise is so unpleasant, the designer of an A/D encoding system must insure that the signal is complex. One way to do this is to inject some analog white noise into the signal before the A/D encoding. Now the encoder is always seeing a complex signal. It turns out that a small amount of real analog noise is usually enough to transform the granulation noise to white noise. The digitized signal is thus the sum of the analog noise, the quantization noise, and the real music. Generally, the analog noise need only have a value which is approximately 1 LSB.

This technique is sometimes referred to as dither. This is a term from the old radar servo-mechanism days when the backlash from gears was a problem. The dither makes the quantizer, or gears, jiggle around so that it appears as if the quantization levels are never exactly at the same place. As a consequence, there is an averaging process which can smooth out the effect of sharp quantization levels as referred to the signal. It turns out that the mathematics of this technique is very complex, but the hardware implementation is very simple.

If you wish to test a piece of digital audio equipment for this effect, just place a low level mid-frequency sinewave into it and listen to the output. Being low level it will, of course, be noisy, but it should sound undistorted. Similarly, low level music should sound clean, albeit noisy. Because this issue was only discovered in the early 1970s, some of the early digital audio equipment had significant granulation noise.

DAC ERRORS

Both types of quantization error just discussed are true for perfect A/D encoding. We now need to consider the errors introduced by defects in the actual DAC device. First, let us divide those errors into two classes: random and systematic. The random errors are those which have no pattern. The systematic errors are those which generally occur at the major carry points, as we mentioned in the last article.

It should be apparent that the random errors result in the same type of quantization noise produced by the theoretical quantization. The systematic errors, on the other hand, are much more like the granulation noise. As an example, consider a DAC component with a systematic 0.5 LSB error at the major carry (0 volts). We represent each of the positive quantization levels as being 0.5 LSBs too low. Since the error is a function of the input signal, we can consider the output from the encoding to be the original sinewave plus a small square wave of the same frequency. The square wave's amplitude is fixed, since it is only dependent on the sign of the input, and not the amplitude. The square wave amplitude is determined by the DAC's error. This effect is exactly like the granulation noise, except that it happens at only one quantization level, and not at all levels.

If you want to continue to test your digital audio equipment, place a low level sinewave at mid frequencies, with an extremely low frequency signal. Take 5 Hz as the ultra-low and 1 kHz as the test signal. The 5 Hz is only used to sweep the test signal over enough of the DAC range to find the DAC's 0-volt major carry. Typically, the designer will add some extra DC just to prevent the rest condition from being the DAC's 0 volt. We use the 5 Hz to remove this DC. Have fun! ■

N. I. WEINSTOCK

Film vs. Video, Continued

• A letter to the editor in this month's **db** sparks the direction of this column, which is more-or-less a continuation of my reply to the writer. The "films" I presently find myself making are being shot on film, but then they are transferred to videotape. This is a common compromise between production quality, expense, and distribution. I don't have to enumerate the reasons for this—especially for anyone brawny enough to lift a professional-quality camera in one hand and a film camera in the other—nor for anyone who has compared their "rental weights."

There are differences in the "look" of film and video. In the viewing room at production's end, one sees a film with greatly different effect than it will have later, when transferred to video. This column will concentrate on the different miking, mixing, and visual processes that must be employed, in making film for video.

If one's distribution system involves professional format audio tapes, played over full-range sound systems into high quality monitors, production concerns are quite conventional: bring back the highest-quality sound, period. Unfortunately, the realities of playback are often different. And, even in the best of audio playback situations, two visual situations must be considered.

First, unless the audience is to see the film on a projection screen, the standard film vocabulary must be condensed to close-ups. Fortunately, (or unfortunately, depending on how you look at it) most good (video) cinematographers are entirely familiar with this style, and some are familiar with no other.

Second is the more-diffficult situation—similar to dynamic range in audio—of lighting latitude. On film, one can get away with a very high-key (strong primary) lighting source and little or no secondary fill. The film's total latitude of white-to-black may be as great as a hundred-to-one in black and white (6½ f-stops). In color, new stocks are bringing latitudes of eight- or sixteen-to-one (3 or 4 f-stops) to situations of very low total light level. By contrast, the latitude of video is from two-to-one, to four-to-one. In order to record anything at all in video, the minimum light level (measured in foot-candles) is much higher than it is in film, and that complicates things even more.

The difference in foot-candles needed to adequately light a scene is not absolute, because there are higher-and-lower-quality video cameras, recorders and tape, and because there are different film stocks in different formats that may be employed on different occasions. Minimum light needed to shoot video may be roughly gauged by the number of horizontal lines a camera is sensitive enough to attain. Manufacturers usually quote this statistic for optimum lighting conditions only, but the number of lines, or resolution, decreases directly as the amount of light does. The best small home-video camera can register 300 lines under optimum conditions, and needs at least 7 foot-candles to shoot at all. (The professional line standard is of course 525, and even that's not too good. A foot-candle is a measure of the amount of light intensity from one "international candle" falling on a one-square-foot surface at a distance of one foot.)

The foot-candle intensity needed to illuminate a film stock is usually referred to as that film's speed, and is given a rating by the American Standards Association (ASA) or in Europe the *Deutsche Industrie Norm* (DIN). As might be expected, there is a constant relationship between foot-candles, ASA rating, and all the other variables of exposure—which happen to be aperture setting, or

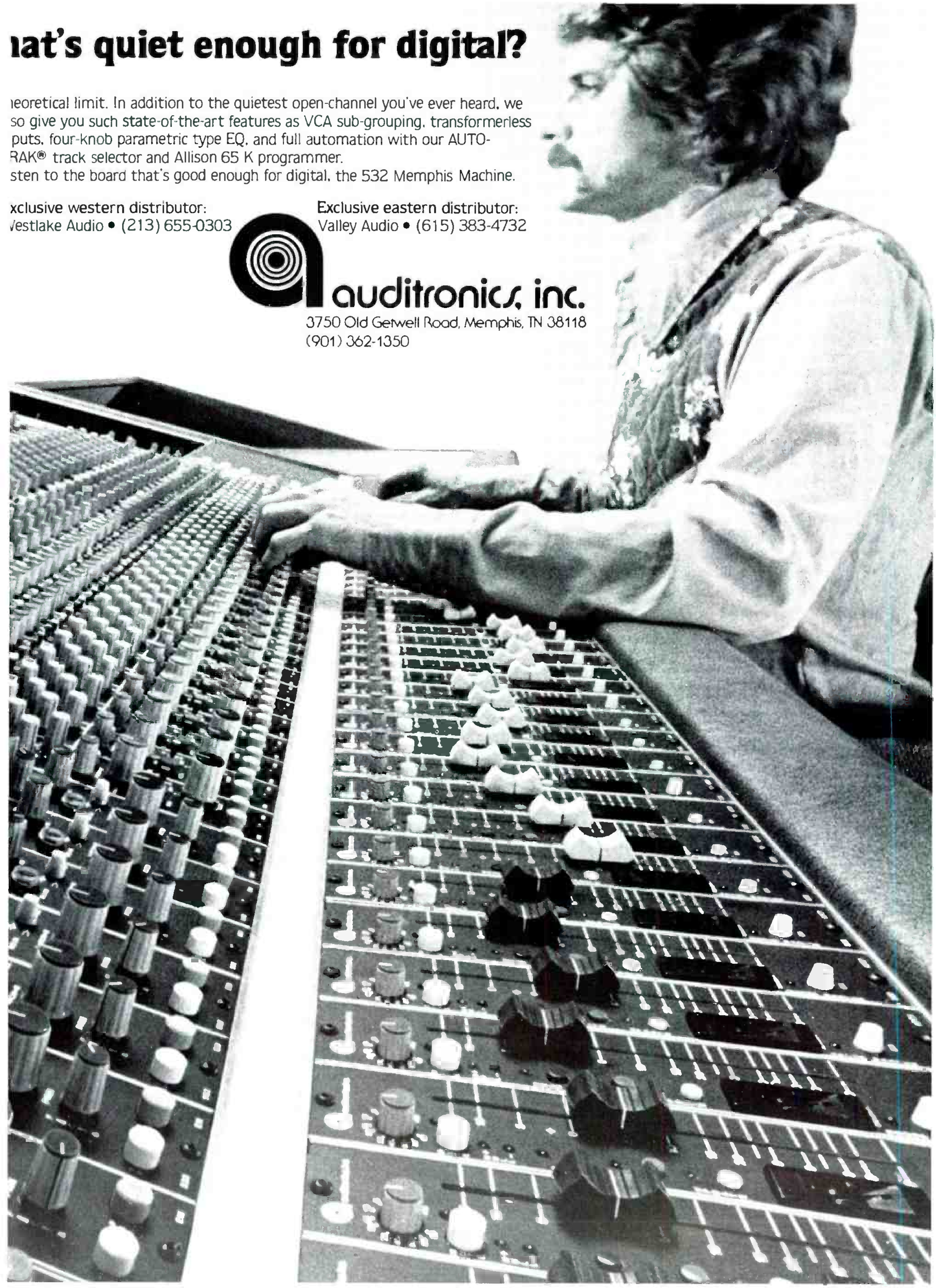
f-stop, and shutter speed (rarely a variable though: sound-film runs at twenty-four frames-per-second, and the shutter is open on each frame for half of that time).

But why go into all this? We come down to a need to shoot a film differently if it is to be seen on television or some other video system. Fast films must be generally avoided, for their graininess will add to video's lack of resolution. Slow films give the best contrast and light-to-dark ratios, but contrast must be kept to a minimum: visual dynamic compression, if you will. But the only way to dynamically compress a film is to do so *before* the scene enters the camera. Otherwise, you must employ some very tricky optical effects (priced so as to be a last resort) for cleaning up mistakes. (We hereby promise optical effects for a future column). The subject to be filmed, and the way that it is to be lit, must be chosen so that foot-candle levels are even and flat: bright, but not too-bright, and fairly boring. There are no Sternbergs of TV lighting.

Why then put something on video, if not for the quality of its image? Remember that television is also talk, talk, talk, and the reason becomes clear; the medium is video-plus-audio, of if you like, Sound with Images. However, when we switch to the audio side of the equation, we find that, limited as the dynamics and frequency response of motion picture film is, video is even worse. Even back in the early days of optical sound, film-sound frequency response had an advantage of a good couple-of-thousand cycles (they weren't called hertz then) over today's TV response. As for dynamics, under optimum conditions a video show *can* be accompanied by some very high-quality sound. But this would seem to call for a big room, so why not project film and get the higher-quality image as well? No, video is valued by the viewer expressly because it *is* small and it is limited. There's no projector to set up, and the room doesn't have to be darkened in order to watch. Therefore, the audio portion must also be limited—or be expected to be limited. So be prepared for the inevitable: the ubiquitous tiny TV speaker, with its tinny-TV audio.

Just as video film style must be limited to the close-up, so must audio style be limited—not only in range, but to one or two sound-sources at a time. As brilliant as it seems in the booth, it may be unintelligible in the conference room or on the home screen.

So with all of this, how can the honest (audio) recordist's soul accept video as an entertainment medium? Well, the rosy future of video discussed in previous columns does seem inevitable, as does much technical improvement within the medium. On this one can bank one's hopes and look forward without pessimism. That does not preclude shedding a tear at the wholly unnecessary demise of film. ■



NORMAN H. CROWHURST

db Theory & Practice

Developing Creativity

• Did you think about that question we asked at the end of last month's T & P column, about getting creativity going again, to re-establish American "know-how?" The problem presented reflects itself in engineering, perhaps as much as in any vocation. In one sense, progress tends to discourage creativity, but does it have to be that way?

In our own youth, electronics used tubes, the most advanced of which were triodes and pentodes. They were pretty basic elements, but they were fairly costly. Designing something that used them required that economy be carefully practiced: never use two tubes where the job could be done with one.

Today, micro-circuits have replaced tubes, but before that, transistors replaced tubes more directly, practically on a one-to-one basis. But even the advent of transistors had what might be termed a liberating effect. Tubes were not only more costly and much bigger than the early transistors, they also had functional limitations.

The fact that a tube needs a heated cathode, means that it must have its circuits organized so all the cathodes are low-potential points in their respective circuits. And when grids are used to control cathode-to-anode electron flow, the voltage applied to the grid must always be referenced to the cathode, not the anode.

Transistors changed that. A tube anode—controlled by appropriate voltage(s) on its grid(s)—only conducts when it is positive with respect to the cathode. But transistors can be made either way. Some transistors conduct when the collector is more positive than the emitter, some when it is negative, according to which type they are: PNP or NPN.

But today, transistors are only seen as a transition from discrete circuitry to integrated circuitry. ICs (integrated circuits) are used for almost everything today. And an IC may have anything from half-a-dozen transistors and diodes of mixed type, up to several hundred, all produced in a "chip" that will sit on the tip of your pinky without you knowing it's there.

Today's engineers don't bother much with basics any more (by which we mean items like transistors and diodes), but think in terms of available ICs. All the functions built into your digital wristwatch or your pocket computer, which can include log and trig functions, are contained in a relatively few ICs, possibly in a single large-scale IC, still so small that, when you look inside you are apt to say, "Is that all there is? Where are the 'works'?"

About the smallest unit today's engineer will think of is an "op amp," short for operational amplifier. ICs usually come with at least four of these on a single chip. Back in the days of tubes, a whole bank of tubes and assorted equipment could never have duplicated the performance of one cheap op amp today. That's progress!

All this has had the effect of making the engineer look for, or wait to be introduced to, a variety of pre-packaged ICs. All he knows is that for a certain variety of inputs to any particular IC, it will deliver a predictable variety of outputs. You put a tiny battery into your wristwatch or calculator and lo, it tells the time, or calculates, or maybe both at once! What more does an engineer need to know? Anyone can replace a battery, and if the IC "goes," (which it seldom, if ever, does) the whole thing's "shot," anyway.

If you want to do something different, you think of putting together parts, usually ICs, possibly with some control circuits, that together do what you want. Even before this trend got started, the approach used in electronics education, like everything else, became one of learning "facts." A device has certain characteristics, which are used in certain ways. New devices are developed with different (usually better in some way) characteristics, and everyone has to "go back to school" to learn a new set of facts.

In the days of tubes, perhaps the main

fact was $\mu = g_m r_a$. Given the mutual conductance, g_m, which is the rate at which plate current changes when grid voltage is varied at constant plate (anode) voltage; and the plate resistance, r_a, which is the relationship between plate voltage and current, at constant grid voltage; multiplying the two together yielded the amplification factor, μ, which is the rate at which plate voltage changes when grid voltage is varied, with the plate current held constant.

When I took my registration examination, that was the principle question, which implied that if you knew that, you were a fully qualified electronic engineer! Things have probably changed since then. My point here is that, in the real world, a tube never operated at precisely any of those neat quantities that were supposed to describe its characteristics.

All this comes back to our problems with education, which may not be a subject for this column addressed to audio people, but it cannot be avoided altogether either. The indoctrination aspects of education, where we are conditioned to look for information we can apply to our specific problem, without understanding it, results in this whole approach. And as line after line of new devices appear on the market, we have to "go back to school" to get ourselves "updated," when that wouldn't be necessary for a creative engineer.

If we were truly creative, all we would need, when something new came out, would be a data sheet on the new device, to let us know its capabilities. We would be able to take it from there. But we can't, generally speaking. Even knowing the right questions to ask helps. We have been to update seminars where, after asking some of the right questions, other attendees came to us for answers, instead of going to the seminar organizers. Doesn't that tell you something?

So what is creativity? If we are to develop it, or stimulate it, we must know what it is. And there's the rub: how do you describe something to someone who has never seen anything like it? As someone once made an analogy, "How do you describe sex to someone who hasn't experienced it?"

Well, we won't get into that, but creativity is important. In our schools, the arts are regarded as the main avenue for creativity, so let's take it from there. In teaching the arts, whether it be painting, sculpture, music or whatever, the first step is copying, learning to do what someone else has done. That is certainly not creative. A computer could be programmed to do that, and by error reduction to gradually get closer to the perfect copy.

What a good arts teacher is looking for, is some sign of individuality in the student; a student who shows his own individuality in his artistic expression. This is where the real artists come from. But, engineers and architects are traditionally creative people too. They create what has not been before, just as an artist does.

Again the question comes back to what constitutes creativity: when does it begin to show, and what can we do to develop it or stimulate it? My son remembers a Geometry teacher who had no creativity, and expected none from her students. As a result, he was turned off from math and I have met thousands of others with similar experiences.

The day I visited his class, she was handing back homework that had been set on the theorem that proves that, if two sides of a triangle are equal, the corresponding opposite angles are equal. They had been shown the traditional way of lettering a triangle, ABC. I think in that particular book, the B was at the apex of the triangle. A student came up to argue with his teacher that his proof was correct. I looked at it. It was. But he had put a different letter at the apex. So all the lettering in the proof was different from that in the book, although it was absolutely correct.

Nothing particularly creative about that, I suppose, but what was suppressing creativity was the fact that the teacher couldn't see that. The way she saw it, the book proof was correct, and nothing else was. Changing the letters was not allowed. Now this student had reconstructed the proof correctly, but had made the "mistake" of lettering the triangle differently from the way the book had.

When we learned geometry, we not only learned the theorems, but we had riders on them, to which we were expected to develop our own proofs or disproofs, without learning them from the book. That stimulates thought and, eventually, can lead to creativity. At first, such a procedure causes the student to search his memory for something similar that he can adapt. But adaptation is perhaps the first step toward true creativity.

Suppression of creativity is perhaps the worst malady of our school system. It turns the potentially good students into dropouts, and the easily programmable students get rated as good. They cannot be challenged, because they are incapable of individual thought. If you try to challenge them, they will respond, "You show us how to do it, and we'll do it." But finding out how to do something that has not been shown is completely beyond them.

Fortunately, man is naturally a creative creature. He wants to think, to find a way to do something original, that is his alone. Only by programming, so that he responds the way he feels he is expected to respond, does he lose that native capability. So what's to stop man from redeveloping his natural creativity?

Unfortunately, being programmed can become a habit. During the school years, the young adolescent can easily be reactivated. In fact, most of them want that. But later in life, the "protection" afforded by programming inhibits reactivation. It is always "safer" to do what you are expected to do. Eventually, the power of individual thought atrophies. And this is, unfortunately, the state of the vast majority of our teachers. For this very reason, their students are a threat to them: to maintain control, they must get them programmed to respond predictably. Individualists are a particular threat.

There is only one way to circumvent this: bypass the teachers. With very few exceptions, in today's world, teachers are an obstacle to education. If you meet a good teacher, you will find that he or she could not agree with this statement more. And because it is true, many potentially good teachers are working as checkout clerks at supermarkets, or bartenders. You'd be surprised how many people in such mundane jobs have teaching certificates!

So how can we circumvent the prevalently bad teachers, who presently control the schools, to the extent that good ones are not even teaching? We must bypass them. And our technology provides a tool for doing just that. Today, schools have budget problems. The biggest item on any school budget is teacher salaries. As long as we have to have more and more teachers, at higher and higher salaries, the problems will get worse. But if we could phase out the useless teachers—say by not hiring replacements as they leave—we could cut costs as nothing else would.

Mediated instruction is the way to do that. Educational Research Associates (if you want to write for a catalog, the address is 333 SW Park Avenue, Portland, OR 97205) has shown the way. There is a lot of mediated instruction out, but much of it is no better—often worse—than the printed kind, because it was developed from existing textbooks, that are geared to the programming system as it is. These materials are designed to bypass teachers, while allowing existing staff to administrate them. And because the results are good, the dumb teachers who act only as administrators (which suits them fine!) get the credit for what they didn't do. Now isn't that a way to go?

Incidentally, I should warn you that I might be prejudiced on this. You see, I wrote mediated materials for Educational Research Associates on Physics, English, Problem Solving Arts (a three-part series that goes from learning to count through calculus—can you imagine what will happen when third graders are learning advanced calculus?) and Statistics. Also a course on Basic Electronics. We have a saying, "the proof of the pudding is in the eating!" ■

PICKUP SYSTEM

• The W-350, a new professional studio grade pickup system, offers super-low distortion characteristics, high quality electronic specifications, and the additional features necessary to improve performance for the serious musician. There is no key or valve noise, feedback, or water blockage with the W-350. It features an LED indicating battery checker, a phase reversal switch, and a 12 position equalization switch (which includes the flat position) to compensate for the difference between the inside and the outside of the instrument. The W-350 is available for all woodwinds, piccolo, all flutes and all brasses. The type W transducer is a patented pressure-only type and comes in three versions: flute, piccolo and screwmount. The type W transducer for flute and piccolo type replaces the cork assembly, while the screw mount requires a 10-32 threaded hole or threaded insert for placement in the mouthpiece, barrel, bocal, etc.
Mfr: FRAP
Price: $550.00
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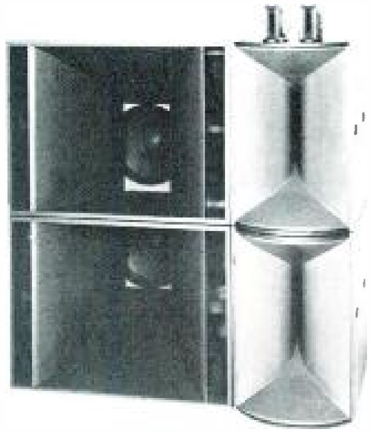
Mfr: Audio and Design Recording, Inc.
Circle 49 on Reader Service Card

FRONT-LOADING HORN SPEAKER SYSTEM

• The Ramsa model WS-712 employs newly-developed Twin Bessel horns to deliver uniform directivity in both the horizontal and vertical sound fields while maintaining uniform frequency response, 45 to 22,000 Hz, in both directions. The Ramsa speaker system has a power handling capacity of 200 watts (continuous program) and a pressure sensitivity of 106 dB (1m, 1 watt on axis). Sound dispersion is 90° (H) by 40° (V). The Twin Bessel horn design is used for the low, middle and high frequency range in the 712 system. In addition to the benefits of the design, the material used for the horn's construction, fiberglass reinforced plastic, further contributes to flatten radiation power at low frequencies. The middle and high frequency compression drivers in the Ramsa speaker feature heat-resistance treated aluminum voice coils to increase power handling characteristics and a magnetic ferrofluid-filled magnetic gap is employed in the voice coils to limit temperature rise. Additionally, Rhombic edges on the diaphragm reduce resonance, resulting in a smooth high-range frequency response.
Mfr: Panasonic
Circle 48 on Reader Service Card



REMOTE LINE DISCONNECTOR

• The Model RLD-5 is a new remote line disconnector that enables broadcast line troubles to be remotely isolated to the telephone line or the broadcaster's remote pick-up equipment without dispatching repair personnel. Installed at the telephone line protector, the RLD-5 is activated by the telephone company test board to temporarily disconnect the broadcaster's pick-up equipment from the line, return a 1000 Hz tone at 0 or −10 dBm to the test desk and open the line for 15 to 20 seconds, thus verifying broadcast line continuity and quality. Following tone return and open time, the line is restored to normal. The entire test sequence is completed in less than 45 seconds. Compatible with all metallic equalized and non-equalized broadcast lines, the RLD-5 provides a positive interface to facilitate trouble isolation and repair.
Mfr: Cidcomm, Inc.
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PRO MIXING BOARD

• The Pro-4 3 series is offered with 12 through 24 inputs and features two true stereo submaster or four mono submaster operation at the touch of a button; separate stereo and mono outputs; balanced and unbalanced inputs and outputs with shielded, oversized mic input transformers; gain status indication; stereo solo and mute; four LED level arrays; four sends; four band EQ; long throw faders; channel patching, and external power supply.
Mfr: Dallas Music Industries
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Electro-Voice's Greg Silsby talks about the Sentry 100 studio monitor

Production Studio WRBR-FM, South Bend, Indiana.

In all the years I spent in broadcast and related studio production work, my greatest frustration was the fact that no manufacturer of loudspeaker systems seemed to know or care enough about the real needs of broadcasters to design a sensible monitor speaker system that was also sensibly priced.

Moving to the other side of the console presented a unique opportunity to change that and E-V was more than willing to listen. When I first described to Electro-Voice engineers what I knew the Sentry 100 had to be, I felt like the proverbial "kid in a candy store." I told them that size was critical. Because working space in the broadcast environment is often limited, the Sentry 100 had to fit in a standard 19" rack, and it had to fit *from the front, not the back*. However, the mounting hardware had to be a separate item so that broadcasters who don't want to rack mount it won't have to pay for the mounting.

The Sentry 100 also had to be very efficient as well as very accurate. It had to be designed so it could be driven to sound pressure levels a rock 'n roll D.J. could be happy with by the low output available from a console's internal monitor amplifier.

In the next breath I told them the Sentry 100 had to have a tweeter that wouldn't go up in smoke the first time someone accidentally shifted into fast forward with the tape heads engaged and the monitor amp on. This meant high-frequency power handling capability on the order of five times that of conventional high frequency drivers.

Not only did it have to have a 3-dB-down point of 45 Hz, but the Sentry 100's response had to extend to 18,000 Hz with no more than a 3-dB variation.

And, since it's just not practical in the real world for the engineer to be directly on-axis of the tweeter, the Sentry 100 must have a uniform polar response. The engineer has to be able to hear exactly the same sound 30° off-axis as he does directly in front of the system.

Since I still had the floor, I decided to go all out and cover the nuisance items and other minor requirements that, when added together, amounted to a major improvement in functional monitor design. I wanted the Sentry 100 equipped with a high-frequency control that offered boost as well as cut, and it had to be mounted on the front of the loudspeaker where it not only could be seen but was accessible with the grille on or off.

I also didn't feel broadcasters should have to pay for form at the expense of function, so the walnut hi-fi cabinet was out. The Sentry 100 had to be attractive, but another furniture-styled cabinet with a fancy polyester or die-cut foam grille wasn't the answer to the broadcast industry's real needs.

And for a close I told E-V's engineers that a studio had to be able to purchase the Sentry 100 for essentially the same money as the current best-selling monitor system.

That was well over a year ago. Since that time I've spent many months listening critically to a parade of darn good prototypes, shaking my head and watching some of the world's best speaker engineers disappear back into the lab to tweak and tune. And, I spent a lot of time on airplanes heading for places like Los Angeles, Grand Rapids, Charlotte and New York City with black boxes under my arm testing our designs on the ears of broadcast engineers.

The year was both frustrating yet enjoyable, not just for me but for Ray Newman and the other E-V engineers who were working on this project. At this year's NAB show it all turned out to be worth it. The Sentry 100's official rollout was universally accepted, and the pair of Sentry 100's at the Electro-Voice booth was complemented by another 20 Sentry 100's used by other manufacturers exhibiting their own products at the show.

What it all boiled down to when I first started the project was that I knew that the Sentry 100's most important characteristic had to be *sonic integrity*. I knew that if I wasn't happy, you wouldn't be happy. I'm happy.

Greg Silsby

Market Development Manager, Professional Markets

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db Editorial

ALMOST TWO YEARS AGO (March, 1979), we surveyed the international audio scene, with studio reports from Central and North America, Europe and Japan. And now, more than enough time has passed for another look around the world, to see what's new in international audio.

We return to Mexico City, this time to visit the RCA Mexicana recording studios, and the scene of the first interface between a (British) Solid-State Logic board and an (American) Ampex tape recorder. SSL boards were described in our August and September, 1979, issues, and the Ampex ATR-124 was featured in the January, 1980, issue of **db**. At that time, we speculated on the interface system that might be needed to allow those two beasts to talk to each other. Well, it all went together nicely in Mexico City, and by coincidence, in Glendale, California as well.

In Glendale, Yamaha recently opened an impressive R&D facility, comprised of two studios designed for hands-on research and product development. Within the studios, American producers and engineers will get the chance to do a little "tire-kicking" of the Japanese company's latest recording and performing hardware. In return, Yamaha design engineers get the chance to see how their design prototypes stand up to studio use and abuse. It should prove to be a splendid way for supplier and user to collaborate, much to the advantage of both. The coincidence mentioned above is that Yamaha also selected SSL and Ampex for its Studio A control room.

By contrast, the Hitokuchi-Zaka studios in Japan chose Automated Processes consoles and Studer multitrack tape recorders. The studios were designed by Takamichi Suzuki, whose work is well-known in Japan.

And in Colombia, South America, Enrique Gaviria and Mario Saraste sought out New York architect John Storyk to help them in the creation of *Fonovision Internacional*. Storyk's participation in the project began several years ago, when Senor Gaviria visited the United States, to study at the Institute of Audio Research. Another coincidence—architect Storyk recently completed the design of the Institute's new control room/classroom, but that's another story.

In England, the APRS (Association of Professional Recording Studios) continues to grow and prosper, with almost 200 member studios now within its ranks. This month, John Borwick brings us up-to-date on the latest APRS news.

In our last international audio issue, we offered a change of pace with a feature story on "Time-Aligned Loudspeaker Systems." This month (another coincidence?), we again conclude with a loudspeaker story. ■

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HOWARD SHERMAN

Fonovision Internacional— Colombia's First World Class Studio

Located high in the Andes, Fonovision Internacional *is poised to become a force in the world of quality recording facilities.*

JOHN STORYK IS A MAN who likes challenges. Regarded as one of the world's leading designers of recording studios, the 34-year-old architect has supervised the the design and construction of studios for the likes of the late Jimi Hendrix, Leon Russell, and Ace Frehley (KISS), among others. His multi-million dollar entertainment center for music industry entrepreneur Albert Grossman in Bearsville, New York, will include a state-of-the-art recording and video production complex, three restaurants and an extraordinary 300-seat luxury dinner-theatre. The Storyk-designed Centel video complex in Boston is New England's newest (November, 1980) and most sophisticated video production center.

One of Storyk's most recent projects (completed in December, 1980) is *Fonovision Internacional*, a 10,000 sq. ft. concrete and masonry showplace in Colombia equipped with two studios, two control rooms and full support facilities. Situated conveniently near the capital city of Bogota, the 24-track facility is already booked solid for local audio and multi-media work, and will be prepared to accept international projects by the beginning of March.

GENESIS

Two and a half years ago, engineer/producers Enrique Gaviria and Mario Saraste, already owners of a small 8-track studio, decided they wanted to open a world-class international recording facility. To that end, the former musicians organized a consortium of Colombian businessmen and acquired three acres of land on the northern outskirts of Bogota.

Situated in the Andes, 8,800 ft. above sea level, Bogota is a major South American industrial city of more than five million people. In the past, the city has not been considered as an entertainment industry center. Nevertheless, Senors Gaviria and Saraste believed that a first-rate, multi-track/multi-studio facility would become a welcome addition to the Bogota community. By the Fall of 1977, funds had been raised and they commissioned Storyk, whose work in the States they had admired, to design and build what would come to be known as *Fonovision Internacional.*

CONSTRUCTION

One of the many unique features of the recording complex is the fact that it was completely built from the ground up, an uncommon practice in these times of urban renovation. The entire structure, including the shell and 80 percent of the interior walls, is constructed of exposed 25-centimeter brick. "That kind of solid construction is economically feasible only in a country where the building vernacular is masonry," comments Storyk. "More importantly," he adds, "the rough-surfaced brick can be an important acoustical asset if used correctly."

In some instances, of course, the more-traditional acoustic construction (wood studs, insulation, baffles, etc.) was used. In instrument stations, for example, some amount of trapping and wooden slot resonators were needed. "Spring isolators and certain kinds of fiberglas were imported from the States," explains California record producer Bob Margouleff, who spent a month in Bogota supervising construction.

(continued on page 28)

Mr. Sherman is the president of Howard Sherman Public Relations.

MICROPHONE APPLICATION NOTES

• The first several issues of PZM Application Notes, prepared by Crown, are now available. The Notes include suggestions on the use of PZM microphones in certain specified situations, based on the early experience of users, and the theoretical considerations of the product design. The first of the Notes, in what will be a continuing series, covers: theory; shaping the pickup pattern; piano; theatre; percussion, and string instruments. Mfr: **Technical Services Department, Crown International Inc., 1718 W. Mishawaka Rd., Elkhart, IN 46517.**

BATTERY HANDBOOK

• A new 20-page illustrated booklet, "The Battery Handbook," describes how a battery works; why cell size, type, load profile and end voltage determine battery applications. Proper battery care including charging, equalizing, gassing, temperature, cleanliness and connections are also discussed in detail. Charts and tables show battery rating curves, discharge load profiles, battery charging methods and recharging characteristics. Mfr: **Ratelco, Inc., 1260 Mercer Street, Seattle, WA 98109.**

PRODUCT BULLETIN

• Switchcraft's new product bulletin, NPB 355, describes the company's new line of adapters, which provide full patching capability from male or female "Q-G" plugs or receptacles to phono plugs or jacks. The bulletin lists features, part numbers, and suggested list prices for the four new adapters. It also includes assembly instructions and details on material specifications. Mfr. **Switchcraft, Inc., 5555 N. Elston Ave., Chicago, IL 60630.**

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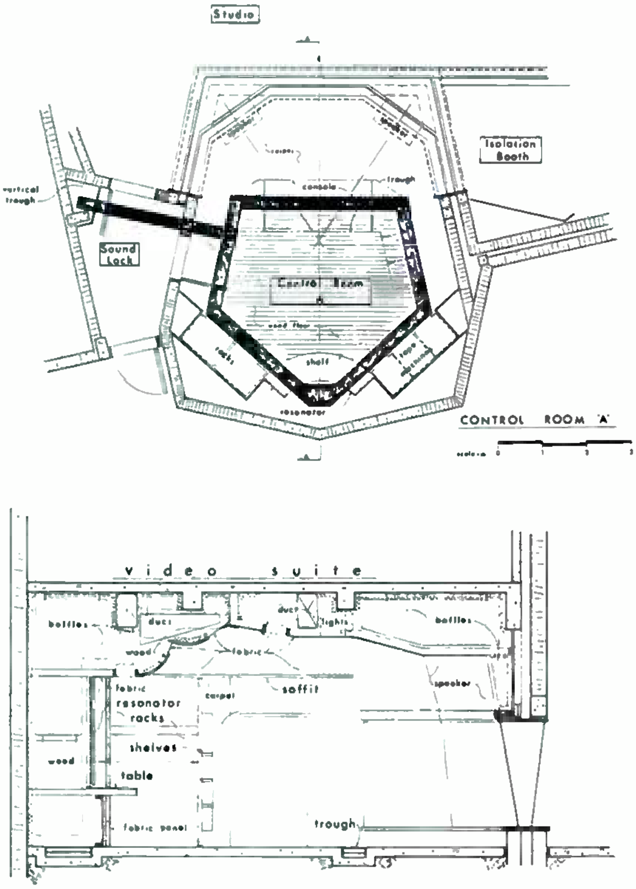

Figure 1. Floor plan and sectional view of the Studio A control room.

The stickiest problem of all was providing an air-conditioning system for the audio complex. Bogota has one of the most temperate climates of any major city. Temperatures rarely fall below 40 or 45 degrees Fahrenheit or go much above 85, so most domestic housing and commercial buildings have no need for heating or air-conditioning systems. Recording studios, however, pose special problems and require careful temperature control throughout the facility. And since there are virtually no professional air-conditioner installation firms in Bogota, experts were called in from neighboring cities.

Figure 2. From the isolation booth, looking into the control room and the studio.

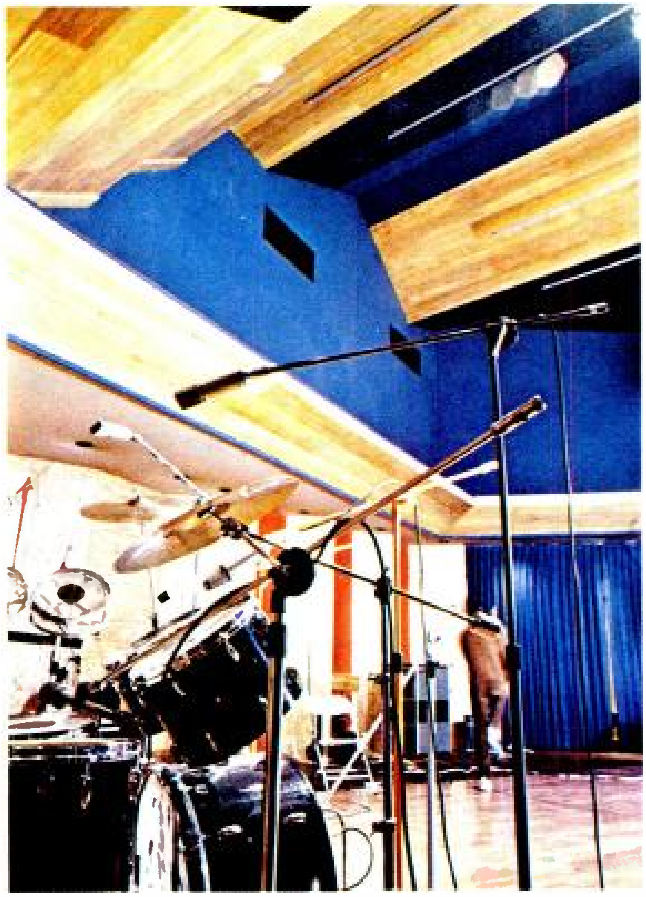

Figure 3. This view of Studio A gives a good idea of the structural ceiling details.

THE PROGRAM

The *Fonovision Internacional* program puts it in a class with the best facilities in the world. It includes two recording studios: Studio A, a 2,500 sq. ft. room which can accommodate from 30-to-40 musicians, and the smaller 800 sq. ft. Studio B. The studios are complemented by identical control rooms, with complete wrap-around vision. These identical rooms are extensions of Storyk's latest generation of "Phase Coherent" rooms, similar to Howard Schwartz' NYC control room and the new Criteria East studio expansion.

Both rooms have monitoring concepts based on the UREI "Time Aligned"™ system, but custom-designed by Storyk and Ted Rothstein. Both are equipped with state-of-the-art Neve consoles, Ampex tape machines and a full complement of outboard equipment. Each control room is capable of comfortably accommodating 15-18 people. Although *Fonovision Internacional* is designed primarily to handle local and international rock and pop work, there is a video control room atop Studio A, offering full video monitoring and camera capability. Plans are to add a ¾-inch video production and post-production facility that will enable performers to produce promo tapes and audio/video demos.

Finally, the recording complex is equipped with all the necessary support systems: maintenance shops, pre-production offices, mechanical facilities, storage space, washrooms, lobby and lounge. A separate building on the property has been renovated to house full kitchen facilities and, on an adjoining acre in a redolent eucalyptus tree setting, a six bedroom mini-hotel for international groups will soon be constructed.

THE INTERNATIONAL LANGUAGE

While music has always been considered *the* international language, quality audio recording facilities have, to date, been limited to a precious few major world capitals. Fonovision Internacional aims to become a part of that small, but growing community and, based on John Storyk's track record, it seems likely to achieve its goal. ■

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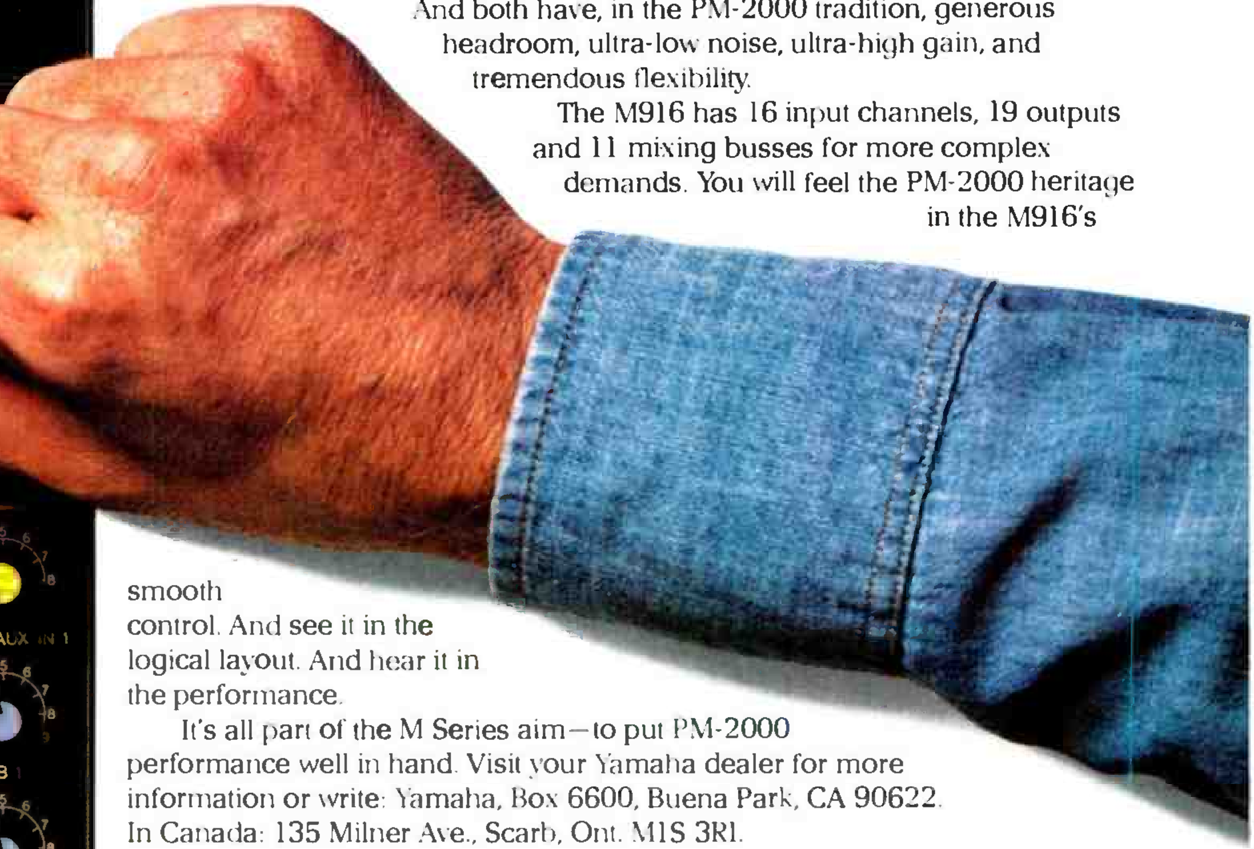

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YASUSHI ONO

The Design of the Hitokuchi-Zaka Recording Studios

More than technical factors go into the making of a first-class studio, as the Hitokuchi-Zaka Studio proves.

IN GENERAL, the quality of a recording studio is evaluated not only by audio effects, but also by visual effects. In Japan, as elsewhere, there is a growing demand that architecture and lighting create an aesthetic atmosphere that will be pleasing to the performers. For this reason, the Hitokuchi-Zaka studio complex has been colored in earth tones, and finished with quality woods, such as cherry, cedar, teak, medlar and rosewood, to produce a pleasingly-soft atmosphere. The lighting facilities can be controlled by the performers to meet their own individualized requirements and personal tastes.

Of course, this is but one of the many important factors that make the production of good music possible. Just as important are the technical factors, which must always be given their due consideration.

Below sound insulation, vibration proofing, room acoustics, and facility-noise reduction, from the viewpoint of architectural acoustics are discussed. This complex consists of three studios, three control rooms, one mix-down room, and other rooms for editing, tape copying, auditions, piano-practice, and reverberation chambers. The plan and sectional views are shown in FIGURES 1 and 2.

SOUND INSULATION AND VIBRATION PROOFING

In general, the items to be examined for sound insulation and vibration-proof design are as follows:

1. Insulation of noise and vibration coming from outside the building.
2. Insulation of noise and vibration coming from the facilities placed within the building, and
3. Sound insulation between individual rooms.

As for the first two items, there was little problem because the studios and control rooms were built away from the sources of noise and vibration. For the last item however, much effort was required, since the studios, control rooms and other acoustic areas were all built within the same building.

The quantity of sound insulation required was calculated by identifying all the routes through which noise would travel to individual rooms. This noise was taken as an incident sound pressure level for each room, on the assumption that the sound is picked up by a microphone placed near the wall, with the sound-receiving point one meter from the sound insulation layer. The calculation was made at the three bands centered at 125, 250 and 500 Hz, in which problems tend to occur.

Since the studios and control rooms were built one upon another, a floating structure design was required. The structure must be made with heavy, rigid material, since otherwise it tends to transmit unpleasant vibrations, depending upon the vibration mode of the floor, and this affects the sound field within the room. In this studio complex, the thickness of reinforced concrete is 125 mm, which was increased to 250 mm for the control rooms. The floating structure presented the problem of insulation of pipes and ducts. Rubber sound-insulation coupling was used for the pipes, with flexible joints for the ducts. To assure maximum sound insulation, the use of ducts on the same system between individual rooms was avoided as much as possible, to eliminate cross-talk problems through the ducts. When the space was too small, a sound arrester was mounted on the duct, as needed.

ISOLATION BOOTHS

Several isolation booths were built in each studio, in order to offer maximum sound separation between groups of musical instruments. The sound insulation between each booth and the studio is approximately 20 dB at 500 Hz. Doors made of a single 15 mm-thick plate of glass were installed to assure a feeling of visual intégration between booths and studio.

The floor of the drum booth was separated from the main studio floor, so that vibration cannot travel through the floor to the microphones near other musical instruments.

STUDIO DESIGN

In the acoustic design of a studio, it is important to meet the contradictory requirements of sound separation between

Mr. Ono is the director, engineering division, of the Hitokuchi-Zaka Recording Studios in Tokyo, Japan.

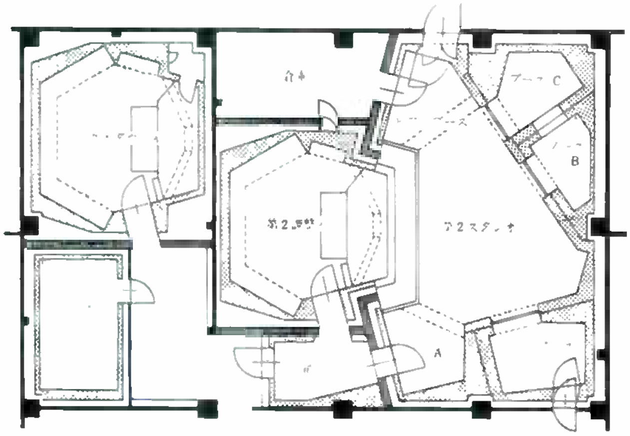

Figure 1. The plan view of Studio 2 and the mix-down room. Isolation booths are labeled A, B, and C. The piano booth is to the left of (C), and the drum booth is to the right of (A). The mix-down room is at the upper left of the figure.

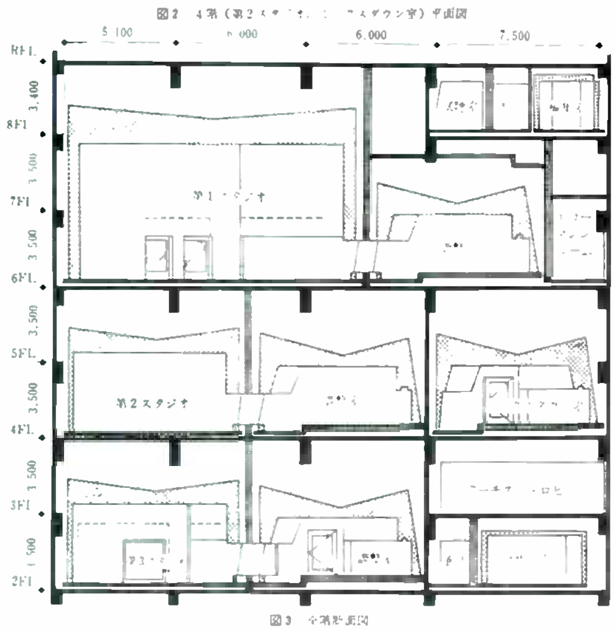

Figure 2. Sectional view. Studio 1 is on the top floor(s). Directly below is Studio 2 and the mix-down room, and Studio 3 is on the lowest level.

Figure 3. The interior of Studio 1.

Figure 4. Studio wall detail.

individual instruments and a visually-integrated feeling between the individual performers. Many booths were built to meet the former, while the latter was met by installing doors made entirely of glass between each booth and the studio, and a small window between two of the booths.

In order to insure good visual connection among musical instruments, particularly between drums and piano, the individual booths were arranged as shown in FIGURE 1. With reference to the arrangement of musical instruments within these booth locations, the wall characteristics (dead zone and live zone) were created according to the specification of sound absorption in the main studio. The booths are classified as drum, piano, dead (1), and live (2).

The drum booth is separated from other booths to suppress very low-pitched vibration leakage through the floor. To eliminate the acoustic "fog" caused by the sound reflected from the ceiling, the entire ceiling area is made to absorb sound. The walls are finished with quality wood ribs, and are composed of sound-absorptive sections made of felt and glass wool, and sound-reflective sections made of plaster board. The reflective surfaces are designed so that reflections do not reach the floor. The desired live or dead zone is created by varying the proportions of reflective and absorptive surfaces.

CONTROL ROOMS

The requirements for a control room are: transmission of the sound from the monitor speaker to the mixer without coloration, correct localization of images, and discrimination against noise. To realize these requirements, the dimensions and shape of the room must be carefully examined. The dimensions are determined by the number and size of the recording hardware, the spatial relationship between speakers and mixers, and other facilities. In the typical studio, the floor area may be from 40 to 50 square meters, and the volume becomes approximately 100 cubic meters if the ceiling height is 2.5 meters. The floor area and height of these studios are 46 square meters, and 2.65 meters, respectively.

Next, the shape of the sound insulation must be examined. In view of symmetry and the spatial relations between the speakers and the mixer, the mixer is usually positioned at about the center of the room. Care should be taken not to create a lack of high pressure in the low-band frequencies at the mixer position due to room modes. In this studio, the length and width of the control room are approximately equal, and room modes tend to focus. Therefore, the sound insulation walls are inclined, so as to avoid creation of parallel-facing surfaces. The sound insulation ceiling is shaped like a ship's bottom. As a result, no booming phenomenon is observed. Although there are many theories on the shape of finished surfaces, the essence is to make perfect symmetry bi-laterally, so that direct and reflected sounds may reach the mixer without interfering with each other. To do so, it is necessary to carefully design the shape around the front monitor speakers. The shape of this studio is also shown in FIGURE 1.

As for sound absorption, the whole was made dead (average sound absorption is about 0.45) to make the speaker reproduction sound clear. The principal specifications are; *Front wall:* Finished with quality wood ribs, and felt-plus-glass wool. *Rear wall:* Felt finishing, and interior glass wool. *Ceiling:* Felt finishing, and interior glass wool, part plaster board, and part perforated veneer. *Floor:* Carpeted.

ROOM ACOUSTIC DESIGN

The table below shows the dimensions of each of the rooms considered important from a viewpoint of acoustic design.

LISTENING TESTS

Listening Tests were conducted in the control room to determine the optimum position of the speakers, the speaker enclosure design, and the appropriate materials for the front surface of the ceiling. According to the test results, the speaker enclosure was surrounded with sound insulation materials, glass wool was filled around the speaker itself, and the enclosure's front surfaces was finished with felt.

In general, reflection from the walls can be avoided by properly designing the shape of a room, but the first- and second-order reflections from the floor and the studio window to the floor cannot be avoided. Consequently, they interfere with the direct sound, making the sound pressure lower over a bandwidth from 40-to-60 Hz at the mixer position. To avoid this problem, the front surface of the control desk is made absorptive, thereby eliminating this dip in the bandwidth. As a result, the measured transmission characteristic shows good balance from the lower part to the higher part of the bandwidth.

The structure, size, shape and sound absorption properties of the three control rooms and one mix-down room are identical, so as to give interchangeable acoustic characteristics.

AIR-CONDITIONING

Acoustic design must take into consideration the mechanical noise of the air conditioning system, as well as the sound of air motion at the system inlets and outlets.

Resonance at a particular frequency may sometimes be noted in a room having high sound absorption. The cause may not be due to faulty design within the room, but rather to the reflection of sound from the air conditioning system. This is caused by resonance of the duct walls, or the humming of a pan at the vent. To prevent these problems, the insides of the vertical ducts, extending from the sound-arresting chambers to the vents, were lined with rubber. The pans at the vents were filled with sand to prevent humming noises.

Based on our past experience, the allowable levels of air conditioning noise were set at NC 15 for the studios and NC 20 for the control rooms and editing rooms. The values measured after the completion of construction satisfied these requirements. ■

STUDIO DIMENSIONS				
	Floor area (square meters)	Height (meters)	Volume (Cubic meters)	Reverberation Time (seconds)
Studio 1	166	6.29	679	0.28
Isolation Booths	8-11	2.89	20-28	0.12-0.14
Studio 2	102	3.68	194	0.18-0.22
Isolation Booths	9-10	2.78	20-25	0.10-0.16
Control Rooms & Mix-down Rooms	46	2.65	91	0.14-0.17

Figure 5. Studio 3 control room. Each control room has a 32 in/24 out Automated Processes console. The 16- and 24-channel recorders are Studer A-800s, and the 2-channel machines are Studer A-80s and Ampex ATR-100s.

Figure 6. This view of the mix-down room shows the auxiliary equipment rack behind the mixer, mounted in the producer's table.

JOHN WORAM

RCA Mexicana Studios

A combination wood and oxygen shortage posed some interesting problems for the designers and builders of RCA Mexicana.

MEXICO CITY is the second largest city in the world, with a population of 16 million, projected to grow to 31 million by the end of the century. It is also the capital of a nation where music has always played a vital role in the everyday routine of the people. So it is not surprising that this sprawling metropolis is also a thriving center of recording activity.

RCA Records Mexico is one of the largest producers of recorded products in Mexico. Last year, the company pressed over 20 million long-playing records for the local market, and provided nearly 300 master tapes to a dozen other Spanish-speaking populations in Central and South America and Spain. RCA Mexico is not only a leader in terms of product output; the company now also seeks to establish a position of leadership in the realm of state-of-the-art recording techniques and technology.

The most recent manifestation of this commitment is a $2,000,000 renovation project, designed to upgrade three RCA recording studios and disc mastering suites. In November of 1980, the first phase of this project was completed. Earlier, the second-largest of the live rooms had been totally stripped, control room and all. It was then rebuilt with advanced acoustics and equipped with the finest recording gear available.

Such a project is no small task anywhere, but in Mexico City it presents a unique combination of challenges. For example, wood shortages have lead to a strong conservation-and-reforestation program, and the use of wood in construction is now restricted by the Mexican Government. In addition, local economics dictate the use of materials which possess the greatest lifespan at the least cost. The net result is that—unlike most studios—the interior construction of the RCA Mexico studios and control rooms is largely brick and concrete! Beyond the acoustic problem this creates, studio builders must also cope with a shortage of concrete due to the massive highway construction.

All of this adds up to a lot of fun for the man responsible for any studio construction project. In this case, the project manager and acoustic consultant was Senor Mario Sanchez Roldan. Sr. Sanchez Roldan originally came to RCA as a highly-recommended designer, and quickly became Audio Director for the entire complex. He was eager (in between phone calls to locate more concrete) to discuss the acoustic considerations of building in Mexico City.

"First of all, we are using brick and concrete for our interior wallls, floors and even sections of the ceiling. Naturally, the absorption coefficients of such materials are a known quantity to acousticians; but in this instance we were not free to use lumber where we would have liked. This creates problems in terms of structural transmission over a completely different range of frequencies, and also gives us problems in controlling acoustic energy travelling through the air.

"You must remember that Mexico City is 7200 feet above mean sea level. Sound travels faster in this rarefied atmosphere and is also more directional. These factors, coupled with the concrete problem, require some radically different equations and approaches in order to achieve the same degree of excellence found in multi-track studios in 'friendlier' locales.

In the studio, we have constructed the floor on three tiers, like balconies. The highest tier is for strings, the middle section is for brass, and the lower is for rhythm instruments. Of course, there is also a drum booth and a vocal isolation booth. We have actually cast ducts into the concrete floors of each of these sections, and used the spaces beneath each platform as tuned traps. It is necessary to trap the floors like this, because we do not have the ability to create the same degree of absorption in the walls and ceilings. There are traps in these areas, but the limitations imposed on us have required what I call 'Spherical Acoustics,' because we are working from all 360 degrees to achieve the desired isolation.

"These balconies have the added advantage of presenting a variety of apparent room sizes to a microphone, depending on the tier where the microphone is placed. The upper tier provides a solid back wall and proximity to a mostly 'live' ceiling, reinforcing the energy of the strings—yet it faces into a large open space that can be used to the necessary advantage by selection of the appropriate microphone pattern. The second tier has a back wall which is heavily trapped, a greater ceiling height, and a tighter relationship with the front wall which gives a nice 'airy' quality around the brass, with some delayed reflection off the front wall. These can be controlled by the

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Figure 1. A wide-angled view of the "pyramid power" ceiling, and the three-tiered studio floor.

Figure 2. A drum booth is at the rear of the studio, and if you look closely, there's also a piano "room" back there.

choice of microphones. The main floor has the advantage of the trapped floor, trapped side walls, and a trapped rear wall, along with a front wall which is absorptive to either side of the control room window, and of course highly reflective in front of it."

The arrangement provides amazing flexibility for controlling the sound at the source, rather than through later electronic processing. However, those interested in experimenting with such an arrangment should be cautioned that it works largely because of the narrower dispersion of sound, both vertically and horizontally, at this extreme altitude. Visiting musicians, particularly brass players, should also be cautioned that the thin air takes some getting used to. Sanchez Roldan has provided two aids to his musicians for this reason.

The first is "Pyramid Power." Having taken care of the problems imposed by physics, he decided that metaphysics could help out too. Carefully-proportioned pyramids are built into the ceiling at regular intervals. It has been demonstrated that a force, or energy, is generated by this particular convergence of planes, and Sr. Roldan wanted all of his products to benefit from the Pyramid Energy ceiling. (For those who don't believe in Pyramid Power, portable oxygen tanks are also available.)

The final area of acoustic concern is the control room. The same techniques of "spherical acoustics" provide the needed trapping here. But again, the rare atmosphere creates particular problems when working with transducers designed to react to the more-typical atmospheric back pressure found in sea-level studios. The only solution to this, short of pressurizing the control room and installing airlocks, is to fool the speakers. At RCA this has been done using a rather radically-angled Venturi-type compression ceiling. The mathematics of this are beyond the scope of this article (and this author!), but test results indicate that this particular design has reduced the effective altitude of the control room by almost one mile, at least across the front of the console.

The traditional methods of achieving a desirable RT-60-versus-frequency curve in the engineer's and producer's position now work quite nicely, with a clean punchy sound being quite apparent. Outside of this "pressurized" area, the effects of the high altitude are dramatically apparent.

But perhaps the most dramatic point of the control room is the equipment. The primary specification for the entire renovation project at RCA Mexico was to put together studio systems the equal of the best in the world. It is clear that no expense has been spared to do this, but Sanchez Roldan points out that his company is in business to make money, not to spend it, and that all of his equipment specifications are designed to do this.

For starters, the heart of the control room is a 40-input Solid State Logic Master Studio System. RCA Mexico has the distinction of being the first studio in the world to install an SSL with the new Total Recall Studio Computer. This system not only provides fader automation, it also stores and recalls the position of every other control on the console, using a flashy high-resolution color-graphics display to indicate the differences between stored positions and current positions. The computer also controls the tape machines, keeps track of all sorts of session information, and provides some fairly incredible mixing capabilities.

Figure 3. The control room interior. Note the angled ceiling, and the SSL's video monitors mounted above the window.

Figure 4. Sr. Roldan keeps an eye on the video monitors while setting up the board.

All of this comes at a price—in this case, nearly $250,000 *plus* almost 35 percent import taxes levied by the Mexican Government on all imported electronic equipment. Sanchez Roldan feels it is one of the best equipment investments RCA Mexico has ever made. "We are happy about the prestige in having the first SSL Total Recall System, but as this studio is used mostly by our own artists, prestige is not as important as a drawing card. This equipment gives us the sound and the flexibility we need. We have also calculated the savings which Total Recall gives us by opening up our rooms when they would otherwise be locked up for one artist. And, the computer allows us to eliminate a second engineer just to operate the tape machines. Over the life of the machine, it will more than pay for itself.

"Even more important in a country like Mexico is the reliability of the equipment. You would not believe how impossible it is to get emergency parts through our customs. We could not find anything with the features we needed which was built better or more easy to service ourselves. And because of our import taxes, we may use equipment longer than other places. The SSL Computer is all software-based, and we can effectively get completely modern updates through the mail this way."

One interesting thing about RCA Mexico's SSL Computer is that it speaks English and Spanish. The common keyboard is engraved in Spanish, and the video display confirmation and interrogation can be in either Spanish or English, depending on which program disc is loaded. Ths handy feature taught the SSL installation and training crew their first 25 words in Spanish; it also makes the Mexican artists and producers feel more comfortable with the computer.

For similar reasons of flexibility, quality and reliability, RCA Mexico chose the Ampex ATR-124 as their multi-track recorder. This presented the studio with another "first," namely the first SSL/ATR-124 interface in the world. Normally, this would not be a big deal, but the SSL is an unusual console, in that it contains complete tape recorder controls, which are interfaced to any tape machine using opto-isolated relay contacts. The ATR-124 is an unusual tape machine, in that it uses serial code to address the track select and other recorder functions.

Sanchez Roldan was aware of the problems, but he told each manufacturer, "I want the best, you have the best, now it is up to you to make this work!" Ed Engberg and Norm Major of Ampex, and Colin Sanders and Grey Ingram of Solid State Logic all got together, traded information, and *Voila!*, they came up with a translator which enables the SSL computer to talk to and understand the ATR-124 computer, making RCA Mexico perhaps the only studio in the world where the equipment talks in Spanish, English, and to itself! Indeed, when the Ampex and SSL engineers first connected their respective machines together and they worked, someone was heard to comment "Thank God they aren't arguing with each other."

Rounding out the equipment selection are several ATR-100 two and four tracks, Telcom Universal Noise Reduction frames capable of accepting Telcom, Dolby and dbx noise reduction cards, a variety of delay and reverb devices including Lexicon 224s, Prime Times and Delta-Ts; two live chambers, nine EMT plates and three AKG BX-20s, along with a fine collection of microphones including some beautiful old RCA ribbon mics and tube condensers.

Many "Gringos" here in the States have a completely false impression of Mexico and the Mexican recording business. The country is often viewed as a market for second-hand equipment and technology. The RCA Mexico story is only one example of how wrong this perception is. (For another view, see our "Report From Mexico City," in the March 1979 **db**—Ed.) The Mexican recording industry may not be as hot as the Mexican cuisine (but then, nothing is—Ed.), but it is most clearly in the vanguard of studio techniques, powered by enlightened artists, engineers and management dedicated to world-class recording excellence. ■

NORMAND L. MAJOR

The Ampex/SSL Interface

A brief look at the logistics involved in the interface between the Ampex ATR-124 and Solid-State Logic's computer-controlled audio board.

FOR DECADES, computer and automation sytems have used multi-wire interconnection for short distances: for longer distances, data and commands are coded and sent serially on one pair of wires. Conversion from the serial code to a multi-wire scheme is only done when necessary—for example, to control a piece of equipment designed for simple contact closure.

The code recognized and generated by all computer systems is the American Standard Code for Information Interchange, or ASCII. The ASCII code accommodates 95 characters, including upper and lower case letters, numerals, punctuation marks, plus 32 machine control signals such as line feed [LF] and carriage return [CR], appropriate for controlling teleprinters. (Bracketed characters [CR, etc.] identify machine control signals—Ed.)

Modern multi-channel tape recorders offer an expanded list of operational features that need to be remotely controlled, and it is not surprising that the new Ampex ATR-124 multi-channel analog audio recorder used this code for remote control, making it compatible with computer-driven editing systems and automation systems.

The first interface between the micro-processor controlled ATR-124 and Solid-State Logic's computer-controlled audio board featuring "total recall" occurred last October, at the RCA recording studios in Mexico City. This article will describe that interface at the RCA studios, with particular emphasis on the software and hardware.

Mr. Major is Associate Engineer, Audio Products Group of the Ampex Audio-Video Systems Division, Redwood City, CA.

THE SERIAL INTERFACE

The serial line feeding the ATR-124 is the same one that reports the status of all the buttons on the ATR-124's front panel to its main CPU. Not only are all the Safe, Ready, Input, Repro, Sync and Mute commands for every channel readily accessible, but so are the Tape Speed, VSO, Group and Panel Memory commands.

All of the above commands can be initiated by sending a sequence of ASCII characters. The format is listed in FIGURE 1. For example, to place channel 1 into Safe, the following is sent: The letter, F (denoting that a function is selected), then the number, 3 (Safe), and a [CR] to indicate the end of the command. Next, to specify Channel 1, the characters, CØ1, are sent, and this is followed by a [CR]. As the end of every sequence, the [CR] indicates the end of the command.

The serial ASCII code must be sent at a rate of 19200 baud and contain 8 bits with even parity to communicate with the main CPU. The receiver interface is an optical isolator. A typical interface for TTL levels is described in FIGURE 2.

Brief as the above description of the serial interface is, it is easy to imagine the ATR-124 obeying commands generated by a home computer. Of course, on the recorder itself, the ASCII commands are generated by depressing the customary machine function buttons, rather than by typing out the code. The ATR-124 has several useful outputs to provide feedback information to the interfaced system. These include all the transport modes, true direction, tachometer pulses (either from the capstan or tape time), electronic record status (indicating that at least one channel is in record), etc. All are available at remote or accessory connectors.

Button	Command Character	Next Character(s)	
AUX	A	[CR]	
ALL	B (block)	[CR]	
CHANNEL	C	0-9 (tens)	0-9 (units) [CR]
REC	F(function)	1	[CR]
READY	F	2	[CR]
SAFE	F	3	[CR]
INPUT	F	4	[CR]
SYNC	F	5	[CR]
REPRO	F	6	[CR]
MUTE	F	7	[CR]
GROUP	F	8	[CR]
GROUP 1	G (group)	1	[CR]
GROUP 2	G	2	[CR]
GROUP 3	G	3	[CR]
GROUP 4	G	4	[CR]
GROUP CLEAR	K (kill)	[CR]	
MASTER RESET	I (initialization)	[CR]	
7.5 ips	S (tape speed)	1	[CR]
15 ips	S	2	[CR]
30 ips	S	3	[CR]
VSO	V (VSO)	S (select)	[CR]
DECREASE	V	D	[CR]
INCREASE	V	I	[CR]
%	V	P (percent)	[CR]
TONE	V	T	[CR]
A	P (panel memory)	1	[CR]
B	P	2	[CR]
C	P	3	[CR]
D	P	4	[CR]
R	M (monitor memory)	1	[CR]
P	M	2	[CR]
F	M	3	[CR]
S	M	4	[CR]
PANEL MEM	X (transfer)	P	[CR]
MONIT MEM	X	M	[CR]
REC*PLAY	R	P	[CR]
REC*STOP	R	S	[CR]
REC	R	H	[CR]
STOP	T (transport)	1	[CR]
PLAY	T	2	[CR]
FAST	T	3	[CR]
SERVOS UNLOCKED	F	4	[CR]

Figure 1. Tape recorder instructions and ASCII command characters.

RCA MEXICO CITY

Not all controlling devices provide serial ASCII code. Sometimes, only an array of relay contacts or some other form of parallel interface is available. This is indeed the case with the SSL interface, even though it is driven by a mini-computer.

Therefore, RCA's recording studio required an interface system such as that described above. This first parallel/serial interface was designed by Ampex, using an off-the-shelf micro-processor system. Essentially, it is a micro-processor with input and output devices, plus read-only memory and random-access memory. A description of this interface will serve to underline how this approach is an excellent general solution to any interface situation.

PROBLEM: SSL supplies 24 relay contacts, each contact representing one of the 24 audio channels. An open contact indicates the channel is to be in Safe; closed, to be in Ready.

These relays are activated by SSL's mini-computer. One other relay contact is supplied to indicate that either all channels are to be in Sync, or all are in Repro. These contacts must be sampled, processed and then sent in the appropriate serial form to the ATR-124's receiver.

SOLUTION (see FIGURE 3): The relay commons have been grounded with respect to the interface, and the contacts (along with pull-up resistors) are connected to standard input/output circuits, which are interrogated by the micro-processor. The

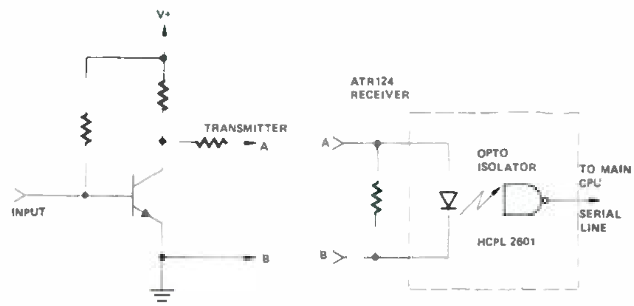


Figure 2. The console/tape recorder interface.

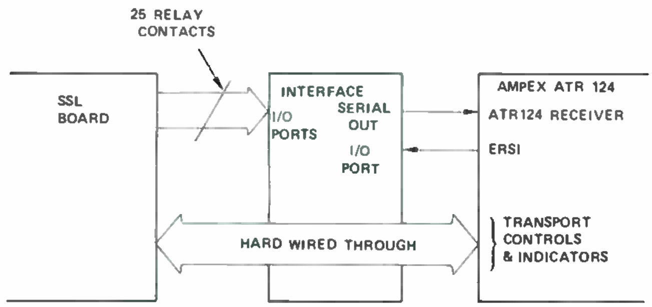


Figure 3. The parallel/serial interface allows tape recorder functions to be controlled from the console.

micro-processor (instructed by the program resident in read-only memory) debounces and processes this information and delivers it to the USART (Universal Synchronous/Asynchronous Receiver/Transmitter) which in turn sends the command sequence to the ATR-124 in the serial form mentioned earlier. This method is software intensive, giving the advantage of easily changing and expanding its capabilities by installing another read-only memory. Inputs to the interface from the ATR-124 are the ERSI (Electronic Record Status Indicator) upon which the interface bases its ready-record decisions, TDR (True Direction Sense), and the TTAC (Tape Timer Tachometer). Transport controls and indicators are hard-wired directly to the SSL board, as it can provide dedicated switch closures for these.

The interface gives the SSL board control over the ATR-124 to the extent that its designers intended. In the future, it could also control tape speed and other functions of the ATR-124. All of this can be implemented with a change in software, not hardware.

The success of the interface was immediate and the studio held its first recording session within a few days. As a point of interest, the tape transport servo response of the ATR-124 exceeded the maximum setting in SSL's software. It is understood that SSL is raising the maximum setting to take advantage of the ATR-124's transport capabilities.

With the ATR-124 as an example, it can be seen that the integration of a complex device into an even-more complex system of devices is best facilitated by the use of a standardized, computer-oriented interface. In this way, adaptation of the device to the system, or even to individual preference, is transferred from a hardware design problem to an intellectual definition of function, implemented in software in less time and at less cost. ■

TOM LUBIN

A Studio For Research

A guided tour through Yamaha's R & D base.

AMONG INTERNATIONALLY-KNOWN Japanese corporations, the Yamaha name has become one of the most noticeable because of the diversity and quality of its products. Yamaha musical instruments have become a standard by which other brands are judged. During the past five years, Yamaha loudspeakers, amplifiers, and live mixing consoles have received equally wide acceptance.

An important reason for their ability to produce high-quality products is that they have consistently invested a good deal of time in research and development. Most of the Yamaha products that reach America have had their prototypes tested and critiqued by professional musicians and engineers. Often, through this process, a new Yamaha product has generated a word-of-mouth reputation even before it was put into production.

Three years ago, to further expand this approach to new product development, Yamaha decided to establish a permanent R & D base near Hollywood, a musically active area where valuable professional opinions are readily available. Hiro Kato, the present manager of the R & D base, was assigned to study the feasibility of such a facility. During the course of several trips between Japan and Los Angeles, Hiro selected a site in Glendale, and expanded the plan from just a music studio and lab to the inclusion of full state-of-the-art recording facilities. In this environment, Yamaha product engineers and U.S. experts could listen, measure, record, experiment, and exchange opinions.

The original acoustic specifications for the facility were then made by Takashi Fujita, a top acoustics expert at Yamaha's electro-acoustics laboratory in Japan. Those specifications included the number of rooms, functional electronic requirements, sound isolation, average absorption coefficients, and noise criteria in each room.

Hiro then selected George Augspurger to be the acoustic designer consultant who would convert Takashi's specifications into a working design. At George's suggestion, Peter T. Creamer & Associates were retained as the project's architects.

LAYOUT

To meet Takashi's specifications, there were two important issues to consider. First was the proper layout of needed rooms within a limited space. Besides the two control rooms and studios, there had to be sufficient office, lab, and storage space, as well as a lounge and a place for a facsimile machine.

Secondly, since the primary purpose was to use the facility for laboratory testing, Takashi specified very stringent acoustic requirements whereby the internal environments of each of the studios and control rooms should have almost total isolation from one another.

In the final design, two main studio areas are separated by an experimental lab. The control rooms for the two studios are acoustically identical. It was specified that when a product evaluation was being conducted in Studio B, its control room could be used simultaneously for other assignments. To meet this functional requirement, very stringent sound isolation was required between control room B and studio B.

BASIC CONSTRUCTION

The facility was built from the ground up. Decoupling was achieved with conventional materials, by constructing totally-isolated rooms within a massive basic shell. The most difficult isolation problem would exist between control room B and studio B. George, after consulting fellow acoustician L. W. Sepmeyer, concluded that an isolated three-wall system of two stud-and-sheet rock walls with a solid 8-inch concrete block wall between them would do the job. Three different thicknesses of plate glass (½", ⅝", ¾") are used, one pane per wall.

Concrete block construction was specified for the entire outside shell and interior separation walls. As each wall went up, all of the holes in the blocks were filled with mortar. All of the block walls have their footings connected to the base slab, and are in no way connected to the floating slabs on which the interior walls and ceiling are built. All the studio and control room block walls were plastered with one-half inch of mortar to improve the performance of the walls, by sealing the surface pores on the blocks. The roof consists of four inches of light-weight concrete, poured on top of corrugated steel decking, supported by steel I beams that were tied to the block walls.

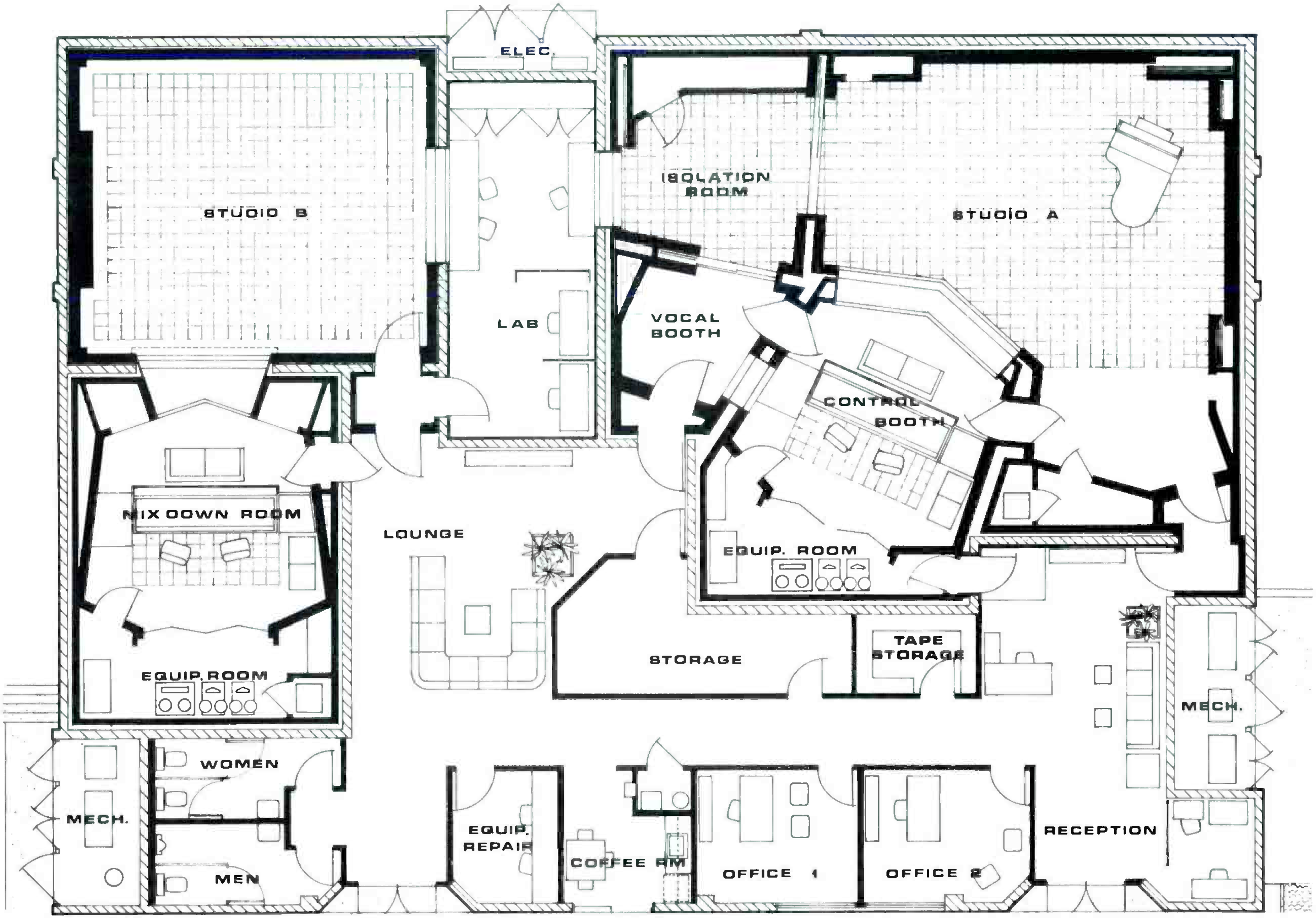
After the shell and all the interior separation walls were completed, the six different floating floors were poured. These rest upon Consolidated Kinetics vinyl-coated pre-compressed fiberglas cubes.

Standard 2-by-4-inch frame construction was used to finish the interiors, which are supported by the floating floors. Great care was taken to make sure that none of the interior walls or ceilings touch the exterior shell, and that all adjacent systems remained free of any mechanical coupling. Between all the studs, batten insulation was stapled. All the studio and control room surfaces were sheathed with four layers of ⅝ inch gypsum board, with each seam staggered. Also, every seam and joint of the wood walls and each layer of gypsum were caulked.

MEASUREMENT

Because the Yamaha specifications were so stringent, the isolation performance put forth in the design could not be positively verified until all the doors and windows were mounted. When finally it could be measured, performance was

Mr. Lubin is the president of Creative Space, eight self-operated recording suites designed specifically for songwriters developing material for future record productions. Formerly, he was editor of Recording Engineer/Producer.


ELEC.
STUDIO B
LAB
ISOLATION ROOM
STUDIO A
VOCAL BOOTH
CONTROL BOOTH
MIX DOWN ROOM
LOUNGE
EQUIP. ROOM
EQUIP. ROOM
STORAGE
TAPE STORAGE
MECH.
WOMEN
MEN
MECH.
EQUIP. REPAIR
COFFEE RM
OFFICE 1
OFFICE 2
RECEPTION

as expected, except for the wall between the mixdown room and studio B. It was good, but not good enough. All concerned could not understand why the three-wall system was not meeting its specifications. The walls were somehow mechanically coupled. The three panes of plate glass were removed and the dead air gaps between the three walls were analyzed. The isolator material used with the floating floors had also been used between the two stud walls and the middle block wall. This was diagnosed as the problem. It seemed that the isolation board was excellent if sufficiently compressed by the poured concrete, but—as was the case between the walls—the vinyl coating acted as a rigid connector when lightly loaded. The two gaps were cleared of the board, and the windows were re-installed.

Measurements were again made. The specs were impressive, but there was still a bit of transmission. A mechanic's stethescope confirmed that there was still a slight amount of coupling. The glass was again removed, and the two spaces once again checked. The only thing that connected any of the three walls together was the pile carpet that covered the gaps, so the carpet was cut, and a special wood moulding was used to provide a visual cover. A small air gap exists between the overhanging wood lip and the carpet on the other side. Thus, there is virtually nothing connecting the three wall systems together. When the windows were once again installed and measurements made, the system now met the transmission loss requirements.

CONTROL ROOMS

Takashi had also specifically defined the acoustic criteria for the control rooms and studios. The two control rooms, of course, had to be identical. They had to accept the wide variety of speakers which at some future date Yamaha might want to analyze. It was important that the machine area could be closed off from the rest of the control room. But at other times, the machine area should open directly into the control room for those engineers who not only want to see the tape machines running, but need to have quick access to them. To satisfy both criteria, the machine area to the rear of each control room was designed to be closed off with large folding doors that extend the full width of the control room. Long double-layer glass was installed in each door so that visibility is maintained. When in place, adequate attenuation of machine noise is achieved, while low frequencies from the control room can pass through the machine area which acts as a bass trap.

True to the intent of the project, a newly-designed four-way speaker system is installed in control room A. All the Yamaha crossovers and amplifiers are mounted in an air-conditioned rack at the back of the machine area. (Both stock and prototype units are being used.) A Solid-State Logic console was selected for this room. The decision to purchase this particular board was based on the feeling that the Solid-State Logic design would most likely be adaptable to the increasing digital interface requirements.

In addition to the same main monitor speaker system above the window, control room B used UREI 813 speakers at both sides of the window. Because of the small size of the window between control room B and studio B, this gives the added bonus of having sound localization at ear level.

STUDIO A

Studio A was designed as a state-of-the-art multi-track recording space. Two mid-bass slat absorbers cover most of the two outside walls of A and use varied sizes and spacings of wood planks. Behind the slats are 8-to-12-inch deep insulated cavities. These two wall surfaces give good bass absorption and, at the same time, high-frequency reflection and diffusion. Tucked in the walls on both sides of these two surfaces are absorptive panels that can be slid on tracks to any place in front of the slat absorbers. With panels in position, the reflection from the wood absorbers is eliminated, and since the panels are thin, the amount of bass absorption remains constant.

The corner areas where these panels retract also serve as bass traps. In keeping with the conservation-of-space theme, some of the other bass traps were also designed to double as storage closets. It was George's feeling that there is little acoustic advantage to completely packing a bass trap with fiberglas, so the valuable space it occupies can also serve other purposes.

The rest of the acoustic interiors are made up of a combination of three different types of absorptive panels. (These are essentially the same as those used by John P. Edwards in a number of his studio designs.) Type C is fairly broad-band but does not absorb mid and low bass. Type B is reflective at higher frequencies, but absorptive at lower frequencies since it acts as both a panel absorber and a Helmholtz resonator. Type A is exactly like B, only the panels are positioned with the soft fiberglas facing into the room. It provides both low-and high-frequency absorption, since the top end is absorbed in the exposed fiberglas, while the bass continues on through, to be dissipated by the perforated-masonite panels. All of the panels were covered with felt and are indistinguishable once mounted.

How many of each type panel, and where placed, determines the characteristics of the room. If changes become necessary, one type of panel can be replaced with another, since they are all removable after installation. The panels were also made in a variety of sizes so they could be used wherever needed in both studios.

The studio A ceiling is totally non-parallel to the floor. This was difficult to achieve, but it was Peter Creamer who finally came up with an approach that is both acoustically acceptable and visually exciting. The main area of Studio A is covered with hardwood parquet and as an option, a carpet may be put down if a deader room is desired.

The room also has two sound chimneys for guitar and bass guitar miking. One of them is located near the acoustically-damped area of the room, and another at a rather-lively corner of the studio.

The soffit above the window in the studio is covered with grill cloth to create another space for acoustical adjustment. This is also where the air conditioning supply and plenums are placed, along with a pair of Yamaha NS1000 speakers.

Between all the rooms, acoustic double-doors and sound locks were used, with the exception of the two isolation booths. Interleaving glass sliding doors made by Arcadia Manufacturing were used for separating these areas. Many designers feel these particular units are the best glass sliding doors for acoustic application. They seal very well and feature a threshold which can take heavy machinery being rolled over it without the tracks being damaged.

STUDIO B

Since studio B was built for product evaluations and other experimental usage, the interior design is a bit less exotic than A. Again, walls are faced with three types of acoustic pancls, and a wood mural with a large slatted bass absorber at one end is similar to that in studio A. The ceiling treatment is a rather conventional suspended ceiling with a great deal of fiberglas absorption above it.

The room is also equipped with a panel that slides up and down in front of the window to control room B. This can be slid down to close off the visual connection between the audio and the control room, when different programs are occurring simultaneously in these two areas.

OTHER CONSIDERATIONS

Needless to say, the air conditioning had to be more than quiet. It had to be silent. Five separate systems, on separate slabs, were used; one for each of the studios and control rooms, and one for the rest of the building. Bruce Walker, special consultant for air conditioning and noise control, made suggestions on the original plans. All of the machinery is decoupled from the concrete slabs, and very large and flexible ducting was used throughout.

The lighting for the studios and control rooms is extensive and can be adjusted at a multi-dimmer panel situated toward the side of each control room. All of the lighting, service lines, and isolated power systems were designed for minimum interference with the audio systems, and there is a large single-point ground system.

The audio circuitry installed by Carl Yancher and Steve Fouce of Lakeside Audio Associates was equally thorough. There are five duplicate input/output panels in studio A, four in studio B and one in each of the isolation booths. Each panel has 48 mic inputs, speaker outputs from A and B, cues from A and B, and a blank portion for future expansion. Tie lines were made between studio A and B, with 24 mic lines, 48 audios, 120 controls, and video lines. The lab has panels from both studios for special experimental purposes. The reverberation systems are housed in the attic space above the office, and can be used by both A and B with an electronically-switched priority circuit.

Yamaha's substantial effort, made into a long-awaited research facility, has finally become a reality.

It is likely that years ahead will see this studio gain a reputation built on its quality sound, as well as the new products that are developed through new information and discoveries made there. ■

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JOHN BORWICK

Scene From Europe

APRS Digital Get-Together

Two meetings of APRS members led to face to face discussions between parties representing all sides of the digital industry.

BRITISH RECORDING studios and engineers have often had good reason to be grateful that the pioneers in this mushrooming industry banded together to form a Trade Association nearly 30 years ago. The Association of Professional Recording Studios (APRS) was inaugurated in 1951 and now has about 150 member studios. From the outset, the APRS aimed at fostering good relations with the manufacturing side of the industry, and so there was a "Manufacturing Member" category. At first, this was comprised of just a few custom pressing plants, but later on, other equipment manufacturers saw the advantages of APRS membership. Today, about 60 of them have joined, plus various educational studios and overseas organizations.

One obvious APRS activity which clearly benefits all its members is the Annual Exhibition of Professional Recording Equipment held in London each June. (See Reports in **db** Magazine, September 1978, September 1979.) Other helpful activities have included the preparation of calibrated alignment tapes, guidelines for the leadering and labelling of tapes submitted for disc transfer, a standard form of Artists Contract, legal advice, joint-venture participation in overseas exhibitions, training courses in basic studio engineering, digital techniques and computers, and the sponsorship of a wide-ranging handbook "Sound Recording Practice" (published by Oxford University Press, second edition price $55.00). From time to time, the APRS has also set up meetings and forums on subjects of high topical interest, and the most recent of these have been related to digital recording.

John Borwick is **db**'*s British correspondent.*

NEW WORKING PARTY

The users—present and future—of digital recording equipment are at least as anxious as the design engineers for the industry to go digital in worldwide harmony. While a single studio can go it alone with any of the digital machines now on offer (on sale or lease), the interchange of masters and completed programs surely demands a measure of agreement on the main digital parameters. At the same time, the final shape and performance specifications of future domestic software and hardware units must be taken into account when the professional decks, mixers and transfer devices are being designed and operated.

The need for studio users to make their voices heard on this burning question prompted the APRS (which already acts as the British industry's spokesman in discussions with government and other official bodies) to convene an exploratory co-ordination meeting back in February of 1980. This took place during the AES London Convention and was conducted by an independent Chairman, Dr. Richard Helyer of Digitalent, London. More than 40 people representing all sides of the industry attended, either as observers or participants. Protagonist organizations represented included the BBC, Polygram, MCI, EMI, Ampex, Studer, 3M, Decca, Neve, Sonopress, Sony and BASF.

Part of the background to this meeting had been the frustration felt by some European representatives when they had been prevented from bringing this problem fully out into the open at the November 1979 AES Convention in New York. The present attitude of the AES is that they should not call meetings specifically to discuss *standards* for new technological developments. The fact that such meetings could be construed as running contrary to US Anti-trust Laws had been pointed out, for example, by Stephen F. Temmer, President of Gothan Audio Corporation. His view was that, "Standards have their important place in all areas. They also should be kept out of areas which have not yet reached the marketplace; i.e. while innovation is still in progress and competition is needed to advance the state-of-the-art" (letter published in *Recording Engineer/Producer*, April 1980).

The February meeting discussed at some length the several digital systems now in existence. It was decided to set up a Working Party to review what action should be taken to resolve the problem. It was also decided that the APRS should act as Secretariat for the time being.

WORKING PARTY MEETS IN BRIGHTON

The first meeting of the Working Party was duly organized by the APRS on September 22nd, 1980 in Brighton, during the International Broadcasting Convention. All interested parties had been invited to attend, and indeed the register of those attending read like a "Who's Who" of digital recording. The Chair was again taken by an independent engineer, Hugh D. Ford.

Björn Blüthgen (Polygram, Germany) put forward the view that studio members should be equipped with each of the competing digital systems before any useful assessment of their relative merits could be made—followed by the hoped-for feedback to manufacturers of features most desired by practicing recording personnel. Blüthgen handed out copies of a "User's Questionnaire" which he had dispatched to a wide list of manufacturers and potential users back in January. Responses so far received indicated a wide support for the idea that some harmonization, if not standardization, of essential tape track/signal format parameters was highly desirable. (Herr Blüthgen's questionnaire has since been printed in the *Journal of the AES*, September 1980 issue, along with a report of the May 4th meeting of the AES Digital Audio Technical Committee, held during the AES West Coast Convention in Los Angeles.)

Dr. Toshi T. Doi (Sony, Japan) distributed sheets outlining the format jointly agreed to by Sony and Stüder (Switzerland) for stationary head digital audio recording. On the vexed question of sampling rates, this document suggests three nominal rates: 50.4 kHz for highest quality studio recording, to provide a safety margin and maintain 20 kHz bandwidth on machines having ±10 percent pitch control (said to be a studio requirement for special effects); 44.1 kHz for other studio applications, digital audio discs and machines based on PAL or SECAM video tape recorders; 32 kHz for broadcasting etc., in accordance with the EBU (European Broadcasting Union) Standard.

Guy McNally (BBC, Great Britain) reported that the BBC was working on a general standards converter, on the assumption that 32 kHz, plus at least one other sampling rate, would be needed in the future. F.A. Griffiths (Decca, Great Britain) agreed that a standards converter was already a much-needed item if the plurality of machines now in studio use were to be exploited to the full. He pointed to the need for flexible editing facilities and that these when designed properly, could give studio engineers better editing control than the old analogue techniques.

Thomas Bermingham (3M, USA) stressed that he and his colleagues were attending as observers rather than Working Party participants, pending a ruling on the U.S. Laws situation. He declared his company's keenness to secure the maximum feedback from studio users, with 30 machines already in the field. The 3M view was that, "While adoption of a universal machine format remains a more distant possibility, establishment of a signal standard at an early date would represent a major step forward." The sampling rate proposed in the 3M document was 50 kHz. While this rate could be converted for compatibility with film, TV, and other systems, other parameters were at variance with the Sony proposals.

This led to a general discussion on such basics as the minimum audio bandwidth needed for high quality reproduction of music. Several speakers felt that 15 kHz was adequate, which made a sampling rate of 32 kHz both an economic and a subjectively-acceptable value. Others felt that 20 kHz should be regarded as a minimum, even though this would increase costs, and possibly delay the wide acceptance of a domestic playback format. Björn Blüthgen reported that he had offered (at the May AES Convention) to produce a tape with and without 15 kHz filtering. He had used a filter which was a colossal 140 dB down at 16 kHz; yet, when proper attention was paid to group delay distortion, even "golden ears" found it difficult to identify which portions of the tape were filtered.

The meeting chairman, Hugh Ford, made it clear that full cooperation with the AES was essential. There seemed little doubt that this cooperation was guaranteed by the large number of those present who would also be at the next AES Committee meeting during the New York Convention (October 31 to November 3, 1980).

LONDON, 28th NOVEMBER 1980

Almost the same line-up of about 40 interested parties attended the next meeting of the Working Party in London on 28th November, 1980. One change was an increased attendance by studio engineers, so that the user's point of view was given more prominence.

Editing of digital recordings was discussed at considerable length. While several engineers had found that digital editing, on any of the systems currently in use, took a little longer than traditional cut-and-splice methods, they had been able to achieve phenomenal accuracy. The possibility of introducing a controlled, and rehearsed, degree of cross-fade at the editing point was also useful. The point was also made that digital machines were mechanically more noisy and distracting than analogue recorders, particularly during frequent search and play operations, and often had to be located outside the control room for this reason.

On the question of cost, smaller studios felt that a full digital installation was still prohibitively expensive. The 3M representatives confirmed that a complete package with editor would cost something approaching £100,000. The figure of £10,000 quoted for the ¼-inch Mitsubishi linear recorder with cut editing was a cheaper proposition which might appeal to smaller studios. As for ultimate sound quality, the Soundstream system came in for high praise, but it was a limiting factor that a producer was obliged to visit Salt Lake City to supervise the editing. It was noted that Soundstream was planning to set up editing facilities in Europe and other US centers during 1981.

The next Working Party meeting was scheduled to take place in the Connaught Rooms, London on January 30th, 1981.

APRS STUDIO GUIDE

The APRS has compiled an up-to-date directory of its member studios. The 74-page guide lists each member in alphabetical order, and provides details of studio dimensions, recording equipment and other facilities. Where appropriate, an indication is also given of the studio's expertise in areas other than conventional music sessions, such as radio commercials, speech recordings and audio-visual presentations.

Currently, APRS membership has been extended to some 180 studios. The majority are based in Great Britain, but to date just over 20 European studios have also become APRS members. The newly-published guide will be used to promote the activities of the Association of Professional Recording Studios and its growing membership at important exhibitions around the world.

Copies of the directory will also be sent to record companies, producers, advertising agencies and other similar organizations who regularly book session time at both large and small recording studios.

Copies of the new Guide to APRS Member Studios are available from The Secretary, Edward Masek, 23 Chestnut Avenue, Chorleywood, Herts WD3 4HA. Telephone: Rickmansworth 72907.

The next APRS Exhibition will be held on June 10-12, 1981 in a new venue, the Kensington Exhibition Centre, Kensington High Street, London. Inquiries should be directed to the APRS Secretary. ■

ROY KOMACK and BRIAN MORIARTY

The Bose® 802 Professional Loudspeaker System

An alternative to the compromises of conventional technology.

THE PAST DECADE has seen significant advances in the science of professional sound reinforcement. Increasing control sophistication and improved electronic circuit design have enhanced the realism and expanded the creative potential of "live" sound production.

Yet, speakers are still the same: big, heavy, difficult to install, expensive to move around. Loudspeaker arrays often dominate the visual aspect of live performances, and that artificial, ear-fatiguing "horn sound" is a haunting reminder that the quality of live sound reinforcement is not yet as lifelike and natural-sounding as many fine home music systems.

This is not to say that loudspeaker design has not progressed. But the progress has been in the areas of conversion efficiency and durability, rather than user convenience or naturalness of sound. The problem lies in the continued application of traditional design criteria, adapted from the primitive technologies of the past. Designers have neglected the practical need of installers and performers by perpetuating "tried and true" ideas and materials. Consequently, the industry has learned to live with the inadequacies of available technology, accepting them as fundamentally inevitable and even *desirable*. Thus, we still hear that large speakers automatically deliver bass response, and that horns and multi-way systems arre the only way to get proper performance.

SOME SYSTEM LIMITATIONS

There are limits to what can be achieved with the traditional tools of loudspeaker design. For example, multi-way systems and large wood enclosures are inherently unsuitable in many real-life situations, and their use entails important performance compromises in *any* situation.

Multi-way drivers. Dividing the audible range among a number of specialized drivers was the "only way to go" in 1930. But modern physics and psychoacoustics have uncovered a number of drawbacks to this approach.

For near-field listeners, the physical separation of low-, mid- and high-frequency drivers can produce a distinctly unnatural "spatial discontinuity" effect. Varying dispersion patterns create difficulties with coverage, feedback control and

Roy Komack is Professional Products marketing manager and Brian Moriarty, technical writer for Bose Corporation, Framingham, Massachusetts.

1. A pair of Bose 802 professional speakers. The SS-3 speaker stand is easily folded for storage and transportation.

2. Eight full-range helical voice coil drivers are mounted on four angled facets for smooth dispersion.

3. The 802 helical voice coil. The axial gap in the bobbin eliminates eddy current losses.

wandering stereo localization. Add to this the complexity and unsightliness of typical stacked arrays and the uncertain effects of time incoherence, and the problems of multi-way systems often outweigh the advantages.

Crossover networks. The evils of multi-way systems are compounded by their associated crossover networks. Response coloration, phase shift and/or power losses are unavoidable in even the most sophisticated passive networks. Some of these effects can be minimized by using active crossovers and multiple power amplifiers, but only at greatly increased system cost and complexity.

Wood enclosures. Materials for speaker cabinets should be selected for their rigidity and freedom from resonance, light weight, ease of fabrication and durability, *not* for historical or sentimental reasons. There is no inherent advantage in using wood for a sound reinforcement speaker enclosure. In fact, the only way to construct an acceptably rigid and durable wood enclosure is to sacrifice ease of installation and portability.

Horn Systems. The off-axis frequency response of horns is rarely uniform. Sharp "skirts" in their dispersion patterns make it difficult to achieve good results in areas where one horn takes over from another. These and other factors lead to that characteristic "horn sound."

This brief overview illustrates some of the severe restrictions imposed by traditional speaker designs. No amount of tinkering can completely overcome these basic disadvantages of size, weight and sound quality. It is virtually impossible to produce a truly practical, natural-sounding professional loudspeaker system with conventional technology.

AN ALTERNATE APPROACH

In 1962, a U.S. Patent was awarded to M.I.T. Professor Amar G. Bose for a loudspeaker system consisting of an electronically-equalized array of small, closely-spaced full-range drivers. The faceted front baffle and close spacing of the drivers created a natural, lifelike dispersion pattern without the spatial discontinuity of physically separated multi-way driver arrays. The radiation pattern simulates the broad dispersion characteristics of human voices and musical instruments more closely than traditional speaker designs, which concentrate the sound in a narrow beam. Because the speaker was designed for far-field listening environments, the issue of time coherence (which has become prominent in the studio environment) loses its importance. This concept was introduced to the professional sound reinforcement industry in 1972 with the Bose Model 800 speaker. The pentagon-shaped 800 system utilized eight 4½-inch drivers mounted on two front facets, with a separate active electronic equalizer controlling the frequency response of the system to achieve a flat total radiated power spectrum. Its compact design and lifelike sound made it popular among small performing groups and in various sound reinforcement installations.

Further study suggested that the ruggedness and portability of the 800 speaker could be substantially improved if the wood enclosure was replaced with some more suitable material. And, some additional refinement of the drivers and the equalization system would make possible an entirely new speaker, which will be described below. Because of these enhanced value-and-performance characteristics, the 800 system was honorably retired and, in 1978, the Bose 802 Professional Loudspeaker System appeared in its place.

There are no woofers, tweeters, midrange drivers, crossover networks or wood parts in the system. Like its predecessor, it contains a matched array of eight 4½-inch full-range drivers. Each driver employs a one-piece, forged backplate/polepiece and a high-energy ceramic magnet that is 20 percent heavier than the Model 800 driver magnet. This extra magnet weight improves sensitivity by approximately 0.5 dB. The front plate is permanently molded into the driver frame, which is made of glass-reinforced thermoplastic polyester. Its non-magnetic composition significantly reduces flux leakage, thereby adding another 0.5 dB to the sensitivity. Changes in the surround system and the shape of the driver cone increase the effective piston area by about 20 percent, for an additional 1 dB of efficiency.

The Model 800 voice coil was completely redesigned for the 802 system. Fabricated entirely from aluminum, the new coil consists of flat ribbon wire wound edgewise on a one-inch diameter bobbin. The critical winding operation is performed by custom automated equipment which holds dimensional variations to less than 0.001 inch. The coil's edgewound configuration increases the amount of wire in the magnetic gap by 50 percent, for a sensitivity improvement of 2 dB. The use of aluminum lowers the total coil mass by 10 percent, enhancing the transient response and adding another 0.5 dB to the efficiency. DC resistance is also lowered by 10 percent, improving the sensitivity by yet another 0.5 dB. Taken together, these design refinements yield a net sensitivity increase of 5 dB over the Model 800 driver. Equivalent SPLs can therefore be achieved with only one-third as much amplifier power. Alternatively, the wide dynamic range and durability of the 802 system allows it to radiate two to three times as much energy as its predecessor. In addition, the change from a nylon to an aluminum bobbin, and from conventional enamel to an advanced high-temperature insulating system, increases the power handling capacity of the driver from 10 watts to 20 watts continuous, a 100 percent improvement.

FULL-RANGE SOUND FROM "MID-RANGE" DRIVERS

As in the Model 800 speaker, the 802 drivers are mounted very closely to each other to take advantage of the "coupling effect." Because the driver spacing is small compared to the wavelength of any bass frequency, the drivers act together as if they were a single transducer of cone area equivalent to the sum of the eight individual cone areas plus all of the area between and among the eight cones. The *effective* cone area is therefore about the same as that of a 15-inch woofer.

Wide-range driver design is always a compromise between response extension and efficiency. A simple unassisted driver or array of drivers capable of flat response over the entire audible spectrum (supposing such a device could exist) would be grossly inefficient; any measures taken to improve the efficiency would degrade the response. For this reason, no attempt was made to give the 802 driver an inherently flat frequency response. It was instead optimized for the highest possible efficiency, power handling capacity and durability. The response of the 802 system is broadened and "flattened out" by a separate line-level active electronic equalizer, which incorporates two identical 10-pole processing channels (for stereophonic applications). There are no controls other than a power switch; its fixed equalization characteristics are the inverse of the unequalized response of the 802 driver array (see FIGURE 4). Twelve dB-per-octave infrasonic and ultrasonic filters are included to reduce power waste, slew rate distortion and high frequency interference. The 802 equalizer's detailed response curve takes into account the combined effects of driver response, enclosure characteristics, even the absorption of the grille material. The unit is usually inserted into an audio system between the pre- and power amplifiers, or into the effects loop of integrated mixer/amplifiers.

The frequency response of loudspeakers is usually designed to be flat when measured on-axis in an anechoic chamber. This practice is only appropriate for studio monitor speakers because it assumes that the listener will be located directly in front of the speaker in the near field. In virtually every other listening situation, the dominant sound energy comes from the reverberant field, and what the listener perceives is a function of the average frequency response at all points on an imaginary sphere surrounding the loudspeaker system. This is known as the *total radiated power spectrum* of the system. Speakers designed for flat on-axis response may "look better" on paper, but do not sound realistic because they do not radiate enough high-frequency energy into the reverberant field. For this reason, the Bose 802 loudspeaker is designed for *flat total power radiation*, which more closely corresponds to the way frequency response is actually perceived in real environments.

IMPEDANCE CONTROL NETWORK

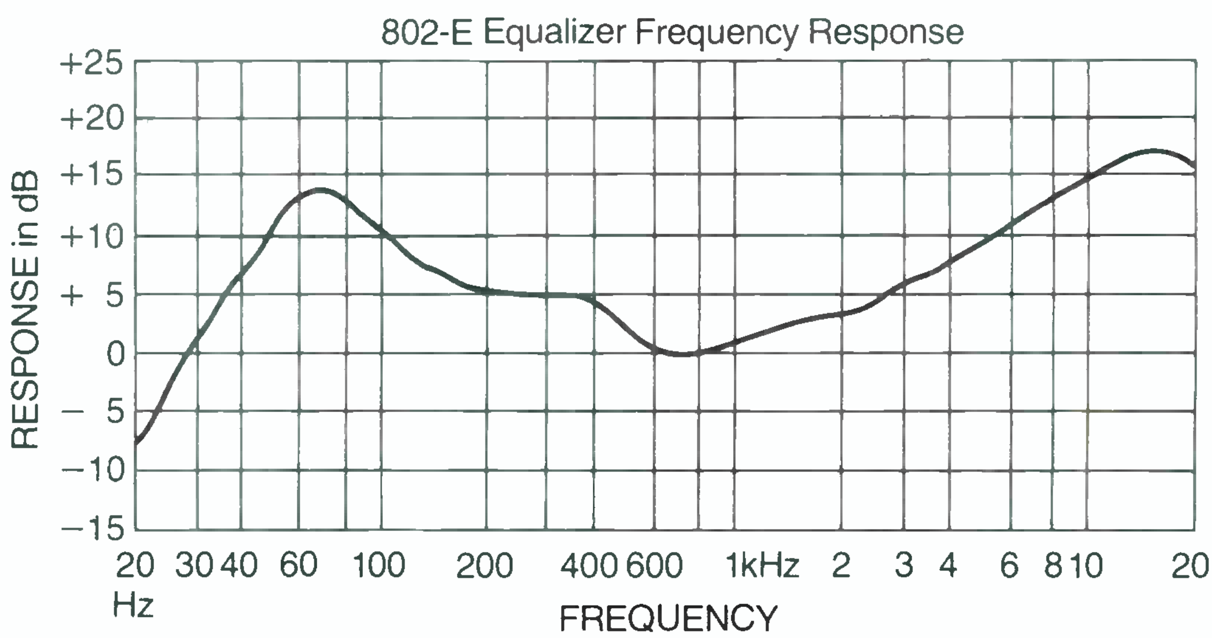
Bose engineers ran into an unusual problem in determining the proper equalization curve for the 802 system. It was found that the prototype eight-driver array exhibited an undesirably-high impedance in the region above 8 kHz—so high that it required more than 22 dB of compensating boost at 16 kHz to achieve accurate response. This amount of equalization would have used up too much amplifier headroom and caused possible instability problems. So a special impedance control network was developed to tame the rising high-frequency impedance characteristic of the array.

The network consists of a 10-microfarad capacitor wired in series with a current-sensitive protection device similar to an incandescent lamp. This circuit is connected in parallel with four of the eight drivers in each enclosure. Under normal signal conditions, the capacitor partially bypasses these four drivers so that the high-frequency impedance of the system is lowered. If presented with a level of high-frequency energy that would endanger the other four drivers, the impedance of the protective device rises, effectively removing the capacitor from the circuit and allowing all eight drivers to share the load. This simple network substantially reduces the equalization requirement and also protects the speakers from the high frequencies generated by amplifier clipping or current limiting.

COUPLING CONTROL INSERTS

A prominent feature of the Bose 802 loudspeaker is the dual reactive air column system, designed to reduce excessive cone excursion at low frequencies. This "portlike" system, tuned to

4. The equalization characteristics of the 802-E active equalizer are accurately matched to the drivers and enclosure of the speaker to produce a flat total radiated power spectrum.



55 Hz, provides most of the energy radiation in the bottom octave of the musical spectrum. Since the driver cones do not have to travel as far, low-frequency distortion is considerably lower than in sealed-box speaker systems. This resonant system improves the bottom-octave sensitivity by about 1 dB.

Prior to the development of the 802 system, there was no method of stabilizing a ported multiple-driver speaker system. The problem was that slight efficiency differences among the drivers caused them to force each other out of phase during powerful bass passages. The air columns incorporate a unique solution to this problem.

Each air column is loaded with a thin disc of polyester batting (a material of randomly-oriented fibers) which slightly obstructs air flow in the columns. These coupling control inserts lower the Q of the enclosure vent system and decouple the drivers from one another at resonance. Both column assemblies are easily removable without tools so that the inserts can be changed if they become clogged over years of operation in dusty environments. This convenient and completely airtight method of user-replaceable vent damping was recently awarded U.S. Patent Number 4,180,140.

STRUCTURAL FOAM ENCLOSURE

Perhaps the most unusual and innovative component of the system is the enclosure itself. Seasoned professionals shake their heads and mutter unkind things about plastic when they see it. But when they lift a speaker in each hand without turning red, the advantages of structural foam in loudspeaker fabrication become very apparent.

The enclosure is composed of high-density polyethelene copolymer, reinforced with 10 percent mica for improved ruggedness and impact strength. It is manufactured by structural foam molding in three separate pieces: main case, back panel and cover. Interior surfaces of the case and back panel are heavily ribbed to eliminate vibration, and assure high mechanical rigidity. After system wiring, the back panel is permanently fused to the case by the EMAbond RF welding process for a perfect airtight seal. The one-piece removable cover features a deep integrated carrying handle, space for various mounting accessories and spare parts, plus a storage compartment for the 802-E Active Equalizer. A system of rails and grooves on the enclosure allows it to be safely stacked on top of other enclosures or cradled in its own cover at different angles for use as a stage monitor (foldback) speaker.

The Bose Professional Products Division has prepared a comprehensive Engineers' and Architects' Design Guide which reviews some of the basic sound reinforcement system design criteria. In addition, the guide documents the physical and acoustical characteristics of the Bose system. Direct sound field contour maps and design equations may help the system designer predict and control maximum SPL, coverage and articulation loss for various combinations of stacking, placement and amplifier power. (Excerpts from the Design Guide are presented at the end of this feature—Ed.)

IN CONCLUSION

The time has come for loudspeaker designers to become as open-minded and innovative as the rest of the industry. The enormous potential of new materials and production techniques should be turned loose on this rather hidebound segment of the audio community. The improvements it is possible to make in this area can be heard and appreciated by anyone, while so-called "breakthroughs" in the minutiae of amplifier and mixer operating specifications offer little, if any, real benefit to listeners. Growing public interest in noise reduction techniques, digital recording, and motion picture sound is a powerful indicator that audiences can discriminate between "live" and "canned" sound and *care* about the difference. Manufacturers must be willing to meet this challenge with fresh ideas and bold, innovative products. The Bose 802 loudspeaker system is only the beginning of what will hopefully become a new generation of speakers that eliminate needless compromises in the quality of professional sound reinforcement. ■

Engineers' and Architects' Design Guide

SOUND REINFORCEMENT SYSTEM DESIGN

In designing a sound reinforcement system, three of the most important performance criteria to be satisfied are: maximum achievable SPL, uniformity of coverage, and articulation loss (intelligibility).

DEFINITIONS

AL(%)—
Articulation Loss (in percent). That percentage of consonants from a standard word list that will be misunderstood when the list is spoken through the sound reinforcement system. It is generally agreed that up to 15 percent loss is acceptable, in these context-free lists.

RT_{60}
Reverberation Time. The time (in seconds) required for the sound level in the room to decay by 60 dB, once the input is stopped.

Q
Directivity of the speaker system. The ratio of the sound pressure on axis of the speaker to what it would be if the speaker were an omnidirectional radiator of the same conversion efficiency.

V—
Volume of the room in cubic meters.

S—
Sensitivity of the speaker system. The SPL at one meter on axis with one nominal watt applied to the terminals of the speaker.

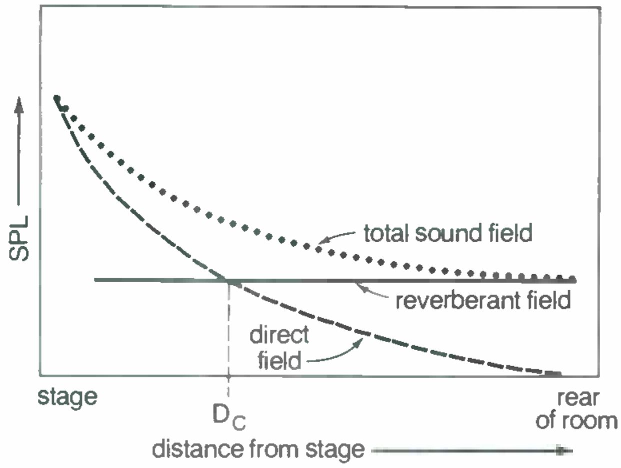
A—
Amplifier power, in watts.

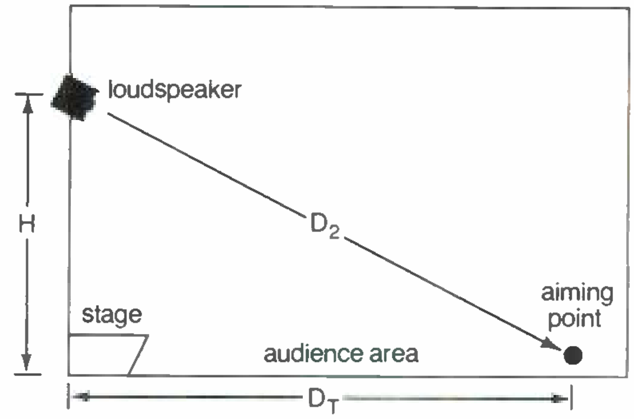
D_c—
Critical Distance. The distance, in meters, from the speaker, on axis, at which the strengths of the direct and the reverberant sound fields are equal.
$D_c = .0565\sqrt{QV/RT_{60}}$

D_2—
The distance, in meters, from the speaker, on axis, to the point at which it is aimed.

H—
The height, in meters, from the floor to the speaker. (H is zero for stage-level speakers.)

D_T—
The "throw" distance. The distance, in meters, from a point on the floor directly below the speaker to the aiming point.





MAXIMUM ACHIEVABLE SPL

First, calculate the maximum achievable sound pressure level at the rear of the room for the direct field and for the reverberant field. At least one of these values must equal or exceed the job requirement.

For the reverberant field:

$SPL_{max} = S - 20 \log(D_c) + 10 \log(A)$.

For the direct field:

$SPL_{max} = S - 20 \log(D_2) + 10 \log(A)$.

If the SPL is not sufficient to meet the job requirement, either the amplifier power or the number of speakers, or both, must be increased. Remember that each doubling of the total system power will increase the SPL by 3 dB.

UNIFORMITY OF DIRECT-FIELD COVERAGE

To determine the ratio of direct-to-reverberant sound throughout the seating area, follow the following procedure:

1. Calculate the critical distance, as described previously under Definitions.
2. Determine the distance from the speaker to the aiming point, D_2. Remember that $D_2 = \sqrt{H^2 + D^2}$.
3. Calculate the difference, in dB, between the direct and reverberant fields at the aiming point by, $SPL(D/R) = 20 \log(D_c/D_2)$.

ARTICULATION LOSS

Before embarking upon the calculation of articulation loss, it is important to remember that intelligibility is *not* the same as fidelity, and that the concept of articulation loss is relevant only to speech reinforcement systems, not to music amplification systems. The human singing voice is to be considered music, not speech; and the traditional approach to minimizing the articulation loss in speech may actually degrade the quality of amplified singing or other music. With this caution understood, we will proceed to the calculation of articulation loss in a speech reinforcement system.

The articulation loss at the aiming point of the speaker is calculated by one of the two following formulas:

$AL(\%) = 0.9\,(D_2/D_c)^2\,RT_{60}$, when $(D_2/D_c)^2 \leq 10$.
$AL(\%) = 0.9RT_{60}$, when $(D_2/D_c) > 10$.

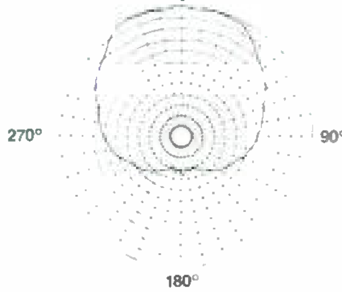
It is generally agreed that the maximum acceptable articulation loss in a speech reinforcement system is 15 percent. If the formulas above yield a value greater than 15 for the rear of the audience, one of two approaches must be taken. One alternative is to use a secondary speaker system (possibly time-delayed) to cover that portion of the audience for whom $(D_2/D_c)^2$ is too great.

The other possible solution is to use a speaker system with a higher Q; for example, changing from a single 802 speaker to a stacked pair. This has the effect of increasing D_c, the critical distance.

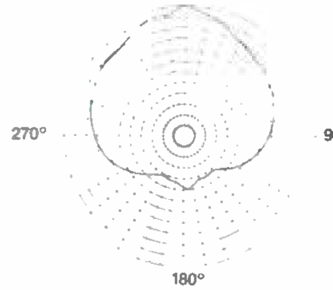
The polar plots below demonstrate the significant increase in projection capability provided by stacked loudspeakers.

NOTE: The complete Design Guide is available on request from the Bose Corporation, Dept. ADG, 100 The Mountain Road, Framingham, Mass. 01701.

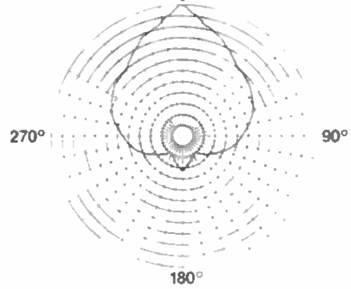
Polar Dispersion Patterns



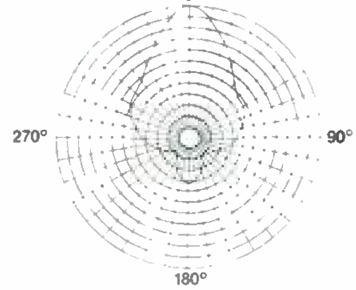
[illegible] Coverage
([illegible] speaker, stacked [illegible]
[illegible] of four
Scale [illegible] dB per division
[illegible]



Vertical Coverage
(single speaker)
Scale 2 dB per division
Bandwidth 300 Hz – 14 kHz



Vertical Coverage
(stacked pair)
Scale 2 dB per division
Bandwidth 300 Hz – 14 kHz



Vertical Coverage
(column of four)
Scale 2 dB per division
Bandwidth 300 Hz – 14 kHz

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People/Places/Happenings

• **URSA Major, Inc.**, of Belmont, MA has announced the loan of a Space Station SST-282 Digital Reverberation System to each of the five schools for use in their audio programs. The schools which submitted successful proposals to the URSA Major School Loan Program were **Fredonia State University College, the University of Iowa, the University of Miami** (Florida), **Purdue University**, and **San Francisco State University**. Starting in January, 1981, each school will keep the Space Station for the duration of a school term or proposed project, and will then document what it has learned about reverberation and signal processing for URSA Major.

• **Michelle Meisner** has become the new studio manager of **The Automatt** recording complex in San Francisco. Prior to joining the Automatt staff, Ms. Meisner worked in national record promotion for **Fantasy Records**. Also new at The Automatt are recording engineers **Susan Gottlieb** and **Maureen Droney**. Susan Gottlieb's experience in engineering includes work with **Vance Frost & Associates**, and a teaching post at San Francisco's Music **Recording School**. Maureen Droney was chief audio technician at **San Francisco Lightworks**.

• Each year the **Audio Engineering Society** makes a Publication Award to authors of outstanding papers published in the Society's Journal. This year's award was made to **Laurie Fincham**, Director of Engineering at **KEF Electronics**, and his KEF colleague **Mike Berman**, for their joint authorship of an archival paper on Digital Measuring Techniques. Entitled "Application of Digital Techniques to the Measurement of Loudspeakers," the paper was published in the AES Journal, Volume 25 Number 6 (June 1977). During the same evening at the Awards Banquet, Mr. Fincham also received the **Board of Governors Award** for his chairmanship of the 65th Convention held in London during the past year.

• **Le Studio**, the first East Coast studio to install the computerized (floppy disk system) **Solid State Logic** Master Studio System, has now expanded to 48 tracks with video interlock. The first special using the video sync, which is obtained by interlocking the multi-track recorders with a **JVC** ¾-in. videocassette, was the **Peter, Paul and Mary** special for the **Canadian Broadcasting Corporation** and **Home Box Office**. In keeping with the fast-developing video cassette and videodisk market, Le Studio has a room for video cassette editing, using the JVC ¾-in. system for post-production pre-editing on TV specials and videodisks. Currently recording at Le Studio are **Anthem** recording artists **Rush**. Rush is recording the follow-up to their successful Permanent Waves LP which was also recorded at Le Studio.

• **Aphex Systems Ltd.**, developers and manufacturers of sound enhancing equipment for the professional audio and music recording industry, has signed an agreement with **AKG Acoustics**, of Vienna, Austria, for the European firm to market its professional and consumer audio equipment. The international marketing agreement was announced in Los Angeles by **Marvin Caesar**, president of Aphex Systems, and executives in Austria of AKG Acoustics, which develops, manufactures and markets professional and consumer audio equipment. AKG will begin to market six different sound enhancing audio products manufactured by Aphex in the broadcast, music and consumer markets in Austria, Germany, England and Africa. The agreement also calls for AKG to market Aphex products in Eastern Europe, including Russia, Hungary, East Germany and Czechoslovakia. Among the first professional audio equipment AKG will sell will be the Aphex Aural Exciter for recording studio and broadcast application; Equalizers, for professional musicians and recording studios; Compressor-Expanders; and "The Grouper," for recording studios.

• **Gerry Block**, well-known studio engineer and inventor of the **Compudisk** disc mastering computer, has been appointed general manager of the New York recording complex of **Sigma Sound Studios**. The move was announced by **Joseph D. Tarsia**, president.

• The Casino of Montreux, Switzerland which annually stages the famous **Montreux Jazz Festival** will acquire a permanent and complete sound reinforcement and stage monitoring system. The new system, which is made up entirely of **Electro-Voice** components, will be installed by December 31, 1980. The Casino's sound reinforcement and stage monitoring system will be almost identical to the 1980 Jazz Festival installation. It will include nearly 70 E-V speakers and speaker systems, E-V and **White** equalizers, **BGW** and E-V/TAPCO amplifiers, **Yamaha** and E-V/TAPCO mixing consoles, an E-V electronic crossover equalizer and a **Klark-Teknik** digital time delay.

• During October 1980, the Working Group on Digital Control Interface under the **SMPTE** banner finished a successful field test of the proposed SMPTE standard for control of television studio equipment, it was announced by SMPTE Engineering Vice President **Roland J. Zavada, Eastman Kodak Co.** A progress report prepared by **G. Little, Ampex Corp.**, was presented at the recently concluded 122nd SMPTE Technical Conference in New York City. The documents involved a proposed ANSI standard and two SMPTE Recommended Practices which are well along in their preparation. This control technique is expected to have a broad application in the television, film, and audio areas.

• **Soundcraft, Inc.** subsidiary of **Soundcraft Electronics, Ltd.** of London, announced the recent installation of a Series 1624 24/16 recording console by **Opryland Productions** of Nashville, Tennessee for use in their post-production center.

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