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JULY 1984 VOLUME 18 NO. 6

FEATURES

COLUMNS

DEPARTMENTS

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Letters

FOR THE RECORD

TO THE EDITOR:

It has come to my attention that there may be some confusion arising from my article "Audio/Musical Design For Animated Shows: Part II. The Bullwinkle Experiment" (March '84). To illustrate my point that dimensional animation has its roots in cartoon animation, I described a restaurant show that, indeed, grew from a cartoon show: Bullwinkle's.

For the record, my involvement was with the original designers and in the creation of the *prototype* system. Within a year this prototype was replaced with a production version manufactured by A.V.G. Productions and Paramount Sound Systems. This version is currently in operation in all Bullwinkle's Restaurants. I hope none of your readers have mistakenly visited a Bullwinkle's restaurant specifically to hear Aural Image Animation™ in operation. This was only possible during the first year.

JIM WELLS

MORE FOR THE MONEY

TO THE EDITOR:

We very much appreciated John Borwick's factual article, "An AKG Update." To the chapter describing "The Tube" we wish to add that these units are practically handcrafted and, including the remote pattern selector, power supply, suspension, cables, connectors and flight case, cannot possibly be purchased for "around $1,000." To be exact, the Professional User Net price for the complete system is $1,700.

This price, by the way, is less than the cost for a "trusty 30-year-old AKG C-12" on the used equipment market. We have never subscribed to the "whatever the market will bear" philosophy.

ANDREW BRAKHAN

Index of Advertisers

About The Cover

ABOUT THE COVER

• This month's cover shows the legendary Edgar Bergen and Charlie McCarthy entertaining a young fan. The microphone is a Unidyne I model #556B, which began production in 1938 and was discontinued in 1946.

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GLASS IN THE STUDIO

To The Editor:

Judging by its Part I (db, April 84) F. Alton Everest's article on sound-leakage through studio windows is to be highly commended.

It is a timely review of a design detail that becomes very critical when tight budgets and accelerated deadlines leave no room for error, which is almost always. The demand for early studio completion seldom leaves time for experimentation.

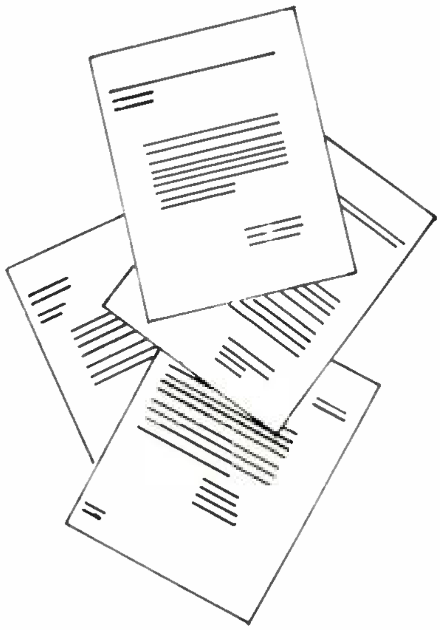
There is one variation that 50 years of involvement did not let me try. So I am most interested to see if it will be mentioned in Everest's Part II.

The polycylindrical studios of 30-odd years ago made excellent use of the sound-dispersion characteristics of convex surfaces. So why not use similarly curved glass in studio windows, both to eliminate plane reflections, and to provide a degree of arched-barrier rigidity that flat glass cannot equal?

The idea encountered non-availability of suitably curved glass on the few occasions in the past when a test would otherwise have been feasible. So it remains an enticing idea with no empirical data to support it, and as far as I know, with no collected data to permit theoretical analysis.

The apparent superiority of the curve still fascinates me. It will continue to do so until specific information becomes available on its practical value in studio design.

J. K. Gordon



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KEN POHLMANN

Studer Digital: A Prototype Recorder

• Digital technology is proliferating throughout the audio industry. Since the introduction of digital technology into the recording studio in 1971, under the gise of the digital delay line, most facets of the industry such as acoustic design and testing, signal processing and enhancement, musical electronic instruments, automation, synchronization, recording, and consumer reproduction are increasingly using digital devices to accomplish their tasks. It is safe to say that most of the basic audio applications have already been impacted by digital technology. However, the success of digital technology has quickened growth of both the sophistication of its products and the scope of their market. In fact, a curious phenomenon is taking place as digital technology begins to redefine the nature of the audio industry's practices. Not only have existing products been re-designed with digital techniques, but entirely new products have been developed, such as the digital audio processor and sampling frequency converter. Thus a second wave of digital design is beginning to appear with new concepts, proposals, and products whose ideas have originated independently of pre-conceived analog design notions. Digital audio design is emerging in its own right.

One company long noted for never emerging with its designs until they are right is Willi Studer International, AG, Regensdorf, Switzerland. Although Studer has devoted considerable resources to studying digital audio technology from the very beginning, it is only recently that the company has chosen to actively enter the marketplace. This is in complete accordance with the company's stated policy and known reputation for only offering fully consolidated products, a practice which successfully combines pragmatism with high technology, and conservatism with engineering innovation.

Dr. Roger Lagadec, with other Studer engineers, has recently authored three papers for the Audio Engineering Society which clearly state Studer's philosophy and inten-

Figure 1. Studer 2-track digital prototype recorder.

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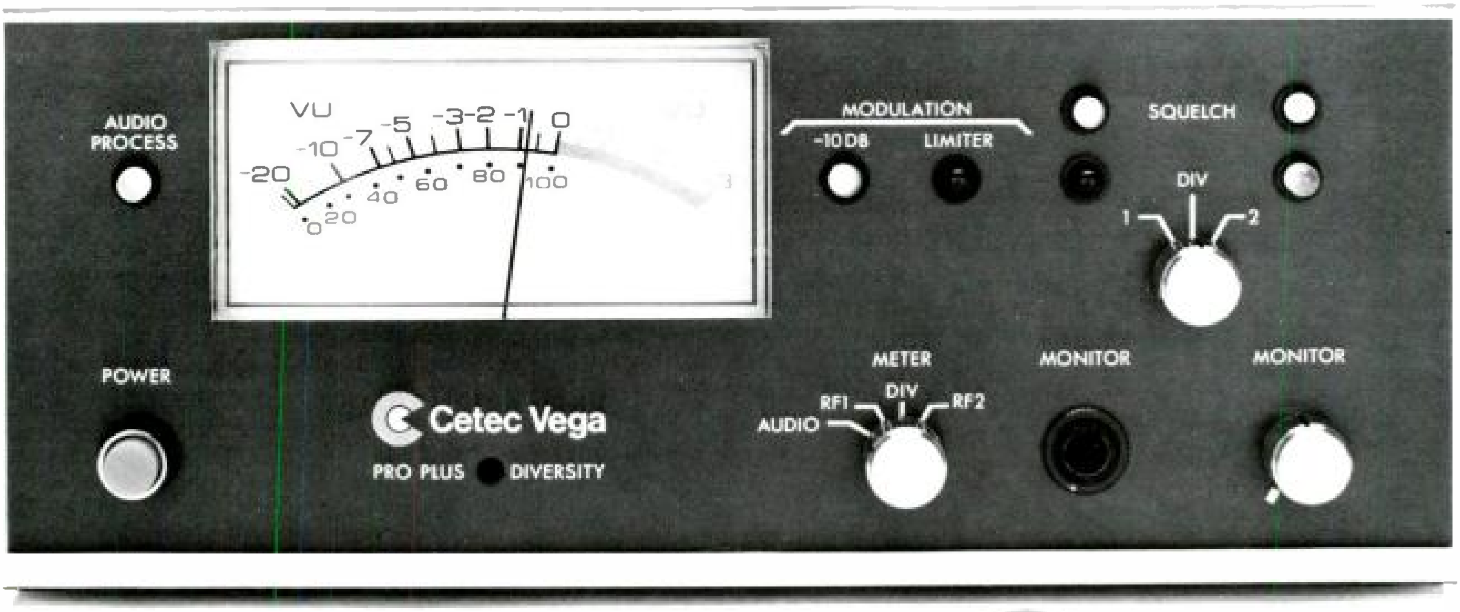

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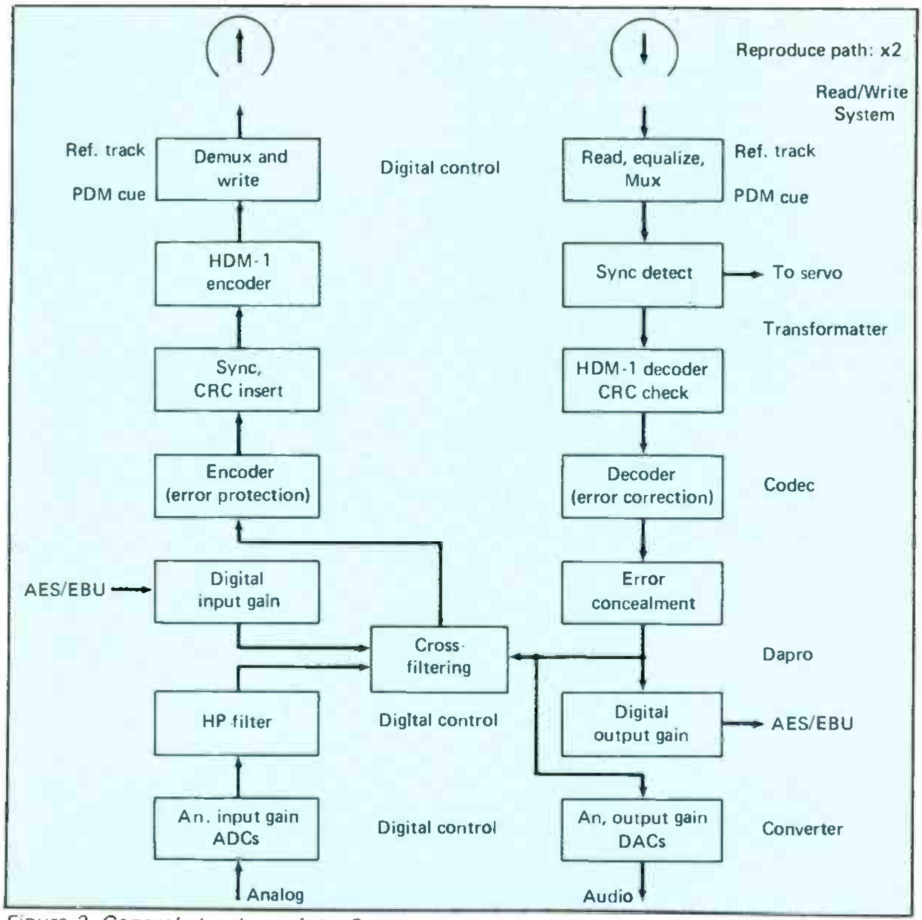

Figure 2. General structure of the Studer prototype recorder.

tions toward digital audio technology. Two papers presented at the Paris AES convention, and one paper presented at the Anaheim AES conference present a prototype, a concept, and a proposal respectively, which brilliantly exemplify the second wave of digital design. In this month's column and the next two, I will discuss these papers, both to further acquaint readers with their contents and to illustrate how these papers act as signposts, pointing toward digital audio's future.

The Studer professional 2-channel stationary head digital audio recorder shown at the Paris AES Convention is a prototype; thus the features and design concepts presented in its description must be treated as possibilities, rather than eventualities. It is clear that the design engineers were given liberty to design this recorder from scratch, dismissing any carry-overs from analog tape recorders, as well as other digital recorders. Of course, practicality imposes considerable constraints, as do gentlemanly agreements, such as DASH. Nevertheless, this recorder is a remarkable example of ingenuity; its innovations include an enhanced use of the DASH format, improved transport, triple digital head design, double-speed dubbing, circuit compatibility between 2-channel and multi-channel recorders, automatic tape playback interchangeability, single board encoding and decoding, on-board digital audio processing, radically new A/D and balancing circuit designs, digital cue tracks using PDM, comprehensive error protection display, and diagnostics logging. Although each of these features deserves mention, space limitations permit discussion of only the most salient aspects.

Studer has been particularly aggressive in establishing a reliable digital recording format. This prototype recorder utilizes the DASH-S format which calls for 8 digital audio tracks and 4 auxiliary tracks, one of which, the Reference Track, is specified by the standard. The prototype's two outer tracks on the quarter-inch tape are used for digital cueing. The remaining inner auxiliary track is used for labels and other data such as SMPTE/EBU time code. The use of time code on digital machines is problematic; utility is lost when time code is nonsynchronized with samples, and synchronization drop frame

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THE FEATURES

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This spherical representation of the original soundfield allows a stereo signal to be extracted pointing in any direction and of any first order polar diagram. The angle between the two microphones may be varied between 0° (mono) and 180° and the apparent proximity to the original sound sources may also be adjusted.

The control unit also provides a four-channel output signal, known as "B format," which exactly represents the first order characteristics of the soundfield. Recordings stored in "B format" allow the POST SESSION use of all the aforementioned controls. The advantage of being able to set such critical parameters as image width, direction of point and tilt, polar patterns and distance – all in the peace and quiet of the dubbing studio – cannot be over-emphasised. "B format" is also the professional signal format for Ambisonic surround sound and may be encoded directly to domestic transmission and consumption formats.

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formats are inherently incompatible with any sampling system. With this prototype recorder, Studer legislates for use of the AES/EBU Interface format; the AES/EBU Interface (or SMPTE/EBU) time code could be written in binary form on the auxiliary track as part of the auxiliary data. This could be recorded, along with labels and other auxiliary data, at the same recording density as the audio data tracks; identical error protection could be used for both types of tracks. An auxiliary data track fits within the DASH format, yet greatly expands the recorder's functions, permits more upward compatibility, and establishes the idea of intelligent tape recorders, making it better able to function in a studio network.

The Studer prototype recorder incorporates a number of improvements lacking in existing digital audio recorders. The recorder uses three heads (two reproduction heads) to permit read-after-write in both recording and sync recording modes. This is accomplished with a sync (pre-read) head, write head, and a read-after-write head. To improve electronic editing capabilities, double-speed dubbing is specified; the recorder can thus operate in read or write modes at double the sampling rate.

Pending standards for read and write circuitry, and standard manufacturing tolerances for heads and tape, interchangeability of tapes even between identical model recorders is nontrivial for any digital tape recorder. To help guarantee reliability, the Studer prototype specifies digital self-adaption of the entire read/write system. Differences in recorded tape parameters may be monitored and used to control this circuitry to maintain consistency. Similarly, work is underway on adaptive equalizing, which would permit data read-out independent of tape speed, a prerequisite to advanced digital audio synchronization and editing techniques.

The Trans-formatter circuit accomplishes all error-protected data/channel data transitions. For encoding, the Trans-formatter receives eight synchronous error-protected data streams and formats them to eight channels with synchronization pattern and CRC words, encoded for HDM-1. For asynchronous decoding, synchronous patterns are detected, CRC is checked, HDM-1 is converted to NRZ, erroneous blocks are marked for subsequent processing, and the eight streams are time-base corrected. This single circuit board design promotes compactness, and its independence from the Reference Track is highly significant, which further ensures data reliability.

The Codec circuit accomplishes all audio coding and decoding, which consists of the matricing/de-matricing, check sum computations, interleaving/de-interleaving, and error correction. During decoding, dropouts up to a theoretical maximum of 32 blocks-per-track are detected and fully corrected, as are interleave errors from tape splices, and error patterns from reproduction amplifier failure. Two-dimensional array processing of error flags permits processing of errors in correlation on several tracks, as might occur with a severely damaged tape.

A digital audio processor system performs numerous functions including digital gain adjustments to inputs and outputs, input high-pass filtering to remove DC components, and error concealment. The processor

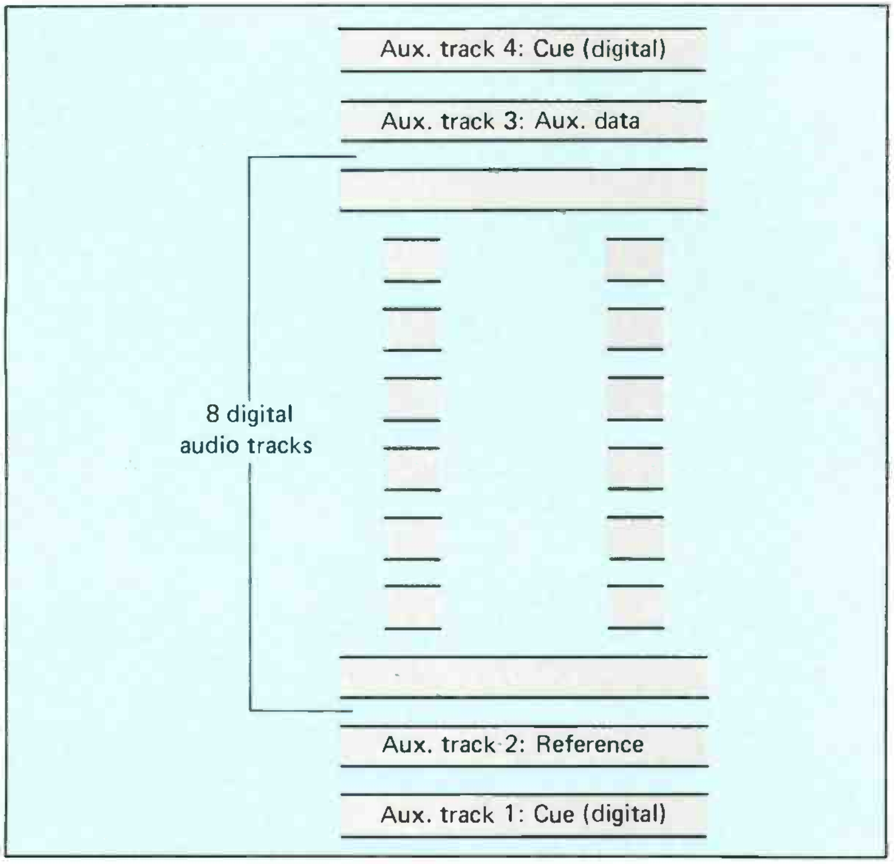

Figure 3. Assignment of functions to the auxiliary tracks.

also manages punch-in, punch-out, and tape splice processing, which is accomplished with digital cross filtering, claimed to be more flexible than cross-fading.

One of the few carry-overs from existing technology conceptions in this prototype is the inclusion of A/D and D/A converters. Obviously, professional tape recorders of the future will omit these devices, but Studer has consented to include them in this model. Although details are not yet forthcoming, it appears that a new approach has been employed in the design of the A/D converter. A two-stage converter uses a D/A converter operating uncritically, thus offering the possibility of conversion beyond 16 bits. This is more evidence of Studer's interest in greater dynamic range systems; their proposal for interface data includes ranging bits.

Analog cue tracks are used in existing digital recorders. However, this requires extra heads, risks interference by bias to the digital tracks, and results in poor analog sound quality on the digital tape, suitable only for manual cueing or fast speed searches. The Studer machine formats the cue signals digitally; the

PCM signal is recorded on the cue track with PDM (Pulse Duration Modulation). With an index of 0.5, the PDM's minimum and maximum durations correspond to the transition intervals of the HDM-1 digital audio data. Thus, the same head and associated circuitry can be used for both. With 7-bit resolution and a static compander, good quality is obtained from the cue tracks. Parenthetically, some compatibility exists between analog recording and PDM. With a tracking filter to attenuate the sampling frequency component, PDM may be reproduced with an analog head, and with equalization an analog track may be read by a PDM reproduction circuit.

The prototype machine may be controlled in a variety of ways, including direct remote control, or digital serial control via the AES/EBU Interface. SMPTE/EBU time code may be recorded on the cue tracks, or the auxiliary data track may be used via the AES/EBU Interface time code. Digital displays include level, gain, and comprehensive LED diagnostics with eight levels for signal quality of each of the two channels: very low CRC error rate, moderate CRC error rate (all errors corrected), high CRC error rate (all errors corrected with third and fourth pass error correction), and moderate interpolation rate (incomplete correction, splice or long drop-out detected, track loss, muting, or no valid data detected).

The Studer prototype recorder thus embodies many design conceptions and features which are certain to become standard on professional recorders of future generations. Additionally, it represents a highly important precedent in matters such as use of the auxiliary data track to carry AES/EBU Interface data. Clearly, the second wave of digital audio products has left the stand-alone concepts of analog audio behind, and is moving toward the next generation of recording studios. The Studer prototype recorder thus signals the coming of the future networked, intelligent studio, unified by bit streams of both audio and control data. A tape recorder in such an environment would be a data storage device, a peripheral to the central processor.

Are we omitting something? Should we include a discussion of how the prototype sounds? No, that is no longer a meaningful criterion for a digital audio storage device. Its only job is to write and read data reliably. Any alterations to the sound will be left entirely to the recording engineer. ■

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1. Lagadec, R., McNally, G.W., "Labels and Their Formatting in Digital Audio Recording and Transmission," 74th AES Convention, New York, October 1983.
2. Lagadec, R., Girsberger, H.P., Brandes, C., "Design of a Professional 2 Channel Stationary Head Digital Audio Recorder," 75th AES Convention, Paris, March 1984.
3. Lagadec, R., "Proposed Guidelines for an Auxiliary Data Track on Digital Audio Tape Recorders," submitted to the AES Technical Committee on Digital Audio, Working Group on Labels, Paris, March 1984.

JOHN EARGLE

Developments In High Frequency Pattern Control: The Defined Coverage Horn

• In the early days of the motion picture sound art, exponential horns were commonplace for both high- and low-frequency applications. The reason was simply that the exponential design offered the most uniform acoustical coupling of the driver to the air. It behaved as an efficient acoustical transformer, a necessity in those days of relatively modest power amplifiers.

EARLY HORNS

The exponential horn exhibited poor pattern control: the angular coverage was wide at low frequencies and narrow at high frequencies. A second generation of horns was soon to be introduced. Groups of exponential sections were clustered together to make a multicellular horn, and this represented the first attempt to maintain constant coverage. While the splayed cells did, in fact, direct the coverage at mid-frequencies, the tendency of each cell to beam at high frequencies produced "fingering" along the common walls between adjacent cells.

Soon to follow were radial, or sectoral, horns. These were of exponential cross-section in their vertical plane, but had straight side walls.

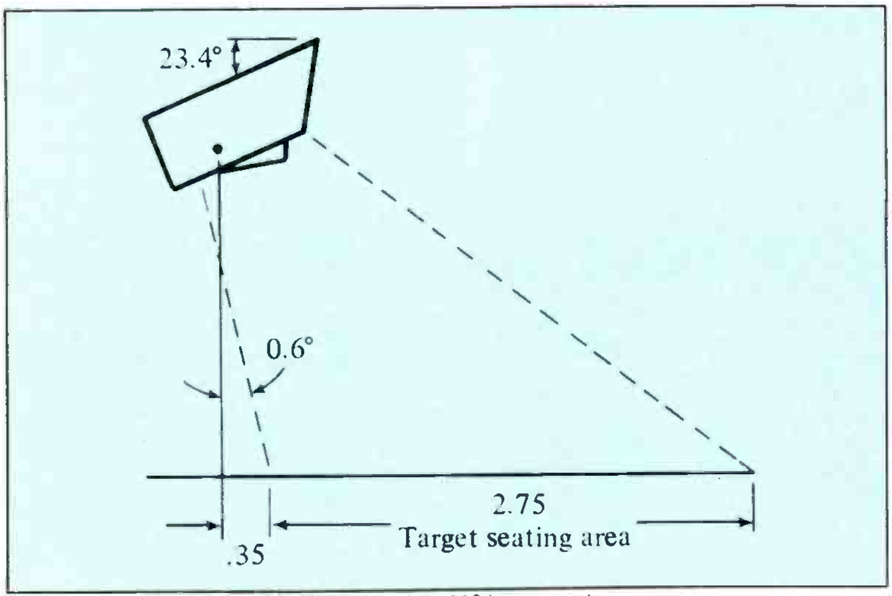

Figure 2. Mounting the 4660 Loudspeaker System.

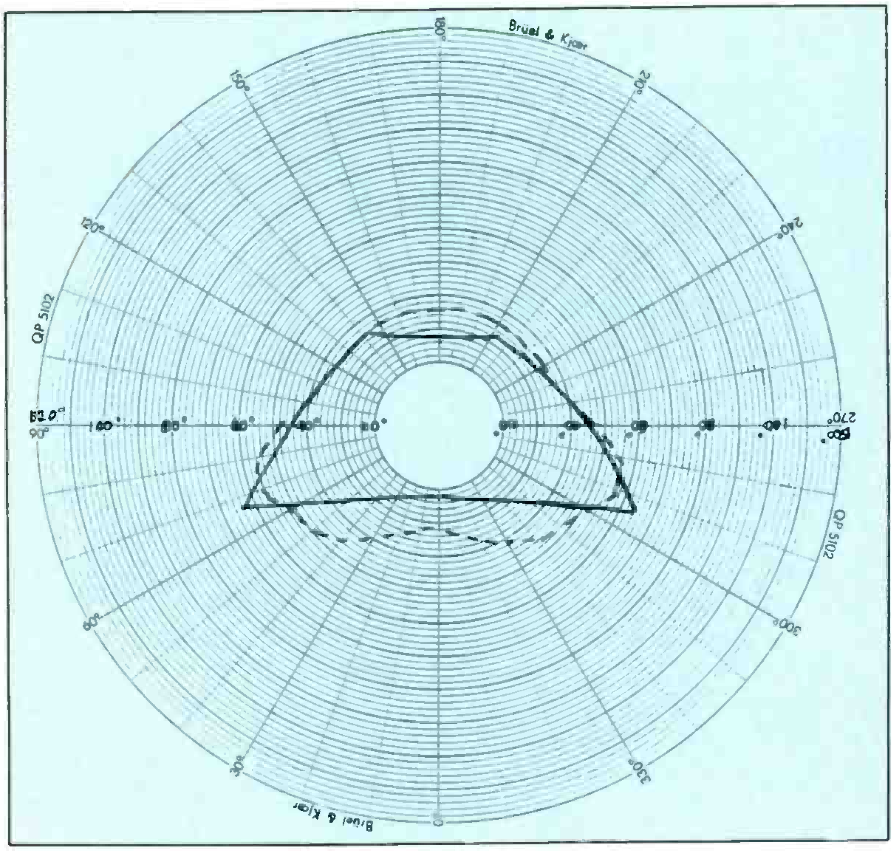

Figure 3. Solid line represents seating area as seen from the 4660. Dotted contour is the -6 dB isobar at 8 kHz.

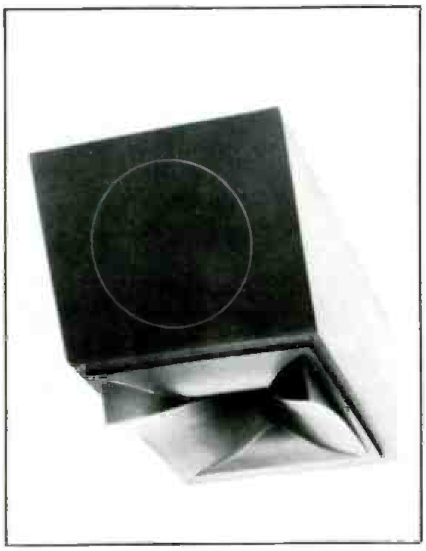

Figure 1. JBL 4660 Defined Coverage Loudspeaker System.

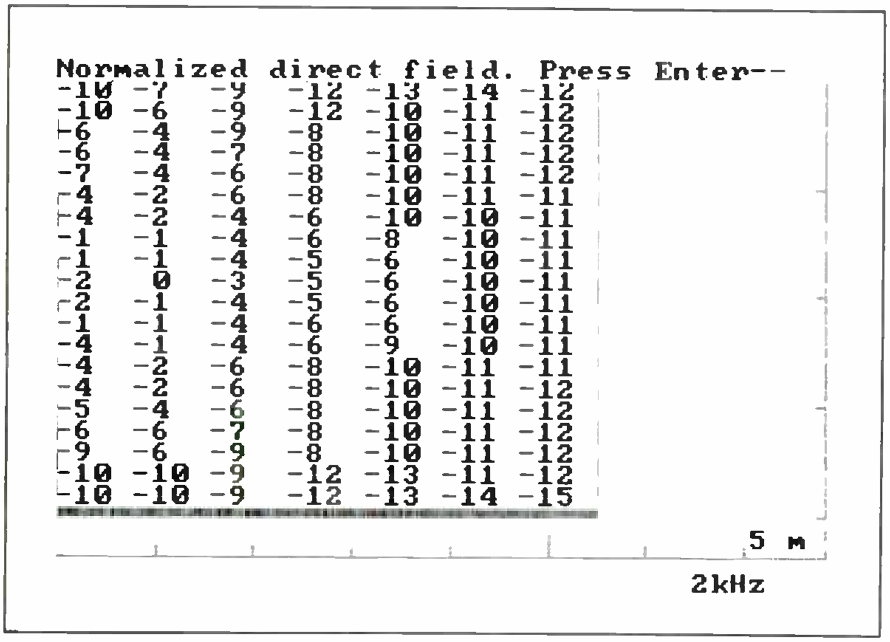

Figure 4. CADP plot of direct field response at 2 kHz. Loudspeaker is mounted at center left, 3.5 meters behind boundary.

When properly designed, they exhibited excellent horizontal pattern control. However, in the vertical plane, due to the exponential flare, the pattern narrowed progressively with rising frequency.

CONSTANT COVERAGE

Things stayed pretty much the same until the mid-'70s, when Electro-Voice introduced a family of constant coverage horns. This third generation approach was adopted by Altec and JBL, among others; these devices have since become staples of modern sound reinforcement system design.

High-frequency horns have generally produced patterns symmetrical about both vertical and horizontal axes. However, most applications require some degree of skewed coverage in order to properly cover an audience area. Working from this premise, D. B. Keele set about designing a "defined coverage" horn. Such a decision has a fairly broad coverage angle toward the front of the room and a narrow angle toward the back of the room. The basic design was discussed in detail at the Fall 1983 AES Convention in New York (Preprint number 2052).

THE NEW GENERATION

JBL, Inc. has just introduced a new loudspeaker system, the model 4660, which makes use of such a horn design. This "fourth generation" system is shown in FIGURE 1 in its normal mounting attitude. The target coverage is a rectangular space of aspect ratio 2 × 2.75, with the system mounted 0.6 unit high and 0.35 unit back from the front edge of coverage, as shown in FIGURE 2.

The front coverage angle is nominally 110 degrees, while the back coverage angle is nominally about 38 degrees. In practice, the elevation angle can be varied, and spaces of different aspect ratios can be covered.

FIGURE 3 shows the view of the target space, in spherical coordinates, as seen from the mounting position for the system (solid lines). The dotted line shows the -6 dB isobar at 8 kHz. Note that the two lie very closely to each other. The isobar plot shown here was measured at a fixed distance from the horn, so there will be an inverse square falloff in level from front to back. However, with careful orientation of the device, level variation along the sides and back of the room will be no more than plus or minus two or three dB. FIGURE 4 shows a CADP display of direct field coverage produced by the 4660 system. Note that coverage around the sides and back of the room is quite smooth.

In addition to obvious applications in sound reinforcement, the defined coverage horn has great promise in other areas. Experimenting is now underway using the design in stereophonic systems, where the unique pattern control can be used to stabilize phantom images for off-axis listening positions. ■

BARRY BLESSER

The Digital Fader And The Question

• Several months ago, a friend called me from California to ask a question. He had just been playing with a new digital audio mixing console and he had the feeling that he could hear the faders in the music. Since he did not really believe this was possible, he was embarrassed to ask if there was any technical basis for his perception. Unlike some critics of digital audio, he understood enough of the issues to be very cautious about making an assertion. Nevertheless, he trusted his ears and was looking for some explanation.

I listened to his description of the equipment and the listening experiments, and I came to the conclusion that there was a real technical issue which needed to be examined. It is difficult to imagine that with all the high-power engineers working on digital (after all, it is not an activity which can be easily done in the garage by high school students), a simple device like a fader could present a problem. The basic element of a digital fader is simply a multiplier. The audio appears at one input, and the required gain appears at the other. The equation which describes the digital fader is the following:

$$a_o(t) = g \times a_i(t)$$

where:

a_o is the output signal, a_i is the input signal, and g is the gain.

ON BEING SMART

As I was thinking about the issue, I began to philosophize about how to approach the problem. As a college student, I remember that I was very impressed by all the smart people that I met. They appeared to know all of the answers. By studying hard and learning the lessons of my professors, I began to learn the answers.

Such is the life of a student: learn what somebody else has decided is important. This experience has a very severe limitation which does not become apparent until one asks how the professors learned what they know. Did they become intelligent by learning the answers from *their* professors? If we answer yes, we would be led to the conclusion that it would be very difficult for people in the world to create new ideas.

Ten years after starting my professional career, I realized that the answers are usually not very interesting; it is the question which is hard. A very carefully presented question actually contains the answer or points the way to the answer. Consider, for example, the following question: How long does it take to stop a 2,000 pound car going 60 MPH on wet pavement, when the brakes have been applied so that the wheels lock on a newly paved road? You may not know the answer but the question contains all of the information which is needed to do the experiment. It may take some effort to create the

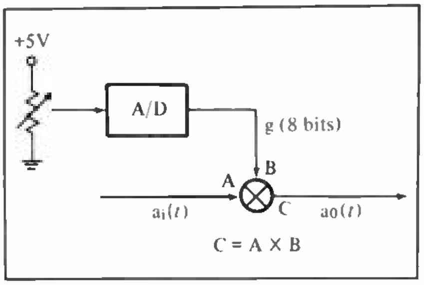

Figure 1. Model of a fader in a digital mixer—a linear potentiometer with 5 volts feeding an 8-bit A/D converter, which drives the B port of the multiplier.

experiment but it does not take much intelligence. The intelligence was required to create the question.

The philosophic distinction between the question and the answer is what separates intellectual men from boys (or women from girls). The telephone call from California required me to create a question. By my definition, a complete question is one that contains enough information to make the answer a simple matter of experimentation. I need also point out that, in addition to the laboratory experiment using real equipment, there is also something called a *gedanken* (german for thought) experiment. This is an imaginary experiment whose results can be proved based on other knowledge already available. Since I am basically lazy, I first tried to deal with the issue by creating questions which could be answered by a gedanken experiment.

THE MODEL

To begin the process, I asked my friend several questions about the hardware. The fader was constructed of a simple voltage potentiometer whose output was fed to an A/D converter. This digitized voltage represents the gain. The block diagram is shown in FIGURE 1. The digital word then entered the equivalent of the B port on a multiplier. The digital audio was 16 bits in and 16 bits out, but the gain word was only 8 bits. We all know that gain resolution does not have to be that great compared to the audio signal, since the ear can just discriminate a 1 dB level change. An 8 bit gain would allow for 256 gain states. This would appear to be enough.

The model looks simple, and there is essentially no place for any problem. We might thus be tempted to respond that the "golden ear" was deaf or that the hardware was broken. As an old timer, I am very hesitant to jump to such a conclusion if I have any trust in the person presenting the issue. The fact that I could not think of an explanation does not prove there is no problem. Remember: You can never prove the negative; you can never prove that there is no problem. If you do find the problem you can prove that it is a problem by running an experiment. I can prove that there is a needle in the hay stack by finding the needle, but I cannot prove that it is not there if I don't find it. (Maybe I didn't look carefully enough.)

So, I began to make up issues and imagine things. Through the process of fantasy, I came up with a list of possible issues. Each was examined and discarded when considered carefully, except one.

THE FADING ACT

When the mixing engineer is fading a signal from full level to low level, he is moving the gain continuously from 1 to 0. But, he is also making the gain go through a continuous series of small step changes. If it takes 1 second to go through the 256 states, then there will be a state change every 4 msec. A repetition rate of 4 msec. is 250 Hz, clearly in the audio band. By extension, if the fader takes ¼ second to go from 1 to 0, then the state changes will be at a 1 kHz rate. These repetition rates are possible candidates of concern.

FIGURE 2 shows us that the true signal from the A/D converter is actually a staircase rather than a perfect ramp. By thinking about this

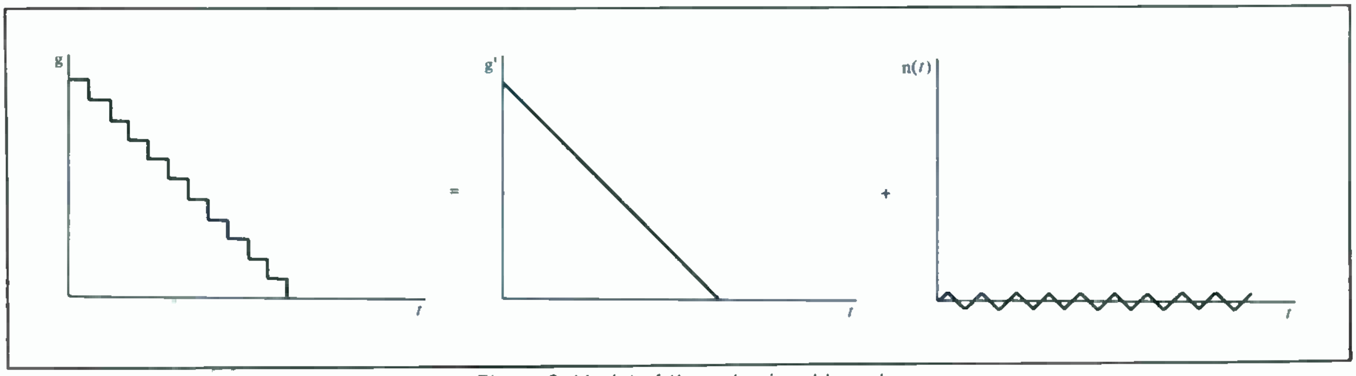

Figure 2. Model of the gain signal based on the actual signal from the A/D converter (left), which is then decomposed into a pure signal (center) and a noise-like signal (right).

signal as being two signals—a perfect ramp and a residue—we can gain some insight. The ramp signal is that which corresponds to our normal experience with analog faders. The residue is a noise-like signal which might very well result in audible distortion.

The residue will be periodic if we pull the fader at a constant rate. Now consider a gedanken experiment. Let the audio be an extremely low frequency like DC (we can do anything in our mind). The result of multiplying the residue by the DC will be simply the residue itself. Moreover, its amplitude will be only 52 dB down and its frequency could easily be 1 kHz. A ¼-second burst of 1 kHz tone at -52 dB is audible if the audio is very low frequency. We used DC in our gedanken experiment not because DC is a likely signal in an audio system but because we could prove what would result without going to the laboratory. The degree to which this can be extended to a real world situation may have to be concluded in the laboratory, but we have a feeling for the phenomenon and we have some idea what to expect.

We have learned that low frequencies will result in a tone when the fader is pulled at a fixed rate. We have also learned that the rate of pulling determines the frequency. We can make a model of more complex situations, as, for example, moving the fader up and down continuously.

If you have followed us to this point, you may be impressed with what you have learned. But notice that the answer is obvious and easy to understand when the question is created in such a detailed fashion.

WRONG THINK

Part of the disadvantage of being smart is that you can easily do "wrong think." Wrong think is a way of coming to a conclusion by using information incorrectly. At the beginning of this article I misled you with the statement that 256 states should be enough because the ear can only hear a gain change on the order of 1 dB. This has been well reported in the psychophysical literature and it can easily be confirmed. This would allow us to prove that the changes in gain from the A/D converter could play no role until I was at the lowest 10 level when a step was 1 dB. This is "wrong think" because the assumptions, ie., the questions, are not the same. All of the psychophysical experiments consider either a single change in gain or a comparison between a signal at two different levels. Our question is that of perceiving a continuous series of small gain changes which are step-like and repeat at an audio rate. Our model is better composed of a noise residue rather than a gain. It is the difference in the question that leads to a very different conclusion.

FADER DESIGN

Having demonstrated the defects in this fader design, we must now revert to finding the set of questions that would give confidence in our understanding of the problem. Should we think of the fader's A/D as having to be an audio quality A/D with 16 bits of accuracy and 90 dB of S/N? Should we think of the fader as being sampled? Does it need an anti-aliasing filter the way the audio signal required?

There are often many ways at looking at a set of questions, and these are not easy. Let us consider a few. There is no need for an anti-aliasing filter because that filter removes very high frequency components above the sampling rate and the fader is a very low frequency signal (full scale to 0 in 250 msec). There is no sampling process in the fader channel other than the fact that there must be a new gain value for each audio sample. Hence, the fader signal is really sampled at the audio rate. The fact that the fader signal is so much lower frequency than the sampling rate means that we need not consider it as being sampled. The residue noise comes from the act of quantizing. (For a discussion on quantization without sampling, see some of the early **db** articles in this series or my article in the October 1978 *JAES*.

The Quantization process is like a simple non-linearity. And like all non-linearities, it is able to create harmonics. In our example, the nonlinearities are able to turn a ramp (fader signal) into a distortion which is a periodic triangle. Because the resulting frequency is a function of the rate of fading (uncontrollable) and the number of steps (controllable), we may only consider the latter. Doubling the number of levels will double the frequency but the result becomes unchanged if we also halve the rate of fading.

In simple terms, the frequency of the residue cannot be controlled in the design. Observe, however, that the amplitude also decreases by a factor of 2 for each doubling of the number of steps. Our discussion is leading us to a very simple question: what level of *multiplicative* noise is inaudible? I happen not to know the answer but I could easily find it out in the laboratory. A gedanken experiment might allow me to conclude, however, that the number is extremely low. I might be able to find a case for a requirement of 90 dB. Do we really need a 16 bit A/D? In principle the answer is yes. Since the consequences are not desirable, I need to consider other ways of solving my problem.

Since the g signal of FIGURE 1 is the one that contains the noise, we could place a simple lowpass filter in this path to remove the residue. Notice that the filtered result would have more than 8 bits. With enough smoothing (filtering), the g signal would be a smooth ramp in 16 bits even though the original version was a staircase. The filter has interpolated the values between samples. Without the filter, the gain values for each audio sample might have been the following series:

1.000 1.000 .990 .990 .990 .990 .990 .980 .980, etc.

whereas after filtering they might be:

1.000 1.000 .998 .996 .994 .992 .990 .988 .986, etc.

In the first case, there were large steps of 0.01 units after 5 samples of the same value. With filtering, the gain changed by .002 units on every sample. It is as if the fader had a higher order number of bits. Notice that the filtered fader signal might not correspond exactly to the analog voltage from the potentiometer, but we are not interested in accuracy for its own sake; we are interested in a smooth signal even if there is a 0.001 dB error in the actual gain.

TESTING

This article was not intended as an academic lesson. To test a digital mixing console or an analog mixing console that uses an A/D on the gain control, try the following experiment: Use a 20 Hz sinewave at full level as input and place a highpass filter in the monitor channel. Try moving the fader up and down at varying rates. This will tell you if the designer understood the question. Since the highpass filter removes the 20 Hz, you will only hear the residue. ■

JESSE KLAPHOLZ

Buying Hardware and Software: Shopping Tips

• The first place most people look for advice when shopping for a computer is in a magazine. The question then becomes, "Which magazine?" What I have found to be the general rule is a ratio of at least four-to-one of computer magazines over hi-fi, video, and music combined.

According to the *Standard Periodical Directory*, (8th Edition-1983-84), there are currently 514 computer magazines in print! (And you thought there were a lot of audio magazines?) There are two main types of computer magazines: generic, which deal with computers and computing in general, and publications which deal with a specific brand or model, such as *Macworld* (MacIntosh), *PC World* (IBM PC), *Apple Orchard* (Apple IIe), etc.

Selecting the right magazine for your needs can be quite confusing. The words 'compute' and 'personal' are used in just about every combination imaginable, so that many magazines begin to sound alike. For example, I was browsing through an article last month in *Popular Computing*, or was it *Popular Computer?* No, I think it was *Personal Computer*, or was it *Personal Software?* Maybe it was *Personal Computer Age*, or could it have been *Creative Computing?* Oh, well, I forget what the article was about anyway.

Perhaps one of the most intriguing virtues of the more popular computer magazines is the reader service card. The purpose of the reader service card is to aid the reader in obtaining information relating to his needs or curiosities. However, by the time you sift through hundreds of ads and fill out the reader service card with 1,000 numbers on it, the mailman is already delivering the next issue!

BOOKS, BOOKS, & MORE BOOKS

Our secondary source of computer knowledge is the computer book. However, attempting to catalog all of the computer and computer-related books published to date would be as monumental as cataloging everybody doing anything at all related to audio. My guesstimate is that there are several thousand books currently in print. Computer mania has definitely struck us; a visit to almost any bookstore will prove it, with titles ranging from *You Just Bought A What?* through *So You Think You Need A Small Business Computer* to *Science And Engineering With VisiCalc*.

There are a number of publishers which deal largely with computer books. Some of the more popular computer publishers include: Osborne/McGraw Hill, Byte/McGraw Hill, Hayden, Sybex, Micro Text, Brady, and Que. The basic inventory of any computer store's book rack usually includes several introductory books to computers, 20 or so books on BASIC, another 20 books on every other programming language, books about the hardware that the particular store sells, and books about the software that the store sells.

One of the most popular software packages is WordStar by MicroPro. WordStar is just about the defacto standard of word processing software packages. A cursory look at the shelves will reveal at least several books on WordStar. I have seen in one store, *Introduction to WordStar*, *WordStar Made Easy*, *Applications of WordStar*, *Practical WordStar Uses*, and *Last Word On WordStar*. Which one do you buy?

On the other hand, some computer stores answer this dilemma of the saturation of computer publications by simply not selling computer books at all! They maintain, "If you want to read about it, buy the book at a book store. If you want to do it, come here!" Either way, the time comes when one must take that big step forward into the big world of computing.

PRACTICAL APPLICATIONS

These days, computer debates are

commonplace. However, the one point that most computer salesmen and users seem to agree upon is that applications are the primary purchasing criteria. As mentioned in the last column, a computer is a useless heap of chips without software telling the computer what to do. Software can make your life easy or it can be a nightmare.

For example, I know of one manufacturing company in the Philadelphia area that purchased a Deck PDP-11 (a highly respected mainframe computer) with a comprehensive business applications software package; to date that company is using that mainframe computer strictly for word processing. It turned out that the software included with the purchase could not be adapted to the environment of this particular company, nor could any "off-the-shelf" software for the PDP-11 do the job. Hence, this company maintains its operation on a manual bookkeeping system.

On the other side of the fence, a sound systems installer and rental service in the Washington, D.C. area maintains its inventory control on a dedicated computer system. The system is Timex/Sinclair based with several add-on peripherals; the total cost of the system is around $500. In the words of Albert Einstein, "Things should be made as simple as possible, but not any simpler." The age-old cardinal rule, do more with less, has proven itself for us in sound system design time after time. When it comes to computers, these rules should be equally applied.

Word processing is a universal application of computers. Whether you realize it or not, your operation, whether you're in broadcasting, recording, or live sound, can benefit from word processing. Reports, pricing schedules, specifications, proposals, letters, etc., can be generated on a computer, printed out, and saved in a file so that it can be used in part or its entirety at a later time.

If you're currently doing any or all of the above tasks manually, the time and precision of generating these documents can be improved dramatically. If you're not currently generating any of these documents or are depending on an outside source, then a computer system can really help your operation take off.

Another popular computer application is the use of data-bases. The defacto standard here is d-Base II. Like word processing, a data-base is also universal to any operation. It can be used to track inventory in a rental business, track repairs of equipment, log customer activity, keep account of studio time down to specific engineering times, etc., do large system bids, and in general produce reports from your data relating quantities and items. Once you have purchased a general database program, you can then generate your own program or as many programs as you like.

SPREAD-SHEETS

Spread-sheets are the single program application responsible for introducing the "personal computer" to the big business office. Sure, you could find eight-bit micro-processor based systems with a couple of floppy drives and at least 48K of memory, but these were dedicated "word-processing machines" from people like IBM, Xerox, Exxon, Victor, etc.

When a salesman came around to demonstrate a computer system, the typical businessman generally wasn't interested. But when the office supplier showed him a new word-processor that could do half as much as the personal computer for twice as much money, the businessman fell for it. He just wasn't ready to buy a "personal computer" which he felt belonged at home. However, he finally realized that electronic spread-sheets were a solution to the typical businessman's problem: "what if?"

The business application of spread-sheets can easily be explained with the "what if" scenario. There was once a radio station called WXYZ. The station manager was Joe Blowkowski. Joe was confronted one day by the station owner about the station's ratings and sales. After lots of arguing they finally agreed it might be wise to invest in a personal computer to help them sort through the whole mess.

Joe returned from the corner computer store with his new computer in one hand and a bag of disks and books in the other. Joe got down to brass tacks and was soon developing spread-sheets about the station's projected sales, "if..." In less than a week, Joe returned to his boss with a stack of charts and reports generated with his new spread-sheet/graphics/word processing package.

"According to my calculations, we're not spending enough in engineering and advertising," Joe exclaimed to his eyebrow-raised boss. With his curiosity enticed more by amusement than seriousness, the station owner queried Joe further. Mr. Blowkowski explained, "We're presently spending 12 percent of our budget on engineering and 18 percent of our budget on advertising." Joe continued, "If we increase our engineering budget to 20 percent and advertising to 28 percent, we can anticipate an increase of sales in excess of 150 percent!"

Impressed with all the hard facts and figures, the owner gave the go-ahead to Joe. Engineering had some engineering to do and decided to go digital, making the local sound equipment distributor very happy with the large sale out of the blue. Advertising decided to capitalize on the whole digital thing, and the station's revenues did in fact increase by some ungodly number.

Spread-sheets are a little more complicated than data-bases. They not only perform the basic arithmetic functions, but also have in their bag of tricks almost any scientific math function encountered. Yes, even log functions!

Thanks to the big business world's need for electronic spread-sheets, through the trickle-down evolution of this software, we in the audio world can benefit by applying spread-sheet techniques to our environment. Spread-sheets have been very successfully applied to many phases of audio design and project costing.

Like data-base applications, once one has purchased a spread-sheet software package, many different applications can be developed. Designing with spread-sheets allows many "what ifs" to be asked; for example, "What if we change the value of a particular resistor?" This feature alone can save many hours of manually re-computing the entire reiterative equation processes involved.

Spread-sheets also allow you to re-use old data or designs by simply plugging in the new parameters, greatly economizing the engineer's time. Any circuit or function based on mathematical relationships can be designed or analyzed with spread-sheets, including pads, antennas, Fourier analysis of vibrations, filters, dB conversions, Helmholtz resonators, standing wave frequencies, reverb time, and loudspeaker coverage, just to give you a basic idea.

THE CATCH

Unfortunately, there's a hitch to all of this: in general, the more powerful the program, the longer it takes to learn how to use it. It is not uncommon at all to find hords of "third party vendors" selling software packages that help you to use the original program. If you've never used a spread-sheet before (manual, that is), and don't understand the concept, don't expect to jump right into it. However, if you are experienced with spread-sheets, you'll kick yourself for waiting so long.

For those of us who don't have the time to learn these high-powered systems in depth, perhaps the "ready-to-use" software packages are for you. Even though the smaller packages don't have quite the capabilities of the often three-to-four times as expensive programs, in most cases they do more than fill the bill. As a result, these smaller packages allow the user to immediately use the computer for the reason he bought it in the firt place: to make life easier.

Once you have decided which software packages can help you the most, it may be time to enter the door of your computer store. The salesman that greets you, as in any other business, may either turn you off or on to that store. It is important to select the store that can support you the best, and that means asking some serious real-life questions, like, "If I buy my computer here and I get it home, will you help me get started with the software I buy from you?" See what the general reaction is from other customers in the store and customers phoning in.

After you've made the purchase and the original dealer can't help you, it's going to be pretty difficult to get help from another dealer, as a general rule. Another important feature to look for in a dealer is whether the salesmen have sufficient "hands on" experience to help you out in case you have problems. A strange thing about the computer business is that most software houses (software is the area in which you'll most likely need help) refuse to talk to the end user and will refer you to your dealer!

Until someone like Apple or IBM proves me wrong, a computer is still a glorified calculator/storage/display device, incapable of doing your job for you. However, used judiciously, a computer can enhance and greatly improve the quality of your work and you'll enjoy doing your work much more. ■

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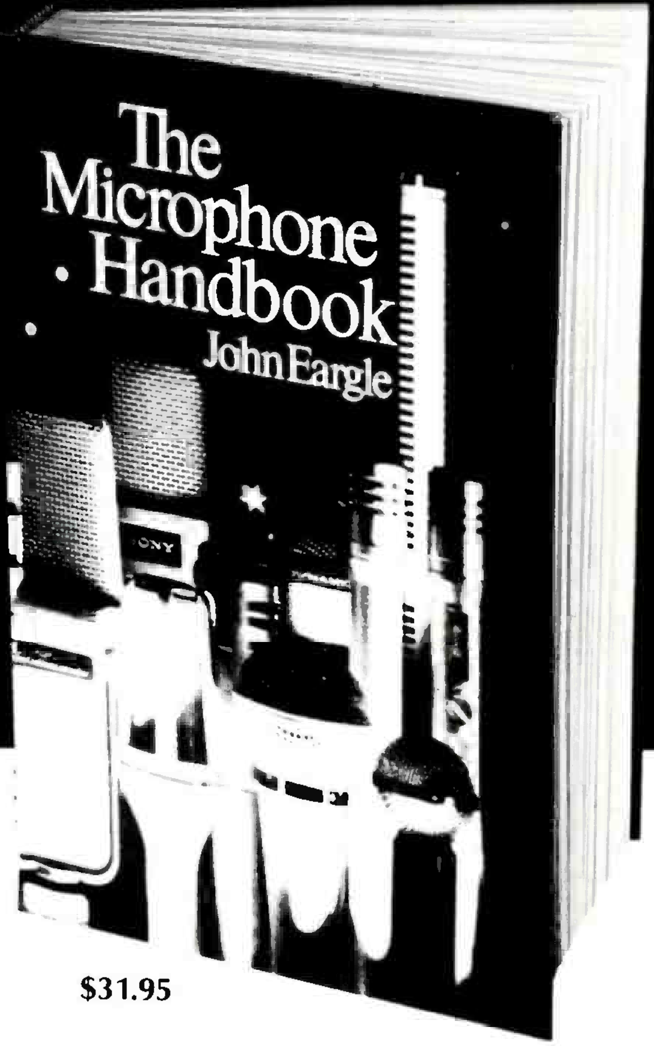

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Editorial

A Shure Thing

OUR COVER THIS MONTH is intended to remind you that while the topic of this issue is microphones, the microphone as we know it is not a new item. Nor is it an endangered species.

Digital electronic recording is here, and, no doubt, is here to stay. But happily, our ears are still not digital, so we need to input and output to our digital domain in an analog form. This means speaker systems to monitor with, and microphones to record with.

The Shure Unidyne that Charlie McCarthy is speaking into (with Edgar Bergen's thrown voice) was introduced as an advanced cardioid design back in the late '30s, and it was still being manufactured by Shure until the late '40s.

These famous mics are still in use in many a recording studio and on sound-reinforcement stages. It's a testament to the fundamental dynamic design and the genius of a then young engineer named Ben Bauer (later to gain more fame at the CBS Technology Center) that these mics continue to work and, indeed, work well.

Finally, about the picture. We found it in a dusty box in Shure's Evanston, Illinois offices, and we thank Shure's Lee Habich (Manager, Marketing Communications) for his patience as we thumbed through hundreds of other pictures. No exact date can be established for the photograph (at this time), but Shure believes it to be from the late '40s to early '50s. It is, of course, a publicity shot of the times. The child in the background is not identified, but we do not think is a young Candice Bergen, or is it?

* * *

With our sister publication, *Modern Recording & Music*, we recently attended the NAMM (National Association of Music Merchants) Convention in Chicago. If there is anything to report from that show, for you the **db** reader, it is that there is a continuous blurring of the fine lines between the pro audio markets and the so-called MI (Music Industry) markets.

For example, music synthesizers are obviously music performance instruments, but they are products of high engineering technology and, more to the point, every recording studio either must have several, or have access to them. And, they require the same kind of technological support in the studio as a digital recorder. So, don't be surprised if you start reading more about them, and products like them, in these pages.

We serve the recording studio, broadcast audio and sophisticated sound-reinforcement disciplines. As these products come our way expect to be reading about them here. L.Z.

ROBERT F. HERROLD

New Concepts in Microphones—The Age of the Miniature Cardioid Electret Condenser

Microphone of the future? Here the author outlines the advantages and disadvantages of this mini miracle.

SEVERAL YEARS AGO, someone predicted that the electret condenser would be the microphone of the future. His reasoning was that the electret has certain important advantages over other microphone types, a few of which are:

1. That it has smoother, wider range frequency response for an equivalent price level.
2. That it offers better uniformity and rejection of off-axis sound, hence better control of feedback in sound reinforcement or less coloration of sound in recording and other applications.
3. That its physical size can be made significantly smaller without loss of sensitivity or low-end frequency response.
4. That, if properly designed, it will withstand the rigors of everyday use.
5. That its lighter weight allows more freedom of placement.
6. That its shapes are generally more aesthetically appealing than the larger shapes of other types.

Along with all these undeniable attributes, however, electret microphones do have some disadvantages. It may be worthwhile to examine some of them here and assess their overall importance.

POWER REQUIREMENTS

All condenser microphone types require a voltage source to power an internal impedance-matching network or other necessary electronics. Battery-only types offer limited application opportunities. Fortunately, this is a problem of minor proportions when one considers the superior overall performance of the electret condenser.

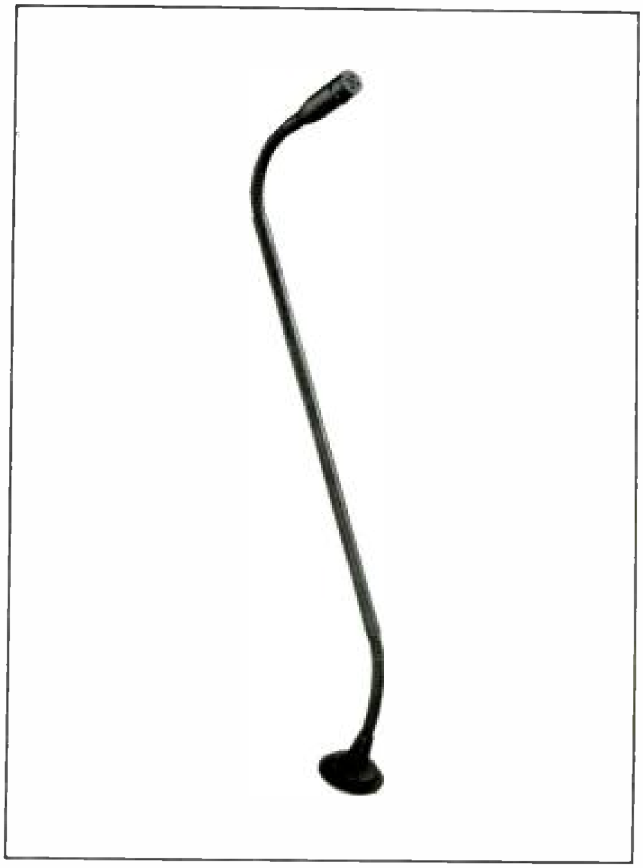


Figure 1. Audio-Technica AT857 double gooseneck microphone.

Robert F. Herrold is Product Manager, Microphones, at Audio-Technica.

The well designed, moderately priced, electret condenser microphone should offer powering flexibility to the end user. I refer to the ability of the microphone to function on internal battery, external battery, or external phantom power. The internal battery should be inexpensive and readily attainable. Furthermore, the internal circuitry of the microphone should be designed to yield high performance and yet be miserly with regards to the amount of current drawn. A battery life of several months would eliminate the constant worry about changing batteries often. Also, an On/Off switch (battery only) would be convenient when the unit is not in use, thus vastly increasing the life of the battery.

Most modern audio (mixing) consoles now feature phantom power capability. The voltage may vary from one manufacturer to the next, with 24 VDC or 48 VDC typically available. Therefore, the electret condenser microphone should be ideally capable of functioning on a variety of external DC voltages with the most flexible electrets accepting a voltage range of 9 to 52 VDC. In those instances where the use of a battery is not feasible, external powering will do the job.

SIGNAL-TO-NOISE

Condenser microphone noise, or "self noise," as it is sometimes called, results primarily from the amount of current drawn by the electronics. Of course, the use of low quality electronic components also contributes to noise. Diaphragm size and the capsule-polarizing voltage are also factors to consider. The moderately priced, well designed electret will yield an admirable signal-to-noise ratio in all but the most demanding situations, as in a recording setup in which the microphone is placed at a considerable distance from an orchestra playing an extremely quiet passage. In such situations, a very low level hiss may be heard in the background of the audio signal. In comparison, non-battery microphone types are usually considered to have no self noise. On the other hand, most orchestral distant pickup recordings today are accomplished with expensive condenser microphones which require a source of power, an indication that recording engineers do not consider the noise problem to be a terrible one.

OVERLOAD (DISTORTION)

Most audio people agree that non-battery microphones do not distort when exposed to the typical sound pressure levels encountered in television, radio, recording, and sound reinforcement applications. Not true for condenser types.

Distortion from a condenser microphone resulting from exposure to high sound pressure levels is directly related to the frequency of the sound and the bias voltage available to the appropriate circuit components. An electret microphone circuit biased with 1½ volts will typically have 1 percent T.H.D. at a sound pressure level of 130 dB (1 kHz). With a 9-volt bias, this figure can easily increase by 10 dB. The 130 dB SPL figure is more than adequate except for close mic'ing of loud brass, percussion, or vocals. The normal speaking or singing voice will present no problem when mic'ed close up. Applications of microphones in the market areas outlined above typically call for them to be used at a distance of 6 to 12 inches or more, which drastically reduces the sound pressure level (SPL) at the diaphragm, and hence precludes the probability of distortion.

So much for the supposed disadvantages of the electret condenser microphone. To balance the equation, let's now look at its considerable advantages.

FREQUENCY RESPONSE

A condenser microphone reproduces sound by charging and discharging the capacitor created by the movable diaphragm (one plate of the capacitor) and the fixed back plate (the other plate of the capacitor). The change in capacitance results from the motion of the diaphragm in response to impinging sound waves. The reference, or rest, voltage is the polarizing voltage of the capsule supplied in modern designs by a permanent charge affixed to the non-movable back plate. The motion of the diaphragm causes a voltage swing above and below the rest voltage.

Because the polarizing voltage is affixed to the back plate, the material selected can be optimized to maintain the permanent charge. This allows the material selected for the movable diaphragm to also be the optimal choice. It can be thinner and more compliant than older designs, thus allowing better low-end frequency response and at the same time providing the stability required for faithful high-frequency reproduction. Another benefit of the low mass moving system is vastly improved transient response, which means more faithful reproduction of the desired sound.

In comparison, designing a dynamic unidirectional (cardioid) moving coil microphone to yield a smooth, wide-range frequency response is extremely difficult without using more than one element. It is impossible to design one at a price equivalent to the electret condenser. The difficulty lies with the fact that the coil attached to the diaphragm makes the moving system very heavy in comparison to the ultra-low-mass electret diaphragm. This makes the moving system of the dynamic sluggish in comparison to the electret, thus making high frequencies difficult to reproduce; this results in lack of smoothness and poorer transient response. Also, low-frequency reproduction of the dynamic mic is more critically associated with diaphragm size than that of the electret.

OFF-AXIS UNIFORMITY AND PHYSICAL SIZE

Laboratory microphones are small in diameter and cylindrical for a reason, which is to minimize the disturbance of the sound field trying to negotiate the obstacle suddenly placed in its path. The original sound wave not only passes initially on a direct path from the sound source, but it can be counted upon to return a number of times as it bounces off various boundary surfaces and obstacles in the room. The shape of the microphone will affect both on-axis and off-axis sound. The smaller the microphone, the less the sound field will be disturbed and the more accurate the sound will be reproduced. Off-axis sound will also have less coloration. (It will sound more like the on-axis sound.)

Audio-Technica has long recognized the need for a line of miniaturized unidirectional electret condenser microphones of a functional, versatile design. A recent technological breakthrough has allowed them to develop the UniPoint™ microphone series, designed to be significantly reduced in size without sacrificing low-end frequency response, sensitivity, or signal-to-noise. To date, the best performance has been obtained from a unit approximately 7/16 inch in diameter. A ¾ inch length is easily attainable, but for aesthetic reasons, a 1⅜-inch length was selected. Miniature omnidirectional electrets smaller than this have been around for years, but the unidirectional design is more difficult due to the rear opening to the diaphragm.

Due to the physics involved, it is impossible to build a uni-dynamic microphone that even approaches these dimensions. The reason for this is that both sensitivity and low-end frequency of the uni-dynamic depend upon the size and compliance of the diaphragm as well as internal acoustical considerations. A uni-dynamic diameter diaphragm cannot provide a good low-end frequency response, and sensitivity suffers significantly, having 6 to 8 dB less output.

With regards to basic sensitivity, the electret condenser mic can always provide considerably more output than a dynamic. There are situations that demand less output, possibly equivalent to a dynamic, and this can easily be accomplished in the unit's electronics.

Using small diameter electrets for certain types of applications outlined below, we observed some interesting advantages inherent in this design.

OVERHEAD PICKUP OF CHOIR OR ORCHESTRA

In many sound reinforcement applications today, the number one priority seems to be that the people seated in the audience or congregation should be able to hear the music, singing, or person speaking without being able to see any "unsightly" microphones sticking up or hanging down. Microphones hung or flown cannot be hidden or they will not do the job they are intended to do effectively. A microphone used on a pulpit or lectern can be "buried," but in the majority of situations, adequate gain before feedback cannot be obtained using this method. The sound quality and performance obtained by using a microphone up close remains unparalleled. As usual, compromise is in order, but with the advent of the small diameter uni-dynamic, it is not as painful as it once was.

A very small microphone has a tendency to blend into its surroundings. By comparison, a standard size dynamic or electret can loom large on the horizon. A shotgun looms larger, and a boundary-type microphone mounted on a surface large enough to provide adequate low-end frequency response is larger still. Couple this with the fact that gain before feedback is typically a problem in sound reinforcement, which dictates that the microphone needs to be placed relatively close to the sound source as a function of its loudness. In the case of a person speaking, the microphone may need to be within a few inches, and a normal size microphone tends to hide the speaker's face.

As pertains to a microphone or microphones flown from the "rafters," the distance above the choir will also be a function of loudness as will the required balance from front to back of the choir. The louder the source, the farther away the microphone(s) can be before feedback occurs. From the standpoint of the typical choir, perhaps 20 to 30 voices, the microphone(s) needs to be approximately four feet above and in front of the heads of the front row to maintain overall balance. This necessitates using a very small microphone if you desire it to "disappear."

The orchestral application differs, somewhat, from the choir application. For the most part, an orchestra performing in a concert hall or auditorium does not require sound reinforcement. If it does require reinforcement such as in an entertainment showroom, individual instruments or pairs of like instruments need to be close mic'ed. Again, miniature microphones would tend to "disappear" when used in this situation.

Another possibility for the orchestral application might concern the necessity to place microphones for recording purposes. A miniature stereo pair suspended from above would be most appropriate in this situation, or a multitude of miniature microphones for reproducing individual instruments.

SPEAKING OR SINGING VOICE PICKUP, FIXED MICROPHONE

There is a multiplicity of situations calling for a fixed (non hand-held) microphone, especially for the speaking voice. Some of these applications are: pulpit, lectern, podium, boardroom, news desk, award show, game show, teleconference, variety show, and countless others. These applications can be broken down into two basic categories: sound reinforcement situation A, where the loudspeaker and microphone share the same environment, and sound reinforcement situation B, where situation A prevails, but sound is also picked up and broadcast for TV or radio use.

In situation A, we are faced with the restrictive factor of feedback. As mentioned previously, the well designed, small diameter uni will yield the smoothest on-axis and off-axis response, thus providing the best gain before feedback characteristics as compared to larger microphone types. This provides the possibility of increased working distance over other microphones, a highly desirable attribute. The miniature size makes the person speaking, as well as the audience, less aware of the presence of the microphone.

In situation B, feedback is not as much a factor as sound

Figure 2. Audio-Technica AT853 choir microphone.

quality, background ambience, and pickup of unwanted sound.

When the listener is in the same environment as the microphone and loudspeaker (situation A), extraneous noise, leakage, etc. are not noticeable unless they are out of the ordinary. However, when the listener is sitting at home in a quiet environment listening over a loudspeaker, ambient noise becomes noticeable to the nth degree. Background ambience can be controlled through close mic'ing techniques. Close mic'ing on TV requires aesthetic appeal which, in turn, requires a smaller microphone.

When the situation calls for a lapel (lavalier) microphone, a miniature uni can provide improved gain before feedback and/or a noticeable reduction of background ambience as compared to the usual omni-type. One word of caution: because the uni is obviously more directional than the omni, it should be worn in the center of the body. If it is placed to one side and the talent turns his head to the opposite side, a change in level will be noticeable.

RECORDING

Anyone who has visited a recording studio is aware that there are usually a wide range of microphone types available. The reason for the multitude of microphones stems from the fact that each microphone has its own particular sound character. A microphone used to reproduce the sound of a trumpet may not be appropriate for reproducing the sound of a violin. Individual mic'ing of instruments becomes highly subjective, sometimes even a tradition. Therefore, since the mini-uni is new and "untried," judgement about its ability to "faithfully" reproduce a particular instrument must be delayed for a while.

Size is not quite as important in recording, for obvious reasons, but a microphone that weighs only ½ ounce is a lot easier to attach to a boom, hang, etc., than the traditional microphone. Also, it won't get in the way as much or take up as much room.

To date, appropriate UniPoint models have been used mainly for stereo-pair and distributed overhead mic'ing applications (due to limited availability). Performance has been exceptional. There is reason to believe that these microphones will also excel in many individual instrument recording applications.

SUMMARY

The small diameter, wide-range, electret condenser allows use of a microphone for some purposes one has always wanted to apply. The mini-uni is clearly audible, but barely visible. Its advantages are as follows:

1. Smooth, wide-range frequency response
2. Superior off-axis rejection
3. High sensitivity
4. Lighter weight
5. Easy to suspend
6. Small size makes it "disappear" into the background.

The mini-uni can be used in the majority of applications; however, it is not recommended for use in the following:

1. Very loud vocal
2. Inside a kick drum
3. Mic'ing an amplified instrument speaker up close
4. Too small for most hand-held applications ■

RICHARD ELEN

Ambisonic Mixing—An Introduction

CRUCHFIELD Manor is situated on England's A330 between Ascot and Maidenhead in Berkshire, a few miles from Bracknell, and is an impressive building once owned by the Cadburys. Adjacent to the Manor is a small building, once the blacksmith's forge for the stables next door, which the present owner, composer/arranger and musician Keith Mansfield, has converted into a 24-track studio which is jointly owned by him and top recording engineer John Timperley. Principal recording equipment consists of a Soundcraft 2400 24/24 console and 24-track recorder, and a Studer B67 stereo machine.

Now that the equipment exists to produce Ambisonic mixdowns from multitrack and the fact that the Ambisonic Technology Center is "down the road" (10 miles away in Reading) made Cruchfield an ideal location for experiments with the new prototypes.

The primary requirement for mixing multitrack material into Ambisonics is that the tracks on tape can be localized individually into their required positions in the soundfield image. This requires quite simply, the Ambisonic analog of the common panpot. The simplest way of achieving this is to build a unit based around a 360 degree potentiometer with a sine/cosine law, allowing one to localize a signal source within the soundfield and giving a B-Format output. B-Format is the standard format for Ambisonic signals: it consists of four signals in its complete form, these being labelled X, W, Y and Z, where W is effectively mono, X represents front-back, Y left-right and Z up-down. As we are only considering horizontal surround-sound (without height information as in full "periphony") in this instance, we can dispense with the Z signal, although the circuitry is easily modified to cope with it. The present prototype "pan-rotate" unit is a 1 U rack-mount device with eight continuously-rotatable sin/cos panpots plus a "rotate" control which can be used to rotate the entire soundfield continuously through 360 degrees. The panpots can be switched to come before or after the rotate control and there are two external B-Format inputs—one pre-rotate and one post—which may be used to chain these or similar modules together. The rotate control may also be switched out altogether. The single B-Format output may be taken to a 4-track recorder, or to an encoder to produce a UHJ 2-channel-encoder master tape which can be cut to disc or broadcast in the usual way.

Unfortunately, sin/cos pots are very expensive, and in addition the pan-rotate unit requires individual mono inputs to be fed from the desk—this could make the console patchfield rather a mess—so other methods of localization had to be found. One such is the "converter" unit, which enables the console panpots to be used for localization, thus freeing the pan-rotate unit for dynamic panning effects, or where a higher degree of localization control is required. The production versions of the pan-rotate unit will be 2 U, high and will include a "radius vector" control for each panpot. These controls are used with the panpots to determine the distance of the source from the center of the image, so the angle can be set with the panpot and the apparent distance away can be adjusted with the radius vector control, which is a 270 degree pot with a switch at the end of the track, varying the distance from "normal" (switch position), corresponding to the periphery of the field, through "zero" (center) to "minus," thereby allowing panning across a diameter of the soundfield. The two controls can be used together, of course, to produce "spiralling" effects between the periphery of the field and the center.

CONVERTER

The converter unit is another 1 U rack-mount device with no controls at all other than a power switch. Inside are two independent B-Format groups with separate outputs, each group having five inputs. These are designated LF, RF, RB, LB and ES and are designed to be fed from four console groups plus an echo send (post-fader). The echo send level supplies the W (mono) reference while the console group inputs provide level information which is "converted," with the W signal, into B-Format. Thus any console panpot can be used as an Ambisonic localization control merely by turning up the echo send level (which must be the same level as the main channel output at any fader position) and routing to two of the four groups.

The converter unit assumes two things: first, that you can pan between odd and even groups; and second, that the panpots have a "constant power" characteristic. This latter gives you a signal level 3 dB down in the center in stereo, and is quite a common panpot characteristic (more on this subject later). FIGURE 1 shows the way in which the console panpots are used. If we assume that the four

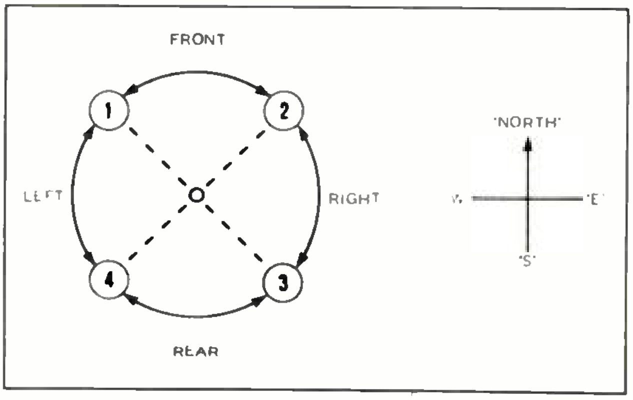


Figure 1. Panning with the converter unit. Panning across each 90-degree quadrant is achieved by routing to two of the numbered groups (one odd, one even). Note that while panning across left or right quadrants, panpot and signal move in opposite directions.

Richard Elen is the former editor of Studio Sound.

groups are 1, 2, 3 and 4, then selecting 1 and 2 allows panning across the front quadrant (LF to RF); selecting 2 and 3 pans across the right quadrant (RF to RB) and so on. The converter unit is thus ideally suited to the task of localizing most of the channels in B-Format, especially those which do not need the dynamic effect capability of the pan-rotate unit. And the fact that there are two such B-Format groups, both entirely separate, adds to the flexibility of the unit.

TRANSCODER

There is a further unit which has localization applications too, the Transcoder. This was one of the first Ambisonic units produced for mixing applications, and the original prototype which we used for mixes at the 1981 Cunard Hi-Fi show was based on the UHJ 2-channel encoder made by Calrec to accompany the Soundfield microphone. In essence, it takes four inputs which correspond to the four corners of a "quad" setup (generally four groups again) and "transcodes" them into a 2-channel UHJ-encoded signal. The unit offers two "soundstages," front and rear, which may be varied in width between 0 and 180 degrees at the front and 0 to 150 degrees at the rear. Panning between the front two groups with the console panpots pans across the entire width of the front stage, and so on. The main drawbacks are that the unit cannot produce a B-Format output, and that it is impossible to localize a sound in certain places (i.e. at the sides between the extremes of the two soundstages). This rather limits its effectiveness in normal mixing applications: its primary use would be to transcode 4-channel "quad" tapes into UHJ for Ambisonic release. It does, however, have application in basic mixing and is a useful place to bring back echo returns in a mix to be performed with the other units if the output is to be UHJ rather than B-Format.

DECODING AND MONITORING

Having covered the units used for Ambisonic localization—all of which were designed to allow a conventional stereo multitrack desk to be used for Ambisonic mixdown—we now come to one of the vital parts of the system: decoding, and monitoring the result. A number of Ambisonic decoders are currently available from any one of half-a-dozen manufacturers, although you won't find many in the shops—yet. We used the top-of-the-range decoder from Minim, which is a tastefully-furnished 1 U device with the minimum of controls, which may be rack-mounted with optional "ears." It has a screwdriver preset on the left to set up the decoder for the preferred speaker layout and is designed to drive four speakers arranged in any rectangular configuration between 2:1 in either direction—i.e., the ratio of the sides of the rectangle (with the speakers at bypass—normal stereo on the front pair of speakers—when the power is removed. A further pushbutton compensates the signal if the speakers are more than three meters away.

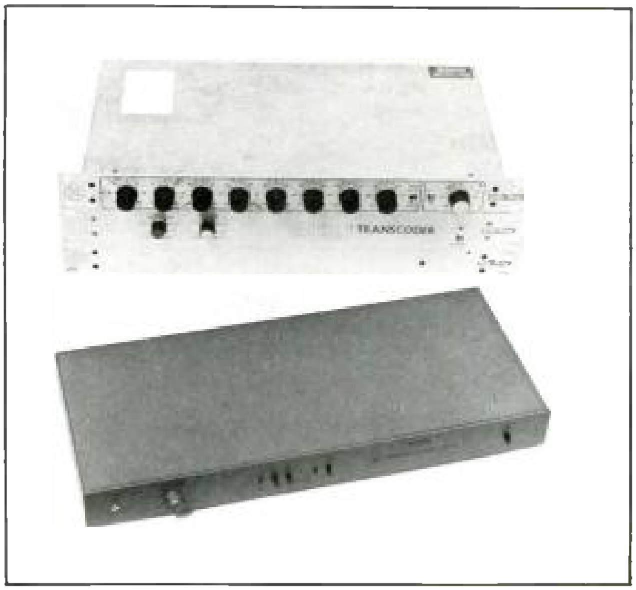
The speaker feeds were sent to a pair of HH V-200 MOSFET amps driving four Keesonic Cubs mounted on mic stand bases, enabling their height to be adjusted easily. Keesonic kindly supplied two extra Cubs with threaded baseplates to fit standard mic stands. There is no disadvantage in using small speakers for Ambisonics, as long as they are not too directional, particularly at HF.

SETTING UP

The first part of the setup procedure was to position the speakers in sensible places. The easiest way of doing this is to find a practical rectangular layout in the control room, position the speakers to point at the listening position (which is rather less critical than for stereo, although careful pointing of the speakers is more necessary if the HF dispersion is limited) and then measure the sides of the rectangle and set the layout control on the decoder accordingly. It is misleading to try to adjust the layout control by ear, especially if you are new to the system. The speaker height should be adjusted to normal monitoring criteria, with the provision that it is helpful not to have them too high, to avoid the image forming slightly disconcertingly over your head! But then again, the same is true with normal stereo monitoring. The decoder controls are then adjusted accordingly.

We checked the layout with a 2-channel UHJ "walkround" test recorded with a prototype Ambisonic panpot in 1981. There is also a "walkround" band on the current Hi-Fi Sound test record, made with a Soundfield mic. These tests simply consist of a voice calling out cardinal points around the sound field: North (center front), East (center right) and so on, plus the points in between. The cardinal points are in many ways more useful than the old style "quad" labels LF, RF, etc., as often the speakers will not be 90 degrees apart, and unlike "quad," there is no "pulling" of the image into the speakers. Thus "North West" might not be near a speaker at all, but it *will* be 45 degrees left of center front, if all is well. Minor compensations can then be made with the layout control if needed.

In our case, we derived the 2-channel send for the UHJ input of the decoder from the "alternate control-room monitor" outputs of the Soundcraft 2400, and here we had our first taste of a problem which kept on cropping up during the setup procedure. It is a problem that has nothing to do with Ambisonics—it is simply that Soundcraft, aiming for the US as well as the domestic market, have decided to wire their XLR connectors with pin 3 hot. All our gear, of course, right down to the HH power amps, is wired with pin 2 hot. This means that precisely nothing came out when we first tried the monitoring, and later the Transcoder. A few minutes work with the soldering iron perverted a few XLR leads to phase-reverse mode but the whole exercise was a pain.



Ambisonic Transcoder and AD 12 Decoder

Having sorted it out, we were left with an Ambisonic monitoring system which could be switched out without disconnecting anything. As John Timperley uses the Keesonics as secondary monitoring anyway, this was an essential consideration—indeed one that was designed into the whole "Ambisonic Mastering System"—one should be able to use the studio setup for ordinary stereo with the minimum of effort and complexity.

We next decided to set up the Transcoder. This has XLRs in and out, including the B-Format input which uses a 5-pin XLR wired to an agreed standard. All the Ambisonic units to date are unbalanced, and here again we were expecting conflicts with the XLR pinouts—and got them. Luckily the ADR Transcoder has links on the PCB to allow for curious XLR wiring "standards"—as will production units of the other devices—you simply remove the top cover and replug the links. Through all our experiments, this XLR perversity was our only problem. It took several minutes to sort it out every time it occurred.

The Transcoder was provided with XLR-to-Bantam leads which enabled us to patch it into the comprehensive patchfield on the 2400, for test purposes deriving the inputs from group insert sends 1 to 4, thus bypassing the group faders to ensure the same levels from each group. The 2-channel UHJ output was patched into the mix bus insert returns, to enable the main mix fader to be used to control the overall level of the UHJ-encoded signal to tape. An experimental mix was performed, which immediately indicated that the ADR production Transcoder was more accurate and flexible than the original prototype we had used back in 1981.

A couple of weeks later we tackled the interfacing of the converter and pan-rotate units, having used the intervening period to order up bantam to ¼-in. jack leads from Future Film Developments. These were brought down together with some 5-pin XLR leads to interconnect the various B-Format outputs and inputs. We used a speech signal from the radio as a source, patched into a console channel for testing (speech is very useful for checking localization as it contains a narrow range of frequencies: we also *expect* a voice to be coming from a particular direction). The post-fade line out from the channel was routed into the pan-rotate unit's number 1 panpot, with the rotate control switched out, the B-Format output from the unit being patched directly into the B-Format input of the decoder for test monitoring purposes.

We were immediately struck by the accuracy of localization offered by the pan-rotate controls, and by the fact that localization in B-Format is somewhat superior to that achieved via 2-channel UHJ. The controls proved easy to use, and the ability to send a sound right around the room with one control was fascinating, although some engineers might find 360 degrees rotation on one 360 degree pot a little coarse compared with the conventional panpot, which offers approximately 60 degree localization with a 270 degree control. This soon becomes natural, however, and the highly-accurate one-to-one correspondence of knob position to sound position is very satisfying. We then experimented with the rotate control, using it to rotate an entire soundfield derived from several panned channels through 360 degrees. Rotating the entire soundfield is quite remarkably disconcerting, although this isn't something one would normally want to do. Rotating certain elements within the sounfield is more subtle and useful.

It is amazingly easy to spot localization faults caused by incorrect wiring. The first B-Format lead had a fault in it: the Y signal line had come adrift in one of the plugs. This caused a rotated signal to move from back to front rather than around the circle, and was easy to spot as an error condition. The fact that there was no localization to left or right immediately pointed to the Y signal being missing, which was found to be the case. This having been rectified, we next tackled the converter unit.

We expected problems with the converter unit, as it is designed to take an auxiliary postfade cue send as its W source, which must be the same level as the feed to the groups. In addition, the unit is designed to work with constant-power panpots, as stated earlier. Soundcraft consoles do not use constant-power panpots. They use instead a compromise between constant-voltage (6 dB down at the center in stereo, which gives a good, wide image) and constant-power (3 dB down at center, giving good stereo/mono compatibility). This results in a law which gives about 4.2 dB down at the center in stereo. It is quite easy to modify these panpots to constant-power; you simply add resistors across the existing ones in the wiper of each gang of the pot, the added resistors being of the same value (3.3 kΩ) as the existing ones. However, I did not relish the thought of persuading John that it was a good idea to add dozens of extra resistors all over his console! So we decided to try it without console modification to see what happened. Neither did we measure the aux output to determine at what level we obtained the same level as from the main output. We just plugged in.

Setting a panpot to hard right, we patched the first group of the converter lines into group insert sends on groups 5 to 8, and derived the ES line from the Aux 1 output on the patchfield, turning the Aux 1 send on the channel to full and the Aux 1 master level control to 0. Routing the channel to groups 5 and 6 (to give panning across the front quadrant) we faded up the channel fader, again using speech as a test signal. A non-localized sound emerged which couldn't be said to have been situated anywhere. Turning up the Aux 1 master level, however, caused the sound to start "gathering itself together" at the North-East edge of the sound field.

With the Aux send level at 6 (on a scale of 0 to 10), the signal was accurately localized in the N-E corner, exactly as it should have been with the desk panpot in the far right position. This position, incidentally, was nowhere near the speaker: we had a rectangular layout slightly wider than it was long, and the sound in this instance was at an accurate 45 degrees to the listening position, rather to the left of the RF speaker and a good few feet beyond it! We had already noted with the pan-rotate box that the sound field extended some way beyond the limits of the speaker layout. Turning the Aux pot beyond 6 and towards 10 tended to drag the localization into the room and towards center-front. Rotating this control up and down produced a well-defined point (setting 6) at which the sound was accurately localized in the desired position. On checking this later, we found that indeed, this position corresponded to zero level with respect to the main output.

Then came the bigger test, which was to evaluate the law of the desk panpot operating Ambisonically via the converter box. Panning from right to left, we perceived a very smooth panning law, with the pot covering the front quadrant with a great deal of accuracy, and no discernible changes in level across the quadrant. Thus we did not have to modify the console: neither should mods be needed on consoles which have this kind of panning law compromise, which includes a good number of consoles currently available. Some consoles with constant-voltage panpots have a "compatibility" button which switches the panning law to constant-power, and these present no problems either if the "compat" button is pressed. We do not yet know what the effect would be of using a constant-voltage panpot through the converter, but we would expect a non-linearity of the panning law and slight changes of level across the quadrant, plus changes in the precision and quality of localization. As it was, however, the Soundcraft panpots exhibited a very smooth localization action which was comparable to that experienced in normal stereo. Panning across the other quadrants was equally impressive, one disconcerting (and unavoidable, unless you can pan between *any* pair of groups) factor being that panning over the side quadrants, the panpot is moving in the *opposite direction* of the sound! Familiarity with this slight idiosyncracy comes pretty rapidly, though.

MIX CONFIGURATION

Having ascertained that all the units were operating correctly, we set up the console for a mixdown. This was done in the normal way, but with the Ambisonic units set up as shown in FIGURE 2. Thus the "A" group of the converter box was fed into the post-rotate input of the pan-rotate unit, deriving its feeds from console groups 5 through 8 and Aux 1; the B group of the converter was patched into the pre-rotate B-Format input of the pan-rotate unit, with its signals derived from console groups 9 through 12 and Aux 2 (although this was not used in fact). The eight pan-rotate inputs were patched direct into channel line outputs as were required later in the mix, and the B-Format output of the pan-rotate unit was fed into the B-Format input of the Transcoder—this simply encodes it into 2-channel UHJ.

The fact that our Soundcraft had only 24 input channels led us to some rather unusual wiring when it came to echo returns. We wanted to use the converter box for primary localization, the pan-rotate unit for dynamic panning and rotate effects, and the Transcoder to return two reverb units—a MicMix spring system and an EMT Gold Foil—to the UHJ master output, spreading the EMT over the front stage and MicMix across the rear, fed from Aux 5 and 6 respectively. Normally we would have patched the echo returns into the monitor panel inputs, which return to the mix bus and whose faders can be "swapped" with the group faders (which are full-length as opposed to the monitor faders, which are small). However, in this configuration, the mix bus was to carry the 2-channel UHJ-encoded master signal, rather than straight stereo, so we couldn't return to there! Instead, we patched the first four group outputs into the Transcoder, returning the reverb units into the group insert returns on those

Fig 2 Mix configuration with Soundcraft 2400 and ADR Ambisonic units

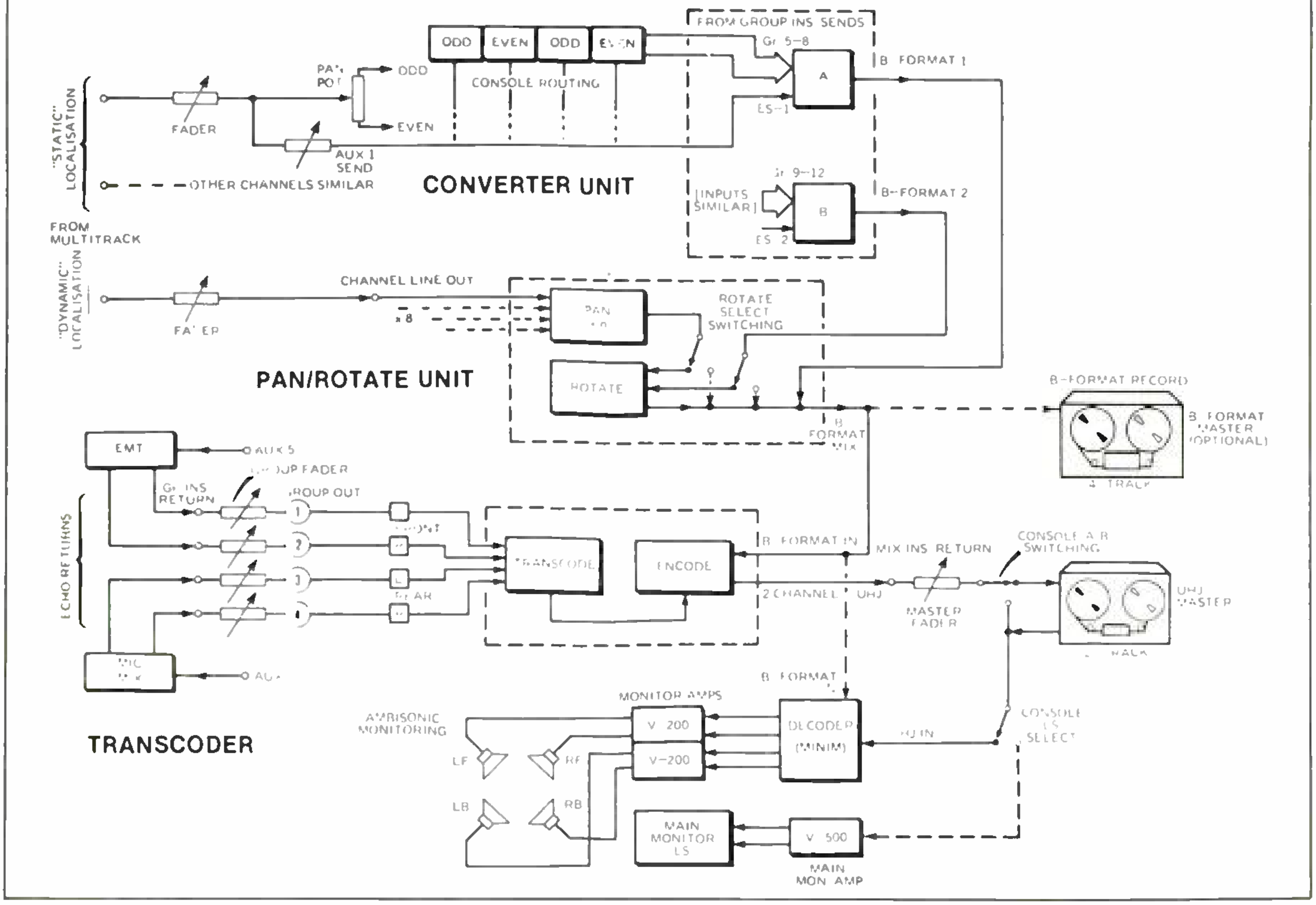


Figure 2. Mix configuration with Soundcraft 2400 and ADR Ambisonic units.

groups. This enabled us to use the large group faders to control the echo return levels. We could adjust the spread of the reverb returns in the sound field with the width controls on the Transcoder (which affect only the signals applied to the four transcode inputs, and not the B-Format input which is merely being encoded and mixed at unity gain with the UHJ from the transcode inputs to provide the main output from the unit). This enabled us to spread the EMT across up to 180 degrees at the front and the MicMix across up to 150 degrees at the rear. We had previously discovered, in our 1981 experiments—the first Ambisonic mixes derived from 24-track—that two reverb returns spread across front and rear stages like this is very effective: while only mono sends from the channels are used, you can send echo from a given channel and have it return either in the same sector as the source or in the opposite sector, just as it is sometimes useful in stereo to have a signal on the left with its echo on the right, for example.

The UHJ 2-channel mix output from the Transcoder was then patched into the mix bus insert return to allow the master fader to be used as before, and the mix was sent to the stereo master recorder in the usual way. This had the added advantage—as the decoder was patched out of the alternate monitor outputs—of allowing easy replay of encoded tapes, just by pressing the stereo replay buttons; plus the capability to select bypass on the decoder to check stereo and mono compatibility as well as being able to switch in the main monitors instead—with a single button—and hear the UHJ signal in "stereo" or "mono" on the main system. This proved to be very useful, and again fulfilled one of the design goals of the system, that of ensuring that the system could be returned to normal stereo use with as little replugging as possible—in this case, it was simply a matter of pressing either bypass on the decoder, or the speaker select button on the board, to return to stereo monitoring.

MIXING TECHNIQUES

The Ambisonic boxes were placed on a table next to the console, where the 24-track remote is positioned, for easy access to the controls. This is shortly to be replaced by a rack cabinet which will enable all the Ambisonic boxes plus the extra monitoring amps to be neatly installed, and still leave room on top for the 24-track remote at a reasonable height!

Almost immediately a number of mixing techniques were evolved to make best use of the system. One of the great strengths of Ambisonics is that there is a very high degree of compatibility between UHJ, stereo and mono. Indeed, we feel that the compatibility in these respects exceeds that of stereo. When an Ambisonic signal is "collapsed" into stereo—simply by replaying a 2-channel UHJ signal into two monitors with no decoder, for example—the "rear" images are simply projected on to the front, so that they are perceived at the same distance to the side of the stereo sound-stage as they are in surround. Thus a sound positioned, say, "South-South-West" in surround will end up half-left in stereo. Also, a signal rotating continuously around the field in surround will simply pan from left to right and back again in stereo.

Additionally, a UHJ signal replayed in stereo has a number of interesting characteristics. First, as Ambisonics uses phase and level to localize a source, rather than simply level as we are used to with conventional stereo panpots, the image is much less dependent—either in stereo or surround—of listener position, very much like a coincident-pair recording (indeed, Ambisonics is very much like a "three-dimensional coincident pair," and this is exactly what the Soundfield mic is). This means that you can wander about in the listening environment without losing the image. It also means that producer and engineer, for example, will hear the same thing. In stereo, as well as surround, the image is more stable, and sources can be more "tightly" localized. For some reason yet to be properly explained, the width of a UHJ recording played in stereo on loudspeakers is wider than the speakers, and stereo effects can be produced which go round the room without having the benefit of an Ambisonic decode system! On headphones the effect is even more pronounced: instead of the usual "line between the ears" of conventional panpotted stereo, some kind of "brain-decoding" produces the illusion of a sound field, albeit less deep (front to back) than Ambisonic speaker-decode.

All this means that stereo-mono compatibility is built into the Ambisonic system: indeed it is more compatible and more flexible than ordinary panpotted stereo. And those exciting effects will cut on an analog disc with no trouble, too, unless they are really loud and bassy, which means situation normal as compared with conventional stereo.

As much Ambisonic material will be played back on conventional stereo systems (they may even be mixed specifically for stereo applications!) initially, this compatibility must be borne in mind. There are no technical problems with compatibility, as has been discussed above; but there are also creative considerations

Soundfield Control Unit *Ambisonic UHJ Encoder* if the mix is to sound good in stereo. For example, newcomers to Ambisonics may find it very tempting to localize sources in places where you could never go in stereo—round the back and sides. It must be remembered that in stereo, these will be "collapsed" on to the front, and careful positioning is necessary (and easily practicable) to avoid cluttering of the stereo image. Impressive sounds which spread right across the soundstage from East to West sound very good Ambisonically, and you may like to put several stereo signals across from E to W—but in stereo they will all be split left and right (albeit somewhat beyond the speakers). Impressive front-back stereo splits will end up in mono when replayed conventionally, and this should also be borne in mind. This means that care should be taken in what signals get positioned where. Use the all-round positioning that Ambisonics offers by all means, but bear in mind what the relationships will be in stereo. Just as you choose what to locate near or far from center in stereo, choose in similar manner when mixing with Ambisonics. Here the ability to check stereo (or mono) easily at the push of a button proves itself.

We found rapidly that the best approach when balancing for Ambisonic/stereo compatibility was to localize sounds in decode mode first via the locator unit, later choosing which channels were to go to the pan-rotate unit, checking them in stereo on the small speakers (which will give a better image) as and when needed, and then switch to the main monitors to get sounds and EQ (because, in stereo, little speakers *are* little again) and to obtain the right *stereo* balance. You will find, as we did, that a mix balanced in this way—that sounds great in stereo—will also sound excellent through the decoder in surround. Consider especially—if you are using dual reverb systems like we were—the relationship between a source and its reverberation in surround. Here reverbs like the Quantec and the Lexicon—which have four outputs and two inputs—really come into their own. The Quantec, especially, with the capability to add early reflections to the first two channels (these should be wrapped across the front stage of the Transcoder), is exceptionally effective, and the Lexicon can of course be split to offer two stereo-out/mono-in systems with differing characteristics.

Of course, on some material, all the sources will sound best localized around the front of the sound field, with only reverb/ambience at the rear—this is obviously no problem. While rock music sounds impressive coming from all points of the compass, multitrack-derived orchestral material would sound a bit unusual performed in this way, with the listener in the middle of the orchestra! On such material, it is useful to have recorded a set of ambience tracks in B-Format on the multitrack from a Soundfield mic if you have one, although a stereo pair or two could be routed via the Transcoder if desired, along with the reverberation returns. If you have effects returns with routing, or spare channels for them, it is best to allocate four groups to the Transcoder inputs and route the returns and effects to them, bearing in mind that you can position returns with the ordinary panpots anywhere between the extremes of the two soundstages available with the Transcoder. They can also be routed via the converter box groups in a more flexible way, and if channels are available with normal routing for effects and reverb returns, localizing them in this way may prove more useful. The idea of the soundstages on the Transcoder does, however, give a useful impression with returns from reverberation devices, although of course it cannot be used if a B-Format master is required.

Generally speaking, it is preferable to mix Ambisonically to B-Format, and store the master on 4-track tape. It can then be encoded later through an encoder or via the B-Format Transcoder input for conventional 2-channel purposes, and 2-channel UHJ is the current conventional "release format" for Ambisonic recordings whether for FM broadcast, conventional analog record release, or Compact Disc. However, B-Format can later be encoded into other forms for transmission, for example "2½" or 3-channel UHJ, which can be transmitted via FM radio by using phase-quadrature modulation for the third channel (which may be bandwidth-limited, hence the "2½") as had been demonstrated by the IBA. Such a system is now permitted by the FCC in the USA. Equally, B-Format can be used to derive future formats for Compact Disc—provision is made in the system for various types of extra information including 4-channel release, and subcoding could be used to switch a decoder automatically when a suitable record is played. Thus 2-channel UHJ is fine for now, but B-Format will have applications in the future. In addition, one obvious application for Ambisonics will be in the audio-visual market: here it will be quite possible to use B-Format audio tapes on-site, played back via a decoder in the viewing area. Four channels of tape facilitate three channels of B-Format (assuming horizontal-surround only—hence no Z channel) plus one track for slide sync or even timecode.

SUMMARY

Ambisonic mixdown involves no techniques which are either difficult or time-consuming to learn. The existing units, which will be available shortly, are not expensive compared with modern outboard gear and add an impressive extra dimension in stereo, let alone surround-sound, to even an averagely-equipped studio. With more console facilities, as are available on many modern boards, even more flexibility is possible. VCA grouping, for example, would enable the various B-Format group levels to be adjusted easily with a single fader—for example, the two B-Format groups in the converter box could then be balanced independently and used for subgrouping, as could the group of signals feeding the Transcoder if it is in use. As it stands, all the Ambisonic units mix at unity gain, and thus need external control for group balancing. Of course, VCA groups can also be used normally, and this may be a preferable alternative, controlling the individual groups of faders on the console irrespective of which Ambisonic device they are feeding.

While care needs to be taken in the setting up of the equipment, especially the decoder/monitoring section, this requires no more concern than usual in such areas. The system has the big advantage that returning to stereo is remarkably easy, whether during the course of an Ambisonic mix, or after the session when conventional stereo needs to be restored. All those who have heard or experimented with the system to date—including experienced musicians, engineers and producers—have commented on the remarkable compatibility of Ambisonics with stereo and mono, and the flexibility and ease of use of the system. ■

This preceding article was reprinted from ***Studio Sound****, September 1983 issue, with permission.*

ALBERTO I. ESCRINA

Recording Studios of Buenos Aires

Argentina's studios offer modern concepts and equipment at cut-rate recording costs. Here is a rundown of some of the best in Buenos Aires.

ARGENTINA, the southernmost nation in South America, has a population of 28 million. Due to the fact that 40 percent of the country's inhabitants live within a 500 km. radius of Buenos Aires, its capital city, nearly all industrial and commercial activity either takes place in, or is managed from there. The world of recording studios is no exception to this rule.

The object of this article is to list the main studios of Buenos Aires and to inform db readers of their technical characteristics, equipment, and the services they offer.

Readers may be surprised to learn that Argentine studios are designed and built in accordance with the most modern acoustic concepts (two of them incorporate LEDE treatment) and that they are equipped with truly state-of-the-art electronic technology.

Leading this "new wave" of design is Carlos Piriz, Interface SRL's exclusive studio designer, who is in charge of all the company's building and remodeling projects.

Argentina's low rates will probably be another surprise. This is due to the exceptional economic circumstances the country is going through. Experts believe these conditions will remain the same for at least two more years. In addition, low rates are charged by studio musicians, resulting in a constant stream of foreign productions from such countries as Japan, Spain, Mexico, France, and the Latin American market. These foreign producers and record companies are able to lower their costs substantially by recording in Argentina. It is worth pointing out that Argentine studio musicians are considered on a par with the world's finest.

A list of Argentina's studios and their main characteristics follows.

RCA ARGENTINA

Year of construction: 1971
Design: RCA U.S.A.
Studio Size: 12m × 18m
Control Room Size: 6m × 9m
Studio Rate: $25/hour
Services: Recording up to 8-track
Mixing
Cutting

Equipment/Recording Studio:
Neve 48 × 32
Studer 8-track
Studer A800 24-track coming this year
8 EMT 140 with remote
32 Dolby units
RCA RD500 monitors

Equipment/Cutting Room
Neumann VMS-70
Scully 545 w/Neumann VMS-70 electronics
Studer 2-track

PHONOGRAM

Year of remodeling: 1975
Design: Phonogram studio technicians
Studio Size: 18m × 12m
Control Room Size: 6m × 3m
Studio Rate: $20/hour
Services: Recording/Mixing up to 16-track

Equipment/Recording Studio
Polygram 24 × 16 Board
Studer A80 MK II 16-track
AKG BX 20
EMT plate
dbx 16-track
Dolby 2-track
JBL 4333 monitors

MUSIC HALL

Year of remodeling: 1981
Design: Carlos Piriz
Studio Size: 16m × 11m
Control Room Size: 7.60m × 5.80m
Studio Rate: $25/hour
Services: Recording/Mixing up to 24-track
Cutting

Equipment/Recording Studio
MCI JH528D Automated
Ampex ATR 124 24
EMT 140
EMT 140ts
UREI 815A monitors

Equipment/Cutting Room
Neumann VMS-70
UREI 815A monitors

CBS

Year of construction: 1962
Design: Federico Malvarez

Alberto I. Escrina is the marketing manager of Interface S.R.L.

Studio Size: 14m × 10m
Control Room Size: 4m × 3m
Studio Rate: p.o.a.
Services: Recording/Mixing up to 16 track
Cutting

Equipment/Recording Studio
API 2488 24 × 16 board
Ampex MM1100
4 EMT 140ST
EMT 240
16 Dolby
UREI 813A monitors

Equipment/Cutting Room
Scully/Westrex

FONEMA

Year of construction: 1978
Design: Oscar Bonello
Studio Size: 13m × 8m
Control Room Size: 5m × 2.5m
Studio Rate: $35/hour
Services: Recording/Mixing up to 24 track

Equipment/Recording Studio
Solidyne Series 2000 board
3M M79 w/autolocator 24-track
EMT 140
EMT 240
JBL 4343 monitors

MOEBIO

Year of construction: 1982
Design: Carlos Piriz
Studio Size: 8.80m × 8m
Control Room Size: 7m × 7m
Studio Rate: $25/hour
Services: Recording/Mixing up to 32-track
On location Recording up to 32-track

Equipment/Recording Studio
Soundcraft 2400 board
Two Soundcraft 381-16 (synchronized)
Ursa Major 8 × 32
16 dbx
JBL 4435 monitors

DEL CIELITO

Year of construction: 1983
Design: Carlos Piriz
Studio Size: 12m × 7.5m
Control Room Size: 6m × 5m
Studio Rate: $20/hour
Services: Recording/Mixing up to 16-track
On location recording up to 16-track

Equipment/Recording Studio
Tascam Model 15 board
Tascam 90-16
Orban 111B
16 dbx
Yamaha NS1000 monitors

FRANCIS SMITH

Year of construction: 1980
Design: F. Smith studio technicians
Studio Size: 7m × 4m
Control Room Size: 4.50m × 3m
Studio Rate: $15/hour
Services: Recording/Mixing up to 16-track

Equipment/Recording Studio
Yamaha PM2000 (modified) board
Tascam 85-16
Master-Room XL305
16 dbx
JBL 4315 monitors

ION

Year of remodeling: 1978 (control room and cutting room)
Design: Carlos Piriz
Studio Size: 18m × 7.60m
Control Room Size: 6.50m × 5.40m
Studio Rate: $30/hour
Services: Recording/Mixing up to 24-track
Cutting

Equipment/Recording Studio
MCI JH528 automated board
MCI JH24 w/autolocator
EMT 240
EMT 140
JBL 4343 monitors

Equipment/Cutting Room
Scully/Ortofon
MCI JH110
JBL 4343 monitors

PANDA

Year of construction: 1982
Design: Carlos Piriz
Studio Size: 11m × 4m
Control Room Size: 7m × 5.50m
Studio Rate:$20/hour
Services: Recording/Mixing up to 16-track

Equipment/Recording Studio
Tascam model 15 board
Tascam 85-16
AKG BX-15
16 dbx
JBL L300 monitors

ATC

Year of construction: 1978
Design: Federico Malvarez
Studio Size: 14m × 13m
Control Room Size: 7m × 6m
Studio Rate: p.o.a.
Services: Recording/Mixing up to 16-track

Equipment/Recording Studio
Dimerson 24 × 24 automated board
MCI JH16
EMT 140
16 Dolby
Altec A-7 monitors

Any inquiry about recording studios and/or services in Argentina will be promptly answered. Write to:

INTERFACE SRL
Solis 1769
1134 Buenos Aires
Argentina

■

JOSEF DORNER

From 15 MilliMaxwell to 1,200 NanoWebers

Need an explanation? Here's a look at the evolution of fluxivity and level standards.

AN EYE-CATCHING HEADLINE to attract or confuse the reader? Well, certainly not the latter, because one is often confused enough when trying to understand the data printed on the specification sheets accompanying many tape recorders. We find distortion performance and signal-to-noise ratios referred to 185, 200, 250, or even 370 nWb/m. In some cases, flux values as high as 1,000 and even 1,040 nWb/m are mentioned, while Europeans use such odd values as 320 and 514 nWb/m for reference fluxivity. Where do all these different values come from? Maybe some light can be shed on this matter by looking back into the history of magnetic recording.

A STANDARD IS CREATED

Let's go back about 30 years, to a time when Geman technicians were already talking about a standard tape flux, well before their cohorts on the other side of the Atlantic were. Their definition read something like this: "...for the purpose of program exchange, a reference value for remanent tape magnetization has to be established. When using general purpose tapes, this level shall be approximately 6 dB below maximum output level. *(In reality, the span was only 4 dB at that time—author).* Fortissimo passages shall modulate the tape up to that reference level. This is of importance for the purpose of program exchange. Only in applications where program exchange is not a criterion, modulation up to 3 percent of third-harmonic distortion may be tolerated; this is in order to achieve a higher signal-to-noise ratio and better utilization of the tape. For class 38 (15 ips) the reference level is set to 200 milliMaxwell and for class 19 (7.5 ips) to 160 milliMaxwell." (Draft for DIN 45 513).[1] Remember this dates back to 1955!

In America, all that was known at that time, as far as a recording standard was concerned, was the calibration tape made by a well-known manufacturer of magnetic tape recording equipment (Ampex) with a reference level recorded on it, which was named the "Operating Level." That Operating Level was used to calibrate the VU-meter to obtain a 0 VU deflection. By digging a little deeper, one was able to learn that this Operating Level

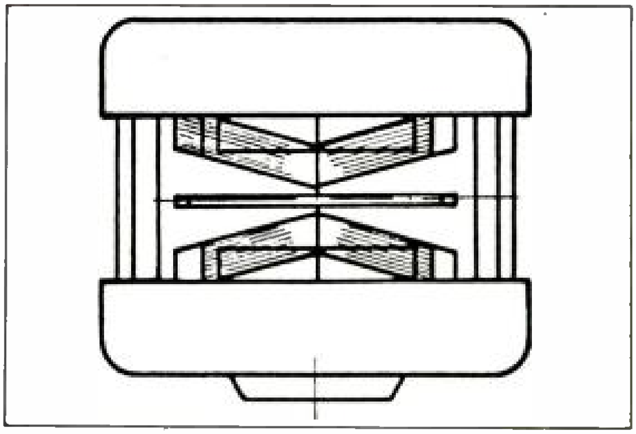


Figure 1. Butterfly Head.

corresponded to 1 percent of third-harmonic distortion on the then most widely used (general purpose) recording tape in the USA.

MOL AND THE VU METER

At this point it may be of interest to note that this general purpose tape produced 3 percent of third-harmonic distortion when it was modulated to a point some 6 dB above Operating Level. Many a studio (particularly some European studios)—where the VU-meter started to make its appearance in the early '60s—may have been misled by this fact to think that a VU-meter has to be operated with a 6 dB lead. By 1966, however, the Deutsche Industrie Normen (DIN) had already recognized that this was not quite correct, because it is stated in the explanatory note accompanying DIN 45 406 that "....on average the lead required is about 8 dB (8 VU). Deviations from this average by ±5 dB, however, are not exceptional."

If one compares this with the old RETMA TR 105 B standard (1951) for Audio Facilities for Broadcasting Systems, one can read the following in section V.2.a: "If a VU-meter is incorporated, it shall remain as normally connected, and its multiplier shall remain set for a signal which is 10 dB below standard output level" (Standard output level is +18 dBm).

Can one not conclude from this that signal peaks, as recorded on tape, produced flux values up to some 8 to 10 dB beyond the 1 percent distortion level, in other words, far in excess of the 3 percent distortion point? Yes, because in 1965 the NAB standard for reel-to-reel recordings has the following to say in a footnote to section 2.04, which relates to the standard reference program level: "It is well established that at least 10 dB margin is required between the sine wave load handling capacity of a system and the level of program material as measured by a standard volume indicator." The NAB standard reference level is described in section 2.03 with a footnote which reads as follows: "The recording was made...at an output level 8 dB below that which produces 3 percent third-harmonic distortion." (This is not contradictory to the above statement because it simply defines a level of tape magnetization which is to serve as a reference.) So, where do we go from here?

THE AMERICAN REFERENCE FLUX

Fortunately, John McKnight in the United States seemed to have been bothered by this lack of a precise value for recorded tape flux. As a consequence, he investigated this situation and prepared his findings for publication in the *Journal of the Audio Engineering Society*.[2] A reference flux of 100 nWb/m is mentioned or suggested in that investigation, and one reads for the first time 210 nWb/m for the earlier discussed Operating Level and 165 nWb/m for the NAB Standard Reference Level. Later on, these values were downward corrected slightly, and from a 1972 data sheet of a manufacturer of calibration tapes, one can read 185 nWb/m for the Operating Level and 150 nWb/m for the NAB Reference Level.

At this point, we should pause to take a closer look at the units of measurement.

UNITS OF MEASUREMENT

NanoWeber-per-meter is the value of fluxivity that would be measured if the tape was 1 meter (or approximately 39⅜ inches wide. Reducing this to a more realistic width, namely 1mm (or 39 mil), the unit became pico-Weber-per-millimeter, which was 0.1 milliMaxwell per millimeter in the days before the ST units came into force. In the case of the NAB Reference Level, the result is 15 mM/mm, which explains one of the values mentioned in the title of this article.

Since we are already doing some calculations, let's look at the previously mentioned German reference of 200 milliMaxwell for ¼-inch tape. If we divide that figure by the metric equivalent of ¼ inch, which is 6.25mm (tapes today are 6.3mm wide), then we get the figure of 32 mM/mm. Converting this to nanoWebers, we arrive at the standard 320 nWb/m.

It may be worth mentioning at this point that in a comparison of U.S. and European levels one must be aware of the fact that the ANSI S 4.6 method of measuring remanent flux yields a value which is lower by 0.8 dB, as compared with a measurement performed in accordance with DIN 45 520. In practice, this means that when comparing calibration tapes of U.S. and European origin, the U.S. tape will yield a higher signal level because what is 200 nWb/m in the U.S. would measure 220 nWb/m in Europe. (This also explains the previously cited downward correction from 165 to 150 nWb/m.)

STEREO-MONO COMPATIBILITY

After this digression into levels and their history, let's continue on. Magnetic oxides were improved over the years, making higher levels of magnetization possible without adversely affecting distortion performance. This made it feasible to raise the operating level (0 VU) to 250 nWb/m for the so-called High Output tapes. In Europe (more precisely in Germany), the advent of stereo made those exacting engineers reach for their slide rules, because stereo/mono level compatibility was their goal. Music productions were already recorded in stereo, yet broadcasts were still in mono. Such a stereo recording, when played back on a full track head, did not produce the same signal level as that which resulted when playing a mono recording; there was some unused, unmagnetized "land" between the stereo tracks, and left and right signals were not adding up correctly. One can live with reduced cross-talk performance in stereo, so the tracks were widened until they were spaced only 0.75mm apart, making each track 2.75mm wide. As a result of this, the core sections of the head spread out at an angle to accommodate the windings. With this, the Butterfly Head was born (see FIGURE 1).

The tape's width was utilized to a possible maximum, but stereo/mono level compatibility was still not reached. A few quick calculations and one can see that a stereo recording has to be modulated to 514 nWb/m in order to produce the same signal level as that which is obtained from a 320 nWb/m mono recording when playing the stereo tape on a monophonic reproducer.

Total flux, mono on ¼-inch (6.25mm) tape:

$$320 \text{ nWb/m} \times 6.25 = 2000 \text{ nWb/m}$$

Stereo played on full track head:

$$\sqrt{(514 \text{ nWb/m} \times 2.75)^2 + (514 \times 2.75)^2} = 1999 \text{ nW/m}$$

The goal was reached: The fader on the mixing desk did not have to be moved, regardless of whether mono or stereo recordings were played! At the time it was a bit strange, perhaps, to see blank tape appearing on the market which was labelled "stereo," though this simply meant that such a tape could be modulated to the higher stereo level without any increase in distortion.

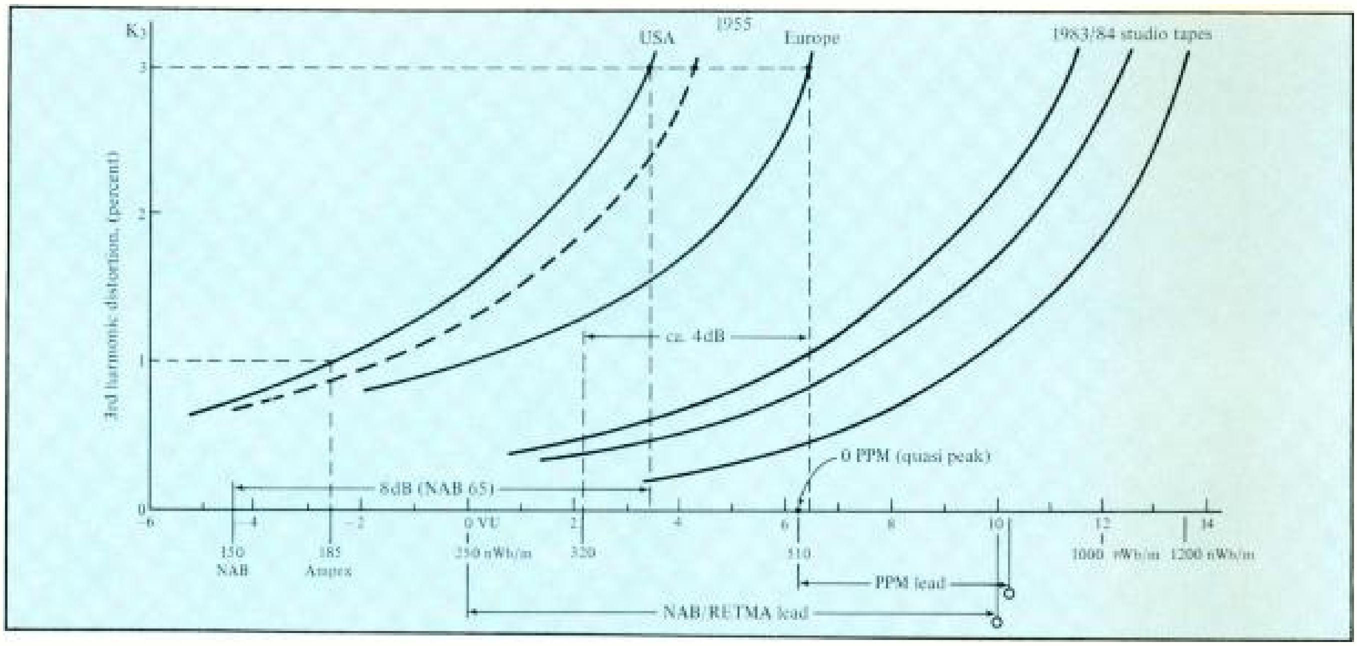


Figure 2. Maximum Output Level (MOL) performance of magnetic recording tapes at 15 ips. O = Theoretical peak flux values when aligning VU meter or PPM as described in text. The dashed line (1955) represents the performance of the old U.S. tape if flux values were measured in accordance with DIN.

Stereo/mono compatibility—which is not of much interest anymore—is thus explained, but what about universal compatibility of recorded levels in general?

VU VS. PPM AND PEAK FLUXIVITY

In America the VU-meter is still favored while in Europe the peak program meter (PPM) is predominant. The performance characteristics of the latter are described and specified in IEC 280-10 and in DIN 45 506. It is a quick-acting meter, and because of this, it is also called a "quasi peak-reading meter." However, as suggested by the word *quasi*, it is not a true peak-indicating device. A closer examination of its characteristic behavior suggests that short modulation peaks may overshoot by 1 to 4 dB.[3]

A graphic presentation (FIGURE 2) of the maximum output level performance (MOL) of various tapes, including the most modern oxides, shows how tape performance has improved over the years. The point of maximum modulation, which is universally considered to be the level at which the third-harmonic distortion content measures 3 percent,[4] has shifted gradually to higher flux values, with 1,200 nWb/m being reached by at least one state-of-the-art tape. This explains the second figure in the title. Quite a wide range from the NAB reference of 150 nWb/m via the high-output operating reference to the German DIN levels for mono and for stereo, up to the MOL which is possible today.

Attempts to establish references of even figures have been repeatedly made. For example, there is the EIA standard RS-400/1972 containing a reference to CCIR 79-1/1966 at which time the value of 100 nWb/m was recommended, and in more recent times, one finds 400 nWb/m mentioned in a newer EIA standard. But all this is of little help to a studio's maintenance engineer when faced with the decision of how he should calibrate his level meters. So, in analyzing this historical retrospect, it comes almost as an automatic conclusion that 250 nWb/m (or even 320 nWb/m) would be a good reference for calibrating a VU-meter to its 0 VU reference deflection, as it would allow the modulation peaks to reach up to 800 or 1,000 nWb/m. In the case of a quasi peak-reading meter or PPM, however, the 510 nWb/m (or 500 for simplicity's sake, being twice 250) would be an equally good reference because its assumed 4 dB overshoot would again result in a peak magnetization in the range of 800 percent nWb/m, still well below the accepted MOL of 3 percent third-harmonic distortion.

It's up to the individual engineer's discretion, of course, as to how hard he intends to drive his tape into saturation. It should be borne in mind, however, that for every dB gained in signal-to-noise, one must pay with a disproportional increase in distortion, a fact which was discovered long ago by a pioneer in the development of new recording techniques.[5]

Analog recording may still be around for a while, and so it is hoped that useful conclusions can be drawn from this article which help to ensure that the inherent quality of analog is not given away unwisely, as may all too often be the case. ■

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5. Langevin, Robert Z. "Intermodulation Distortion in Tape Recording," *Journal AES* (July 1963), Vol. 11, pp 270-278.

MINIATURE ELECTRET LAVALIER MIC

• Sennheiser's MKE 2 miniature electret lavalier mic, which may be sewn in place or attached to clothing with an alligator clip, measures 0.24 inches in diameter by 0.43 inches in length. It will be available in several versions: The MKE 2-0 has a 10-foot cable terminating in stripped-and-tinned leads for connection to various wireless microphone transmitters. The MKE 2-3 has a connector for attachment to the K3U powering module. When used with the K3U, it may be phantom-powered (9 to 52 volts) or operated from the mercury battery contained in the K3U. It includes a three-position 12 dB/octave low-frequency rolloff switch, and a 10 foot cable. The MKE 2-6 has an in-line battery supply with on/off switch and a 13 foot cable terminating in a 3.5 mm diameter miniplug.

Mfr: Sennheiser

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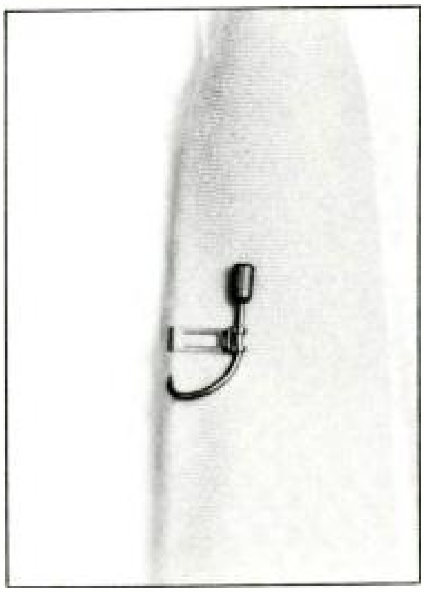


PORTABLE WIRELESS MICROPHONE RECEIVER

• Cetec Vega's Model 66A PRO battery-powered portable wireless microphone receiver has an attractive new appearance, and lower power consumption than the previous version for a longer battery life. It can operate from a 12-volt camera pack (or other external DC power source, 10.5 to 18 VDC) or from its internal battery pack (four 9-volt alkaline cells). Designed for on-location and portable use, the 66A is fully compatible with all previous PRO transmitters. With the optional Dynex II audio processor, the 66A is also fully compatible with Cetec Vega's new T Series transmitters. The receiver operates on any crystal-controlled frequency between 150 and 216 MHz. The receiver pre-selector is a true two-pole helical resonator filter. The 66A uses a combination of LC and multiple-pole crystal IF filtering to provide outstanding IF selectivity and adjacent-channel rejection. The wide-band FM demodulator has low distortion (system THD is typically 0.3 percent or less) and excellent dynamic range. With the Dynex II audio processor, usable dynamic range is in excess of 100 dB.

Mfr: Cetec Vega

Circle 17 on Reader Service Card

MICROPHONE HEADSET

• Nady Systems' new HeadMic™ is a miniature, ultra-light microphone headset ideal for singing drummers and keyboard musicians. The HeadMic™ allows complete freedom of movement, with audio performance and feedback isolation comparable to the best hand-held mics. The unit is virtually invisible on stage. The mic boom and windscreen are available in tan or black to match skin or beard, and the unit's headband is hidden completely in the performer's hair. The malleable headband bends to fit the head and retains its spring to hold the HeadMic™ in place even during rigorous movement. The headband and mic boom are adjusted through holes in a rubber adjust hub for a one-time personalized custom fit. The HeadMic™ comes standard with a state-of-the-art Countryman Associates' Isomax™ directional microphone. The Isomax is noted for ultra-acoustic isolation (freedom from feedback), full range frequency response, high SPL capacity, and ultra-miniaturization. The HeadMic™ is also available with Audio-Technica's ATM-831 unidirectional microphone or other comparable mics upon request. The unit comes standard in two configurations: HM-1, HeadMic™ with switch box and LEMO connector for use with a wireless transmitter (Nady's VHF 900 is ideal); or HM-11, HeadMic™ with switch box and 10-foot cable with XLR connector for direct connection to mixer.

Mfr: Nady Systems, Inc.

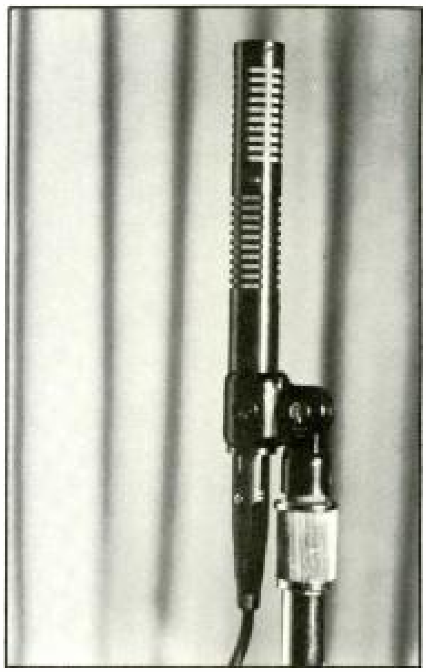
Circle 18 on Reader Service Card

COINCIDENT STEREO RIBBON MICROPHONE

• Speiden & Associates' new SF-12 Coincident Stereo Ribbon Microphone possesses superior stereo separation and imaging, wide-band peak-free response with negligible off-axis coloration, and superb transient response. The microphone employs two 1.5-micron ribbons of pure aluminum, each weighing in at ¼ milligram. This results in the open, airy sound quality for which ribbon microphones have become justly famous. Polar patterns are symmetrical figure-8, positioned at mutually right angles for classic Blumlein stereo pickup. Stereophonic accuracy is excellent with either the X-Y or M-S techniques. High sensitivity is maintained by an efficient magnetic circuit having very low flux leakage. The SF-12 is ruggedly built, employing high quality workmanship and materials. It is supplied with 18 feet of cable (four conductors plus shield) and an adapter terminating in two standard XLR-type connectors for direct connection to two professional 250-ohm preamplifiers. A standard adapter is also supplied. Microphone dimensions are one inch in diameter by eight inches in length, and its weight is 12 oz.

Mfr: Speiden & Associates

Circle 19 on Reader Service Card



DYNAMIC HYPERCARDIOID

• Peavey's HD-40 is a hum-compensated dynamic hypercardioid, ball-type with swivel adapter and foam-lined tote bag. The mic is available with 25-foot low-noise high- or low-impedance cable with quality audio connectors and optional rotary On/Off shorting switch.

Mfr: Peavey

Price: HD-40, $129.00
HD-40L, $149.00
HD-40LS, $157.00
HD-40H, $161.00
HD40HS, $168.00

Circle 20 on Reader Service Card

CARDIOID CONDENSER MIC

• Electro-Voice's model BK-1 microphone has recently been added to its product line. The new microphone departs from EV tradition in a number of respects, including its unique physical profile and handsome black design, while combining popular features from other mics in the EV product line. Dubbed the "Black Knight" by EV, the BK-1 is a cardioid condenser mic especially suited to the needs of vocalists, though it is also excellent for other applications. The Black Knight's notable mix of features offers the performer usable bass-boosting proximity effect that tailors low frequency response with changes in the working distance. The mic's high sensitivity, smooth, peak-free frequency response, and cardioid pickup pattern assure excellent rejection of unwanted off-axis and reflected sounds that cause feedback in a live entertainment situation. This electret condenser-type mic can be powered by either a battery or phantom power. The mic is fitted with an On/Off switch to greatly prolong battery life.

Mfr: Electro-Voice

Circle 21 on Reader Service Card

CONDENSER MICROPHONE

• Beyer Dynamic's MC 734 vocal condenser microphone is designed for use in on-stage and recording situations. Hissing and "popping" noises are suppressed, as is handling noise of the mic. The MC 734 features flat frequency response from 20 Hz to 18 kHz, and is equipped with a 3-step filter to compensate for the proximity effect in close mic'ing situations. Rumble and other stage noise is suppressed by a built-in stage resonance filter. The MC 734 withstands sound pressure levels of 138 dB at 1 MHz, and can be fed by any 48-volt phantom power source.

Mfr: Beyer Dynamic, Inc.

Circle 22 on Reader Service Card

SOUNDFIELD MICROPHONE

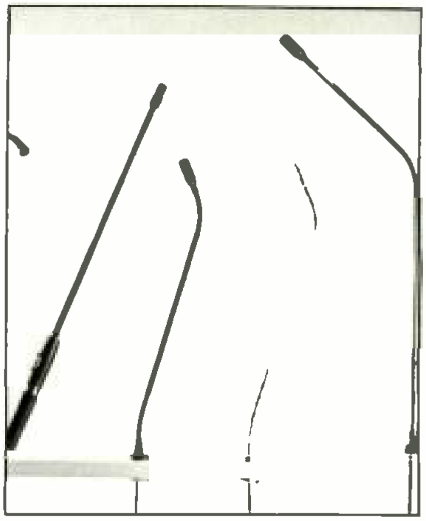
• While other microphone designs necessarily incorporate phase error in their sound pickup to compensate for non-coincident capsules, the design of Calrec's Mark IV Soundfield Microphone is based on the theory of sampling on the surface of a sphere, which allows reconstitution of signals present at the center of the sphere when the outputs are summed—thus achieving true coincidence. The microphone head incorporates four condenser capsules mounted in a regular tetrahedron and individually matched to the associated control units. This design reduces phase error in the signal to previously unobtainable levels while maintaining linear frequency response in all directions. A particular significance to audio engineers is the fact that in the B-format signals of the Soundfield Microphone, the pressure and gradient components of the apparently single diaphragm are presented as separate electrical signals which may be optimized independently. An additional and unique advantage of the Soundfield design is that the mic can be steered in all directions and zoomed in or out at the time of the session or post production stage, as well as varying the polar patterns from omni through all the intermediate cardioids to figure-8. The microphone's B-format output may also be encoded for Ambisonic Surround Sound reproduction; for conventional stereo reproduction, the microphone's stereo output may be used again at the time of the session or in post production.

Mfr: Audio + Design-Calrec, Inc.

Circle 23 on Reader Service Card

ELECTRET CONDENSER MICROPHONES

• Audio-Technica's new UniPoint series contains five ultra-low-profile back electret condenser microphones: the AT837, AT853, AT855, AT857, and AT859. According to the company, the five new models are said to reproduce sound in a highly natural manner with minimal coloration, and, because of their cardioid pickup patterns, may be used at considerable working distance from the sound source. This feature, coupled with their slim, unobtrusive outlines, makes them good choices for pickup of lecturers and entertainers from either a podium or floor stand, and for unobstructed pickup of choirs and similar sound sources. The AT837 is a double-gooseneck model with frequency response tailored for close-up use. The unusual double-gooseneck configuration, featuring a straight pipe section with a flexible gooseneck at either end, permits the same freedom of movement that would be possible with a long goosneck, but it is far easier to keep the microphone from developing twists and kinks characteristic of the conventional gooseneck model. The AT857 double-gooseneck model is nearly identical in shape to the AT837, but with frequency response tailored for natural sound reproduction at an extended working distance from the sound source. It may, of course, be used close up as well. The AT853, though designated as a choir microphone, can be used in a variety of situations requiring a highly inconspicuous microphone that can be suspended almost invisibly over a sound source. Another Unipoint model, the AT855, is similar to the AT857, except that it is mounted on a long, slim tube with 27 ⅝-threads at the lower end so it can be fastened directly to a microphone stand and used where a podium is not provided. The single bend in the tube permits placement of the microphone stand at a comfortable distance from the sound source. Completing the series is the AT859, a wand microphone designed especially for interview applications, which features a handle and a telescoping "wand" that may be pulled out to enable the interviewer to carry on a conversation with another person, even when it is not possible to get close. The microphone is normally 11½ inches long, but may be extended to a length of 18¼ inches. All of the UniPoint microphones feature broad, flat response curves, and all are balanced



low impedance. Each member of the new series may be operated from a variety of power sources. They may all be operated from a variety of power sources. They may all be used with a 30 VDC source, single-ended without a power module or with any 9 to 52 VDC phantom power source. In addition, the AT853 and AT859 may be used with a self-contained 1.5-volt battery.

Mfr: Audio-Technica

Circle 24 on Reader Service Card

PROFESSIONAL MICROPHONE

• Crown's new Model 12SP microphone is well-suited for recording, sound reinforcement, broadcasting, and electronic news gathering. The 12SP must be phantom-powered by a supply providing 12 to 48 volts. The unit features a transformer-balanced, low-impedance output and an integral XLR connector; wide, smooth response with high-frequency emphasis for brilliance; low noise and high output level; a hemispherical pickup pattern; and high sensitivity and excellent reach for clear pickup of distant sounds. In operation, the ruggedly-constructed 12SP can be placed on a surface such as a floor, table, or lectern, used as a hand-held microphone, or affixed to a surface near a sound source such as the underside of a raised grand piano lid.

Mfr: Crown International
Price: $249.00

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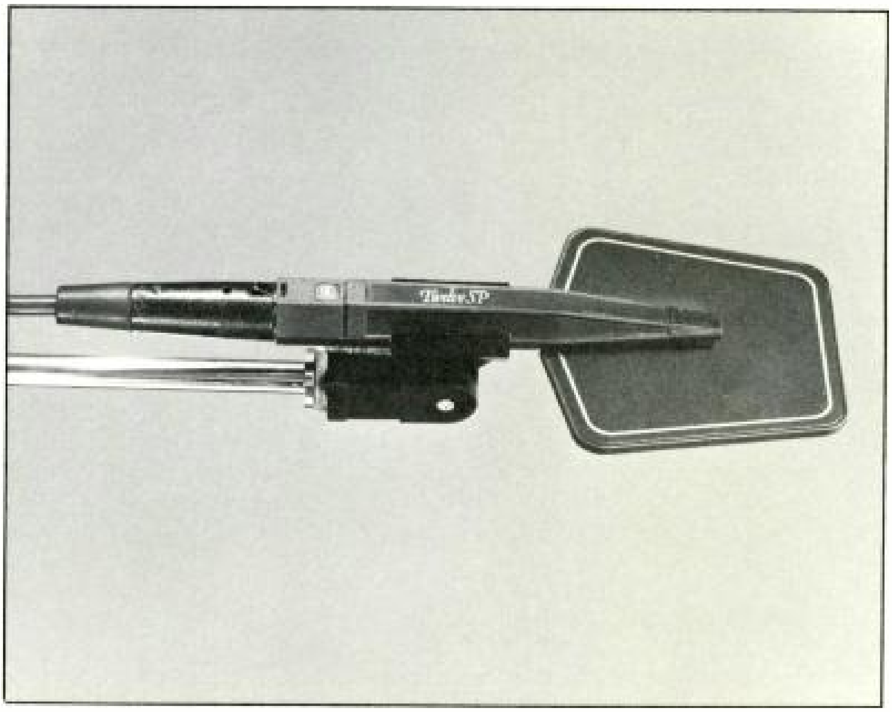
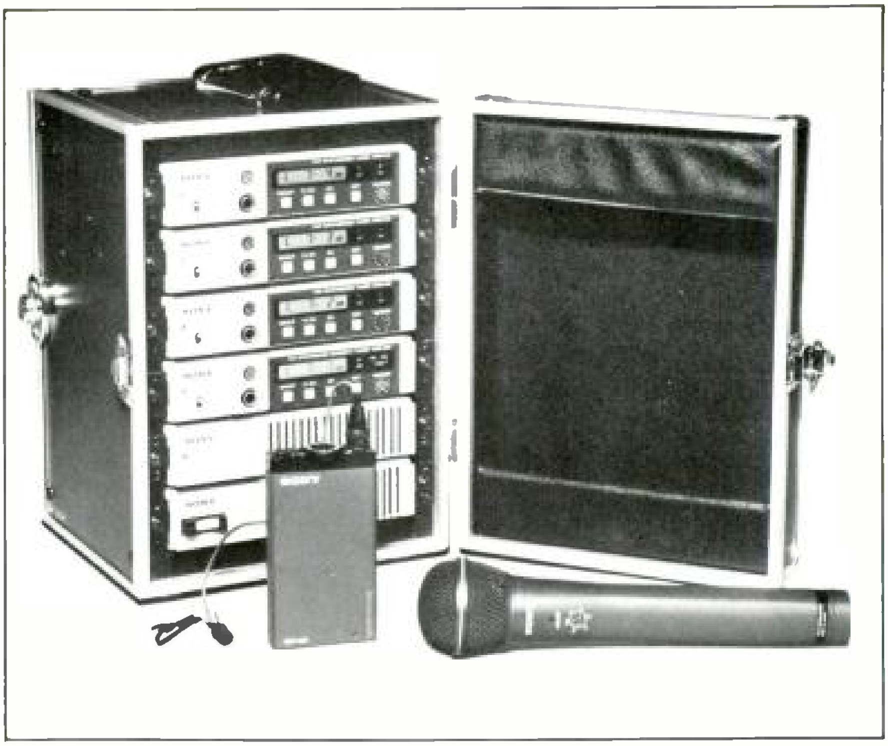


VHF WIRELESS MICROPHONE SYSTEM

• Sony's new VHF wireless microphone system has frequency synthesis in both transmitters and receivers. The exclusive Sony design brings flexibility of operation while providing smooth frequency response and wide dynamic range. Miniaturization of frequency synthesis components is the key factor in the new VHF design. Each microphone transmitter in the system is capable of operating on any of 48 separate frequencies; the system's receivers are equipped to handle all 168 channels in the assigned band. Professional VHF wireless microphone systems operate in the 174-to-216 MHz range, the same transmission band as television stations 7 through 13. Each TV channel contains 24 wireless operating frequencies. The major components of the new system are the WRT-210 hand-held microphone transmitter, the WRT-220 bodypack transmitter (with supplied lavalier microphone), and the WRR-210 and WRR-220 receivers. The frequency range of each transmitter covers two adjacent television channels or 48 individual transmitting frequencies. Thus only four transmitters are needed to cover the entire range between 174 and 216 MHz. The WRR-210 and WRR-220 (diversity) frequency-synthesizing receivers each cover the entire range of frequencies between 174 and 216 MHz, a total of 168 available channels. A pushbutton system with LCD readout allows easy tuning to desired frequencies, and re-tuning if interference is encountered. The new VHF system also offers the choice of diversity reception. When wireless microphones must operate in confined spaces, the combination of direct and reflected waves causes phase cancellation and loss of signal. By utilizing two tuning sections on the same frequency and a "smart" switcher that samples both inputs, the WRR-220 receiver is able to furnish uninterrupted transmission. The Sony system employs true space diversity with two tuners each fed by its own antenna. The diversity circuit automatically selects the stronger of the two incoming signals or mixes them to maintain stable reception. Where more than one wireless microphone system operates simultaneously, the possibility of interference from other channels inevitably arises. The Sony VHF system is designed to minimize these problems in operation. Computer analysis has been utilized to select the most compatible groups of transmission frequencies. Each receiver is programmed with this information and will recall, via the microprocessor, the compatible frequencies once a group has been selected. This method provides six simultaneous channels of clear operation within each television channel. The basic components of the Sony VHF system are complemented by a complete array of antennas and accessories. In addition to the supplied antenna, Sony offers the option of an AN-210 doublet antenna and the AN-220 shoulder antenna. The doublet antenna folds for easy transport, and the attaching clamp is designed to provide maximum flexibility in antenna placement, whether temporary or permanent. The shoulder antenna is designed for use in combination with the portable WRR-210 synthesizing tuner in typical ENG and EFP applications. The WD-210 antenna divider/booster is also available for long range operation of several systems through a single antenna. The system offers the choice of DC power or AC operation with the ACP-210 power supply. The SC-210 carrying case completes the list of accessories.



Mfr: Sony Professional Audio Products, Inc.
Price: Transmitters, $995.00
Receivers, $1,195 and $1,995.

Circle 25 on Reader Service Card

DUAL DIVERSITY WIRELESS MICROPHONE SYSTEM

• Telex Communications' wireless microphone has an exclusive system that utilizes two antennas and a "smart" Dual Diversity circuit in the FMR-2 receiver. This receiver analyzes the signal-to-noise ratios as well as the phase relationship of both signals received. The circuit then compares the two signals, corrects for any major differences in the phase angles, and combines them for maximum signal strength. Each new system includes the FMR-2 Dual Diversity receiver, two ⅝ wave antennas, a microphone, and a transmitter. One system includes the WHM500, a new hand-held condenser microphone with self-contained transmitter. This microphone model has two switches, one of which acts as a "stand-by" to control the RF. The WHM410, a hand-held dynamic microphone/transmitter without switches, is available as part of another system. The WT-200 is a small, beltpack transmitter containing the same switching options as the WHM500. This transmitter has phantom power, making it compatible with most popular lavalier mics. It can be coupled with the miniature Telex WLM-200 lavalier to become part of another Dual Diversity Wireless system. The new FMR-2 receiver has a peak-reading volume indicator that displays audio level on a segmented LED color bar. There is also a display to indicate field strength with a dot that moves back and forth across a 10-segment LED scale.

Mfr: Telex Communications

Circle 26 on Reader Service Card

TRANSFORMERLESS CONDENSER MICROPHONE

• Neumann's TLM 170 i condenser microphone is the first transformerless microphone of the fet 80 series. The direct, balanced signal output was achieved through the use of a new kind of electronic circuit, while maintaining a high degree of interference freedom and low current consumption. It has been possible to reduce the self-noise level of the microphone significantly compared to similar types. The TLM 170 is able to handle sound pressure levels up to 140 dB with minimal distortion. This represents a dynamic range of 126 dB. Five directional characteristics may be selected: omni, wide cardioid, cardioid, hypercardioid, and figure-8. A future option will provide remote controllability of the directional characteristic. Excessive output levels caused by high sound pressure levels may be reduced by a 10 dB attenuation slide switch, while another switch rolls off frequencies below 100 Hz to eliminate low-frequency interference. This microphone may be operated from the usual 48-volt Phantom Powering circuits but can be operated from 24-volt Phantom Powering circuits as well. The TLM 170 i is equipped with a tiltable, elastically-suspended mounting bracket, which isolates the microphone effectively against mechanical noise interference.

Mfr: Neumann

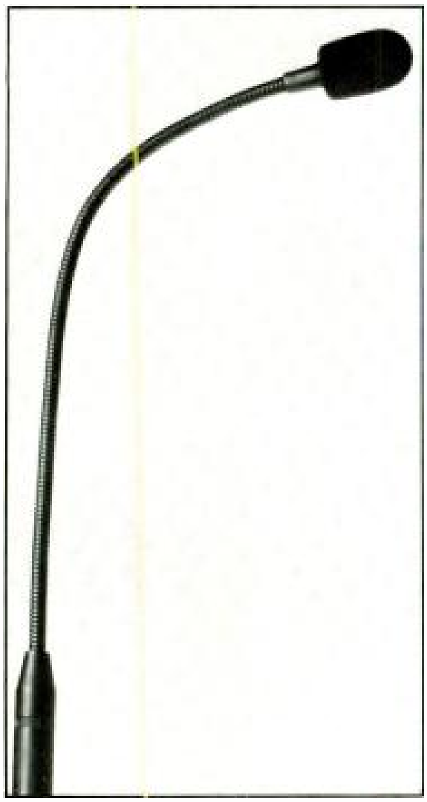
Circle 27 on Reader Service Card

LAVALIER CONDENSER MIC

• Shure Brothers' new SM83-CN omnidirectional, lavalier condenser microphone has been specially designed to provide quality sound reproduction in professional broadcasting and sound reinforcement applications. One problem addressed in the design of the SM83-CN design is the "chest resonance," phenomenon ofen encountered in using lavalier microphones. The SM83-CN's wide frequency response has been specially tailored to compensate for this problem with an electronically-created dip at 730 Hz and an acoustically-generated high frequency boost above 3 kHz. The result is natural sound without boominess or excessive brightness. In addition, the SM83-CN's controlled low frequency rolloff substantially reduces clothing, handling, and room noise. These sonic characteristics are made possible through the use of a specially developed amplifier supplied with the unit. This compact, lightweight amplifier measures $3\frac{3}{4}$-in. by $1\frac{5}{16}$-in. by $\frac{29}{32}$-in. and can easily clip onto the user's belt or fit into a coat pocket. It may be powered by a standard 9-volt battery or by simplex power from an external source (such as Shure Models PS1 and PS1E2), or any microphone power supply providing 5 to 52 VDC simplex voltage. The Shure SM83-CN is supplied with a unique system of hardware that permits a wide variety of unobtrusive mounting techniques. Four mounting clips are provided: a single tie-mount bar (four mounting two microphones simultaneously), and two multi-purpose mounting clips that may be connected to a lanyard, or sewn, pinned, or taped onto clothing. The microphone cable may be completely hidden behind neckties, sweaters, or other clothing. Other features include low noise, minimal RF and magnetic hum susceptibility, a field-replaceable cable that utilizes steel conductors for strength, a dark, non-reflective finish, a foam windscreen for outdoor use, and rugged construction for durability and reliability.

Mfr: Shure

Circle 28 on Reader Service Card



MINIATURE ELECTRET CARDIOID MIC

• Astatic's 827 is a miniature electret cardioid mic permanently attached to a low-profile 13- or 17-inch gooseneck. It is perfect for talk-back, paging, and lectern applications. The mic is phantom-powered and a windscreen is included.

Mfr: The Astatic Corporation
Price: $68.50 suggested retail

Circle 29 on Reader Service Card

HAND-HELD MICROPHONE SYSTEM

• Samson Music Products' new Concert Series microphone system offers the advantage of high-band FM technology at an affordable price. Unlike the typical low-band system in the $475 list price range, the Concert Series broadcasts on the 168-to-199-MHz-band instead of the usual 49 MHz. The use of the higher frequency band allows the Samson Concert Series to offer a more powerful transmitter (50 mW vs. the 10 nW maximum allowed on 49 MHz by the FCC), more usable frequencies (nine channels can be used simultaneously), full bandwidth, frequency response to 18 kHz, longer reception distances of 300 feet under adverse conditions and up to a 700-foot line of sight. A further advantage of the higher broadcast frequency band is freedom from noise and interference from sources such as garage door openers, cordless telephones, and toys, all of which broadcast on the 49 MHz band. The new Samson Concert Series HH 100 system includes two units. The HT-20/ATM microphone/transmitter has an ABS plastic case and an internal antenna (no more dangling wires). The RH-1 receiver utilizes cavity construction for optimum shielding and low noise. The standard mic capsule is an Audio-Technica. Optional capsules available are the Shure SM58 (dynamic) or SM85 (condenser) capsules at additional cost. Standard 9-volt alkaline batteries give six to eight hours of continuous performance. The Concert Series HR-1 receiver uses a folding antenna. LED arrays display RF and AF signal strength, with additional LED indicators for Power On and TX (transmitter) On. A mute control on the back panel prevents the units from broadcasting even when the mic is off.

Mfr: Samson Music Products

Circle 30 on Reader Service Card

FLEXIBLE GOOSENECK MICS

• Countryman Associates' new Isomax IV G Series of flexible gooseneck podium microphones may also be hand-held for interviews. It incorporates active vibration isolation for low handling noise, eliminating the need for shock-mounting. The mic is hypercardioid (supercardioid) and has a frequency response of 70 Hz to 18 kHz, optimized for voice pickup. The mic also features a built-in high-pass filter that cuts wind noise and rumble. The inconspicuous matte black chrome extension tube may be bent and rebent many times.

Mfr: Countryman Associates, Inc.
Price: $310.00 (pro net)

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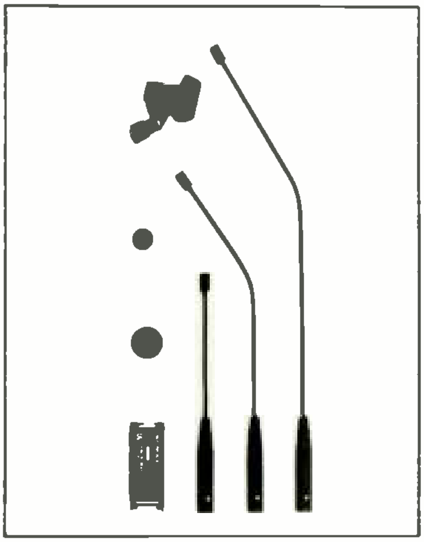

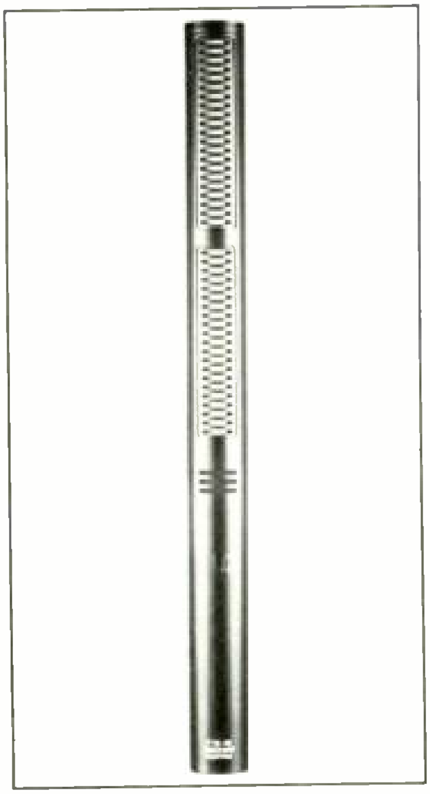

SHORT SHOTGUN MIC

• AKG's C-568EB is a short shotgun microphone (10 inches long) of exceptional directivity. It features full-range response and incorporates a low-frequency roll-off switch to eliminate environmental rumble. Applications for the C-568 range from extremely tight drum-kit mic'ing to on-stage theatre, long-reach choir, and ENG pickup. The C-568 requires phantom powering in the 9 to 52-volt range, or any AKG accessory AC or battery power supply.
Mfr: AKG

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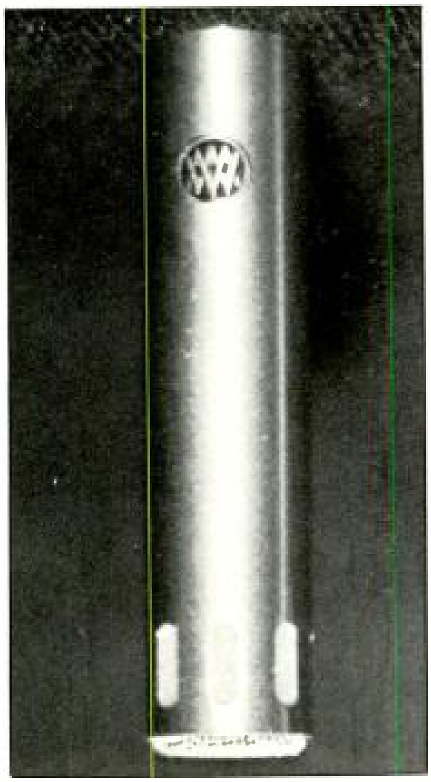

NEW CONDENSER MICROPHONE

• Wright Microphones' new Model SR-1 condenser microphone has an extended-angle cardioid polar pattern, and operates on 12 to 48-volt phantom power. Its frequency response is 20 Hz to 30 kHz, while its maximum sound pressure level is 130 dB. The SR-1 uses a high-compliance shock mount and is guaranteed for one year against manufacturing defects.
Mfr: Wright Microphones

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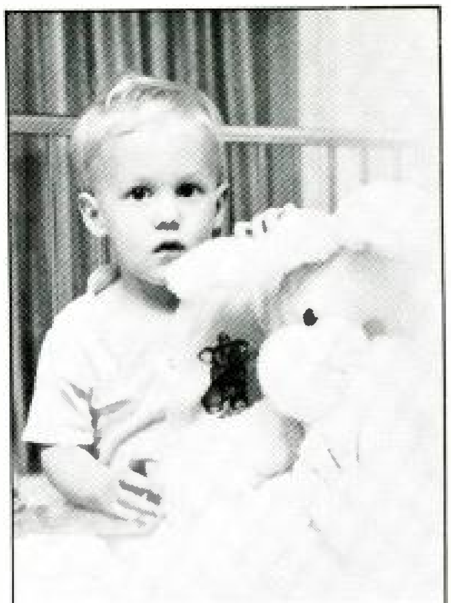
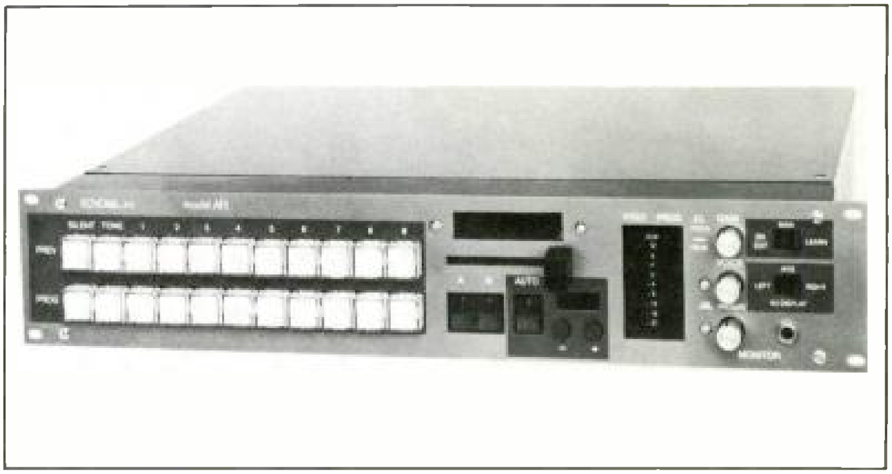


COMPUTER-CONTROLLED AUDIO FOLLOWER SWITCHER



• ECHOlab's new computer-controlled switcher for audio mixing in a video environment, called the Model AFS, was originally designed for use with the ECHOlab SE/3 switcher. However, it can also be controlled by other video switchers or edit controllers and can operate as a stand-alone programmable mixer. Parallel or serial (SMPTE) links are available. When used with a video switcher, the AFS automatically switches the audio input to match the video source. It can select from eleven input sources, including a low-distortion 450 Hz tone and silent. Voice-over can be added to the output, and a stereo option is available. Mixing can be done either manually, with a slider control, or automatically, with pre-selected frame rate, via an Autotake button. Up to two user-created mixing profiles up to 15 seconds in length can be programmed and stored. These can be recalled and executed with a single pushbutton. The AFS employs a new Equilinear circuit that uses LED/dual-photoconductor attenuators said to provide significantly improved distortion (less than 0.02 percent) and noise (greater than 100 dB S/N) performance. Input circuits are transformerless, balanced, with high looping impedance for noise-free operation. Built-in diagnostic software performs self-test and calibration functions on power-up and at intervals without operator intervention. Displays include two 20-segment LED VU meters, one 20-segment Take display, and several red/green LEDs. Autotake rate is shown in a three-digit, seven-segment display.
Mfr: ECHOlab, Inc.

Circle 34 on Reader Service Card

DIFFERENTIAL PREAMPLIFIERS

• Precision Filters' Series 600 high input impedance, differential preamplifiers are said to feature excellent low noise and common mode rejection performance over a frequency response range of 0.5 Hz to 500 kHz. These preamplifiers are available in single-ended and differential versions. Standard fixed gains of 0, 10, 20, 30, and 40 dB, or switched gains are available; gain deviation with power supply and temperature changes is typically .002 dB per-volt or per-degrees centigrade respectively. The Series 600 provides full output of ±10 volts peak to 100 kHz. Full output response is maintained at up to ±15 mA load current, with total freedom from peaking effects with complex loads. The differential version can operate from ±9 VDC to ±18 VDC power supplies. Single supply versions, positive and negative, are available for supply range of 18 VDC to 36 VDC. Power supply noise rejection is excellent, greater than 80 dB over the entire frequency response range.
Mfr: Precision Filters, Inc.

Circle 35 on Reader Service Card

NEW MONITOR AMPLIFIERS

• ProTech Audio Corporation's new audio power amplifiers, designated the 610 (10 watts) and the 625 (25 watts), have been added to their line of professional audio equipment. The new units are designed to provide high-quality audio with increased reliability, and 24-volt backup compatibility. The new units have 3 × 5¼-inch mounting panels and offer the following features: On/Off switch (circuit breaker), self-contained power supply, conductive plastic control pot, binding post termination of backup battery supply voltage, screw terminal connections for Audio In and Out, and discrete voltage gain stages for increased circuit longevity. Both the 610 and 625 Audio Power Amplifiers are designed to be used in applications requiring high reliability of program audio, even during AC power failures.

Mfr: ProTech Audio

Circle 36 on Reader Service Card

THE BOSS

• The Boss™, by Alpha Automation, offers a new, comprehensive approach to the sound recording, editing, assembly, and audio-for-video post production process. It combines the previously separate activities of sound synchronization, console automation, and other control room systems and provides central control of, and communications between, virtually all studio equipment. The Boss is the nucleus of a complete system. This system will interface to mixing console automation, programmable equalization and digital reverberation systems, Musical Instrument Digital Interface (MIDI), track-select, the integration of production paperwork, and a local Area Network (LAN) to tie together multiple control room installations and the front office. Integral Edit Decision List (EDL) software calculates up to 999 user-defined events and executes the preset sequence, allowing you to work out long routines before the session. You can trim values, insert or delete entries, and automatically readjust entry points. Programmable Cue Menu stores up to nine location points where a user-defined series of events will occur when the master plays past each cue point. Up to four functions may be assigned to individual slaves at each location. These locations may be assigned and manipulated from the Scratch Pad Register. SFX Cue automatically runs through a series of user-defined steps for sound effects insertion. Cycle (loop) repeats a user-defined sequence. The system also has a number of Synchronizer Control features, as well as many other features. The Boss 8400 basic system includes the Master Processing Unit (rack-mountable, 16/8-bit microcomputer with two 5¼-inch floppy disk drives), 13-inch RGB/Composite Monitor, and a 95-key typewriter-style remote keyboard. Also included are two software packages—Automated Production Tools and the Synchronizer Control System.

Mfr: Alpha Automation
Price: $16,900

Circle 37 on Reader Service Card

GAUSS 200 WATT COAX

• Cetec Gauss' first coaxial model loudspeaker, the Gauss 3588, is said to be the first compupter-designed coaxial. The new model features a conservative power rating of 200 watts RMS, and has been tested to continuous program levels of 750 watts delivering clean sound. Metric sensitivity is 95 dB for the low frequency and 109 dB for the high frequency. Because of proprietary design parameters, both drivers are virtually in the same acoustic plane, well within the Blauert and Laws criteria, thus eliminating the need for time compensation networks. The new coaxial is designed to work with any standard professional quality crossover. A new Computer-Aided Time-Spectrometry program was developed to design the unique cosh horn for the coaxial. This design provides an extremely stable image, reduced second-harmonic distortion, and virtually no mid-range shadowing.
Mfr: Cetec Gauss

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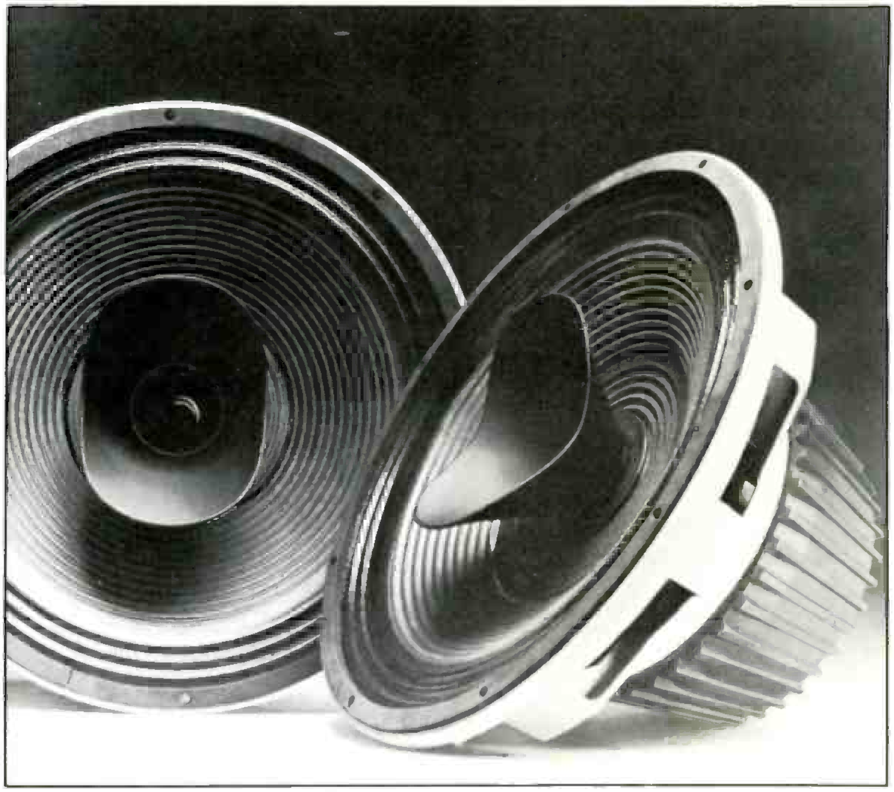

• **LaSalle Music** of Hartford and Boston has announced the establishment of its professional audio division in Boston. Joining the division are three respected veterans of Boston-area professional audio sales and marketing: **Les Arnold**, who is general manager; **Mark Cuddy**, who handles sales to recording studios and production houses; and **Mark Parsons**, who oversees sales to broadcasters, corporations and institutions. LaSalle Professional Audio offers system sales, design, installation, service and financing. Over 250 product lines are available.

• **Robert A. Moog** has been appointed chief engineer of **Kurzweil** Music Systems, Inc., company officials have announced. Moog is well-known as a pioneer of electronic music synthesis. The Moog synthesizer, developed in the late '60s, was the first such instrument to achieve wide commercial acceptance. His company, Moog Music, was acquired by Norlin Industries in 1973. In 1978, Bob Moog founded Big Briar, Inc. to design and build custom electronic musical instruments. He will continue to serve as president of Big Briar. Kurzweil Music Systems manufactures and markets the Kurzweil 250, a digital keyboard capable of accurately recreating the musical sounds of acoustic instruments, including the grand piano, through the use of artificial intelligence and other advanced computer technology.

• **Aphex Systems, Ltd.**, a manufacturer of signal processing equipment, has announced the appointment of three new rep firms in the United States: **Peter M. Schmidtt & Co.**, for metro New York; **Secom**, for Tennessee, Georgia, Mississippi and Alabama; and **Mike Chafee Enterprises**, for Florida. In addition, **Atlantex Music Limited** has been appointed distributor for all Aphex products in the United Kingdom; **AKG Acoustics (Canada)** will distribute the line in Canada: and **Audio Systems PAS** will cover Switzerland.

• **Shure Brothers Inc.**, of Evanston, Illinois, has announced the appointment of **Richard T. Murphy** to the position of vice president, marketing and sales. In his new position, Murphy will oversee all of Shure's distributor and OEM sales and marketing activity. Murphy comes to Shure from the Electra Division of Masco Corporation, where he was responsible for the marketing of electronic scanners and wireless telephone products. Prior to that, he held executive positions with several advertising agencies and worked in field sales for General Electric Company.

• **Edwin E. Pessara, Jr.** has been appointed director of business management for Ampex Corporation's Magnetic Tape Division, Donald F. Bogue, division general manager has announced. According to Bogue, Pessara will have overall responsibility for product line strategy and marketing in Ampex's tape business worldwide. Most recently director of marketing and sales for the industrial products division of TDK Corporation of America. Pessara is a 16-year veteran in the audio-video industry. He began his career with Ampex's former Educational Industrial Products Division in Chicago in 1968 and subsequently held various national marketing and sales positions with JVC of America.

• **Michael K. Solomon** has joined **Electro-Voice** as marketing development manager/music performance products, a new position in EV's music products division. As marketing product manager of Jensen's International Division, Solomon was responsible for overseas marketing for Advent, Phase Linear, Jensen Component Audio/Video Systems and Jensen Car Audio Products. Prior to that, he was marketing product manager for Microphone and Accessory Products for Shure Brothers in Evanston, Illinois.

• Owners of the OB-8 synthesizer are currently participating in a worldwide "Patch Hunt" sponsored by **Oberheim Electronics, Inc.** The purpose of the "Patch Hunt" is to encourage creative programming of new sounds on the instrument, and to foster increased communication between the synthesist and the manufacturer. The submitted patches will be judged by a select group of Los Angeles-based musicians and synthesists. The best patches will then be compiled by the Oberheim staff and made available on data cassette to all participants and other interested OB-8 owners. Creators of the selected patches will receive credit for their efforts in a Patchbook which will describe the front panel settings used in creation of the sounds. Due to numerous requests, the Oberheim International OB-8 Patch Hunt entry deadline has been extended until October 31, 1984. Interested parties may contact Oberheim or its authorized dealers for entry forms and guidelines.

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