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MARCH 1982 VOLUME 16, NO. 3

DEPARTMENTS

FEATURES

About The Cover

• This month's cover features (from left to right) the Tonschreiber B ("Berta"), made in 1939; the Magnetephone FT-2, made in 1936, and the Telegraphone, circa 1910. Photos courtesy of the Ampex Museum of Recording.

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db Calendar

APRIL

24 **Midwest Acoustic Conference.** Hermann Hall, Illinois Institute of Technology, Chicago, IL. For more information contact: Hugh Pearl, Shure Bros., Inc., 1501 W. Shure Drive, Arlington Heights, IL. Tel: (312) 259-7700, ext. 313.

29-30 May 1 **Electronic Distribution Show and Conference.** New Orleans Hilton, LA. For more information contact: David L. Fisher, Executive Vice President Electronic Industry Show Corporation, 222 South Riverside Plaza, Suite 1606, Chicago, IL 60606. Tel: (312) 648-1140.

JUNE

10-13, 24-27 **National Video Festival.** Sponsored by American Film Institute and Sony. June 10-13, Kennedy Center, Washington, D.C., June 24-27, AFI Campus, Hollywood, CA. For more information contact: Television and Video Services program of the American Film Institute, Kennedy Center, Washington, D.C. 20566.

JULY

21-23 **The Second Annual Annual Broadcast Engineering Conference.** The Fawcett Center for Tomorrow, Ohio State University, Columbus, Ohio. For more information on papers and other details contact: John H. Battison, WOSU-AM-FM-TV, 2400 Olentangy River Road, Columbus, OH 43210.

Index of Advertisers

Coming Next Month

• In April, our topic is Broadcast Audio. UREI's Gary Margolis takes a close look at broadcast vs. recording applications; our own John Woram heads south to give us a report on radio station HCJB in Ecuador, and we'll also feature a Broadcast Audio survey. In addition, we'll have our usual columns, departments, and more—all in **db—The Sound Engineering Magazine.**

McCoy Tyner's string section on Bose.

When John Blake and Avery Sharpe aren't performing with McCoy Tyner, one of the world's foremost jazz pianists, they conduct seminars and master classes on theory and technique at major colleges and music schools nationwide. Here's the advice John and Avery give on amplifying acoustic instruments:

John: "My violin has a naturally sweet and mellow tone, and it can also be very dark. Ordinary speakers don't do this instrument justice—they add an unpleasant edginess, a kind of glare that makes it sound like an electrified instrument.

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John Blake, c/o Joanne Rile Management, P.O. Box 27539, Philadelphia, PA 19118.

db Letters

To The Editor:

I have back issues of db for the years 1973 through 1980 which I must dispose of. Would you have need for these or know of any organization, library or other that can use these copies? I am reluctant to put them in the rubbish! Please advise.

Henry L. Messerschmidt

db replies:

How about it readers? Anyone out there have any suggestions? (Please keep the sarcastic ones to yourself.) If you have a legitimate *idea, please contact us here at* db *and we will forward your suggestion on to Mr. Messerschmidt.*

db New Literature

20th ANNIVERSARY DIRECTORY

• The Who, What, Where, When and Why of acoustical consulting is reflected in the new 1981-1982 Directory of the National Council of Acoustical Consultants (NCAC). Published bi-annually, the directory gives practical advice on how to select an acoustical consultant and explains the broad categories of acoustics which members practice. Mfr: **National Council of Acoustical Consultants, 66 Morris Ave., P.O. Box 359, Springfield, NJ 07081.**

RECORDING INDUSTRY LISTING

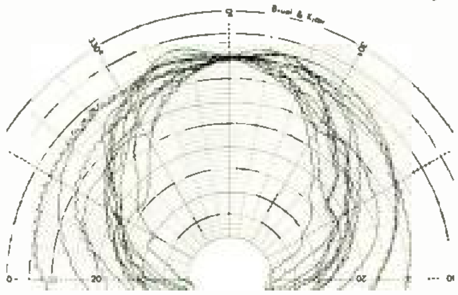
• The JVC Cutting Center has prepared a comprehensive listing of the Southern California recording industry. The "menu" categorizes studios, disk mastering facilities, digital rentals, major record labels, and studio support services such as equipment sales and rentals. The two-page directory comes complete with addresses and phone numbers. Mfr: **JVC Cutting Center, RCA Bldg., Suite 500, 6363 Sunset Blvd., Hollywood, CA 90028.**

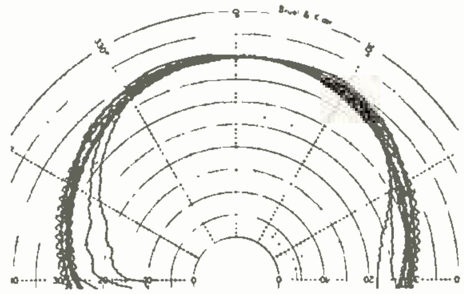
MICROPHONE CATALOG

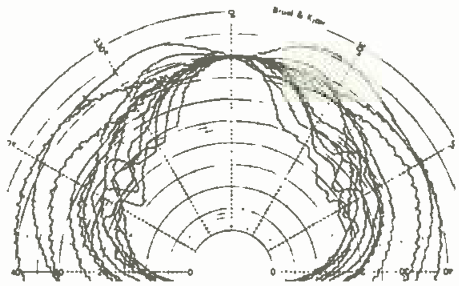
• The complete Audio-Technica 800-series line of microphones and related accessories is presented in a new four-color, 24-page catalog. Pictured and described are A-T studio electrets, studio dynamics, and remote powered special-purpose electrets. A broad array of mic cables, windscreens, line matching transformers and power supplies are also shown. The catalog features several pages of user information to assist the microphone buyer in making a choice of microphones for a specific application. Mfr: **Audio-Technica U.S., Inc., 1221 Commerce Drive, Stow, OH 44224.**

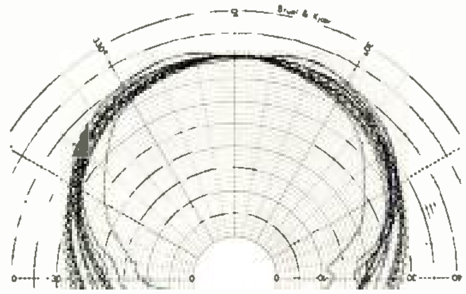
Before you invest in new studio monitors, consider all the angles.

Polar response comparison of a typical two-way coaxial studio monitor and JBL's new 4430 Bi-Radial studio monitor from 1 kHz to 16 kHz.


Typical horizontal


JBL 4430 horizontal


Typical vertical


JBL 4430 vertical

No one has to tell you how important flat frequency response is in a studio monitor. But if you judge a monitor's performance by its on-axis response curve, you're only getting part of the story.

Most conventional monitors tend to narrow their dispersion as frequency increases. So while their on-axis response may be flat, their off-axis response can roll off dramatically, literally locking you into the on-axis "sweet spot." Even worse, drastic changes in the horn's directivity contribute significantly to horn colorations.

Introducing the JBL Bi-Radial Studio Monitors.

At JBL, we've been investigating the relationship between on and off axis frequency response for several years. The result is a new generation of studio monitors that provide flat response over an exceptionally wide range of horizontal and vertical angles. The sweet spot and its traditional restrictions are essentially eliminated.

The key to this improved performance lies in the unique geometry of the monitors' Bi-Radial horn.[1] Developed with the aid of the latest computer design and analysis techniques, the horn provides constant coverage from its crossover point of 1000 Hz to beyond 16 kHz. The Bi-Radial compound flare configuration maintains precise control of the horn's wide 100° x 100° coverage angle.

1. Patent applied for.

And the Bi-Radial horn's performance advantages aren't limited to just beamwidth control. The horn's rapid flare rate, for instance, dramatically reduces second harmonic distortion and its shallow depth allows for optimal acoustic alignment of the drivers. This alignment lets the monitors fall well below the Blauert and Laws criteria for minimum audible time delay discrepancies.

But while the Bi-Radial horn offers outstanding performance, it's only part of the total package. The new monitors also incorporate JBL's most advanced high and low frequency transducers and dividing networks. Working together, these components provide exceptionally smooth response, high power capacity, extended bandwidth, and extremely low distortion.

Judge For Yourself

Of course, the only way to really judge a studio monitor is to listen for yourself. So before you invest in new monitors, ask your local JBL professional products dealer for a Bi-Radial monitor demonstration. And consider all the angles.

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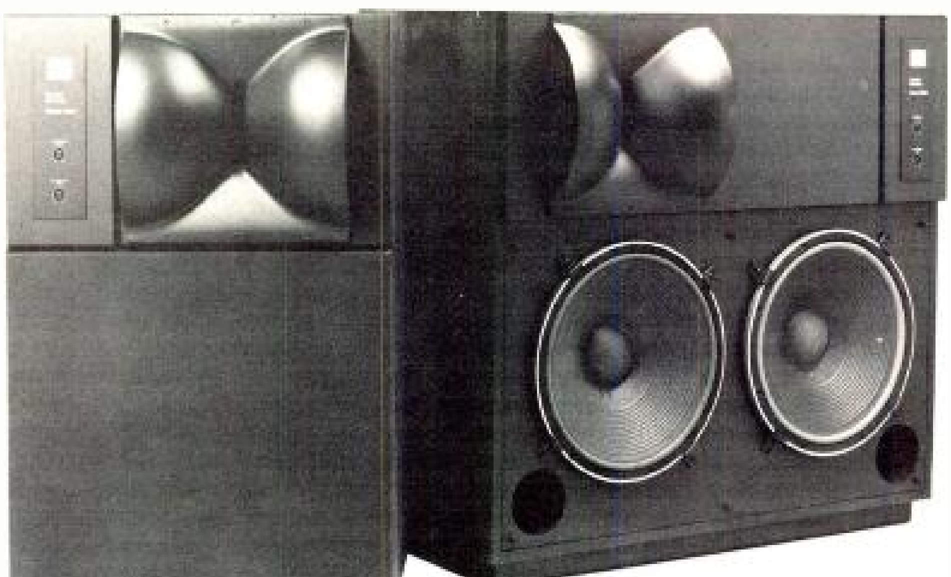

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Compression Drivers

• Virtually all large sound reinforcement systems use horns for high-frequency elements. In part, the horn acts as an acoustic transformer, converting high-pressure sound at low volume-velocity at the throat into low-pressure sound at high volume-velocity at the horn mouth.

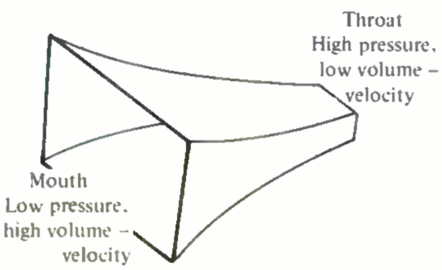

Figure 1. Pressure and volume-velocity relationships in a horn.

This relationship is shown in FIGURE 1. (The term volume-velocity is a reference to the motion of air—not sound—and is measured in m^3/sec). The horn must therefore be driven by a transducer capable of producing high pressures at the throat. Such transducers are usually known as compression drivers, and typical models exhibit electro-acoustical conversion efficiencies on the order of 25 to 35 percent, when appropriately coupled.

FIGURE 2 shows details of a "Western Electric" type compression driver in a cutaway view. In this model, the aluminum diaphragm is 10 cm (4-in.) in diameter, and the voice coil, made of edge-wound aluminum-ribbon wire, is placed in a strong magnetic field. The diaphragm fits within 1 mm (40 mils) of the *phasing plug*. The phasing plug has a set of annular slits, whose area at the diaphragm surface is about *one-tenth* the area of the diaphragm itself. The ratio of these areas, known as the *loading factor* of the driver, affects the efficiency of the device. The slits are tapered and expand away from the diaphragm, towards the 5 cm (2-in.) exit of the driver.

RADIATION RESISTANCE

FIGURE 3A shows an equivalent circuit which describes the mid-band sensitivity of the driver. R_E represents the resistance of the voice coil. The radiation resistance, R_{ET}, of the driver coupled to a horn is found from the equation

$$R_{ET} = S_T(Bl)^2/\rho_0 c S_D^2,$$

where S_T = the area of the annular slits,
B = the magnetic flux density,
l = the conductor length,
$\rho_0 c$ = the acoustical impedance of air, and
S_D = the diaphragm area.

MAXIMUM POWER TRANSFER

In the following efficiency equation, power transfer is maximized (at 50 percent) when $R_E = R_{ET}$:

$$\text{Efficiency} = R_E R_{ET}/(R_E + R_{ET})^2 \times 100.$$

Under this condition, half the input power will be dissipated in the voice coil resistance, and half will be radiated as useful sound output.

The frequency response is uniform up to the frequency at which the mass of the moving system becomes dominant. The equivalent circuit shown in FIGURE 3B describes this. C_{MES} is dependent on the moving mass, and it produces a high-frequency roll-off of 6 dB-per-octave, commencing at a frequency, f_{HM}, given by:

$$f_{HM} = (Bl)^2/\pi R_E M_{MS},$$

where M_{MS} = the mass of the moving system.

In most compression drivers, f_{HM} is in the region of 3 to 3.5 kHz. Two other elements in the equivalent circuit of FIGURE 3B may affect high-frequency response. L_E is the inductance of the voice coil; it can be effectively nulled by plating a silver or copper ring on the pole piece. This "shorted turn" in the vicinity of the voice coil acts like a low resistance on the secondary side of a transformer, of which the voice coil is the primary. The low value of resistance is reflected through, and swamps out the reactance of the inductance. The L_{CEC} element depends on

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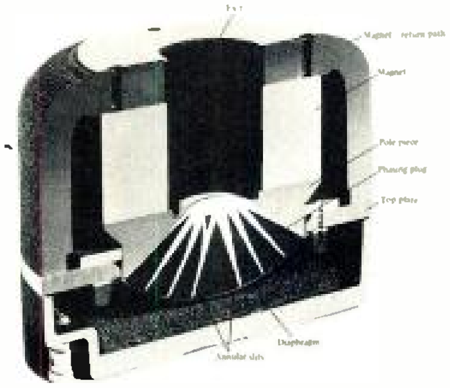

Figure 2. Details of a compression driver (JBL photo).

the volume of the air cavity between the diaphragm and the phasing plug. Keeping the diaphragm-phasing plug spacing as small as possible (consistent with excursion requirements at high power input levels near the lower cut-off frequency of the horn) will minimize its effect on frequency response in the audible range.

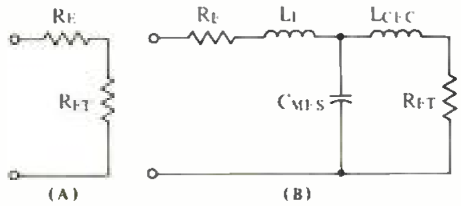

Figure 3. Equivalent circuits for mid-frequency (A) and high-frequency (B) performance of a compression driver.

The high-frequency in the range above 10 kHz can often be enhanced by the action of *secondary resonances* in the diaphragm and its compliance. There are no simple equivalent circuits which describe this. As we will soon see, these secondary resonances can be "nudged" to yield certain benefits with insignificant trade-offs.

MEASUREMENT OF COMPRESSION DRIVERS

In order to minimize the aberrations and directional effects of horns, compression drivers are usually measured on a *plane wave tube* (PWT). FIGURE 4 shows a PWT diagram. The driver is attached to one end, and a probe microphone is located close by. The other end of the tube is filled with a graduated wedge of fiber glass or other absorptive material which attenuates the sound sufficiently so that standing waves, or reflections, in the PWT are minimized. The equation relating the measured sound pressure in the PWT and the acoustical intensity in the PWT is:

$$\text{Sound Pressure} = \sqrt{\text{Power}\ (\rho_0 c)/\text{Area}}.$$

In practical terms, one acoustical watt in a 2.54 cm (1-in.) diameter PWT will produce a pressure of 153 dB-SPL. As stated earlier, the maximum possible efficiency of a compression driver is 50 percent. Therefore, we should never expect to see one-watt driver response on this tube exceed 150 dB-SPL.


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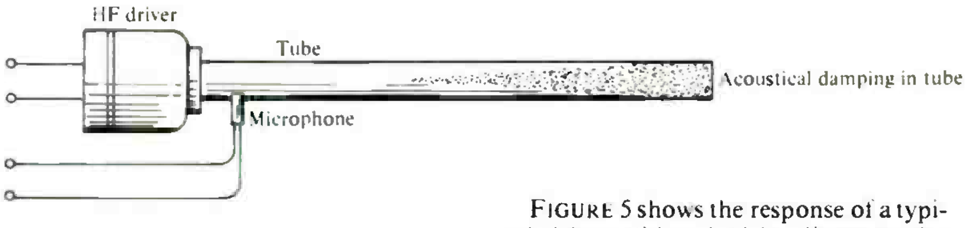

Figure 4. Details of a plane wave tube (PWT).

FIGURE 5 shows the response of a typical driver with a 1.75-in. diameter diaphragm. Here, the input power is 1 milliwatt, 30 dB lower in level than 1 watt. Dashed lines show the transition between the mid-band region and the high-frequency region determined by the moving mass. The region affected by secondary resonances is indicated as well. Note that the mid-band SPL is 117 dB, some 3 dB lower than the possible maximum of 120 (150 − 30) dB, which would indicate 50 percent efficiency. The indicated conversion efficiency then is 25 percent.

POWER RATINGS OF COMPRESSION DRIVERS

At high frequencies, where diaphragm excursion is slight, the power rating of a driver will be determined by its ability to dissipate the heat generated in the voice coil. This is the so-called *thermal power rating* of the device. At frequencies in the 500-800 Hz range, the displacement of the diaphragm will limit the input power. Catastrophic failure of the driver will occur if the diaphragm strikes the phasing plug. In addition, multiple flexures of the diaphragm may result in cumulative metal fatigue in the diaphragm such that after years, or perhaps only months of operation (depending on the input level), a diaphragm will simply fracture, even though specific thermal or displacement limits may not have been exceeded.

Aluminum is most vulnerable to fatigue. Materials such as beryllium, titanium and phenolic-impregnated linen are much less so. Many manufacturers will give several power ratings, dependent upon the lower crossover frequency, and even the slope of the transition.

COMPARING MANUFACTURER'S SENSITIVITY DATA

Most professional compression drivers have mid-band sensitivities within a dB of each other. A general standard in the industry is to present PWT data measured on a 1-inch tube with an input power of 1 milliwatt. A rating of, say, 118 dB SPL will correspond to 148 dB SPL, 1 watt applied to the driver on a 1-inch tube, indicating an efficiency of 31 percent. Remember the 30-dB conversion factor when comparing 1 watt and 1 milliwatt data for constant PWT diameter.

One manufacturer, TAD, provides data measured on a PWT with a diameter of 0.75-in. The reduced cross-section raises the intensity some 2.5 dB, and this factor must be taken into account when making comparisons.

Still other manufacturers present sensitivity data with the driver mounted on a horn. As a general rule, the reference input power is 1 watt, and the reference measuring distance is 1 meter. It is difficult to make an exact conversion from data presented in this form to that normally presented on a PWT, but an approximation may be made as follows: One acoustic watt radiated omnidirectionally in a free field will produce 109.2 dB SPL at 1 meter. The *directivity index* (DI) of a horn is a measure of its ability to focus its radiation along a given direction or axis, as compared to the same sound power radiated omnidirectionally. For example, let us assume that a given driver is rated: 114 dB SPL, 1 watt at 1 meter on a horn with DI = 11 dB. We can work backwards and say that the


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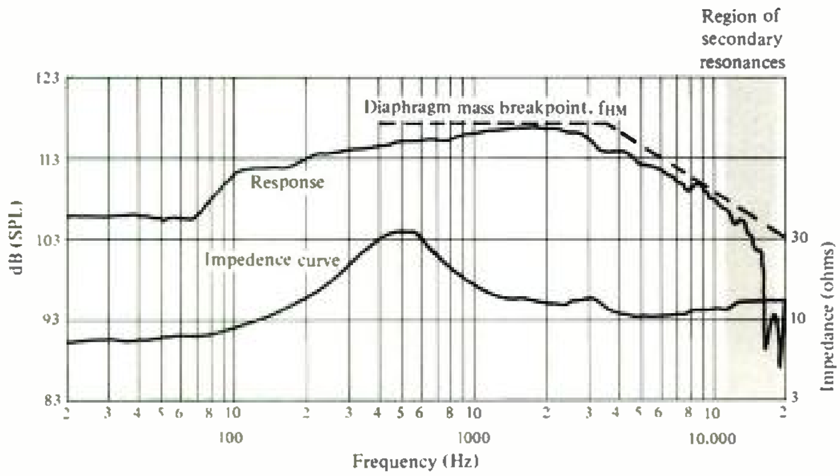

Figure 5. Response of a compression driver (1.75-inch diaphragm diameter), 1 milliwatt input, 2.54 cm plane wave tube.

driver would produce 103 dB (114 − 11), if it were radiating omnidirectionally, at a distance of 1 meter. Since 109.2 dB represents 1 watt at one meter, 103 dB, some 6.2 dB lower in level, corresponds to a power of about 0.25 acoustic watts, or an efficiency of 25 percent. The same driver on a 1-inch PWT, with 1-watt input, will produce about 147 dB SPL, or 117 dB SPL with an input power of 1 milliwatt.

THE ROLE OF SECONDARY RESONANCES

FIGURE 6 shows the response of three drivers mounted on the same horn. All three have a diaphragm diameter of 10 cm but differ in diaphragm material and surround treatment. The JBL 2440 has a half-roll surround which produces a gradual rise in response to about 9.5 kHz, above which point the response drops rapidly. The model 2441, through the use of a different surround treatment, distributes the secondary resonances in such a way that smoother response is exhibited out to about 18 kHz. The TAD model 4001 driver is very much like the JBL 2440 except that its diaphragm is made of beryllium. Because of the greater stiffness of this material, a response peak

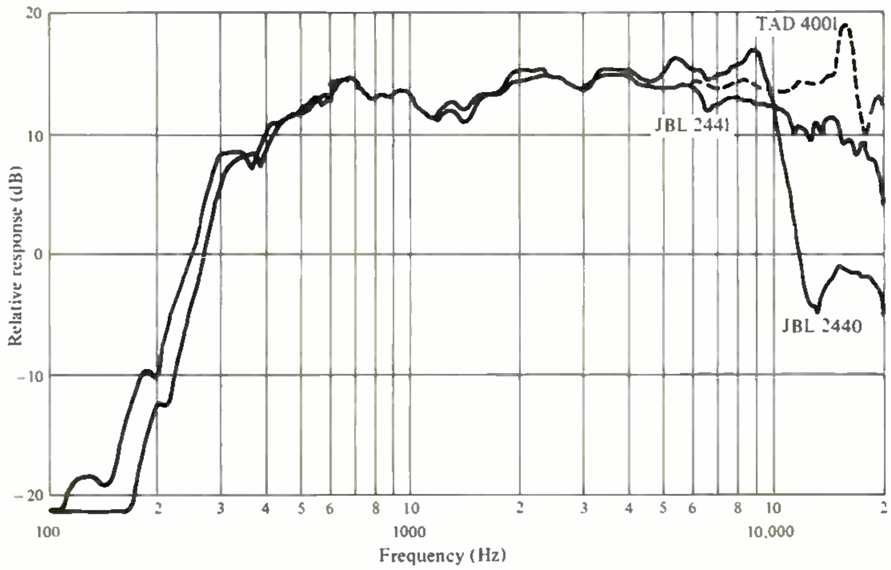

Figure 6. Response of JBL 2440, 2441 and TAD 4001 drivers on JBL 2350 horn (far-field, on-axis measurement).

similar to that of the 2440 is shown; however, it is raised approximately one octave to 17 kHz. Secondary resonances can be used in two ways: to elevate the response out to some cut-off frequency, or to extend the response beyond that cut-off frequency with some reduction in output.

DISTORTION IN COMPRESSION DRIVERS

The moving system in most compression drivers is usually quite linear and is not generally a source of distortion. The distortion measured in a PWT is apt to be fairly high due to the excessive sound pressures in the tube. When mounted on a horn, which is where the distortion really matters, the horn flare rate itself becomes the dominant factor in determining distortion, as a function of frequency and level. In general, a larger diaphragm driver will generate less distortion than a smaller driver, since the initial intensity at the diaphragm-phasing plug interface will be less for the larger driver for a given pressure at the horn mouth. Because of the dominant role played by the horn in establishing distortion levels, we will discuss distortion in horn systems in a later column dealing with various horn types. ■

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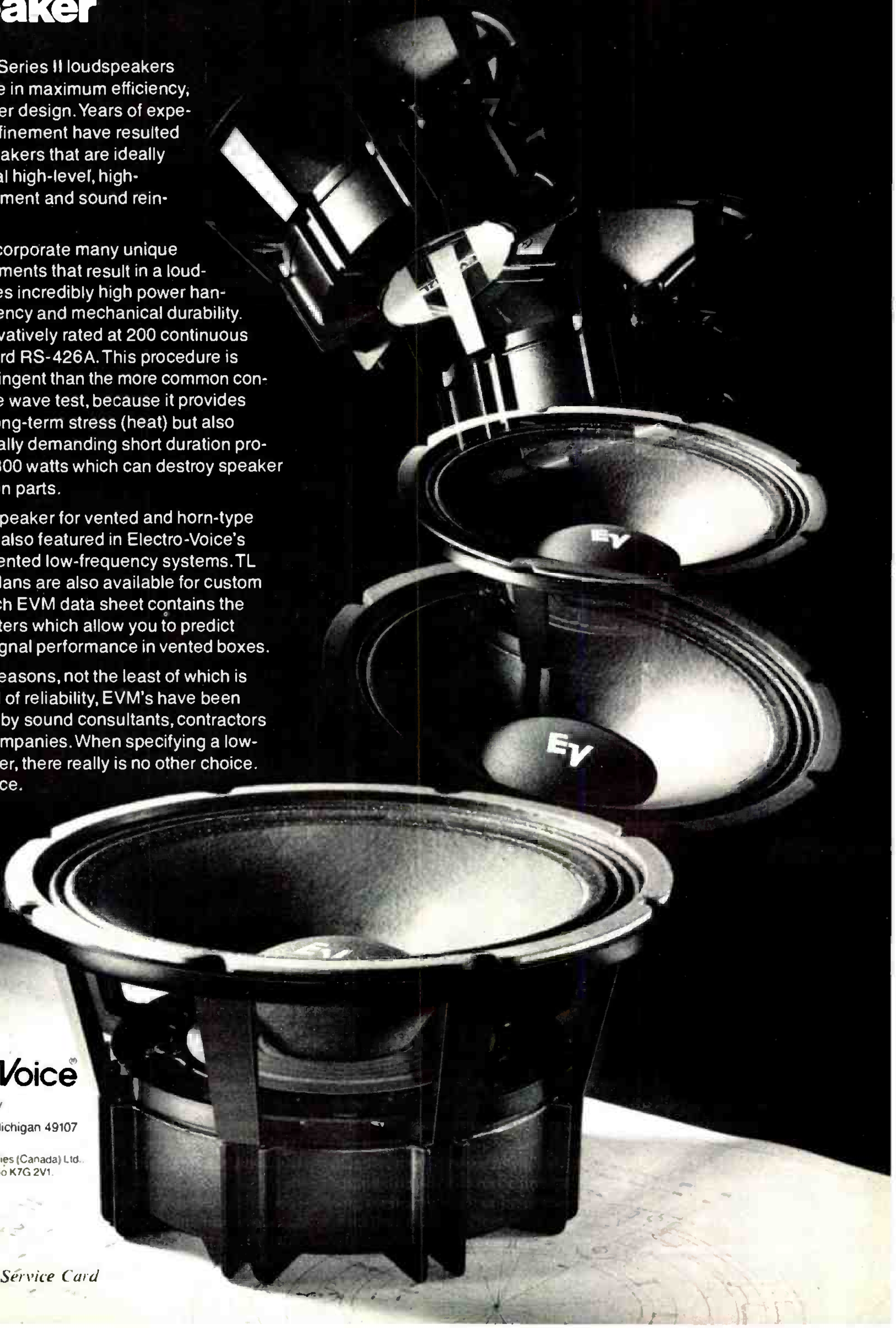

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BARRY BLESSER

db Digital Audio

Flip-Flop Issues

• Recently we talked extensively about the inherent advantages of Schottky and low-power Schottky logic compared to ordinary TTL. The term Schottky refers to a certain type of semiconductor diode, composed of semiconductor material at a junction with metal. A normal diode is made up of a junction of P-type semiconductor material and N-type material (P = excess holes, N = excess electrons). The most interesting properties of the Schottky diode are: its saturation voltage in the forward direction is on the order of 0.3 volts instead of the usual 0.7 for an ordinary diode; it is extremely fast, since there are no stored electrons which need to be removed when the diode is turned off. In other respects (leakage, breakdown, etc.), the Schottky diode is much inferior.

The switching time for a transistor is determined by two active regions: saturated and non-saturated. In the former, the base current drive is sufficiently high to overdrive the transistor, and this creates a large number of excess electrons in the active region. These electrons need to be removed in order to turn the transistor off. Even by temporarily reversing the base current to expedite the removal of these electrons, it still takes a relatively long time because of the intrinsic base resistance. With non-saturated operation, the transistor is still active and there is a much smaller amount of these excess electrons.

We can illustrate some of these points with a few illustrations. Consider the circuit in FIGURE 1 which shows a positive drive current to turn the transistor on, but no way to remove the excess electrons stored in the transistor. These excess electrons will eventually decay by the normal transistor action but it will take a long time, on the order of 100 nsec or more.

With active turn-off, the drive is made to go negative in order to reverse the base current. Thus, electrons are pulled out of the base actively and the device can now turn-off more rapidly. Since no current reversal is possible with TTL, because there are no negative voltages available, an extra transistor must be placed across the emitter-base junction. In FIGURE 2, as long as the main transistor is still on, the emitter-base voltage will still be on the order of 0.7 volts and this voltage

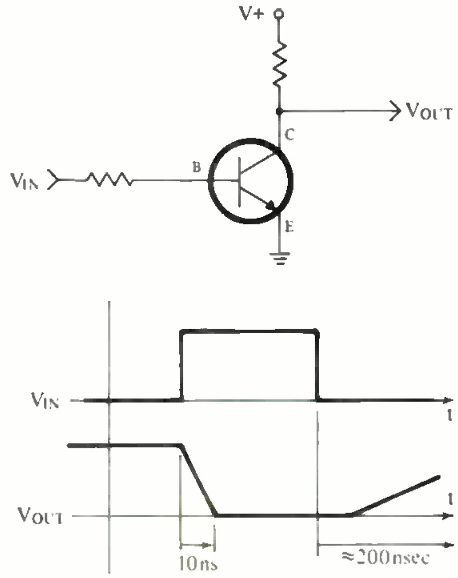

Figure 1. In this circuit, a positive drive turns the transistor on, but turn-off takes a comparatively long time.

corresponds to the collector-emitter voltage of T2. Since 0.7 volts is sufficient to make T2 an active transistor, it can pull current from T1.

The amount of excess electrons which must be removed is a function of the amount of overdrive. A large overdrive is necessary to turn the transistor on rapidly; hence, there is a conflict between fast turn-on and fast turn-off. Ideally, we would like to have a large overdrive to turn the transistor on, but no overdrive once it is on. Moreover, we would like the turn-on drive to be just enough to bring the transistor *near* saturation but not into it.

The Schottky diode will allow us to have our cake and eat it too—without getting fat!

By placing a Schottky diode across the collector-base of T1, we can have as much overdrive as we wish to enhance the turn-on, yet this drive will be removed when the transistor gets near saturation. To understand how this works, we need to review the transistor state at saturation. The collector-emitter voltage goes to approximately zero (typically, 0.1 volts) but the bass-emitter voltage, which corresponds to a diode, stays at about 0.7 volts.

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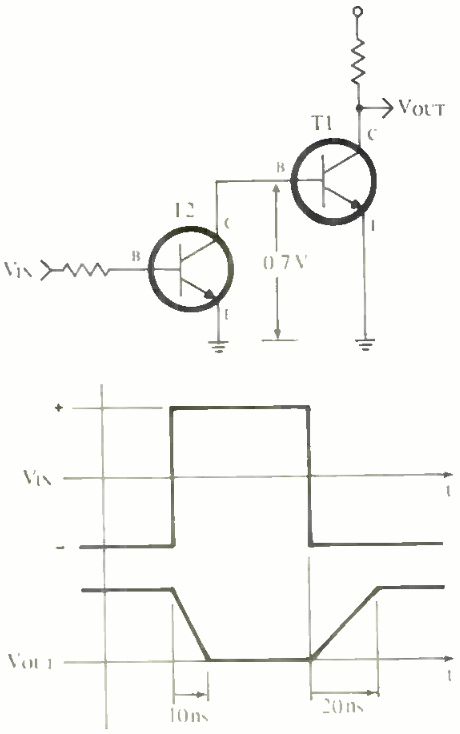

Figure 2. The addition of T_2 allows a more-rapid turn-off.

This means that the collector-base voltage is 0.6 volts at saturation. By placing the Schottky diode across these points, this situation is prevented because the diode has a drop of 0.3 volts in the forward region. Another way of understanding this is that the drive current I_D is prevented from going into the base to become I_B by being shunted to the collector. The circuit is really one of negative feedback. When the collector tries to go below 0.4 volts, the Schottky diode conducts current which is taken from the drive. The resulting base current is kept as low as possible. The transistor is actually not in saturation. Nevertheless, the drive to the base is extremely high until this state occurs since the Schottky diode is reverse-biased until the transistor approaches saturation.

With this kind of circuit, the switching times can become extremely fast because the transistor is always in current mode and not saturated. Typical switching times are on the order of 2-5 nanoseconds.

CIRCUIT EXAMPLES

Now, let's turn to various examples of flip-flops in actual circuits. Most of them are taken from applications in audio signal processing, even though they look like pure digital circuits. The logic required to build a signal processor is viewed as being very complex, yet each piece is really quite simple. A few of these pieces are shown here.

SHIFT REGISTER

The circuit in FIGURE 4 is a shift register, made up by connecting four flip-flops in cascade, with all of the clocks driven from a common clock line.

The data input appearing at the first D-input will appear at the first output after the first clock. Since this is the input to the second flip-flop, it will appear at that output after the second clock. The process repeats until the data appears at the shift register output, four clock cycles later. Although it appears to be obvious that this should work, the function depends on a subtle property of the flip-flops. Since the first flip-flop's output is changing while the old data is being taken into the second flip-flop, it is not obvious that the second flip-flop will reliably get the old data. To really understand the operation, we need to ask if the old data will stay there long enough before it changes to the new data. If we look at the specification sheet for a 7474 flip-flop, we see that the old data must remain present for a minimum of 5 nsec (hold-time specification) *after* the clock has made its transition. This then requires us to ask about the minimum propagation delay for the new data to arrive. When we look at the data sheet we discover an interesting fact: this specification is not given!

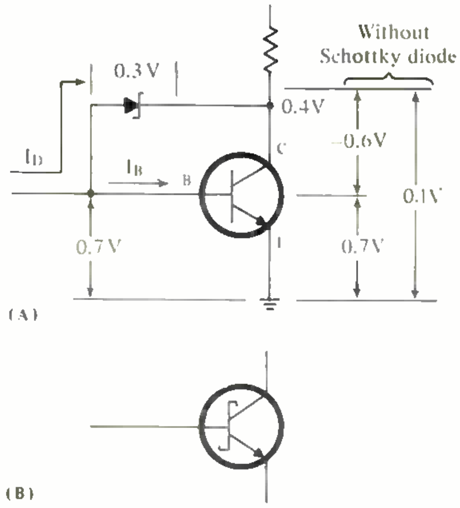

Figure 3. Here, a Schottky diode is placed across the collector-base junction. The Schottky logic symbol (B) is derived from a combination of the symbols for the Schottky diode and a regular transistor.

The typical value is listed as 14 nsec. Could the worst case be less than 5 nanoseconds? The answer is no because the devices are intended to be operated in this mode, yet the specifications do not guarantee it.

The reason is that both propagation delay and hold-time tend to be a function of the same variables: power-supply voltage and temperature. Since it is likely that all of the circuits will be at the same values, if the propagation delay is small, the hold time is also likely to be small. A further check can be made by observing the detail logic for the 7474, but this is a more difficult task. The circuit would not work if the first flip-flop is a 74S74 and the second is a 7474. The new data from the first flip-flop would get to the second in about 3 nanoseconds (expected worst case) which is less than the required 5 nanosecond hold-time of the second. [In breadboard form it would work but never in production (credit Murphy)].

Functionally, the above circuit can be

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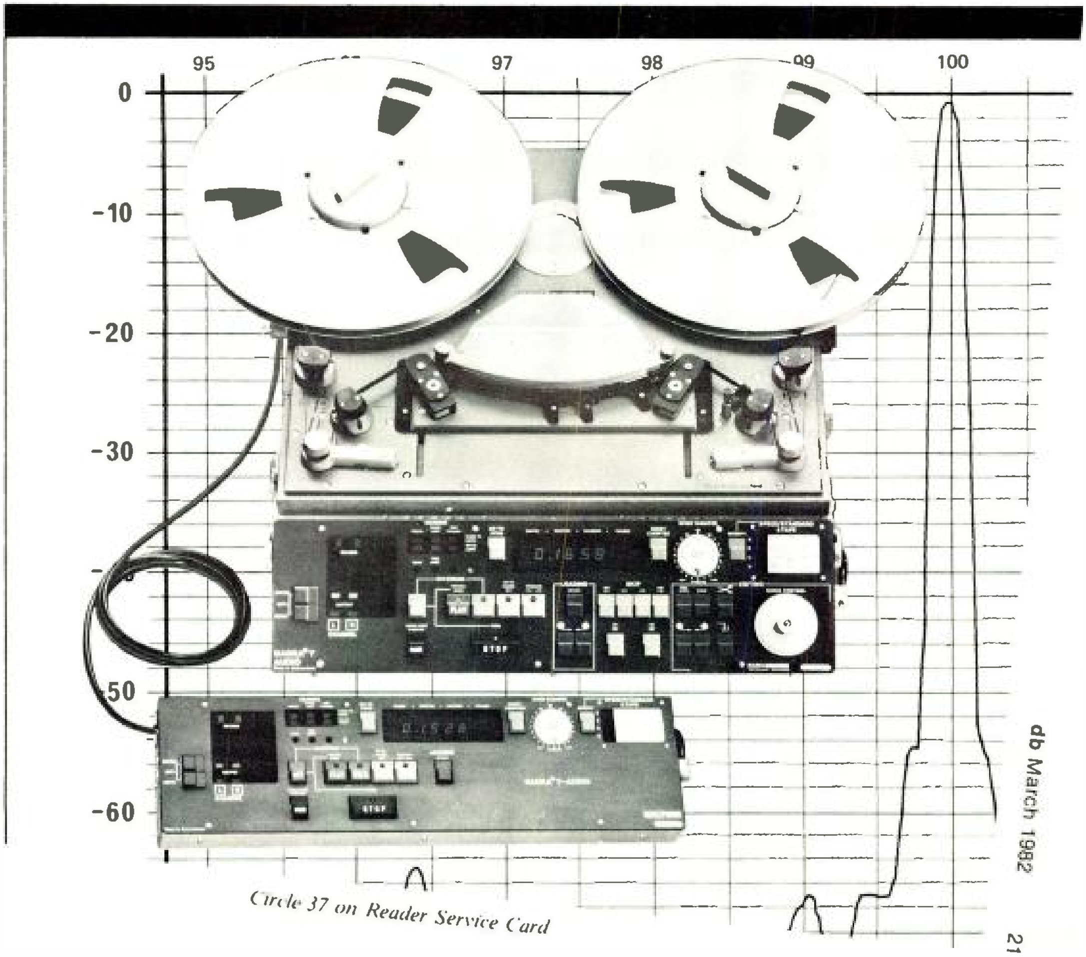

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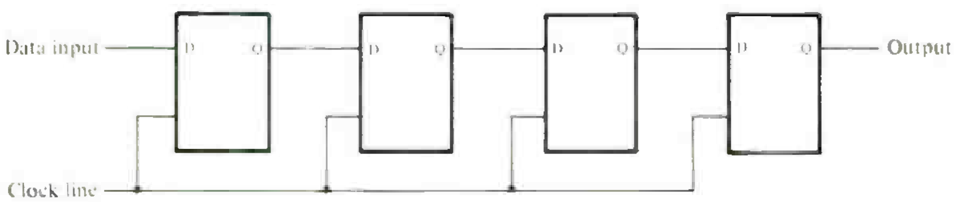

Figure 4. A shift register consisting of four flip-flops in cascade.

thought of as a shift register memory or a 4-unit delay line of 1 bit. Since in audio terms we can also be working with a full audio data word of 16 bits, the above circuit would have to be repeated 16 times, once-per-bit, in order to create a delay in audio terms. This becomes a large amount of circuitry, amounting to some 64 flip-flops. However, built out of simple gates, it would have been at least 384 gates or over 100 IC packages! MSI (medium scale integration) using octal registers, such as the 74LS374, would allow the same 4-word delay to be built with only 8 packages. The primary reason why this structure could not be built with fewer packages is that of pin-out. There are just so many pins that an IC package can have. The new package coming on line for microprocessors and other LSI (large scale integration) have an order of magnitude more pins per package.

Contrary to what you might think, the number of packages in a design is actually a major engineering factor. With too many packages, the size of the board increases or there become a large number of connectors. High-speed digital signals cannot be shipped over long distances because these signals have too large a bandwidth. When the distances are on the order of a foot, one must treat the wires as RF transmission lines! This requires considerations of characteristics impedances, loading, termination, crosstalk, etc.

Another reason for the package-count issue is that the amount of power consumed by a digital processor tends to be approximately proportional to the number of packages, because much of the current is used at the package interface totem-pole output. Internally, the logic signals are not the standard voltages except in degenerate cases. Moreover, the IC designer can keep the internal capacitors small thus requiring less current for the same speeds. In fact, in certain special microprocessors based on IIL (current-injected logic), the internal signals are mostly currents rather than voltages with a TTL combatibility at the interface input and outputs only.

If we were to extend our example to a longer delay, we might initially think that the basic idea should be repeated. However, this would increase the package count proportionately to the delay size. At a certain point, we must change the architecture to achieve a new efficiency. If no intermediate outputs are desired, and if the delay is fixed, one might use a real shift register with 1024 such flip-flops in cascade. One IC could have that many circuits and the pin-out would not increase since there is still only one input and one output for each bit. This, unfortunately, has two difficulties. The first is that such shift registers are being discontinued by IC manufacturers because of a low demand and difficulties in reliable manufacture (the two are related since they would be made reliable if the demand were higher). The second difficulty is that this delay line would be of fixed length determined by the shift register IC and the designer could not easily make a delay of 455 clocks. We refer to such a structure as "hard-wired" since there is no way to change it.

In the next article, we will explore a different kind of architecture for creating delay with the features that a large memory could be made into a large number of different delays with arbitrary length. ■

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NORMAN H. CROWHURST

db Theory & Practice

Engineering Problem Solving

• I often get letters from readers who want advice on how to go about becoming an engineer. So, while this is not specifically addressed to them, it may help answer questions they ask.

The engineer who invents solutions to problems via engineering, has to face the recurring problem of how to keep work coming in. It really doesn't matter whether that work is in research or recording. What counts is that there are clients (producers) seeking your skills. My solution was to write articles, journal papers, and books about the work I had done. This served the dual purpose of sharing my experiences with other engineers, and naturally, of encouraging entrepreneurs with problems to contact me about finding a solution for them, thus bringing me more inventive work to do.

It did far more than that, as engineers who have done likewise can testify: it improved the quality of my own engineering. By following the discipline of setting down what I had done on paper so others could understand it, I understood it better myself. And as time went on, I found I was able to do a better job on each task from the outset.

Later on, this career style led to another advantage: by teaching engineering, as a part-time instructor at two of London's engineering colleges while I was chief engineer of a prestigious company, graduates from my classes became better equipped for a career in "real-world" engineering.

By being in day-to-day contact with engineers throughout the industry, I knew of openings, as well as company philosophy where those openings were, and could help to match students to openings available. I think we all benefitted—the industry most of all. I can't claim the credit for all of it, but it was good to be a part of it.

Contrast that with today's situation. For one person to hold dual positions would now generally be regarded as "conflict of interest." Whose interest, I wonder? The argument seems to be that, if one person holds too many jobs, there won't be enough jobs to go around. Yet earlier, my holding two jobs actually created job opportunities for many others. So the argument is fallacious any way you look at it.

But there's more. By separating job functions, as is done today, we have a situation where everyone does a different job: teachers don't engineer and engineers don't teach. So, graduate students can't engineer; they can only do just what they've been taught, or "tell me how to do it, in A-B-C style."

As I've commented before, Edison didn't pick up a paperback at the corner newsstand on "How to Design a Light Bulb," or "How to Invent the Phonograph." He used his own inventive genius. But people today expect something that ridiculous, in effect.

Architects or engineers "follow the book," which doesn't teach them how to find flaws in their design before they build it, and we've seen the regrettable consequences on all sides. It's not that important in audio—after all, if an amplifier blows up, nobody loses his life over it—but lots of people have lost their lives through mistakes that could have been prevented, had engineers or architects learned their craft properly.

Seminars are held with increasing frequency, to update the copy-cat engineer on the latest hardware. Sales engineers come in, armed with slides or flip-charts, and make a well-polished presentation. They are there to sell the new line of hardware to engineers who can find uses for it. Some engineers go there with the question in mind, "Can I use this for the application I'm presently working on?"

Others attend such seminars to see how the new line fits into the presently-available pattern of devices: applying it will come later, after they really understand just what it is. Come Q & A time, these engineers direct their questions to find that out. But often, the sales-engineer types the company sent are only equipped to answer the kind of questions the copy-cat type engineers ask. They can't answer truly original questions if they weren't "primed" in advance.

But good questions should usually enable one to grasp enough to see how the new line works, compared with what's been available before. For what-

John Stronach started out as a classical pianist and a rock 'n roll drummer. Today, he's a producer/engineer. In fact, he's been a part of the record business since he was sixteen years old. His sixteen years of experience have included work with Diana Ross, The Supremes, the Jackson Five, Bobby Darin, Sammy Davis, Sarah Vaughn, Canned Heat, Alvin Lee, Three Dog Night, John Mayall, Rufus, Jo Jo Gunn, Dan Fogelberg, Joe Walsh, REO Speedwagon and more.

ON BREAKING IN

"As far as recording engineering schools, those things are great for teaching you fundamentals, but don't be spending a lot of money on that. There are people who spend thousands of dollars learning how to be a recording engineer, and they still start as a go-for, which is the same way everybody starts. It's nice to have that behind you, but I don't know. I don't know that it does all that much good. The best way to learn is by doing."

ON REPETITION OF STYLE

"I've seen it ruin people's careers. You can't use the same production style all the time. What works for one group of songs won't necessarily work for another. You have to remain flexible enough to change your production techniques as the music changes."

ON TECHNOLOGY

"A lot of producers and engineers are real spoiled with all this technical gadgetry and wizardry and all the things we can do now. They forget about the music, and the music is the thing we are here for. That's what you have to keep in mind all the time."

ON TAKING OVER

"The producer is there to help. It is not a dictatorial thing. A lot of producers get into a situation such as 'You are going to do it this way,' and it turns out to be the producer's album, not the band's. And I don't think that's fair to the band. It's their music. The act must be able to retain their identity and not just be a vehicle for the producer."

ON PLAYING AROUND

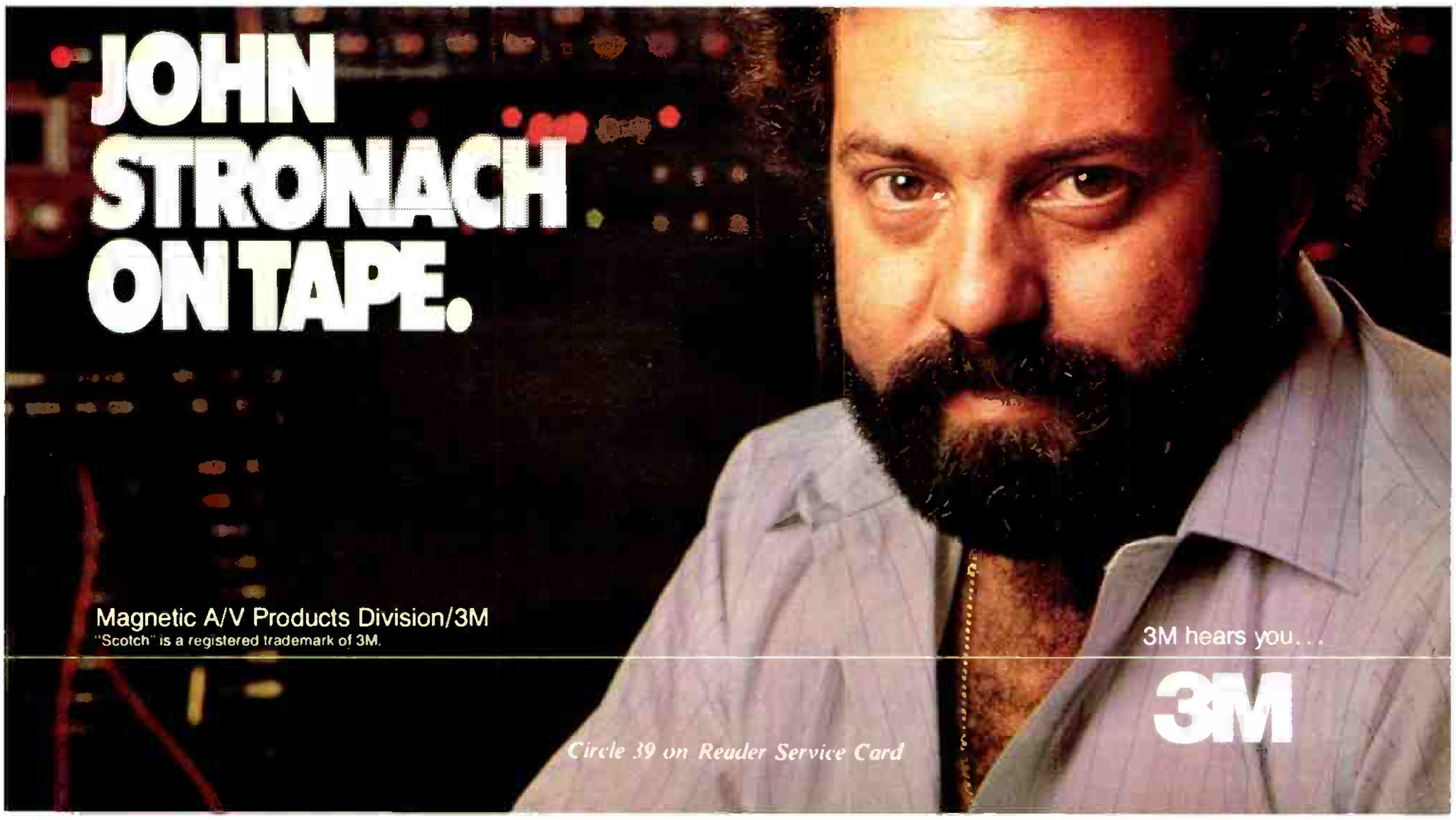
"In today's world, you have to be real businesslike. It's not like the early 70's, where everybody comes in and has a big party. You have to work within budgets, and you have to show up on time. I bring that consistency, and I try to bring a stability to the bands, so they know that they can be as creative as they want, but yet know that they can get a lot of work done and relate with the labels and management and just tie everything together."

ON TAPE

"I used another tape for a time and switched to 3M, because I would make twenty passes and all of a sudden, you would be able to see through the other stuff. They had a bad shedding problem. I just couldn't trust it any more.

"Here at the Record Plant, we give our clients any brand they want. But I recommend to people that they use the 3M, and especially the 226. Their consistency and quality is better. It just doesn't get real good and then drop to bad. You just know that it's going to be okay all the time. You don't have to worry about it. Which is important when you're out there and you're trying to get that magic take."

SCOTCH 226
WHEN YOU LISTEN FOR A LIVING.



ever it's worth, I notice that if I ask a bunch of that kind of questions, come coffee-break time, the copy-cat boys all gather round *me* to get answers, rather than clustering round the sales-engineer types the company sent down. Does that tell you something?

In short, a person who has good grounding in basics, rather than merely knowing answers to a lot of questions, can "catch up" on the latest hardware technology infinitely faster than those who must memorize numerous answers to questions. Connected with this in its impact on today's technology, is the way the technology itself has changed over the years.

Years ago, an engineer was supposed to know the basics—particularly the mathematics associated with design technology. Admittedly, even then, an increasing number merely memorized answers and, given a completely new problem, probably would flunk it. But where, in those days, it was possible to connect basics with the discrete elements of any system built, today that is a remote capability.

You have before you one of thousands of types of IC as an element for your system. You apply the specified voltages to designated pins on this IC to energize it. Next, the appropriate combination of inputs to input pins will cause it to do whatever it's designed to do, and deliver outputs accordingly, whatever they are.

For example, an organ module will generate the 12 master frequencies required for an octave simply by applying a single master frequency to a single input. Changing that master frequency will alter the pitch of the whole organ. That was virtually impossible before the advent of a new generation of hardware. Other pieces of hardware form the basis for extremely-accurate clocks, calculators—even to calculating trigonometric and hyperbolic functions on the spot, not to mention doing everything in audio. You don't need to know how they do it, but merely what they do.

It has been said that if you're out of electronics for two years, you are out of electronics: so much will have changed, you won't recognize anything any more. That is true, regarding the millions of facts you must memorize, or at least be familiar with, concerning hardware. But if you know basics, it's a different story. You are then aware of what is basically possible—even if nobody's invented the hardware to do it yet—as well as what is basically impossible.

For instance, it takes a few seconds for an electromagnetic wave to travel to the moon and back. If someone wants the signal to get back before it leaves, that's impossible, whatever the hardware you may have. Even exceeding the speed of light couldn't achieve that!

At this point in my career, I sometimes feel that I have been "out of it" for far longer than two years. For much of my recent past, I've been pre-occupied with a long-running battle with what is affectionately known as "Big Brother." I'll spare the readers the gory details, but eventually a friendly District Attorney found out what was going on, and made a useful suggestion.

He knew of my engineering credits and said, "With all your credits in engineering, you should find law a piece of cake. And you needn't go to law school to study law. That office in the Courthouse marked 'Attorneys Only' is the Law Library and must be open to the public because it's paid for with public funds. You can study law there."

Following his suggestion enabled me to stay out of jail. I soon found myself applying the same strategy that had worked in engineering and education in a totally new sphere: law. It's a universal principle.

But my first love is still engineering. Studying law enabled me to help others almost as much as I have helped myself: a very necessary service, but I'd still prefer to be engineering new things. I'd watch what's going on, almost from the outside, constantly wishing I was still busy doing it myself.

To me, an encouraging fact of life is the change in political climate. And so I think the time is right for me to move ahead again; to get "back into harness." Best of all, I now have a lot of expertise in the law necessary to back up that effort. And I'll be looking for people who recognize that the basics approach has validity and that much can be saved in engineering costs by adopting it.

The present general philosophy seems to require engineers to learn about the latest hardware, which they apply to design, with little preparation for finding the "bugs" before a great deal of time and money has been invested. However, by using the other approach, we can find the bugs before prototypes are built, but it does require an understanding of basics and of finding errors before they cause failure.

In bridges and structures, design errors result in collapse that may cost lives. In electronic circuit design, today's approach always assumes that each chip will do precisely what it's designed to do, which it does most of the time. But what happens when it doesn't—when something inside it fails? That is still something that is not taught or considered, and a lot of time and money could be saved by having the ability to do that.

So, I'm going back into that kind of work. If President Reagan isn't too old to be President of the United States, I'm not too old to use my lightning catch-up strategy to get back into harness as an electronic design engineer, with a specialty in audio, both analog and digital.

I'll miss writing this column very much. However, I know it will be in good hands with Ken Pohlmann taking over the reins. And, of course, I'll be back from time-to-time with a feature article on some of my various projects that may be of interest to db readers. ■

EDITOR'S NOTE

As Norman Crowhurst embarks on this new direction in his long and distinguished career, we are reminded of a 1959 citation in which the Audio Engineering Society awarded him a Fellowship "...for definitive contributions to audio technology."

Today, Norman looks forward to many more years of much more of the same. We know he'd be delighted to stay in touch with his many well-wishers and possibly to collaborate with some of the more creative types who are still not afraid to try new answers to old questions, and every now-and-then, an old answer to a new question.

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Digital Audio Via The Video Route

The Sony PCM-F1 Digital Audio Processor

• Ask any audio engineer involved in film-making what portable tape deck he or she used on location and the chances are that the response will be "a Nagra portable reel-to-reel machine." Indeed, the amazingly compact, precision-built Nagra professional tape decks are extremely reliable and their speed is so accurate that it's possible to use them for recording the sound track of any film made for commercial, industrial, documentary or even entertainment purposes where lip synch is required. Having been involved in a few of these film endeavors myself (both behind and in front of the camera), I can testify that once the sound of the scene-identifying sticks banging together has been synchronized with the corresponding picture, synchronization remains virtually perfect for an entire take or scene. However, as with all analog tape decks operating at relatively slow speeds and using ¼-inch tape,

both signal-to-noise ratio and distortion levels may leave something to be desired. Then too, as good as the speed regulation of the Nagra is, if a scene is long enough and if no external means of synchronizing the audio deck with the film is provided during filming, it is not unusual to lose synch. The real virtues of the Nagra have been its extreme compactness and its portability. More often than not, even a simple four-input mic mixer is bigger and weighs more than the mastering deck.

Well, from all indictions, some film sound-tracks of the near future may be made in a fundamentally new way. That's because of companies like Sony who, at the recently-concluded Winter Consumer Electronic Show, introduced a new, portable digital audio processor known as the Model PCM-F1. As you probably know, a digital audio processor is a device which converts an audio signal into a specific digital code which is integrated into the standard waveforms of an NTSC television picture format. In other words, each horizontal line of "video" appearing at the output terminals of the processor actually contains a specified number of 14-bit digital words which can then be stored on video cassette tape. During playback, the signals coming from the video tape player comprise the millions of bits which comprise the audio information. These are converted back to a continuous analog audio signal. Late in the 1970s, a committee of engineers from leading electronic companies in Japan agreed upon standards for the signal format to be used in recording digital or PCM (Pulse Code Modulation) using video tape recorders. The format has come to be known as the EIAJ (Electronic Industries Association of Japan) standard.

In the mid '70s, well before the EIAJ format was agreed upon, Sony produced their first PCM Audio Processor, the Model PCM-1. I had occasion to test that unit, which was rather monstrous in size and price (in excess of $5000.00!). When the EIAJ format was agreed upon, Sony quickly up-dated their processor, which became a Model PCM-10, but which was still rather monstrous in size and continued to bear that $5000.00 price tag, give or take a few hundred. The current introduction of the PCM-F1 therefore represents a "third generation" digital audio processor.

That would not be big news, were it not for the fact that the new processor represents a dramatic improvement over previous digital processors not only in its performance but in design, flexibility and price, as well. The price has been reduced to around $1900.00, which at once makes the PCM-F1 a viable alternative to portable analog tape decks previously used for professional audio work in the field. The new PCM processor features two new ICs developed and mass-produced by Sony: the CX-889 analog-to-digital converter and the CX-890 digital-to-analog converter. The latter chip is the same one that will ultimately be used in the Compact Disc digital audio system scheduled to be introduced next year and expected to become the world standard for true digital audio discs. It's interesting to see how once again, products developed for one segment of the audio industry (in this case home entertainment) aid in the manufacture of products for another segment (the semi-pro and pro recording field).

These newly developed large-scale integrated circuits are part of the explanation for the fact that the new PCM-F1 is one-eighth the volume of previous digital audio processors, one-fifth their weight and approximately one-third their cost!

The PCM-F1 works with any NTSC-standard video cassette recorder: professional U-Matic, Beta, VHS, or even the ¼-inch video tape format being offered by Technicolor Audio-Video Division. As you might have guessed, however, Sony would like to see users employ the device in concert with their new portable SL-2000 BetaPak VCR. Together, these units which match each other in size and appearance, make a portable digital two-channel mastering system weighing only 18 pounds.

Complementing the size and weight reduction afforded by the new integrated circuits, the PCM-F1 is, as far as I have been able to determine, the first digital audio processor that can draw its power from three different power sources: AC power supply (available as an optional accessory), a rechargeable battery, and a car/boat adaptor.

What particularly excited me in terms of the PCM-F1's applicability to professional audio was the fact that the unit not only conforms to the EIAJ standard (which was really designed for *consumer* PCM recording), but was also made compatible for use in professional PCM recordings. The difference here has to do with the number of bits per digital word in the digital encoding scheme. The EIAJ standard calls for 14-bit quantization. This level of quantization results in a dynamic range capability of some 86 dB in the case of the PCM-F1, and a harmonic distortion level (referred to maximum record level) of 0.01 percent. Certainly, these numbers are better than *any* analog recording system likely to be owned by small or large studios. But that's really not the point. Professional digital recording equipment has generally favored a 16-bit quantization format which, in addition to increasing maximum available dynamic range to beyond 90 dB, also further reduces theoretical distortion to below 0.007 percent. The Sony processor, by means of a simple, rear-panel slide switch, converts from the EIAJ format to the professional 16-bit format for compatibility with other studio professional equipment. Frequency response in either quantization mode is flat from 10 Hz to 20,000 Hz, within ±0.5 dB. Try to match that with *any* analog recording system you know of—portable or non-portable!

The other well publicized advantages of PCM recording, whether with the aid of a processor and a VCR or using a much more expensive, stationary-head professional digital audio recorder, is the total absence of (or at least the unmeasurably low) wow-and-flutter and the ability to make second-, third-, or more generation copies of a master recording without degrading signal quality. For this latter goal, the Sony processor incorporates a "copy out" jack at the rear of the unit. Available at this output jack is the encoded *video* signal rather than the decoded audio signal which is available at the main output terminals and at the built-in headphone jack.

WORKING WITH A PCM PROCESSOR/ VCR COMBINATION

Having had an opportunity now to work with at least four digital audio processors (three that required a VCR, a fourth that's a complete all-in-one unit from Technics but that uses a video cassette and employs the EIAJ video format), I have had to unlearn some of the fundamental recording techniques that I've lived with for years and years. For example, letting the metering system read above 0 dB is an absolute no-no in digital audio recording. Unlike analog recordings in which tape saturation occurs gradually, when you run out of digital words in a PCM processor/video recording system, distortion increases geometrically. In that sense, there's simply *no* headroom above nominal 0 dB. That takes a bit of getting used to, but remember, 0 dB is a relative level here, and you have so many more dB below 0 dB before you approach the quantization noise floor that dynamic range should never be a problem. It's just a matter of shifting (mentally) the meter ranges downward when you set your levels.

When I first encountered a digital audio processor I felt that the industry would be better off with a dedicated digital audio tape deck so that the VCR would not have to be a part of the system. I confess that after working with the Sony PCM-F1 I've changed my mind. Most of us own some sort of VCR already and as I quickly found out, even though Sony would prefer us to use Beta VCRs with the PCM-F1, my sample processor worked just fine with an early generation portable VHS machine I happen to own. To be sure, that VHS portable weighs about twice as much as the 8 pound 13 oz. PCM-F1, but that serves me right for always buying the first model out! ■

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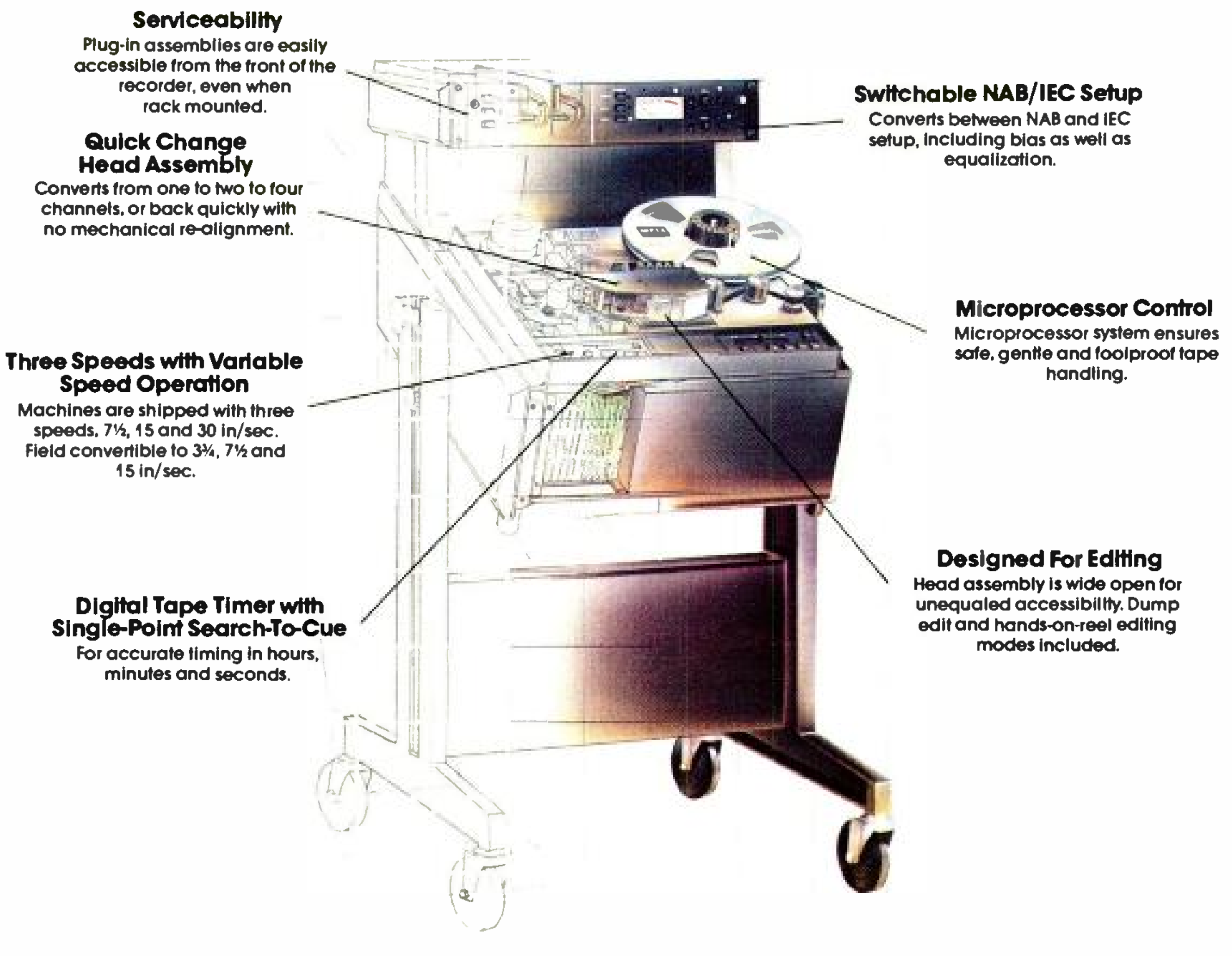

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db Editorial

LIKE SO MANY OTHER publications, this little journal prepares an annual editorial forecast, in which we match up the months of the year with a list of topics of the month. For example: June, 1982: Microphones, and so on through early 1983. So you see we actually do know what will be printed in the next issue, and even in the one after that.

More often than not, our topics are chosen with little or no regard for geography. For example, that microphone is pretty much of an international instrument, and it doesn't really matter where it was built; it works about the same in New York as in New Caledonia. To us, and to our readers, it's the hardware and/or the technique that holds our interest, and not its country of origin.

However, every now and then we like to shift gears and take a closer look at the country in which the action is taking place. While we don't really expect to become the National Geographic Magazine of pro audio, we do find it intriguing to look at how the rest of the world is coping with 20-to-20k.

As you may have guessed by now, this is International Audio month here at **db**. On the following pages, we look back on the early days of tape recording in Germany, explore a present-day concert recording on the island of Madeira, and cast an anxious westward glance towards the far-east (such are the vagaries of the geographic mind) to see what's in store for us from "Japan, Incorporated."

Maybe this is as good time as any to also take a glance a little closer to home as well, and see what's happening—or what could happen—in this little corner of international audio.

So far, the American pro audio industry remains in good health—relatively. We hasten to add "relatively," since the generally rotten economy is certainly taking its toll on just about every industry, including audio. However, this too shall pass (we hope!), and sooner or later we can all get back to business as usual, providing we don't go out of business in the meantime.

To see what could happen, take a look at the US auto industry. The great minds of Detroit have tried just about everything to get us to buy American. (That is, everything except producing a car that is worth buying.) To us, the well-publicized rebate system is a typically-dismal example of the sort of mentality that prevails. Assuming the car buyer has an IQ at least as high as an EPA mileage rating, let's ask him the following multiple-choice question: "Where is the rebate money coming from?" The correct answer is A.) from Santa Claus; B.) from the tooth fairy; C.) from your own pocket, dummy!

Most of us should have little trouble answering this question; we realize that the only way to get a rebate on a product is to pay a lot more for it than it is actually worth. Most of us don't like to do this. Few of us have more money than sense. And so, we retaliate by shopping elsewhere, and getting our money's worth without being forced through the rebate game.

Does this little excursion into automotive economics really have anything to do with pro audio? Possibly. The American audio manufacturer is no longer the only game in town. Just like his automotive counterpart, he now faces competition from around the globe. Perhaps he can profit by studying the mistakes of his counterpart behind the wheel.

Even if Detroit hasn't been able to figure it out yet, most of the rest of us have: give the customer a choice (automobiles, audio or whatever) and he'll select the product that gives him his money's worth. Give the customer no choice, and he'll simply find another source.

So far, we remain generally impressed with the consistently high quality of most recording hardware that crosses our path. But there is always the temptation—especially during hard economic times—to try to cut costs. This can be done in two ways: lower your standards, think "cheap," and hope the customer doesn't notice; or, raise your standards, think "efficient," and hope your customer does notice.

A quick look at our computer industry should convince anyone that we are fully capable of being a world leader. A glance at our auto industry will also show that at times we have difficulty playing follow-the-leader. Which path will we take during this digital decade?

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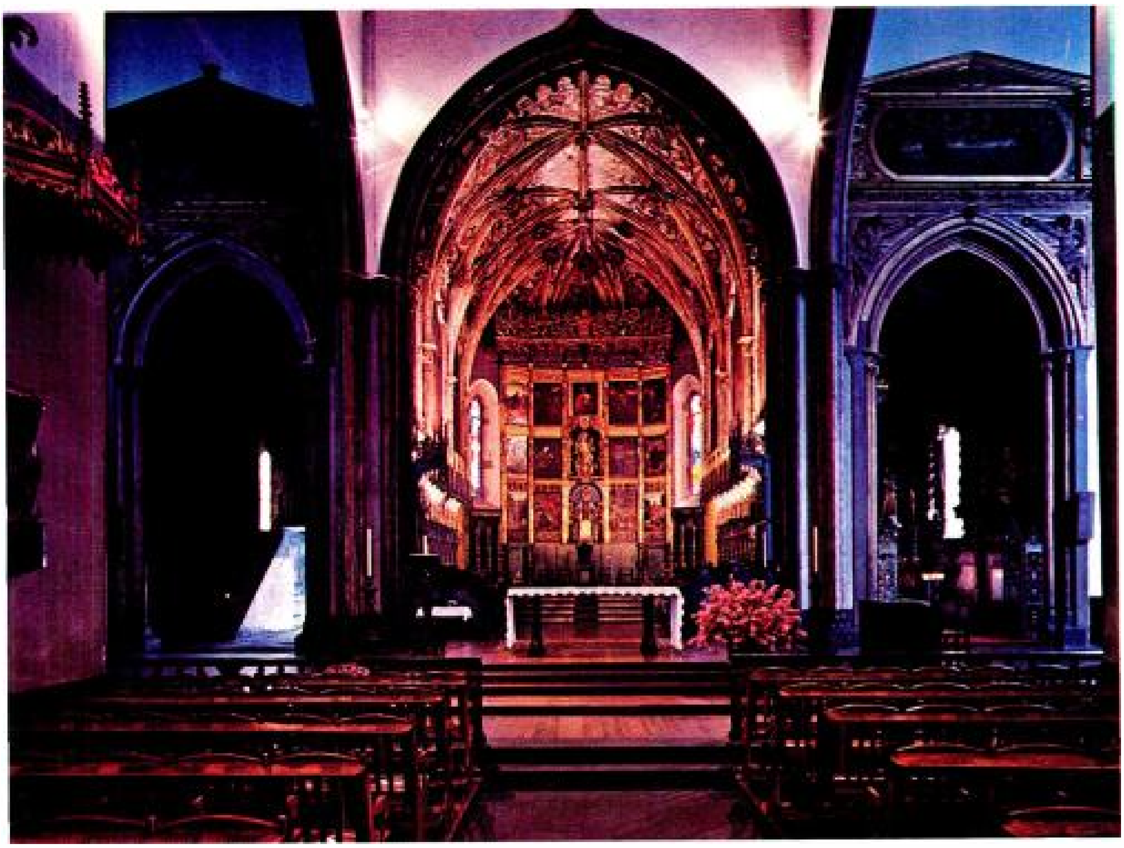

Interior of the Catedral da Sé in Funchal, capital of the Island of Madeira, Portugal. The altar pieces were created in Flanders in the 15th century.

RECORDING THE FIRST Annual Madeira Bach Festival in June 1980 would be an interesting assignment for several reasons. First, I'd be recording nine concerts in seven days, in settings ranging from a 16th-Century cathedral with vaulted ceilings to a studio-like auditorium. The festival would take place in Funchal, the provincial capital of Madeira, the island located off the coast of Portugal. I would not be able to scout the sites beforehand, and because it was a first-time event, I was unable to obtain much in the way of concrete information from which I could determine what and how much equipment to take with me. Even the simplest questions had no clear answers. What electricity would be available? How reliable would it be? What were the acoustic characteristics of the concert venues?

From both logistical and technical standpoints, it was difficult to anticipate the recording situations—and yet equipment had to be chosen, packaged, and shipped to Madeira weeks in advance of my arrival for the Festival.

Richard Koziol is the chief engineer at WNCN, 104.3 FM New York City.

WHEN IN DOUBT, TAKE EVERYTHING

My feeling is that it is best to use the simplest possible equipment for a remote recording (at WNCN, we do about 50 a

View of the 15th century Manuelline Tower of the Catedral de Sé in downtown Funchal.

year). But because I had no idea what I'd encounter that first year of the Madeira Bach Festival, I took more than I needed. Where feasible, I brought duplicate equipment, anticipating the possibility of malfunction or breakdown in shipping. In other cases, I chose equipment with built-in redundancy. And where this wasn't possible, I took some spare parts and simple tools in the likelihood that these small but important items might not be readily available on the island.

Looking back on that first year, I certainly took with me more than I needed. In fact, I knew I would be able to hone down the list of equipment chosen for my return trip, for the 1981 Festival. Yet I was glad to have the extra equipment that first year—if for different reasons than I had anticipated.

In certain instances, for example, I was able to leave a setup intact after a rehearsal, which saved me considerable time later on when the performance took place. (I was a one-man show, with able assistance from my wife, Patricia.)

In addition, I was asked after I arrived if I would provide a stereo audio signal for the local Madeira radio station. I had already (before traveling to Madeira) agreed to provide the audio for both the Portuguese National Television and a PBS film crew which was doing a project on Madeira and the Bach Festival. For these situations, the extra equipment came in handy.

YARDS OF EXTENSION CORD

My basic complement of equipment included an eight input four output mixer; two tape recorders, 12 microphones, stands, clamps, and assorted brackets; and about 2,000 feet of microphone cable in various lengths. I also anticipated the need for power adaptors and converters, owing to the Festival's location in a foreign country. Finally, I brought yards of AC extension cord with which to run the equipment. I had absolutely no idea beforehand where I would be able to set up the equipment in relation to performers, electrical outlets, and so on.

I tried to prepare myself for anything that could possibly go

The Utrecht Symphony Orchestra at the Breicknerhaus in Linz, Austria.

wrong (in case Edsel Murphy had relatives in Funchal), and was surprised and relieved that the worst never did happen. The equipment survived the trans-Atlantic journey with nothing more than a few loose screws here and there. Available power was adequate—although we did experience a blackout both in 1980 and in 1981. I was able to set up relatively close to the performers, so microphone lines did not have to be too long. What little information I was able to obtain in New York, before going to Madeira, had been accurate. That, combined with educated guesses and a few hunches on my part, resulted in a successful recording session; we ended up with 20 hours of music and interviews which were later edited in New York and broadcast in five separate programs.

What I learned that first year enabled me to fine-tune the entire process—from choosing equipment through setting up and recording—when I went back to Madeira last June for the second Bach Festival.

TOOK ALMOST EVERYTHING

As I mentioned, I had taken more equipment the first year than I really needed. Yet in compiling the list for the second time around, I ended up taking almost everything I had brought for the 1980 Festival.

I took two portable tape machines—the Studer A-67 and B-67 models. Since we were recording at 7½ ips, one machine acted as a backup, in case a performance ran longer than an hour. Most of the time, however, the second machine served as a standby unit.

I also brought along the same microphones I had the first year—the AKG 451 Series. I used these condenser microphones because of their conveniently-interchangeable heads, so I could easily alter their pickup characteristics. I brought along several uni- and omni-directional heads for these microphones, as well as some medium and long shotgun heads.

One other type of microphone I used was the Sony ECM 50 lavalier mike. These small microphones are useful in odd situations. For instance, I had a concert in last year's Festival which featured a single guitar backed up by a large chamber orchestra. By clipping the lavalier mike to the performer's music stand, I was able to get a good pickup from the guitar, and yet remain unobtrusive to both the performer and the audience.

As noted, I used a mixing console (TEAC Model 5) with eight inputs and four outputs. Since I normally need only four inputs and two outputs, I figured I would use the extras as standby. I like to use this particular mixing console, which has been modified for lower noise characteristics, because it has lots of redundant features—including several mono mixdown channels. I used one of these for amplification of the guitar mentioned above. This amplification was an after-thought, as I had not known I'd run into this situation.

There were several things to keep in mind here. I was not only recording the concert, but providing amplification to the audience in the *Catedral Da Sé* (Cathedral of the Sea) as well.

This amplification also helped the orchestra because we discovered during rehearsal that the orchestra—located behind the guitarist—was having trouble hearing the soloist. When I fed the microphone on the stand into the regular input channel on the mixer for the recording, I also used one of the mono submixers to feed a Marantz home amplifier which I had borrowed from a local record store. This in turn fed some Acoustic Research AR-32 speakers hidden in the orchestra. This last step enabled the orchestra to hear the guitar and provided enough amplification to give a natural, balanced sound for the orchestra.

This was the kind of situation which was impossible to anticipate. By having what I figured to be extra input and output channels on my mixer, I was able to make the needed adjustments on the spot.

Beyond the equipment mentioned, I again brought lots of cable—some as spare—although I did not bring quite as much as I had the first year. I also brought along the required voltage converters.

REHEARSALS SET THE STAGE

Next to having the right equipment, I found scheduling to be one of the key aspects of the success of this remote recording session. As I mentioned, this was a week-long festival of concerts held in several locations. I used rehearsals to set the stage for recording the performances. Rehearsals were held in the morning and afternoon—but not necessarily for that evening's performances. So I had to maintain copious notes to be certain that my setups coincided with the proper performances.

During rehearsals, I would establish my equipment setup—putting down tape marks on the floor to facilitate gearing up at performance time. However, since on some days the facility was used for religious services between the last rehearsal and the evening concert, I sometimes lost the tape marks during cleanup.

I'd also use rehearsals to set levels in my mixing console. The

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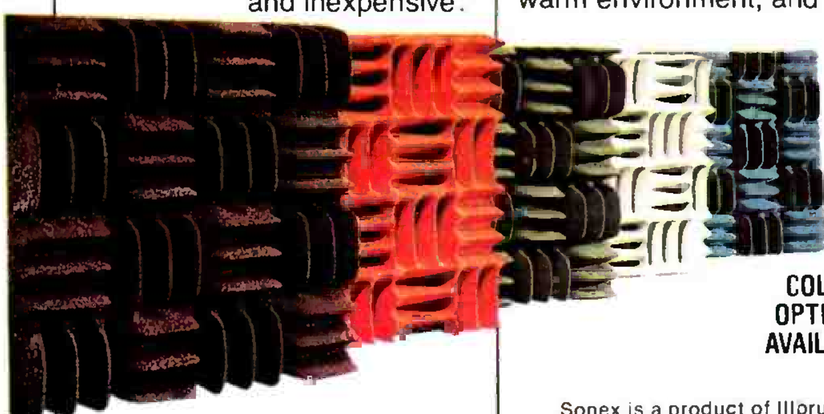

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goal with a remote like this is to make the sound as good or better than what is actually heard in the hall—and yet provide the feeling of the live performance. Finding the right mix during these rehearsals entailed a lot of running on my part. From my outpost behind the orchestra, I would run out into the hall, listen to the music to see how it was sounding, and then run back to the mixer.

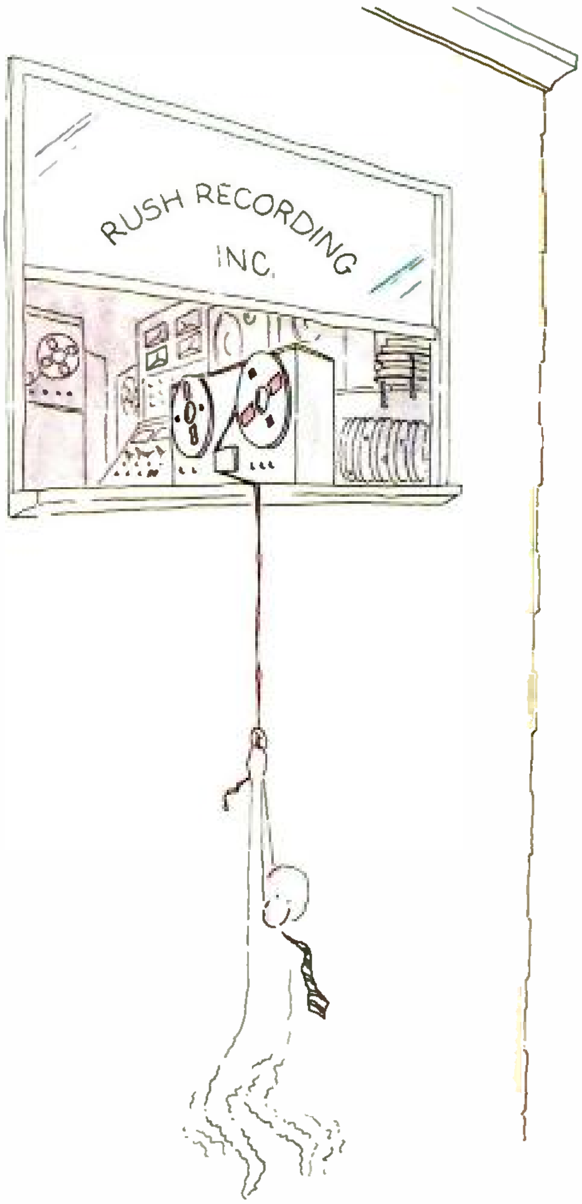
Everything was monitored on headphones—which presented another interesting problem, because sound on headphones sounds quite different than that coming over speakers. Yet in doing the remote, I had to keep in mind that WNCN's listeners would be hearing the music on headphones, on $4,000 speakers, and on a wide variety of systems in between those extremes. It's difficult to keep all of that in mind while setting up for a recording session. Luckily, since I have done so many remotes, I've gotten into the habit of listening to music on headphones, then hearing how it sounds later on in the studio, and then listening to it again while at home as it's being aired. After awhile, you develop a feel which enables you to more closely balance things.

Soprano Elly Ameling, with conductor Yuval Waldman and oboist Allan Vogel after a performance at the Catedral da Sé.

ADJUSTING FOR AUDIENCES AND A SQUEAKY BUS

Even though the balance was set at rehearsal, during the performance I had to keep in mind that when I set the balance earlier, the hall was empty; it would have to be adjusted somewhat once the hall was filled with people. And if I were using amplification for the audience, I would have to adjust to compensate for the filled hall. For these performances, I'd have someone—usually my wife—stationed in the concert hall, to report back to me after the first movement.

Another unknown in doing a remote recording for the first time is the amount of outside noise you'll encounter. The Catedral da Sé—primary location for the concerts—had a reasonably large interior with an ornately carved wooden ceiling, a wood and stone floor, and whitewashed plaster walls. The combination proved to be acoustically excellent for recording. I did not have to get microphones right into the orchestra to avoid being overpowered by the reverb, and yet there was still enough reverb to complement the sound. There was one acoustical problem however—the outside noise of squeaky bus brakes and ever-present motorcycles and trucks. But this was unavoidable, since the cathedral is located in downtown Funchal.



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A performance at the Madeira Bach Festival at the Catedral da Sé.

PLACED MICROPHONES IN COINCIDENT PAIRS

One adjustment I made between the first and the second year of recording was to place microphones closer to the soloists. The first year, placement was relatively far away, which was fine for the orchestra but not when you had a situation where you were trying to pick up a solo cello or a harpsichord accompaniment.

The typical microphone placement consisted of microphones hung in a coincident pair arrangement. These were about 15 feet in the air above the conductor's head. This was the primary pickup. Then I positioned one single microphone on either side of the orchestra. These microphones, with shotgun heads, were used to add presence or accent the pickup of a main microphone pair. I could aim them toward a single instrument or group.

There were additional arrangements depending upon the particular concert. For a performance featuring a cello, I put an additional pair on a floor stand, facing the cello. I used the Sony lavalier microphones for a double harpsichord concerto because they enabled me to get in close without getting in the way of the orchestra or audience. The pickup from these mikes was also used for a small amount of amplification in the hall.

The organ, which was located on the opposite side of the cathedral, required another coincident pair of microphones, mounted on the railing. This put the microphones about 20 feet from the instrument. I found that if I went further out, suspending the microphones into the nave, I found a warmer, more balanced sound from the instrument—but this placement was too susceptible to outside noise.

One of the reasons I rely on the coincident pair of microphones is that it affords good mono compatibility. Many people in the fm audience listen in mono, and any phase cancellation is unacceptable. All mike placement and mixing must always take this into account. So sometimes, I purposely sacrifice stereo effect for mono compatibility. This is only one of many compromises one makes when recording a live performance for broadcast.

THE SETUP IN THE MUNICIPAL THEATRE

While most concerts were performed in the cathedral, there was one performance which took place in the city's municipal theatre—an 18th Century baroque hall which is relatively dry. Acoustically, conditions were much like those of a recording studio and the setup was simple.

The first year, I used a coincident pair, but last year, I used a single Neumann stereo mike with an MS decoder. This was provided by the Portuguese National Radio crew who had brought along some of their own equipment. The first year, however, all they brought with them were two tape machines.

In this regard, the first year's experience was particular interesting. When I arrived in Madeira, I wasn't sure what all of the people who would want audio feeds really needed. Part of it was the language barrier, since I don't speak Portuguese. The PBS film crew, however, was from the New York area, so communication here was relatively easy. The only problem was that what they wanted was to put a considerable drain on my time. They wanted rehearsals, and I had planned only to spot-check rehearsals. But since the film was being shot documentary-style, the crew was using quite a few rehearsals for material. Therefore, in many cases, I had to set up and run my equipment just as if I were recording.

Because the film was being produced for television, they required a mono audio signal. Fortunately, since my mixing console is capable of supplying two separate mono output mixes or balances—independent of the stereo output—this was not a problem.

The Portuguese National Television needed an audio feed—or so I had thought before leaving New York. What it turned out to be was the need for *four* audio feeds. Luckily, I had brought along two signal splitters, with which I was able to divide the signals to feed several devices without interacting. These splitters, connected to the second mono output of the mixer, took care of the television crew.

The final set of feeds, to the Portuguese National Radio and the local Madeiran radio, were totally unanticipated. Fortunately, broadcast audio is international in its terms and definitions. And even though they spoke very little English—about as much as I did Portuguese—we communicated via hand signals, diagrams, and words.

They needed a stereo signal so that they could record the concerts for broadcast in Lisbon over the National Radio facility. In addition, they took the stereo signal and combined both right and left channels to mono, and using an on-location announcer, broadcast the concerts live in Madeira.

RECOMMENDS THE EXPERIENCE

One drawback to the experience of the past two years is that it requires so much work—between setups, rehearsals and performances—that we had little time to enjoy the beautiful island of Madeira. In addition, because of the location of the remote, equipment was tied up for a long time. Although the Festival started the second week in June, equipment had to be packed and shipped in mid-May—and wasn't returned to New York until August.

Yet the benefits far outweighed any disadvantages. I've become friends—despite the language difference—with the Portuguese radio people who I worked with during the Festivals. They gave me a tour of the local station and transmitter facility, and last year, even provided me with a tour of the island after the Festival was finished.

And the Festival itself was enjoyable, in part because I had been able to anticipate the unexpected, so I was able to react quickly and without stress to the many situations which arose. ■

ALMON CLEGG

Report From Japan

Author Clegg heads east to give us a look into the future of Japan's audio industry.

Until about 20 years ago, most electronic products coming from Japan were of very poor quality. Often, reprocessed metals were used, with poor dimensional tolerances, non-uniform finishes, and significant variations from unit-to-unit.

All of that changed during the 1970s. And now, through dedication to long-term planning, Japan has put together one of the most advanced manufacturing and quality control systems to be found anywhere in the world. On a visit to almost any manufacturing operation in Japan, one quickly discovers highly-advanced automation techniques and sophisticated production methods. The investment in equipment, engineering and facilities is conspicuous everywhere.

In a short decade, through determination to surpass the competition, the Japanese have become authorities on quality, reliabilty and dependability. By combining sophisticated manufacturing techniques and high-level design with an almost militant demand on human workmanship, the typical electronic product rolls off the assembly line with a uniformity that is rarely seen elsewhere. (Even the strawberries in the food market have a uniformity from berry-to-berry, basket-to-basket and store-to-store that is hard to imagine.)

Audio hardware manufactured in Japan now has a typical defect rate of less than one percent. In fact, far more damage is caused by improper handling by the customer opening the carton, or during initial use of the device. Yet in spite of the high quality, most Japanese companies still insist on elaborate overseas service networks, where parts and repairs can be obtained. Again, the long-term view is considered important to business, and the image of quality and service is mandatory. Great attention is given to defective parts, and it is not uncommon for a defective part to go back to the design engineer to determine the cause of failure, and what impact this will have on future manufacturing procedures.

WHAT ABOUT INNOVATION?

Some observers claim that the Japanese are not innovators. They merely find better ways to do what someone else has already done. Others point out that labor and government tax favors are beneficial to the Japanese manufacturer. But these arguments are not entirely sound. For instance, labor costs in Japan are actually very high, and automation is the key to keeping these costs down. In fact, Japanese production facilities are far more advanced than those of the overseas competition. They have made it a very careful science to produce high quality at the lowest possible cost. Recent competition from Taiwan, Korea and Singapore and now from China has forced the Japanese manager to be very astute in implementing labor-saving methods. At the same time, quality control techniques have become very effective.

The charge that the Japanese are not innovators needs careful scrutiny. Many of the large companies have research laboratories and engineering staffs that are truly impressive. A look at the Japanese auto industry may be a quick check of innovation. Or if it is not innovation, then it certainly is ingenuity and extraordinary good luck the way they have caused such a market shift. And anyone reading the scientific journals and attending conventions and technical society meetings knows that the Japanese scientist is always there making a contribution.

But the real purpose of this article is not to delve into international economics and business techniques, but to take a look at what is going on in Japanese audio today, and to see what impact this may have on the rest of the world. Before examining the pro' audio field, let's take a quick look at consumer audio, where the Japanese presence has been felt the longest.

HI-FI AND ESOTERIC AUDIO

For quite some time, the high-fidelity market has been saturated by Japanese imports. Pioneer, Technics, Sony, Sanyo, Kenwood, Sansui, Yamaha are all household words by now, and we will continue to see them dominating the dealers shelves in the coming years. With the development of new IC techniques and digital designs, we will see more digital tuners, high-power receivers, more watts-per-dollar and more-sophisticated features in general. Yet, no revolutionary technology is to be seen in this area on a large scale.

Esoteric high fidelity, developed by the Americans, is now being challenged by the Japanese. Denon, Lux, Yamaha, Technics R & B and others have entered this arena with varying degrees of success. However, the greatest contribution may well be the popularization of the new breed of Class-A type amplifiers, and other circuits which provide extremely low distortion and high performance at very attractive prices. However, the esoteric audio buyer still seems reluctant to accept some of these products, and an appeal to the subjective may be the most difficult hurdle for the Japanese to overcome.

PRO AUDIO HARDWARE

Mixers

The great success that Teac/Tascam and Yamaha have had in this field has not escaped the notice of other manufacturers. Look for many new Japanese faces in this busy product area. New features should include: faders that are impervious to dirt, drink and other abuses, meters that are easier to read and more accurate for recording and PA use, and greater flexibility of grouping and assignments.

Microphones

With the development of new materials for the diaphragm and configurations that can handle sound pressure levels of up to 200 dB (!), we will most likely see more Japanese-made microphones around in the coming months. A few models are now in common use, but the majority of popular microphones are still those of American and European companies. The sensitivity that the Japanese design engineer has recently developed for sound quality and tonal balance will make for some interesting new developments.

Japanese manufacturers are now bringing new microphone prototypes to their US sales representatives to get opinions on sound and tone balance before finalizing the design, so we know they are truly interested in the US market.

Loudspeakers and Sound Quality

It is well known that the American listener has not been overly enthusiastic about the sound of Japanese-designed speaker systems. In fact, very few recording studios use Japanese monitors. Will this change as time goes by? It probably will, simply because the Japanese have such a strong desire to narrow the gap between themselves and the competition. So far, cultural and acoustic differences are the major contribution to the lack of acceptance that exists. But as

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time goes by, just as they have done in other product designs, they will surely overcome the subjective quality problems of speaker design.

What are the tonal preference differences? Generally, the Japanese are accustomed to smaller rooms with less-rigid walls and different floor coverings. Also, the close living quarters that most are subjected to require lower listening levels. But the most important difference may be in the type and style of music preferred by the Japanese listener. The musical instruments popular in Japan are quite different from those in the USA.

To get around the problems caused by these differences, some Japanese manufacturers are having the final product "voiced" in the US, using American acousticians for the final sound quality check. In addition, Japanese acoustical engineers are now coming to the USA for training and study on how to appeal to the American taste.

In short, it is not a matter of objective design capability that is lacking: instead, the Japanese must learn more about the subjective aspects of creating a sound quality that is compatible with the American listener.

The NIH (Not Invented Here [Hear?]) factor certainly plays a role in all of this. However, in the long run, pride and prejudice will eventually be overcome as engineers and product managers from both countries work together on product development. This writer predicts that within two to five years we shall see many American recording studios installing monitor speakers made in Japan. Of course, this certainly doesn't mean the end of the American design by any means. With so many personal preferences and musical interests involved, from the studio all the way to the consumer, it seems logical that there will continue to be a wide variety of models from which to choose. But the Japanese will soon have their share of this important market.

As with microphones, new materials are a key to innovative advances in speaker design. For example, RAMSA has introduced a new diaphragm material made from a reinforced olefin material which has much the same properties as conventional paper. It looks like paper, feels like paper, and has the same characteristic "scratch sound" as paper, yet it is stronger and is water/liquid proof as well. Additionally, the new material can be manufactured with very close tolerances and precise control of cones from batch-to-batch, a problem long recognized by cone makers. Because of the much higher rigidity, these new cones can withstand much higher mechanical shock power before fatiguing.

A stage monitor design uses ferro-fluid in the voice coil gap and, with the added capability of heat dissipation from the voice coil to the top plate and magnet, it is reported to be able to handle 400 watts of power! With this type of development, it is likely that some significant improvements in outdoor systems will be forthcoming.

Another development, which is mostly applied to high fidelity and studio monitor systems, is the honeycomb. This new type of flat-piston diaphragm is interesting because it eliminates the "cavity" effect present with cone-type surfaces, and it provides for a higher frequency drive, with the upper harmonic modes being driven out with large voice coils. These systems also have the virtue of close tolerance in manufacturing and matching of sound quality from batch-to-batch, as with olefin. Furthermore, the honeycomb materials are less flammable than paper.

At least one laboratory in Japan is doing development work on controlled-directivity horns. This new breed of horns will have controlled dispersion in the horizontal and vertical planes from the same turnover frequency, as well as uniform radiation impedance for lower distortion and flat loading of the driver. Two or three American manufacturers are also working on such designs—it will be interesting to see who gets there first.

PSYCHOACOUSTICS

Much is being done in this area on the playback side; that is, many companies are developing playback circuits which will "enlarge" the sound field and produce unique perspective envelopes. Acoustic illusions have always intrigued the engineer and listener, and now we may see many such concepts coming into the marketplace.

In Japan, the Acoustic Research Laboratory of Matsushita Electric has developed the art of sound localization to a very advanced stage. They are developing a device which, contrary to the playback circuits just mentioned, is to be used on the recording side during mix-down. The device is capable of taking any track of a multi-track master and locating the apparent source anywhere within the frontal half of the sound stage. In playback (which requires no special circuitry), the track may be heard at any desired location from far left to far right (i.e., ±90 degrees) of the listener, or at any intermediate location in the frontal hemisphere.

This technology should give much more creative freedom to the engineer or producer looking for sound effects which are not presently available. The prototype system includes distance and spatial controls as well.

DIGITAL AUDIO

This is where the real excitement is going to be in the next decade. This writer predicts the following as a result of digital technology: digital microphones and digital loudspeakers. That's right, digital! From the microphone output all the way to the loudspeaker, the signal will be in the digital domain. Furthermore, the power amplifier will be able to time-share, and drive many speaker systems from independent channels. Such concepts are now under investigation in the research laboratories of Japan, and surely in other parts of the world as well.

At the more immediate level, PCM adaptors and fixed-head systems are continuing to be developed. Very-large-scale integrated circuits will do the entire analog-to-digital process in one chip. The chip will be equivalent to hundreds of conventional ICs. Another chip will be utilized for digital-to-analog processing, and a third chip will be for machine control purposes. Hence, we will see a digital cassette deck which will be small, self-contained and with the complete electronics package in three ICs. With a standard 14-bit linear code, the system will probably cost less than a good reel-to-reel semi-professional analog tape recorder.

While standardization of the format for fully-professional recording systems is still somewhat of a problem, it goes without saying that similar integration of the electronics will bring the all-digital studio much closer to reality.

RECORDING STUDIOS

For years, many small American recording studios have been using Japanese consoles and tape recorders. But what's in store for the big studio, where 24, 32 and 48-track work is done? Can Japan find a place alongside MCI, Harrison and others who now dominate the premier studio business?

For an industry that came from far behind in digital electronics, automobiles, and pharmaceuticals, it should be no big challenge to develop sophisticated studio recording hardware. Sony, Mitsubishi and JVC have already displayed incredible strength in their development of digital tape recording systems. And so, the big question is not one of technological capability, but one of desire. Does "Japan, Inc." want a piece of the action? Is the market big enough? Are there enough big studios to make a profitable business of it? Will the growth rate of the big studios be sufficient to make it worthwhile? Will new recording technologies replace the studio of today with large-scale digital storage and editing systems?

It seems a fair assumption that the world of pro' audio is on the threshold of a new wave of technology, which will prescribe a change in the way music is stored and eventually presented to the consumer. A lot of fascinating questions will have to be answered, as Japanese industry decides whether to invest its time, talent and energies on the development of even more audio hardware for the professional recording industry. But based on recent past experience, it's a safe bet that in the field of professional audio, we shall see more and more Japanese influence. ■

RICHARD CHILVERS

A Postscript on British Audio

The following is a brief update from the front on the British invasion (audio-style) of the colonies.

To go from a marketplace about the size of New York State to one the size of the United States can be a daunting prospect to even the most dedicated heart. But this is the prospect facing British manufacturers of audio equipment when they look to the United States. After a while, the sheer size of the geographical area registers as both an attraction and a problem. With all that space, there must be a large market; yet on the other hand, New York is about as far from London as it is from Los Angeles. The way of doing things in the U.S. is so different from U.K. and, although we both claim to speak the Queen's English (although she never seems to speak any of ours), we often end up understanding quite different things.

It was against this background that a number of manufacturers from Britain had been marketing their products in the United States with varying degrees of success. Among them was Keith Monks Audio who, after some years of relative success, realized that more of a commitment to the country and to its individual marketing needs was called for.

Keith Monks Audio was not alone in coming to this conclusion. So after much careful discussion with a number of other British manufacturers, the idea of forming a marketing association of some type was born.

CBA

Initially, three companies were involved in what was to become the Consortium of British Audio (CBA). They were Keith Monks Audio, whose range of professional microphone stands and record cleaning equipment was already well known in the industry; Audio and Design Recording whose range of signal processing equipment is also well known, and the third company Neal Ferrograph, who were interested in the United States for their Reel-to-Reel machines, cassette machines and audio test equipment.

The liason that these companies formed gave them strength beyond that which they could expect as individuals.

In Stamford, Connecticut, a service department, warehouse and office facility was set up so that the Consortium could offer a full service package including importing, warehousing, distribution, servicing and accounting facilities. This took place during the early part of 1979.

By the end of that year, a number of other companies were pressing to join the Consortium. Meanwhile, Audio and Design Recording decided that for personal reasons they would go it alone on the West Coast and so withdrew from the Consortium.

Early on, the decision had been made that the Consortium would be kept as simple as was practical, and the number of member companies would be limited. It had been agreed that, wherever possible, the Consortium membership would be limited to manufacturers who did not make conflicting products.

THE CONSORTIUM GROWS

Mid-1980 saw two more companies being accepted as members of what was fast becoming a very successful and efficient association. A & R Cambridge had built a very impressive reputation for themselves in the United Kingdom in the high-end consumer industry. They, as young men should, wanted to "go west" and so joined the Consortium under the name of their newly-formed subsidiary Arcam (USA).

Canford Audio had been concentrating on laying a solid foundation in the North of England for a business which distributes accessory items and studio furniture to the professional market. It was soon realized that their line of greatest potential was a newly developed range of audio cables. Subsequently, Canford Audio (North America) was purchased by the manufacturer of this high caliber cable. Subsequently, Canford Audio went on from strength to strength.

By now, the Consortium had close contact and a good working relationship with a wide network of representatives and dealers throughout the United States. The remaining original members of the Consortium were seeing a steady and considerable growth. Keith Monks Audio was to produce a 60 percent increase in business in 1979/80 and a further 50 percent increase on that in 1980/81, thus confirming that the idea of operating in a joint manner had been justified.

From the beginning, the CBA proved its worth in the area of Trade Shows and Conventions. Gone were many of the worries for each independent company as the CBA were able to organise the shipment of displays and equipment to and from the exhibitions, as easily as they were able to arrange the importation of the near weekly shipments from England.

In May 1981, considerable weight was added to the Consortium when Trident Audio Developments became members. This immediately gave their new company, Trident (USA), a base of operations from which to increase their already expanding market share whilst providing a direct service department. For many years Trident had enjoyed an enviable reputation in Great Britain and Europe and now, it was felt, was the time to demonstrate their commitment to the American market. They knew that after-sales support and technical back-up was as important to the sale as the initial negotiations, particularly in the section of the market in which they work, that of recording consoles, tape machines and outboard equipment.

Presently, four more companies are interested in becoming members of the newly incorporated Consortium. Three of these companies are solely in the professional industry and one in the Consumer market. Although this will mean that by far the main weight will be on the professional end of the industry, the consumer end is not being neglected and involvement of more companies means a further improvement in the facilities that the Consortium can offer. Studio packages are available from wiring to commissioning and at the other end of the scale, spare parts are available for even early Ferrograph tape recorders.

It is unlikely that more than ten companies will be accepted as members. This prevents conflicting interests destroying an otherwise smooth-working organization. With a combined intent to provide service and support to those involved, the CBA is intent on continuing its policy of increased efficiency and professionalism. For manufacturers and users alike, the future is looking good. ■

Richard Chilvers, formerly the UK Sales Manager for Keith Monks Audio, has headed the Consortium of British Audio since 1979.

J. ROBERT ASHLEY

Audio Conversations—Church P.A. Systems, Part II

Once again we join our fictitious conversationalists as they discuss the very real technical (and political) problems of church P.A. systems.

MISSOURI

JOHN IS ELATED AT his selection as chief resident tinkerer for the local Church of Christ. They are willing to let him make a major rebuild of the P.A. System.

Bob: Why are they dissatisfied with the P.A. System?

John: I'm not so sure they really are dissatisfied. It's 15 years old and the amplifier has a slight hum. The elderly people who often sit in the front complain of not hearing.

Bob: Or is it hearing-without-understanding, because of Auditory Backward Inhibition (ABI)?

John: It certainly could be. In the front of the church you can hear six different speakers with all the sounds within 5 dB of each other. The attention span during sermons is usually 5 minutes or so.

Bob: So you get the chance to cure ABI because they think the system is about ready to break down. But how did you get the job in the first place?

John: Simple: I agreed to donate all the needed equipment. I'm so sick of that P.A. System that it's worth a kilobuck to straighten things out.

Bob: What do you mean—one kilobuck? A P.A. System for that size church with expensive microphones, high-quality wiring, a third-octave equalizer, an 8-in/2-out mixer, 300-watt stereo amplifier, installed speakers, and labor will easily cost more than *ten* kilobucks.

John: First, I'm not going to waste my money on a third-octave equalizer or stereo. A 4-in/1-out mixer will be plenty—we only use two mikes. The old Western Electric amplifier is going to get a new set of valves and fresh filter capacitors and I'll run another 15 years. I expect to spend about a third of my money on a single-array speaker, a third on microphones, and a third on the mixer and wiring.

Bob: Why don't you just cut the power cord off the amplifier and donate the kilobuck?

John: Why don't you forget about that church in Indiana? Last year the microphones were stolen on a Saturday night. I went to the early service and a gentleman you would describe as a timid talker was just terrified. Remember that 5 years ago they put a lot of acoustical tile on the ceiling. It helps absorb the air-handling noises but only a very strong talker can be heard in the back of the church. That timid talker actually talks louder because he will psychologically lean on that microphone. Sorry about that, Bob, your turn-off-the-P.A.system theorem is disproved in this church.

Bob: If it's a multiple-speaker P.A. system, my theorem is still right. The only chance you have to prove me wrong is to correctly design and locate that single-array speaker. Since it will be up front and center, how are you going to handle the complaints that it spoils the looks of the church?

John: I've already sold the preacher—he's tired of seeing people sleep through his sermons. Next week I'm going to rebuild the Western Electric amplifier. The following week, I'll rough in the wiring. I'll check out the mike wiring and the new mixer on the old speakers on a Friday evening. Saturday morning, the old mike wiring gets torn out, the old speakers come down, and the new speaker array gets hung. I'm gambling that it will sound so much better that I'll be able to argue down the complainers with the eyes-and-ears story. If they can understand the timid talker for the first time, I'll win my gamble.

Bob: The preacher will get more compliments on his sermon than ever before—that will help too! Do you really think you can get enough acoustic gain to help the timid talker?

J. Robert Ashley is an engineering staff consultant for Sperry Gyroscope.

John: That's why I decided to use the tapered-array speaker instead of a readily available and less expensive omni-directional speaker. After all, there are all kinds of acoustic suspension home hi-fi speakers that cost less than $300 and would work quite well if omni-directional is OK.

Bob: I don't understand your reasoning. How can you get more gain with a tapered array than with an omni speaker?

John: I've found that the speaker directivity pattern is by far the most important factor in controlling howlback. If the gain is run up for one microphone at a time until the system is almost howling, the sound pressure at the microphone caused by the speaker is just a smidgen less than the sound pressure caused by the talker.

Bob: You can think of it in terms of the total gain through the amplifier being equal to the acoustic loss in going from the speaker to the mike.

John: Yes, but now we've got the cart before the horse. First we need to check time delays to see if anyone will sleep through the sermons. If the closest listener is about 16 feet away, the direct sound from the talker will arrive in about 14.5 milliseconds. Sound from the speaker will arrive at this listener in 18.5 milliseconds. The difference of 4 ms is much less than the 20 ms limit determined by Joseph Henry to make the two sounds add in our hearing process.

Bob: That 20 ms magic number was verified by Haas in 1949 and just recently by Jack Cox at the University of Colorado. He used digital delay lines and modern speakers, and extended the Haas work to include the effect on musicians. His Master's Thesis is a gold mine on this subject.

John: Notice that if we repeat the calculation for the listener a regular church or high school auditorium where this didn't show that a speaker located above and in front of the talker show that a speaker located above and in front of the talker was right. If it isn't run too loud, you'll swear that all of the sound is coming from the talker. It even works for most of the audience when the talker is stage left or right and the speaker is in the center.

Bob: Let's get back to that gain calculation.

John: Since we are first talking about an omni-directional speaker, we assume the sound wave fronts to be spherical and the sound intensity to vary as the square of the distance. On the verge of howlback the gain is

$$10 \log \left(\frac{\left(\frac{1}{1\text{ ft}}\right)^2}{\left(\frac{1}{20\text{ ft}}\right)^2} \right) + 10 \log \left(\frac{\left(\frac{1}{24\text{ ft}}\right)^2}{\left(\frac{1}{20\text{ ft}}\right)^2} \right)$$

or

$$26\text{ dB} - 1.6\text{ dB} = 24.4\text{ dB}$$

Bob: That should be more than enough acoustic gain for sound reinforcement.

John: I agree. It's an optimistic answer because I want to run 3 dB below howling so thay the system will never get shocked into a slowly decaying ring. Also, we have to count on gain decreasing by the square root of the number of mikes open: 3 dB for 2 mikes, 4.7 dB for 3 mikes and 6 dB for 4 mikes. There is another factor that degrades gain. The mikes will pick up reflections from the side and back walls to further reduce the gain by 3 to 6 dB. In the fancy formulas, this shows up by accounting for the sound in the reverberant field.

Bob: Those fancy formulas do a nice job of predicting gain but they sure don't say if the system puts the listeners to sleep. Won't cardioid mikes get you out of this trouble?

John: By a dB or so but not as much as commonly thought. They can reduce the contribution from the speaker to mike path. Then, they add to the gain from the bounce shot from the rear wall. The conventional wisdom about cardioid mikes being the only kind to use doesn't work out like most of the tinkerers think. If a performer takes a cardioid off the mike stand, Murphy's Law says he will point it at the speaker sooner or later. Omni mikes are really better.

Bob: Well, using omni mikes and all the pessimism you can justify, I still see an acoustic gain of more than 10 dB for the talker 1 foot from the mike. In terms of loudness, this means the sound at a listener in front would be the same as it would be 3 feet from this talker in an open field.

John: If you repeat the calculation for the back of the church, you'll see 5 dB less gain. That's another reason for the high center location.

Bob: There is another reason in the church which does not have a balcony. Usually, they have some kind of rear wall which is a good acoustical mirror. If the speaker is located about half way up the front wall, as is usual when twin P.A. columns are used, the sound will bounce off that acoustical mirror and land on the front pews. Crazy as it may seem, auditory backward inhibition will ruin the sound in the front pews and not in the back. Anyway, 5 to 10 dB of acoustic gain with 4 mikes open should be enough.

John: But, if you repeat that calculation for the old system, you find the nearest speaker is 30 feet from the mikes. That gives about another 4 dB of gain. My new system would not sound as loud—especially in the back of the church—as the old electroacoustic disaster. It's not reasonable but everyone expects P.A. systems to be loud and very few understand that loud doesn't automatically mean intelligible. To get at least 10 dB more gain and to get better uniformity of direct sound front-to-back, I intend to use an electrically-tapered linear-array speaker.

Bob: Isn't that just a bunch of two-bit words for a P.A. column? Those usually sound like junk.

John: Many are built to a price tag and to look like what the marketing department thinks will sell because Brand X makes one just like that. They are not tapered and the low-compliance drivers cut off below 300 Hz. The male voice needs flat response down to 150 Hz and the usual columns or horns do not get there. Without tapering, the directional pattern is OK at 400 Hz but really gets to beaming by 1000 Hz. I don't know of a manufactured unit which meets this requirement so I'll have to build my own. To get at least a 3.5 dB directivity index, we need a length equal to a wavelength at 150 Hz. That means my column will be about 2 meters long.

Bob: But at 300 Hz, the directivity index will go up to 6.2 dB and we see a lot of beaming.

John: I can put eight 4½-inch speakers in my 2-meter column with some electrical filters and get excellent uniformity of frequency response and directional pattern from 200 to 1200 Hz. That's what is really needed for natural voice response.

Bob: Will you put some speakers with time delays under the balcony?

John: No. The sound will refract around the balcony edge much better than the tinkerers believe. The sound will be adequately loud back there. The usual complaints about hearing under a balcony are not caused by loudness but by ABI. If I put speakers with time delay back there, I would have to have exceptional directivity in those speakers and do a deadening number on the back wall to keep them from causing ABI in the front of the church. Putting speakers under balconies is witchcraft instead of engineering.

Bob: Will that old Western Electric amplifier have enough power?

John: Yes—it has push-pull parallel 6L6s and will dump 50 watts into a soldering iron all day long. Even with the relatively low efficiency of about 2 percent that I expect from my array, a 12-watt amp will put over 90 dB SPL on any seat in the church. If I were specifying a new amp, I'd go for 100 watts for emergency announcement power since it wouldn't run up the installed price very much.

A couple of months later, we again catch up with our friends.

Bob: How did the new system work in the Church of Christ?

John: It took me a couple of weeks longer to build the speaker than I planned. I still pulled it off so that one Sunday it was the old system without the hum and the next Sunday it was the completely new system. A civil engineer helped me hang the speaker. He taped a mirror on the front of the array and then looked at it through his transit from the rear of the church, just as far under that balcony as he could still see the speaker. When we got the angle of dangle right, he could see his transit scope in the mirror. That impressed the pastor more than anything else. I was able to set it up with two open mikes and 5 dB below howlback. We played a voice tape through it while we walked around the church and the sound uniformity was about 3 dB on a sound level meter and no one walking around could really hear a difference in level. You wouldn't believe the system was turned on until we switched the mike off while the preacher was reading.

Bob: What did the preacher say?

John: He sensed a better attention span from the congregation. Singing was better too. I asked him if he felt the kids were quieter and he said yes and also offered an explanation. The old P.A. system sounded loud and squawky like a TV set at home and thus the kids felt they could talk as they do during commercials. With the clear and not overly-amplified sound, they sensed something different and paid more attention.

Bob: That could be the explanation. I've noticed that kids are quieter in churches without P.A. systems. I guess the same explanation would hold.

For our final conversation on church P.A. systems, we find Bob and Paul walking from a large Catholic cathedral to a convention hotel one Sunday morning.

Bob: Did you notice how slowly and deliberately that priest talked over the P.A. system?

Paul: Yes he has been coached to talk that way. A church with a ten-second reverb time is really a challenge for an acoustical engineer.

Bob: Reverberation time is more of a buzz word than a useful statistic. The church wouldn't sound nearly that hollow if it did not have a dozen P.A. columns scattered about. The theory of that installation is that the multitutde of speakers feed the direct sound field. Each listener is imagined to hear just the P.A. column nearest him. That is pure bullfeathers. Everyone can hear all the speakers and several reflections from each speaker. A person talking loudly wouldn't sound nearly as hollow as that P.A. system. The church has good diffusion—remember the pipe organ—and is large enough that returned echoes are way down in amplitude even if they are in the 150 millisecond forbidden delay range.

Paul: If that cathedral were filled with people, wouldn't a sound reinforcement system be needed because of the absorption and the city traffic noises?

Bob: I suppose the calculations would support your statement. The trouble is, they have a P.A. system, not a sound reinforcement system.

Paul: How would you fix that cathedral?

Bob: Tear out those P.A. columns and use them for Bingo prizes and sell all those cardioid mikes to a rock and roll band. If the priest would mount the stairs to that cupola and preach from there, his sermons would be much better understood. I remember a cathedral half this size. It was before P.A. system junk got inexpensive enough for churches to try using it. I could hear the Bishop and understand all of his English words.

Paul: Wasn't that back in the days of Latin?

Bob: Yes. Maybe P.A. systems are the way the clergy keeps the laity from understanding any more than we understood in those days.

Paul: Didn't you have a prayer book with translation of the Latin?

Bob: Yes. Since the brain was getting the information more from the video channel, we probably understood more.

Paul: We've both lived long enough to know that the Bishop will not even try to get by without a P.A. system. I agree that the one we saw is no good. Assuming that we could convince the Bishop that he needs a better sound reinforcement system, how would you go about it?

Bob: I don't think I can make my usual prediction that a single properly hung speaker array would give intelligible sound. It may require some acoustical absorption before the single speaker will work. The first experiment would be to get a night club P.A. column used by vocalists. I would put the cantor alongside the organ console and move the speaker around in the choir loft. I think that would get the congregational singing going even though the musicians may not appreciate how well. I could play recorded speech through that speaker and walk around the empty cathedral to get a good idea about the room acoustics.

Paul: No wonder you can't sell any acoustical insulting, Bob, you have to put on a show with a real-time analyzer—not just walk around listening.

Bob: The real-time analyzer and pink noise tell about as much about acoustics as about speakers. Those who invented the theory of measurements using real-time analyzers don't know an auto-correlation function from a hole in their head. You can twist the knobs to get any kind of numbers you want—and they don't mean anything.

Paul: They are making more money than you are.

Bob: True—but are the customers getting any benefit from the money they spend?

Paul: I remember that Dave Klepper wrote a paper showing a large single cluster of horns in a large church. ("Sound Systems in Reverberant Rooms For Worship" Journal of the AES, 18:4, 1970.) Wouldn't that work here?

Bob: Probably. But there aren't many horns which get down to 200 Hz and hold their directivity pattern. Doing that requires a huge monster and everyone will squawk about how ugly it is.

Paul: That's certainly true. You can get away with hanging ugly light projectors anywhere because they won't show on the TV screen.

Bob: The linear-tapered-array speaker is much easier to make look acceptable. I'd like to spend a few days with a couple of helpers and a block-and-tackle. I think I could find a location somewhere above the sanctuary where a two-meter-long speaker would give intelligible sound reinforcement.

Paul: And it would block the view of something from somewhere—You'll have a real fight to get permission to leave the speaker where it is needed acoustically.

Bob: I know. I've never understood why priests and ministers will compromise intelligible sound for visual decor. I could try some kind of trick to accidentally-on-purpose not get it down on Saturday afternoon. Maybe the sight of attentive faces during a sermon would convince the Bishop.

Paul: Your problem is selling the first intelligible sound system—especially in a large church. Since the Bishops have never heard anything but honky P.A. systems, they have no reason to believe that an ugly looking speaker array could make every spoken word understood.

Bob: Amen! They would also be inclined to mistrust an honest engineer who wants to do some research in the cathedral before designing the permanent installation. After all, the people who put those P.A. sound columns all over that cathedral were good salesmen—they came on confident that the kilobucks spent would give good sound in the church. The Bishop must suspect that the sound is lousy. He's paid his money and without a decent comparison system, he doesn't realize how the money was wasted.

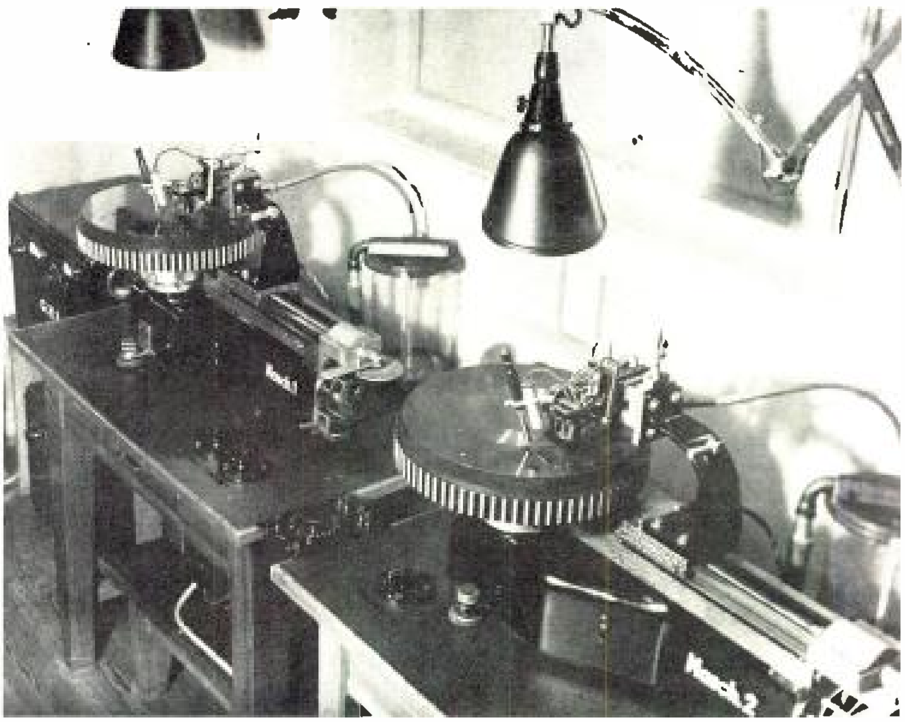
Paul: That's a sad but accurate condemnation of the pro sound people. Most of their church systems do more harm than good. I suppose there is not much for us old fogeys to do but keep giving free advice to turn off that church P.A. system.

Bob: And most of the preachers and Bishops will value that advice at cost. ■

PETER HAMMAR and DON OSOSKE

The Birth of the German Magnetophon Tape Recorder 1928-1945

The following article is based on research done in the past year in Gemany. Author Hammar talked to sources at BASF, Agfa, AEG-Telefunken, the German radio stations, the Deutsches Museum and various retired engineers.



Neumann wax disc recording lathes at Sender Hamburg in 1931. Recording lathes were the forerunners of the Magnetophon in German broadcasting. (Photo courtesy of the Norddeutscher Rundfunk Archives, Hamburg, Germany.)

THE DEVELOPMENT OF MAGNETIC recording was not exactly an overnight event. From its introduction in 1898 as the Telegraphone wire recorder to the controversy of today's digital technology, magnetic recording has gone through stormy times.

Peter Hammar is curator, Ampex Museum and Archives of Magnetic Recording. Don Ososke is with Ampex's Standard Tape Laboratory.

Valdemar Poulsen, the Edison of magnetic recording, invented almost every known form of magnetic storage. His first idea, in 1896, was a magnetic version of Edison's cylinder phonograph. Poulsen spiralled piano wire around a brass cylinder, with a laterally-moving magnetic pick-up head pushed along by the rotating cylinder. Playing time was thirty seconds. A year later the Dane had developed magnetic recorders that used spools pulling wire past the record head at two meters per second, with a recording time of several minutes. Poulsen also

October, 1934 Sender Hamburg remote broadcast recording on a Hamburg commuter train, using the Lorenz Steeltone machine. (Photo courtesy of the Norddeutscher Rundfunk Archives, Hamburg, Germany.)

made a machine that used a steel band to record sound. He even made a magnetic disc recorder whose pick-up head moved along a spiral guide, very much like the magnetic disc video slow-motion recorders developed by Ampex and others in the 1960s. And all this before 1900!

Unfortunately, marketing people regarded Poulsen's technical breakthroughs in magnetic recording as a curiosity, a toy. In 1905 the Danish engineer sold his Telegraphone patents to the highest bidder and went on to do research in other areas of electricity.

Lee DeForest, the inventor of the modern vacuum tube, wanted to perfect magnetic recording—many of DeForest's early Audion tube diagrams used a wire recorder as the theoretical sound source. However, DeForest's efforts were frustrated by lack of cooperation from Poulsen's successor, the American Telegraphone Company.

For shipboard radio recorders in the 1920s, U.S. Navy researchers Carlson and Carpenter improved the Telegraphone with vacuum tubes, and added something new to the record circuit—AC bias. But their Navy sponsors lost interest in communications recording and the two were forced to drop the project. Had the Navy had a bit more foresight (easy for us to say today), we might have had relatively high fidelity magnetic wire recording as early as 1923. The Navy's reaction reflected an attitude that continued from Poulsen's day: magnetic recording was more a curiosity than a practical tool.

The next attempt to commercialize magnetic recording was made almost a quarter-century after Poulsen, when Curt Stille in Germany formed the Telegraphone Patent Syndicate in 1927. Stille envisioned magnetic recorders for dictation, automatic telephone answering, and even music reproduction. None of the members of the syndicate were very successful in their attempt to commercialize magnetic recording, althoug the Lorenz Company in Berlin almost succeeded.

Around 1933, under the direction of Sem Begun (who later headed Brush Development in Cleveland), the Lorenz Company began work improving one of Curt Stille's ideas, using a steel band as the recording medium. Lorenz had enough faith in magnetic recording to design its "Steeltone Tape Machine" for use in radio stations as a transcription device. In fact, by the mid-1930s, several European radio services, including the Germans and the British, had used steel recording on the air. Steel-band recorders had reached a quality level almost equal to the broadcast wax disc.

During the world-wide depression of the 1930s, people relied increasingly on radio at home for entertainment. For broadcasters, the Thirties was a time of tremendous growth in entertainment programming. Most radio stations used recording lathes to cut lacquer or very thick wax discs for use in time-delayed broadcasts. However, the wax discs could only be played two or three times before the grooves were worn. Also, the radio engineer could not easily edit a program recorded on a disc. The necessary disc-to-disc transfers to edit out mistakes led to high generational loss of sound quality.

Naturally then, magnetic recording on a long, thin strip of material offered the broadcaster editing and multiple-replay capabilities that he did not have with discs. But the Lorenz Company's steel-band recorder was out of date before they could get their machine to the broadcast market. Steel as a recording medium was impractical at best. You edited with solder and a welding torch. A fifty-minute reel of steel tape measured over two feet in diameter, and weighed almost 40 pounds!

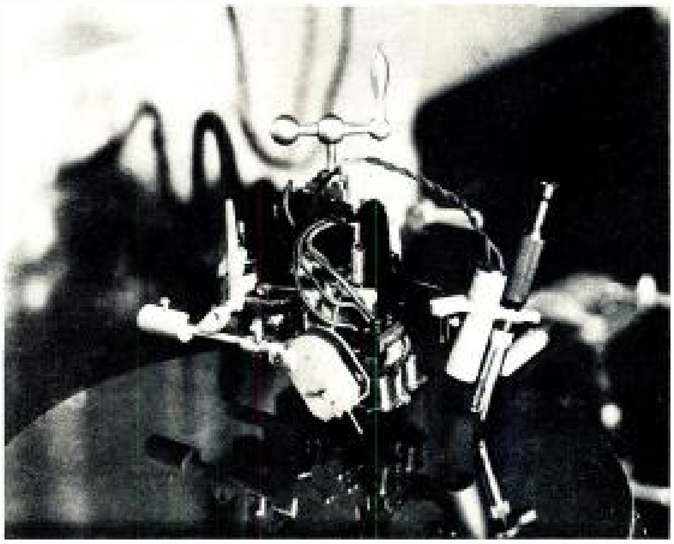
The machines could even be dangerous for their operators. The English version, the Blattnerphone, used at the British Broadcasting Corporation until as late as 1950, was operated in a metal cage so that if the steel band flew off its reel during fast-forward or rewind, the engineer on duty wouldn't lose a hand, or worse.

We have no figures on the cost of solid steel tape, but the expense was high enough to prompt the German radio service's chief engineer H. J. von Braunmühl to look for an alternative to steel. There had to be a better answer to magnetic recording than the steel band.

FROM STRAWS TO CIGARETTES TO MAGNETIC TAPE

In Dresden, Germany, the Universelle Company had been building cigarette manufacturing machines since the turn of the century. One of their engineering consultants in the 1920s was Fritz Pfleumer, whose previous discoveries included drinking straws made of plastic, as well as new forms of foam rubber.

One of the Universelle's machines was designed to make cigarettes with a thin band of real gold around the mouthpiece. Even for 1928, using gold on cigarette mouthpieces was becoming expensive, so the company put Pfleumer to work finding a substitute for the gold. Pfleumer developed a bronze powder that he mixed with lacquer, spread on a wide, long strip of paper, and then slit into tiny pieces for gluing onto the cigarettes.



Neumann disc cutting head, in use circa 1931 at Sender Hamburg. Both wax and lacquer discs were used for the broadcast-quality recordings. (Photo courtesy of Norddeutscher Runkfunk Archives, Hamburg, Germany.)

Pfleumer was somewhat of an audiophile. He liked good-quality radios and recording devices, and did much experimenting on his own. Of course, like most engineers, Pfleumer knew about the wire Telegraphone and the early experiments with steel-band recording.

Around 1928, Pfleumer was in Paris on a business trip. While sitting in a cafe, he was thinking about magnetic sound recordings. He reasoned that, instead of using expensive, heavy steel tape for recording, he could use his cigarette-mouthpiece-label technique to make cheap, lightweight magnetic tape. Instead of bronze powder, iron powder could be mixed with lacquer and spread on a paper tape.

Pfleumer's combined knowledge of paper tapes from his cigarette work, and his understanding of magnetism and electro-acoustics was crucial to his success in making the world's first magnetic tape recorder. He knew, for example, that the iron particles had to be as small as possible to achieve the highest possible frequency response. For Pfleumer, the all-important binder material to glue the particles to the tape was no problem at all. He just used the same lacquer he had used for the bronze on the cigarette mouthpieces.

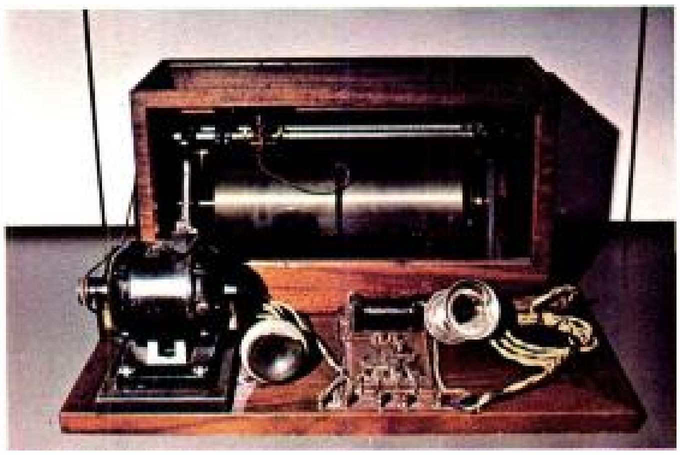
Pfleumer's first tape recorder, built in 1928-29, sounded just awful: distortion, background noise, wow, and flutter. But the point was, the thing *worked!* One did not need a solid piece of ferrous material to record sound magnetically. The engineer described his recording tape as "a 300-meter-long roll of the recording material which lasts twenty minutes and costs only one Mark 50 Pfennigs (about 25 cents) to make. The paper, called Pergamine, is only 0.04 mm thick." He pointed out that, with his new recording system, the tape editor could trade his welding torch for a pair of scissors.

Unhappily for Fritz Pfleumer, the German patent office in 1936 denied him his 1928 patent, finding the American J.A. O'Neill's 1927 magnetic tape patent valid. As far as we know, O'Neill never did make workable magnetic tape or a recording device of any kind.

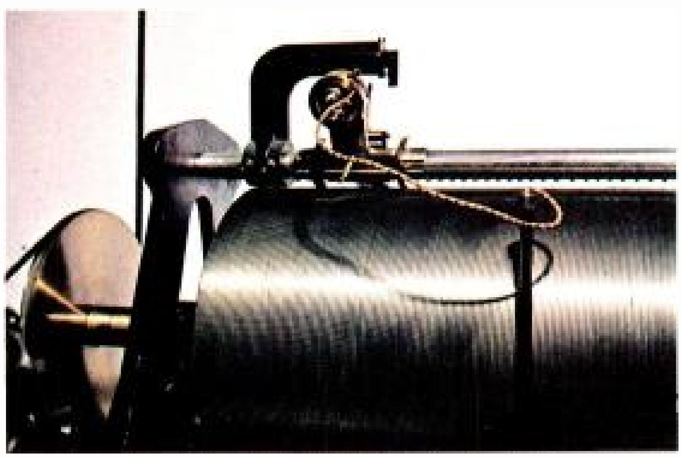
In 1929, Pfleumer took his invention from Dresden to Berlin, to sell it for development. Newspapers there ran stories about the new recorder, after several private demonstrations. AEG *Allgemeine Electricitaets Gesellschaft*, or "General Electric Company") in Berlin, today affiliated with Telefunken, was Germany's second-largest electronics company, after the Siemens Company. AEG designed and manufactured professional and consumer electronic products, much as its business associate General Electric did in the United States.

At AEG in 1930, Pfleumer's first demonstration of his tape recorder, which he called a "sound paper machine," was less than convincing. This magnetic recorder, like others before it, sounded poor. However, for the first time in history, engineers and managers were far-sighted enough to see the potential for tape recordings. By 1932, AEG had signed a contract with Pfleumer to buy his patent outright and develop tape recording.

The engineers at AEG tried to make their own tape at first, according to one account, buying carbonyl iron at the corner drugstore and spreading it on paper "ticker tape." The sound they got from the tape was terrible, and they soon realized that the problems of spreading thin coats of iron-filled lacquer onto strips of paper tape were best left to a chemical concern.



Poulsen Telegraphone, complete (1899). Poulsen intended this device—record/PB time = 30 seconds—to be an automatic telephone answering machine. (Photo courtesy of the Deutsches Museum, Munich.)



Detail of the Telegraphone record/playback head. (Photo courtesy of the Deutsches Museum, Munich.)

THE FIRST TAPES

In the early 1930s, the chief executive officer of AEG, Herman Buecher, heard about his engineers' problem. He called his old friend in Frankfurt, Carl Bosch, who was the head of the powerful IG Farben chemical combine, to see if the two companies could make the development of the magnetic tape recorder a joint venture. In 1932, AEG's Buecher and IG Farben's Bosch arranged for a member of the IG Farben group BASF, (*"Badische Anilin und Soda Fabrik"* or "Baden Anilin [dye] and Soda Factory"), in Ludwigshafen, to begin intensive research into the problems of making good magnetic tape for the new AEG machine.

The first BASF tapes made in 1934 for the Berlin Radio Show were made of pure, powdered carbonyl iron. The iron, which looks like black dust, was mixed with lacquer and spread onto a cellulose acetate film, which was then cut into five millimeter-wide strips (6.35 mm = ¼ inch) several hundred meters long. BASF's first tape had no trade name, and was simply called "IG Farben carbonyl tape."

By 1935, the researchers at BASF had progressed from carbonyl iron to iron oxide with smaller magnetic particles that resulted in better electrical performance. Today's iron oxide tapes are essentially refinements of these early BASF formulations.

At the start of their joint venture with AEG, BASF switched from paper to a cellulose-acetate film. The early carbonyl iron tape was brittle, but much stronger than the first paper tapes. BASF's trade name for their acetate basefilm was "Cellite," so they called their new iron-oxide formula tape "Type C." Manufactured through 1942, Type C tape had a rust-colored oxide with a gray backing.

By 1943, BASF had introduced a third kind of tape, Luvitherm or "Type L," a homogeneous tape basefilm of polyvinyl chloride. Though much stronger than the Type C acetate tape, the PVC did stretch. Type L tape was made by dumping the iron oxide into the PVC vat, and then extruding the mixture into a solid film. Because the iron oxide was mixed throughout the tape, Type L could be recorded on either side. Another IG Farben member, Agfa at Wolfen, later joined BASF in the production of recording tape. In 1944-45, American GIs invading Northern Europe found a lot of Type L tape. Both BASF and Agfa were able to steadily increase their tape production until war's end in May of 1945.

The origin of today's one-quarter inch tape width standard came from a combinatin of good engineering and coincidence. In 1935, just before the introduction of the first AEG/BASF recorder and tape, the companies jointly decided to widen the tape from its original 5.0 mm to 6.5 mm (just a hair over one-quarter inch). The engineers chose the wider tape for greater strength and better electrical performance. We still do not know why they chose the number 6.5 mm.

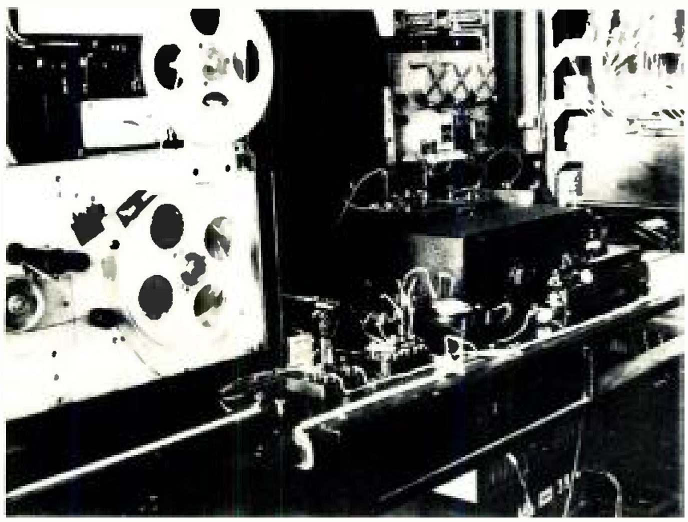
When the Allied engineers examined the captured Magnetophons and their BASF/Agfa tapes, they measured the 6.5 mm width as a quarter inch, plus or minus "a tiny bit." The 0.15 mm difference between a quarter inch and 6.5 mm was really not worth noticing. With the interruption of German tape manufacturing at the end of the war and the importation of American 3M (Scotch), Orradio (Irish), Audio Devices and other tape, the official width of magnetic tape there became 6.35 mm as well.

THE MAGNETOPHON

Our thirty inches-per-second base tape speed also originated in Germany with the Magnetophon. Until 1935, the AEG/BASF R & D team used one meter-per-second as their nominal standard tape speed. However, slight variations from machine to machine in motor performance and capstan diameter made interchangeability of tapes impossible. In 1935, the selection of a newly-designed asynchronous motor for the capstan drive solved this problem. In an effort to simplify future production of Magnetophons and set a world-wide standard, the engineers specified a capstan diameter of 10 mm, ±0. A ten-millimeter capstan with AEG's asynchronous motor and the BASF tape produced a tape speed of 76.8 centimeters-per-second. If the production of the Magnetophons could be standardized, an odd tape speed really would not matter.

When Major Jack Mullin, one of America's tape pioneers, and his U.S. Army Signal Corps engineers measured the Magnetophon's tape speed, they were surprised to measure almost exactly 30 inches-per-second (76.2 cm/s). Mullin's captured Magnetophons inspired the creation of the Ampex Model 200, America's first commercially-successful professional recorder. In 1947, Harold Lindsay, the Model 200's chief designer, used Mullin's 30 ips figure in the American machine's design, which later became the U.S. standard. Mullin had lent Lindsay some of his precious pre-recorded Magnetophon tapes for test purposes, thus the logical choice of a 30 ips tape speed for the American machine.

With the postwar dismantling of the Magnetophon factories, American machines dominated the European recording market in the early 1950s. The Germans adopted the U.S. figure of 30 ips, converting the number back to the metric 76.2 cm/s. No one ever seemed to notice the difference.



The first laboratory prototype of the AEG "Ferrotone" tape recorder in the fall of 1933. The one-motor machine used 5mm-wide carbonyl iron paper tape that was pulled past the heads at one-meter-per-second. (Photo courtesy of the AEG-Telefunken Archives, Braunschweig, Germany.)

From the start of the Magnetophon project, AEG faced the difficulty of making good heads. Both Pfleumer's and AEG's prototypes used record/reproduce heads similar to those originally developed by Poulsen and found on wire and steel band recorders: pole pieces with sharpened points pushed by springs into the surface of the recording medium. Naturally, the pointed head pieces quickly ripped the thin paper apart. Even the later acetate and PVC-backed tapes could not stand more than a pass or two from the points.

AEG's early experiments with the old-style pole-piece heads showed that, in addition to tape destruction, these heads had electrical disadvantages. The magnetic lines of force from the pointed heads with their separate pole pieces were both horizontal (parallel to the axis of the tape travelling past it) *and* diagonal. The lines of flux which intersected the tape were unfocused and mostly unusable, even interacting with each other to create distortion.

View of head assembly and tape path of AEG Magnetophon K-2 (1936), the portable version of the FT-2 shown on the March db cover. (Photo courtesy of AEG—Telefunken and Ampex.)

THE RING HEAD

The solution was an invention by Eduard Schueller: the enclosed ring head. Schueller had worked as a research assistant at the Heinrich Hertz Institute, a technical "think tank" in Berlin, and by 1932 was already experimenting with ideas of magnetic recording. Schueller found that the most important part of successful magnetic recording was the head. He decided to improve on the open pole-piece head design. The result was his experimental ring head. Naturally, as soon as AEG heard about Schueller's work, they offered him a key position on their tape recorder development team.

Schueller's ring head was not only very easy on the early tapes, but also created the nearly ideal magnetic flux pattern necessary for better fidelity recording. The lines of flux were concentrated in their most useful direction, horizontally (in the direction of the tape).

Thanks to the AEG-Telefunken Archives, BASF, the German Radio Archives in Frankfurt, and Hans Westphal of Berlin, we have copies of the earliest recordings made on the AEG prototype recorder in 1933. On the first recordings, the frequency response limit was not more than three or four kHz, harmonic distortion was about ten percent, and the signal-to-noise ratio was quite poor. By 1935, with the introduction of AEG's first production machine, the "Magnetophon K-1," fidelity had been increased, with top frequencies ranging above 5 kHz and with less distortion.

Spinning head (4 gaps, 90 degrees to tape path) of Tonschreiber "Berta" made by AEG, Berlin, circa 1939. (Photo courtesy of AEG—Telefunken and Ampex.)

Although AEG initiated the development of the modern tape recorder, it was BASF who gave the machine its name. The engineers at AEG in 1932-33 dubbed their new machine "Ferroton." At BASF, they were calling their tape "Magnetophonband" or magnetic phonograph tape. The name stuck, and in 1935, AEG started calling the machine "Magnetophon."

By 1935, the Germans had three of the four necessary ingredients of modern tape recording: 1) a stable transport, which the steel band recorders such as Lorenz had; 2) good tape, which the researchers at BASF had created; and, 3) the ring head from AEG's Schueller, with its good magnetic properties and gentle treatment of fragile tape. The fourth element of magnetic hi-fi recording, good electronics, would have to wait until 1939-40, after the Second World War had started.

From Valdemar Poulsen at the turn of the century until the late 1930s, direct-current biasing was the only method known to European engineers to reduce noise and distortion and increase frequency response. As late as 1939, the DC-bias Magnetophon sounded no better than an average 78 rpm transcription disc.

Until 1945, most engineers around the world had not heard of the German tape recorder. It was the combination of DC bias and World War II that kept the Magnetophon in obscurity. Jack Mullin has said that, "Once you hear DC-bias recording, you'll never want to hear tape agian!" Sir Thomas Beecham, having heard his London Philharmonic on tape in November of 1936, reportedly was so horrified by what he heard that he didn't use tape again until 1950.

In 1936, AEG sales people took their new Magnetophon to America for a secret demonstration at General Electric in Schenectady, New York. The DC-bias unit sounded so bad to the Americans that they decided that magnetic recording, at least in that form, was not practical.

The most promising market for the then-unperfected magnetic recording machine in Germany in the 1930s was the Berlin-based German radio monopoly, known as the RRG (*Reichs Rundfunk Gessellschaft* or Empire Radio Company). The chief of the RRG engineering section, H. J. von Braunmuhl, was against using magnetic recording for broadcasting. He liked the tried-and-true wax disc recording lathes with their Neumann heads. However, the progress of the AEG and BASF engineers interested him.

Von Braunmühl bought several DC-bias Magnetophons and put his best engineer, Walter Weber, to work to see if the machines really could be improved enough to be used on the air. Meanwhile, the people at AEG were also hard at work trying to perfect magnetic recording.

Weber at RRG had an idea of how to improve the signal-to-noise ratio. He cancelled some of the noise by adding an inverting bridge circuit with a "dummy" record head to the record amplifier circuit. The resulting 180-degree phase shift reduced tape noise about three dB.

AC RECORD-BIAS

On day in 1939, Weber was experimenting with this circuitry, making recordings of music and speech as well as pure tones. Weber kept logs of which recorder he had used, time of day, and what he had recorded. Later, while playing back one of the tapes, Weber found that the sound was fantastic! He was hearing true high fidelity on tape: extended frequency response, low noise, and low distortion. He traced the recording back to a Magnetophon that used his new noise reduction circuit, checked that circuit, and found that it was in constant oscillation, dumping high-frequency feedback into the record circuit. Weber realized that AC record-bias was *the* answer to hi-fi tape recording. He spent the rest of 1939 and much of 1940 perfecting his AC-bias discovery.

After the boss of the AEG Magnetophon lab across town heard the results of Weber's breakthrough at RRG, he went to his own researchers and said, "What in the world have you guys been doing here, sleeping? Over at RRG, they've just discovered AC bias and turned *our* machine into a high-fidelity recorder. We've got to get on the ball here!"

In fact, AC biasing of the record circuit was nothing new. But times were different, and engineers often missed each other's progress. Back in 1927, the U.S. Navy engineers Carlson and Carpenter, using a Telegraphone, had noticed the improvement of AC bias on wire recordings of telegraph messages. About the same time that Weber discovered AC biasing for tape recorders, Marvin Camras of the Armour Research Institute in Chicago had a similar discovery for use with his improved wire recorders.

The disc transcription room, Sender Hamburg, circa 1935. (Photo courtesy of the Norddeutscher Rundfunk Archives, Hamburg, Germany.)

With the war already in progress in Europe by 1940, it wasn't too surprising that Weber and Camras had not heard of each other's discoveries. In the late 1930s, the Japanese, under Kento Nagai, also discovered the AC bias phenomenon on solid magnetic material.

After the war, the Allied Commissions in Germany and Japan declared all international patents of the Axis powers invalid. That left the quite advanced Armour patent as the finisher in the post-war AC bias license field.

For AEG, the beauty of Weber's discovery was that they could take their existing DC bias design and simply add the relatively simple AC bias circuit, while changing the record head only slightly. The playback of the DC bias Magnetophon was quite good, although its full potential was never realized before AC bias recording. The last production DC-bias

Magnetophon had a specified frequency response of 50 Hz-6 kHz, a dynamic range of 40 dB, and harmonic distortion of 5 percent. The first AC-bias Magnetophon was rated at 40 Hz-15 kHz, with a 65 dB dynamic range, and under 3 percent distortion.

Most of the studio Magnetophons in use at the end of World War II were designed as early as 1938. The first production Magnetophon, the portable K-1, appeared in 1935. ("K" stands for the German word *Koffer* or "portable case.") The machine came in three cases, one for the transport, another for the electronics, and a third holding the loudspeaker. At the same time, AEG produced the cabinet "FT" series Magnetophon *Ferngesteuertes Truhe*, or "remote control cabinet"). The K-2 and FT-2 were introduced in 1936. The only FT-2 in existence that we know of is now a part of the Ampex Museum of Magnetic Recording in Redwood City, California, scheduled to open in the spring of this year.

The K-3 and FT-3 in 1937 were followed by the final Magnetophon in the pre-1945 series, the K-4, in 1938. The K-4 is the best-known pre-1945 Magnetophon. This is the machine that Jack Mullin and his partner, San Francisco filmmaker William Palmer, used to introduce America to the new technology of hi-fi tape recording.

The 1938 K-4 had DC biasing, and after the introduction of AC bias in 1941, a few early K-4s were updated. AEG also made an agreement with the RRG radio people to deliver K-4 decks built to RRG specifications incorporating the AC bias design. The radio station console machines that Jack Mullin first saw at the Radio Frankfurt substation at Bad Nauheim in July of '45 were special K-4 HTS *(Hochfrequenz Truhe Speziell*, or AC-bias cabinet special models.

When the war started, everyone in Germany was ordered to switch over to building military products. That was as true for tape recorders as for coat buttons. AEG produced a very rugged, portable DC-bias version of the Magnetophon that they called the *Tonschreiber* or "sound writer." The best-known of the Tonschriebers was the Type B, or *Berta* machine, which appeared in 1939-40. Berta was unusual because the machine had an extra, spinning head which could be used to compress or expand sound for high-speed transmission of information.

An amazing fact of World War II was that no one on the Allied side seemed to have heard about the hi-fi Magnetophons until the end of the war. This ignorance is even stranger when you consider that popular German magazines and newspapers, publicly sold in neutral Switzerland, printed numerous feature articles about German radio stations. Had the Germans classified all information about the AC-bias Magnetophons as

Telefunken "Reisz" microphone, circa 1930, carbon-type, in solid marble housing, 50-6 kHz. (Photo coutesy of AEG-Telefunken Archives and Ampex Museum.)

"top secret," the Allies probably would have known about the machines before the end of 1940! As it was, they had to wait another five years.

During World War II, the Allies were sometimes confused about Hitler's location. Live-quality broadcasts of his speeches simultaneously came from all parts of Germany. The Allies suspected some sort of high-fidelity recording device, but they overlooked the fact that the Germans had an extremely advanced radio network. A complex web of high quality land lines (10 kHz bandwidth, 600 ohm balanced line, less than 1 dB loss per 1000 km) allowed remote broadcasts from any location to any other location. In addition, time delay broadcasts from magnetic recorders had been standard broadcast procedure in Germany since the mid-1930s. To this day, old RRG engineers are amazed and baffled to hear that Americans thought that the Magnetophons were being used to deliberately confuse the Allies as to the location of high Nazi officials.

In England between 1942 and 1944, Major Jack Mullin and others had been hearing late-night German broadcasts of live-quality orchestral music. Mullin thought that even a madman like Hitler could not compel tempermental musicians to play at three a.m. However, the audio quality of the transmissions was much better than any recording device Mullin knew. What he heard was the routine use of the Magnetophon, which had been developed as a professional and consumer entertainment device.

An AEG Magnetophon K-4, circa 1938. The DC bias version had a frequency response of 50-6 kHz, 5 percent distortion and a dynamic range of 40 dB. Once modified with Walter Weber's AC record bias, the frequency response improved to 40-15 kHz, with a 65 dB dynamic range and less than 3 percent distortion. (Photo courtesy of the AEG-Telefunken Archives, Braunschweig, Germany.)

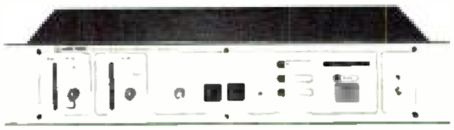
The Magnetophon tape recorder naturally got sucked up into the German war effort. The chief of the AEG Magnetophon lab during the war, Dr. Hans Schiesser, said that he had received specific orders from the Nazi government to work exclusively on the DC-bias military Tonschreibers for use by the army, air force, and navy, and to ignore civilian tape recorder development. However, Schiesser kept a secret set of lab notes which he still has, in which he wrote of his work on high-fidelity magnetic recording. Schiesser's work included the development of stereo record and playback heads, which he quietly did on the side, at some personal risk. For Hans Schiesser and many others at AEG and the RRG, the Magnetophon tape recorder was the exciting way into the future of high fidelity reproduction of sound. ■

The Ampex Museum of Magnetic Recording opens May, 1982. Peter Hammar welcomes participation in the Museum Project. If you have old equipment, information, or are just interested, contact him at the Museum of Magnetic Recording, c/o Ampex Corp. (MS 1-14), 401 Broadway, Redwood City, CA 94063; telephone: (415) 367-3127. Visiting the museum is by prior arrangement. Please contact the Ampex public relations department at the above-mentioned address, or call (415) 367-4151.

db New Products & Services

DIGITAL STEREO PROFANITY DELAY

• The DN772 Stereo Profanity Delay is designed as an obscenity filter for broadcast stations during live formats. The DN772 provides a delay time of 7.15 seconds, increasing to this time by small increments undetectable by the audience. Both audio channels track each other exactly in all modes of operation by using a single control system. The DN772 has a bandwidth extended to 15 kHz and a dynamic range of 80 dB. The unit can also be used as a production delay unit to lengthen pre-recorded programs without using any pitch change. The DN772 is housed in a 3½-in. rackmount unit, and has full remote facilities for convenience.
Mfr: Klark-Teknik Electronics, Inc.
Circle xx on Reader Service Card



FLOOR CONSOLE

• The CS-500 floor console has standard 19-in. rack mount frames which occupy 12 EIA Standard Units (21 inches) and are pre-drilled and tapped 10-32 on EIA centers, so that any cassette deck, open reel deck, amplifier, or other component may be mounted. The console is made of metal and features heavy duty casters for easy mobility.
Mfr: TEAC Corp.
Price: $175.00
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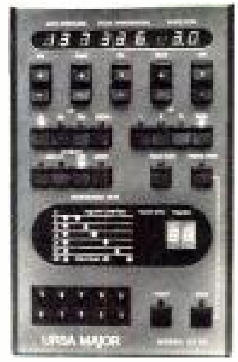
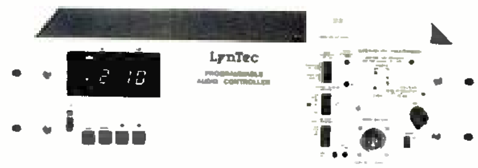
DIGITAL REVERBERATOR REMOTE UNIT



• Ursa Major has announced the production of a remote unit for its 8X32 Digital Reverberator. The remote provides all the controls and displays of the mainframe front panel in a box that measures 5-in. x 9-in. x 1.5-in. The 8X32 now also features capability for computer automated mixdown interface. Early Reflections, Initial Reverberation, Decay Time, and Low and High Frequency Decay are pushbutton adjustable within each of four basic Programs (Plate I, Plate II, Hall, and Space). Non-volatile registers can store 64 complete patches as user-derived programs for immediate recall. LED displays show all settings in use and a readout of the input and reverberation levels. The system will synthesize reverberation over a wide range of natural and unnatural spaces, including decay times up to 20.0 seconds.
Mfr: Ursa Major, Inc.
Circle xx on Reader Service Card

PROGRAMMABLE AUDIO CONTROLLER

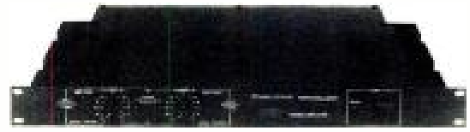


• The LynTec Programmable Audio Controller provides timed level control of up to 7 channels of audio. The output level may be user programmed to rise and fall at specific starting times. The rate of rise and fall are also programmable. All programming controls are on the front panel. The Standby Battery Cover conceals the supervisory programming controls to discourage tampering. The internal quartz clock has standby battery backup to assure timekeeping integrity when the AC power fails. A pulse load battery tester checks the battery once a day to verify battery condition and alarms at 75 percent battery discharge. An optional dual pink/white noise source board provides 2 channels of 0 dBm RMS level noise output for programmed masking noise use.
Mfr: Technical Visuals
Circle 43 on Reader Service Card

VOCAL COLUMN

• The Cerwin-Vega Model V-23 is a compact, 3-way vocal column designed for portable applications. The unit has a wideband, low-distortion sound with good pattern control. The V-23 includes a single 12-in. ER122 driver. This unit uses a copper 2-in. voice coil on an aluminum Quintex (R) former for reliability at input powers in excess of 125 watts EIA. The entire assembly is mounted in a rigid and accurate high-pressure diecast aluminum frame. The bass driver operates in a sealed enclosure with the response characteristic of an optimally damped 2nd-order Butterworth alignment. The bass response has been specifically tailored for accurate vocal reproduction and is essentially flat with a smooth and gradual roll-off below 100 Hz (12 dB per octave). The bass speaker is allowed to roll off beyond 100 Hz to reduce intermodulation distortion and a first-order high pass filter provides transition to the MF-81 8-in. midrange cone transducer. The MF-81 operates in its own sealed damped enclosure and has a 1½-in. copper wire voice coil on an aluminum former for high power handling. At 4 kHz, a 2-pole high-pass crossover feeds the 120 dB SPL H-25 horn tweeter. The relatively high crossover of the horn, nearly an octave above cutoff, increases power handling and lowers distortion. The driver is mounted on a square mouth 60 degree horn which maintains uniform coverage angles over the bandpass.
Mfr: Cerwin-Vega
Price: $325.00
Circle 44 on Reader Service Card

POWER AMPLIFIER



• The Model A15 is a single rack space Professional Series amplifier rated at 65 watts RMS per channel into 4 ohms. The A15 Power Amplifier is equipped with two separate precision variable slope limiters for 15 dB of overload protection beyond the rated power output level, equivalent to over 2,000 watts per channel of clipping headroom. As a result, the A15 can be utilized to power a compression driver in a bi-amped system with no danger of clipping or voice coil damage. Functional features include LED Fault, Signal Present and Thermal Indicators; balance/unbalanced inputs, and automatic mono input. Total harmonic distortion is less than .05 percent from 20 Hz to 20 kHz.
Mfr: Phase Linear
Circle 45 on Reader Service Card

REVERB PROCESSOR

• Designed to enhance any reverberation system, Studio Technologies' reverb processor contains analog time delay, 3-band parametric equalizers with continuously variable Q, two noise gates and the stereo stretcher. The processor is configured so that the variable time delay and one parametric equalizer are in the send circuit to the reverb device. Each of the stereo returns from the reverb is routed through a parametric equalizer, the stereo stretcher, and finally a noise gate which follows the reverb decay down to a −60 dBm level then kills to a −90 dBm level. A noise gate threshold control allows the user to set his/her reverb time while the noise gate time will shorten the "tail out" of the reverb to a more normal 3½ seconds. This effect can be used to fatten certain sounds such as vocals and horns. The stereo stretcher widens the stereo spread of the reverberation creating a wall of sound stretching beyond the speaker boundaries. This effect is made frequency selective by using the two parametric equalizers in the circuit.
Mfr: Studio Technologies
Price: $1500.00
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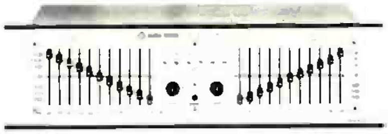
It will discourse most eloquent music.

(William Shakespeare 1564-1616)

Our considerable experience in the field of equalisation coupled with a philosophy of continual research and development has enabled the realisation of a range of high quality Graphic Equalisers which have become standard tools for correcting room acoustics and offer the solution to tricky equalisation problems.

The latest Klark-Teknik equalisers incorporate scale switching which facilitates a fine resolution for small adjustments and yet retains full boost and cut when required. Construction is to the highest standard using selected components and all units are rigorously bench-tested and aligned before a burn-in period and final music test.

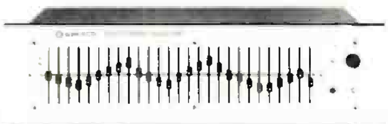
DN22 GRAPHIC EQUALISER



The DN22 is a dual-channel Graphic Equaliser, each channel having 11 filters providing up to 12dB boost or cut at 11 centre frequencies, covering the entire audio spectrum. Separate low and high pass filters are provided on each channel giving 12dB per octave attenuation above and below their respective turnover frequencies.

The DN22 offers an extremely wide dynamic range and negligible channel-to-channel crosstalk.

DN27A GRAPHIC EQUALISER



The DN27A is the successor to the widely acclaimed DN27. It is a ⅓rd Octave Graphic Equaliser, providing boost or cut of up to 12dB at 27 I.S.O. centre frequencies covering the entire audio spectrum.

The equaliser filters are of computer-aided design and consist of actively-coupled L.C. networks of the 'minimum phase' type. The inductors have precision-ground ferrite cores and coils wound to extremely tight tolerances.

DN30/30 GRAPHIC EQUALISER

The DN30/30 Stereo Graphic Equaliser represents a breakthrough in equaliser design, giving two channels of full ⅓rd octave equalisation in one compact unit. In addition to saving on rack space the DN30/30 also means a considerable financial saving for anyone requiring stereo system equalisation.

All-new circuitry developed specifically for the DN30/30 uses ultra-stable N.I.C. minimum phase, combining filter networks to give unequalled performance.

For further information on our complete range of professional audio equipment and application details telephone (516) 249-3660

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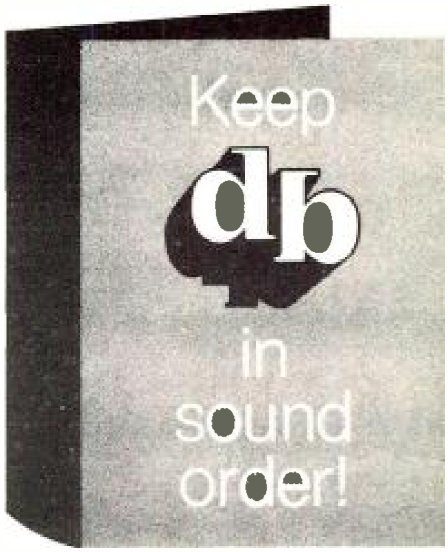

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People/Places/Happenings

• **Harrison Systems** announces the appointment of **Claude J. Hill, Jr.** to the position of vice president of sales and marketing. Mr. Hill comes to Harrison from **Audicon, Inc.**, a Nashville based import marketing and studio design and equipment supply firm where he served as president for two years. Prior to his being with Audicon, Claude was president for four years of **Audio Consultants, Inc.**, the southeastern **MCI** distributor. Mr. Hill has also held sales engineering positions with **Pandora Systems, Studio Supply Co., MCI** and **3M**. Completing the reorganized sales and marketing department are the appointments of **Eric Johnson** and **Brad Harrison** as sales representatives.

• **CAMEO** (Creative Audio & Music Electronics Organization) has announced the addition of five manufacturers to their membership. Joining the association are **Audio Technica U.S., BGW Systems, Fostex Corp. of America, RAMSA-Panasonic,** and **Shure Bros.**

• **Quad-Eight Electronics** announced the appointment of **John Robbins** as manager, Production and Product Support. Robbins will report to **Ned Padwa**, Quad-Eight's executive vice president, who noted that the appointment was made concurrently with the start of production of the company's new 248 Component Series audio console line. Robbins has an extensive background in professional and consumer audio manufacturing and in FM broadcasting. He has been production manager and general manager of the **Marantz Company** and most recently was national service manager for **JBL**. His new responsibilities will cover both standard and custom products at Quad-Eight.

• **Don Richter**, has joined **Broadcast Technology, Inc.** as sales and marketing manager. Broadcast Technology manufactures a variety of audio products for the broadcast industry. Recently, Broadcast Technology moved to larger quarters at 33 Comac Loop, Ronkonkoma, New York. The new and expanded facility is expected to enhance the design, engineering and custom service capabilities of the company, according to its president, **Lou Lindauer.**

• **Edcor**, Irvine, California, announces the appointment of **Ron Dumesnil** as manager of their customer service, order and inside sales departments. Mr. Dumesnil was previously head of customer service for **Altec Lansing.**

• **Sony Corporation of America** has acquired **MCI, Inc.** of Fort Lauderdale, Florida, a worldwide leader in the manufacture and sale of professional recording equipment. The announcement was made by **Kenji Tamiya**, president of Sony Corporation of America. The privately-owned Florida firm, founded in 1955, is the largest manufacturer of multi-track recorders and studio mixing consoles in the United States. Its facilities include a main manufacturing plant and a precision service plant in the Fort Lauderdale area with 440 employees. MCI will be an independent division within Sony Corporation of America. Mr. Tamiya stated that the daily operations of MCI will continue unchanged. MCI founder, **G. C. (Jeep) Harned** will remain as president and chief executive officer of the MCI Division, and **Michael Schulhof**, a director of Sony Corporation of America, has been appointed chairman. Commenting on the acquisition, Mr. Harned stated that the sale was made "to give added financial, technological and new product support to MCI, assuring the company's growth and continued dominance in the worldwide professional recording marketplace." Mr. Schulhof added that the acquisition "further strengthens Sony's capacity for the launch of the compact digital audio disc later this year enabling us to provide the full range of services in support of this dramatic step forward in audio technology." Officials of both companies noted that MCI is well positioned to develop the major new market in professional audio equipment that will open with the introduction of stereophonic AM and television broadcasting and the anticipated expanding consumer market for component television. The new operation will join an existing broadcast video technology from Sony. The company is a leader in videotape recorders for broadcast, industrial and personal use. The Sony Technology Center in Palo Alto, California is a manufacturer of broadcast VTR controls and a major research and development operation in the broadcasting field.

• The Broadcast and Professional Audio Group of **Telex Communications, Inc.** announced the appointment of **Jerry B. Wade** to the position of product manager. Although Wade's responsibilities cover the entire Telex Pro Audio line, he will place special emphasis on **Turner Microphones**, **Audiocom** Intercom Systems and the **Telex** Commercial Music line.

• **Bose Corporation** has announced the appointment of **John Stiernberg** as field sales representative for its Professional Products line in the Midwest territory. Stiernberg has been active full-time in the music and sound industry since 1974. Prior to joining Bose, he was a part owner of the **Morgan Brothers** Music stores in Ripon and Oshkosh, Wisconsin, where he developed and managed the sound reinforcement department. He also has lectured and performed extensively in the upper midwest as Professor Bluegrass and as a member of the Morgan Brothers Band. As a full-time Bose Professional Products representative, Stiernberg will be working exclusively with the Bose Pro sound contractors and retailers in an area that includes Illinois, Michigan, Indiana, the Dakotas, Iowa, Wisconsin, and Minnesota.

• Last spring, **Soundcraft Electronics U.S.A.** moved its headquarters from Kalamazoo, Michigan to a Los Angeles suburb. **Wayne Freeman**, Soundcraft sales manager, embarked on an entirely new marketing program for the company's line of mixing consoles and multitrack tape recorders, adding new reps and rebuilding the dealer organization. The combination has increased Soundcraft's U.S. sales by over 50 percent.

• **Curtis Chan** has been named national engineering manager for Sony's Professional Audio Division, announced **Nick Morris**, general manager of the division. Mr. Chan will oversee all engineering and service of professional audio and digital audio products, as well as conduct consultations in the field. A three year Sony veteran, Mr. Chan's most recent position was Western regional engineering manager for the division. Prior to joining Sony, he was active in developing advanced audio/video editing systems with the **Ampex Corporation.**

Portrait of a Quiet Family

Meet the members of our Studer console family. The compact, very portable 169. The expanded 269. And the full size 369 studio production console. A respected family with three common traits...

Exceptional Audio Quality
Studer consoles are designed for quiet operation, the utmost signal clarity, and precise tailoring of the sound. The inputs feature 3 section EQ, the overload margins are generous throughout, and each master module has a built-in limiter. As for S/N, well, it's no surprise that the 169 has been used for many digital classical recordings.

Total Flexibility
Pick a frame size and outfit it to suit your needs: portable remote, OB van, or studio production. Module selections include basic input (mic/line), stereo high level, master, monitor, reverb/foldback, and aux monitor. Optional linking kits provide extra I/O flexibility. On the 169/269, power supply options allow operation from mains cable, vehicle battery, or built-in NiCad batteries. All three consoles are available with either VU meters or, at no extra charge, one of two PPM types.

Long-Term Reliability
Studer consoles are built for the long run. They'll take years of hard knocks in remote or OB van use. The Studer "family tradition" of reliability is the result of thoughtful engineering and careful Swiss craftsmanship.

Introduce yourself to the Studer family of consoles. They're quietly setting the standards.

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