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THE SOUND ENGINEERING MAGAZINE

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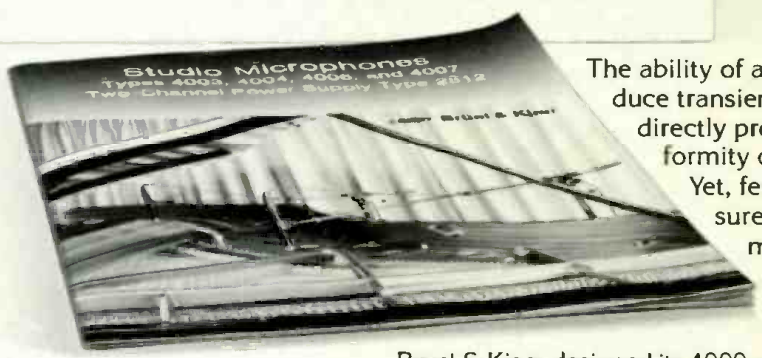
How important is microphone phase response?



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COLUMNS

DEPARTMENTS

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Letters

LOST AND FOUND DEPT.

TO THE EDITOR:
I have a CMD console. I am trying to find an operations/maintenance manual for the console and don't know who to contact. Can you give me the name and address of the manufacturer? I can find nothing on the outside or inside of the console giving me this information. We bought it used.

JOE COULTER

Sorry, Joe, we struck out on this one too. We checked all of our reference manuals and could find nothing on CMD. Perhaps you can give us more information; is there any other information on the console which might indicate a model number or anything else about the unit? Any idea how old it is? If anyone reading knows who makes the CMD console, please pass along the info to db *and we'll publish it.*

TO THE EDITOR:
In your May, 1984 issue you printed a report on a transducer from C-Tape Developments called the "C-ducer." I would like more information on this product and would appreciate it if you could provide an address for the manufacturer or U.S. distributor of this product.

FRED STANKE

Thanks for the inquiry. The address for C-Tape Developments in the U.S. is P.O. Box 1069, Palatine, IL 60078. There is also a toll free phone number at which the company can be reached: 1-800-562-5872.

Index of Advertisers

About The Cover

• This month's cover features Streeterville Studios in Chicago. The view pictured is from the entrance of the control room in the "Suite," which features a Harrison 4032B console, a Lexicon "Larc" (which controls the 224X digital reverb), an Otari MTR 10 4-track deck, an MCI 110 Layback machine (1-inch type C), Studer B-67 (mono and stereo) decks, MTR 10 Locator, Dual MCI Locators, Audio Kinetic Q-Lock 3.10, and various outboard gear.

BARRY BLESSER

Words

• Assuming that I have been successful in convincing you that the key to both understanding and communications is a clear sense of word meaning, I would like to present a glossary of words and definitions. If you think that these are obvious, try giving a colleague a test and compare his answers to those below.

1. BIT

The smallest unit of information in a digital system. A bit is determined as having one of two possible states to carry information. In digital hardware, a bit will be a specific wire, flip-flop, or time slot when time sharing. In communications, a bit is simply a measure of information; for example, that signal has an information rate of 20,000 bits/second. The signal may actually be an analog signal but its information can be measured in bits.

2. BYTE

A group of bits of some specific length forming a unit of information. Historically, the size needed to be specified. Present usage of byte is always referred to an 8-bit group.

3. WORD

Historically, this had been the largest group of bits to form a unit. It then came to mean a 16-bit unit because much of the hardware had 16-bit bus structures. The current usage requires a specification of the size. For example, we can say that the new microprocessors use 32-bit words for internal computation. This means that the largest unit that can be moved is 32 bits. Sometimes the size of a word is measured in terms of the number of bytes. We can say that the system has a 4-byte instruction word instead of saying it has 32 bits.

4. NIBBLE

This is a relatively new term which, like a byte, refers to a specific group of bits; however, it is used to mean 4 bits. It came from the fact that 8-bit microprocessors had instruction to manipulate the upper and lower 4 bits independently. For example, we can say the upper nibble is transferred to the I/O port.

5. INTEGER ARITHMETIC

This refers to the way in which the bits of a word can be used to represent numbers. An integer number means that the bits map directly to a sequence of discrete numbers. A 16-bit word can represent any integer in the range of +32,767 to -32,768. No number outside of that range can be represented, nor can fractions be represented. These cases do not exist in a 16-bit integer representation. Integer arithmetic can use different quantities of bits. With 8 bits, only numbers from 127 to -128 can be represented.

6. MAGNITUDE REPRESENTATION

A number system which does not contain any negative numbers is defined as a magnitude representation. With 16-bit integer magnitude representation, we can represent the numbers from 0 to 65,536. The example above (5) described numbers with both positive and negative signs. The same sequence of binary bits has a different value depending on the definition of representation. Without knowing the representation we can not know the "true" value of the number.

7. NEGATIVE NUMBERS

Negative numbers have had two different representation systems: 2's complement and 1's complement, although the latter is rarely used now. In a 2's complement number, negative numbers are created from positive numbers by complementing the positive number and adding 1 LSB (Least Significant Bit) to the result.

This has many nice properties (including the fact that the negative of 0 is itself 0 since the complement first produces all 1's and the adding of the 1 LSB results in 0 again.

A 2's complement system has only one artifact: there is one more negative number than positive number. This can be seen clearly since there is an even number of integers and one of these is used for 0. There is no positive of the most negative number.

A 1's complement system made negative numbers by the complement

of positive numbers. This posed the major problem of having two numbers for 0: 000000, and 1111111. The computer hardware was thus made more complex and this system was dropped. The advantage of the system was that negation did not require an addition of the 1 LSB. It also had the advantage of having the same number of positive and negative numbers. Nevertheless, this system is rarely used in new equipment.

8. DOUBLE PRECISION INTEGER

When the range of integers represented by one word is insufficient, an additional word is used. This greatly expands the range of numbers. For example, a 32-bit 2's complement integer word (double precision) has a maximum integer of 2,147,483,648. The term double precision is actually only used to mean twice the precision of a lower precision alternative. A 16-bit representation is double precision in a system which can also use 8-bit arithmetic. In systems which have 16- and 32-bit choices, the 16-bit is single precision and the 32-bit is double precision.

Sometimes other terms are used, such as long integers and short integers. Extended precision can sometimes mean the same thing as double precision, although the increase may not be a full doubling. For example, single precision might be 3 bytes and extended precision 5 bytes.

9. FLOATING POINT

In some scientific applications, the range of numbers is much larger than that which can be found in integers. A physics problem might use measures corresponding to the size of atoms or the size of the universe. Therefore we use a scientific type notation in which we have a number scaled by some exponent. This can be written .23343 E12, which means the actual number is $.23343 \times 10^{12}$. The resolution of this number system is determined by the fractional part. There might only be 16 bits in the fraction. This means that the fractional part is quantized and only the following are possible legal fractions: 0000305, .0000610, .0000916, .0001221....., .9999390, .9999695. The exponent of the power of 10 might have a limit from +127 to -128.

Some bits in the word are removed from the fractional part to create a wider range of numbers. The smallest typical floating point number would use 8 bits for the exponent and 24 for the fractional part. Most systems use a full 6 bytes for a floating point number.

The term floating point comes from the fact that the decimal point floats and is not fixed. The exponent effectively moves the location of the decimal point. The two numbers, .23343 E2 and .23343 E3, differ by the location of the decimal point. One number is 23.343 and the other is 233.43.

In the previous discussion we implied that integer numbers have no fractional part. We may define a fixed scale factor to allow for fractions. When we do that we have created a fixed point number. With 16 bits, we can place the binary point (used to separate the integer from the fractional part, as is the decimal point) between the upper and lower bytes. This would create a range of numbers which were the following: .0, .0030, .0078....., .99609, 1.00000, 1.0030, 1.0078....., 127.9922, 127.9961.

10. DOUBLE PRECISION FLOATING POINT

An alternative version of a floating point number with additional bits in the representation to allow for better resolution in the fractional part and wider range in the exponent.

11. BINARY NUMBERS

Numbers which are *written* or *stored* as a set of binary bits. The actual *value* of the binary number is defined by the representation system. If 1,101,111,110,001 is a binary number, we cannot know the value of the number without knowing the representation such as fixed point integer, 2's complement, floating point, etc.

12. OCTAL NUMBERS

These are numbers in which groups of 3 binary bits have been merged together to make the number more complex. It allows for writing a binary number more comfortably. The example above would then become 15,761 in octal format. This is transposed as follows:

binary:	1	101	111	110	001
octal:	1	5	7	6	1

Octal numbers are really binary numbers and vice versa. Each digit in the octal number can have one of 8 values, which we name as the numbers 0 through 7. Each column of digits has 8 times the value of the previous one.

One can learn to count, add, subtract, multiply, and divide in an octal numbering system.

13. HEX NUMBERS

Hex is short for hexadecimal which means that there are 16 numbers for each digit. Like octal it is a group of bits, except that the grouping is 4 rather than 3. Each digit can have values of 0 through 15. The numbers 10, 11, 12, 13, 14, and 15 create a notational problem because they require two digits in our normal notation of 10's. To make these numbers single entries, they are given the symbols A, B, C, D, E, and F, respectively. It takes some practice to think of a number written as 1BF1, but this is the same as the previous example of 15,761 in octal. Hex notation is comfortable when there are 8 bits (two hex digits), 16 bits (4 hex digits), or any number divisible by 4.

14. ASCII

ASCII is a standard international mapping between the letters and

number symbols of a 7-bit word. Each letter, number, and control symbol is given a unique value. When the key on a computer keyboard is pushed, that 7-bit word is generated. The actual word contains 8 bits; the extra bit is often used for parity checking (even or odd), or sometimes fixed as a leading 1 or 0.

ASCII is another representation system for binary numbers. This further illustrates the reason why we cannot tell the value of a binary number without knowing the system. The number could be the symbol for a letter such as Q. We say that the value or meaning is a letter rather than a quantity.

15. OVERFLOW

When the result of an arithemtic operation of two legal numbers gives a number too large to be represented, the result is called an overflow error. The term overflow is a particular kind of error which cannot be avoided. In an 8-bit integer system, there is no legal result when 127 is added to 2 because the number 129 does not exist.

Overflow can apply to both fixed and floating point numbers.

16. UNDERFLOW

This is the reverse case of overflow: the result is too small. It is generally only applied to floating point numbers which have special problems when there is not enough range in the exponent. The fractional part is limited to the range of 1 to 0.5. Therefore, the smallest number might be 0.5 E-99. An underflow would result when we tried to divide this number by 2. The result would be the same as the initial number since there is nothing smaller.

17. CLIPPING

What audio engineers call clipping, digital engineers call limiting. A hardware of software trap is added to the arithmetic process such as that the overflow error case is treated explicitly. The result is artificial, limited (clipped) to the most positive or negative value. We call this an "error handling" algorithm. The overflow is an error, but we define the way in which it will be handled. The previous example, for adding 127 to 2, gives the result 127 when we have clipping. Similarly, 120 plus 12 gives 127 and -110 plus -30 gives -128.

18. TRUNCATION

This is the name for the process by which certain bits in the result are thrown away. The resulting error is called a truncation error. It typically results from multiplication and division since these processes double the size of the result. A 16 bit number multiplied by a 16 bit number result in 31 bits. If the result must be again represented in 16 bits, we throw away the lower 15. The act of discarding means that the result contains an unknown error. It is analogous to quantization error in an A/D converter.

Addition and subtraction do not result in truncation issues because the result has the same number of bits. The exception is the overflow case. There the upper bit is lost because the number is too small. If the addition were to have a *defined* divide by 2, then there could never be an overflow, but there would be a truncation error since the lowest bit is removed. The addition of two 16-bit numbers can result in a 17-bit answer. The extra bit is usually an issue of overflow, but a shift in the binary point can turn it into a truncation issue. ■

JESSE KLAPHOLZ

Analog Vs. Digital

• Ah yes, yet another debate on the touchy issue. Most people assume that when the word computer is used, there is only one type—digital. Wrong, there are analog computers too, and we shouldn't so hastily overlook these devices. In this month's column we'll sit back in the midst of this "computer" age and take a look at our old friend—the analog computer.

To start off, let's backtrack a little to the first **db** Computer Audio column in April. We defined the analog computer as, "An automation computing device that operates in terms of continuous variation of some physical quantities, such as electrical voltages and currents, mechanical shaft rotations, or displacements, and which is used primarily to solve differential equations... Results are measured on meters, dials, oscilloscope recorders, or oscilloscopes." Not much imagination is necessary to see the applications.

People are graphically oriented; that is, we perceive physical relationships much better if we can see them. The dedicated-task slide rule is one such tool that enables us to see these relationships, not only efficiently, but often just as fast as you would have been able to find particular program on a disk. (...now which disk did I put that metric conversion program on?) In addition to these advantages, analog computers are fairly reliable devices that don't fall to the mercy of batteries, "fritzed-out" power supplies, data problems (lost disks maybe?), chips "gone south," etc.

When working over piles of blueprints, it's a lot easier to pull out that old slide rule than to traipse around through a stubborn computer's memory—all those darn protocols! Some people are simply getting hooked on those green monitors, and forgetting the old manual ways that can sometimes not only be quicker but also be beneficial to the project at hand.

USING THE ANALOG SLIDE RULE

Using analog tools often gives a designer a better "feel" for the areas in which they are applied. For example, in calculating the sound pressure levels at a given distance at

an outdoor venue, the designer sees all those in-between values and how they relate to the space very quickly with the use of an inverse-square-law slide rule. With the digital computer approach one would have to boot-up a program, input the venue's physical parameters, tell the computer to look up a given loudspeaker's data, tell the computer where to put the loudspeaker, and finally tell it where you are sitting.

Analog slide rules are available from any sources: component and equipment manufacturers, hardware companies, or stores that sell drafting supplies. Even the corner 5 & 10 usually has at least a metric conversion type on the shelf. These paper and plaster computers have been designed to relate almost any physical relationship, whether it be an electrical measurement in ohms, volts, watts, dBm, dBV, etc., or a physical measurement in units of inches, feet, meters, milliseconds, absorption, reverberation, dB SPL, etc.

As rapidly as the computer industry is growing, so, it seems, are articles such as this one. Some of the most prominent features in a lot of these articles are short "generic" programs, usually about 10 to 20 lines in BASIC, that are transportable among the more popular computers. These "generic" programs usually are written (because of their length) so as to accomplish single tasks, such as RT60 computation, arithmetic functions using the decibel notation system, computing inverse square law relationships, etc. Programs such as these are helpful in learning the techniques of Computer Aided Design. However, as stated earlier, using these basic stand-alone tools in a digital computer environment can become tedious in the long run.

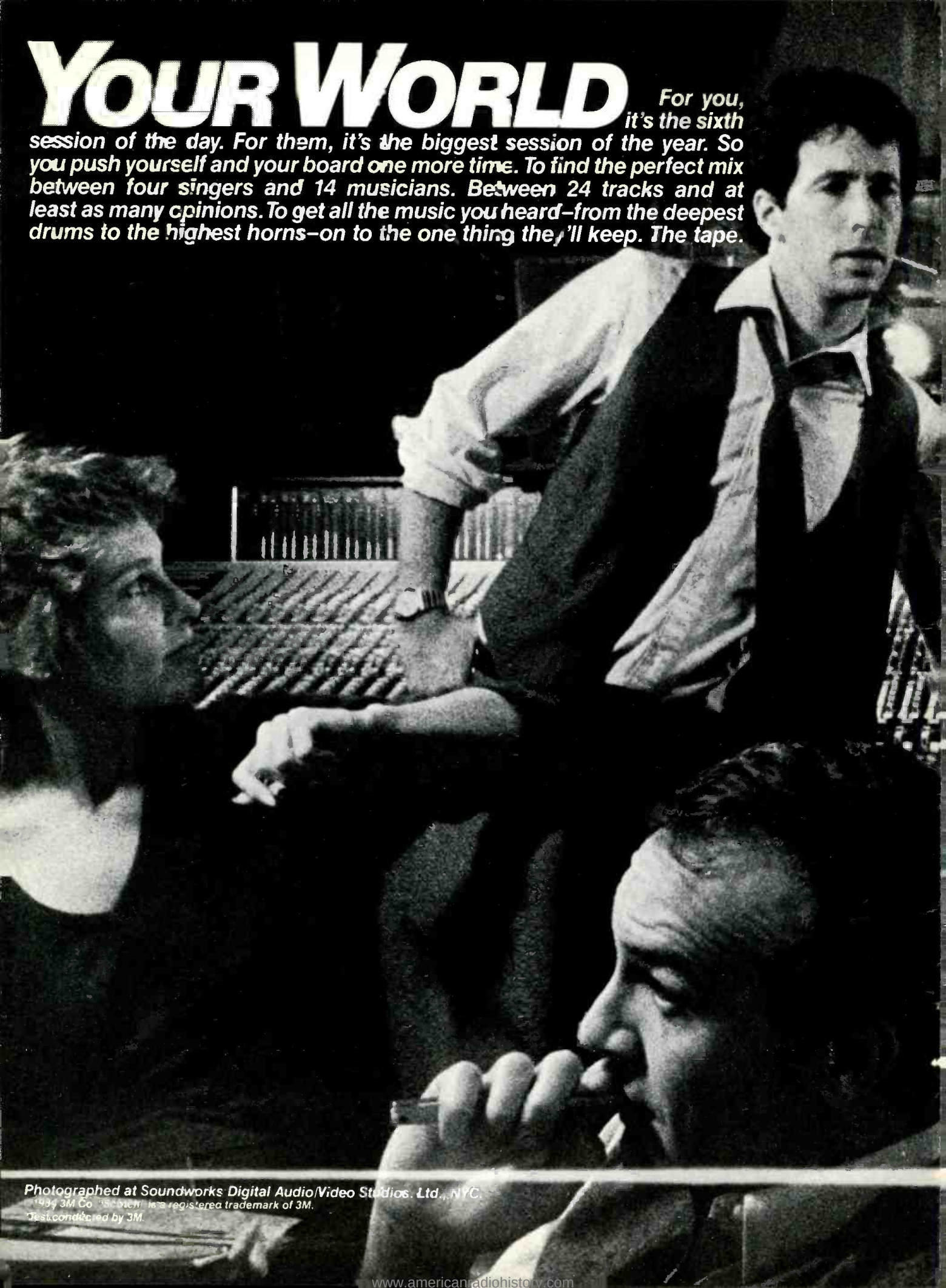
ANALOG LOUDSPEAKER CLUSTER DESIGN

The earliest analog technique for loudspeaker cluster design was developed by Wilfred A. Malmlund and Ewart Wetherill at Bolt, Beranek, and Newman in 1964. Their work was published in the January 1965 JAES entitled, "An Optical Aid for Designing Loudspeaker Clusters." Some 15 years later there was a resurgence of these physical (analog) loudspeaker/room plotting techniques, including: Farrel Becker's polar plot method of loudspeaker design (AES preprint no. 1758, May 1981), and John Proh's global plotting technique (JAES April 1984). Not only can these analog methods be implemented rather inexpensively, but they have also proven themselves to be effective and enlightening as well. In fact, the current generation of digital-computer software is based upon these analog techniques.

A demonstration of the Proh's Cluster Computer shows the impressively short time it takes to project a loudspeaker's coverage in a room. This is accomplished by simply positioning the sphere (with the appropriate overlay) at the loudspeaker position over a set of blueprints, and turning on a light bulb. In comparison to the digital methods currently being implemented, the analog process can be completed in

YOUR WORLD

For you, it's the sixth session of the day. For them, it's the biggest session of the year. So you push yourself and your board one more time. To find the perfect mix between four singers and 14 musicians. Between 24 tracks and at least as many opinions. To get all the music you heard–from the deepest drums to the highest horns–on to the one thing they'll keep. The tape.



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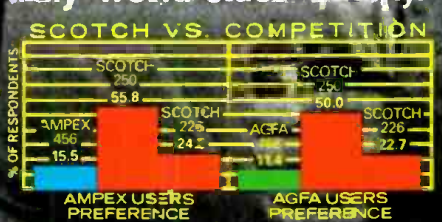

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the time it takes to simply boot-up the program in the computer.

I'm often asked, "Which computer should I buy?". Upon asking those poor lost souls what their applications might be, many reply, "File away inventories and *other* things like that." What ever happened to those little metal boxes with file cards?

One of those "other" things is often manufacturers' list price sheets. Unless you're involved with a substantial operation which uses a minicomputer (or larger) bookkeeping system, simply looking up the price in a looseleaf binder can't be beat for speed and accuracy. A case in point: I recently bought an auto headlight from my local NAPA auto parts dealer, who uses a powerful minicomputer system. After the clerk got the part from the shelf, he went to the computer terminal to input the item, and moments later—*voila!*—a complete invoice, tax and all. But, ask the clerk for the price of a left-handed framistan, and he'll have to "look it up in the book."

Now don't get me wrong, I'm not advocating that all design is better accomplished with analog computing. I'm simply suggesting that digital computers are often too zealously applied. Digital computers are stupid. They have to be told everything—nothing can be taken for granted or inferred. Analog computers, on the other hand, can become part of one's "thinking" process (as was illustrated above). Besides, I'd like to see you spill your coffee on your terminal's keyboard and see if it still keeps on "ticking."

WHY NOT ANALOG AND DIGITAL?

Many engineers have found a combination of both analog and digital computers as their ideal "toolkit." Working with drawings and specifications, using digital computers to run, word-processing, spread-sheets, and/or dedicated CAD (Computer Aided Design) packages can create an efficient working system. The incorporation of analog computers into the digital computer domain aids the engineer.

We should not find ourselves caught up in an analog vs. digital debate to the point that it overshadows our ultimate goals as sound engineers. To the audience of a concert violinist, what is important is not the type of strings or bow he uses, but the music that he makes—that is what they hear. Similarly, to the audio engineer's "audience," what is important is not the computer or program that we use in our tasks, but the effect of the sound system or the recording—that is what they hear.

The medium upon which music is recorded and distributed is not the only "sacred" part of our analog lives that seems to be losing out to the "digital crusade." This summer my six-year-old nephew received one of those new digital watches that not only has your standard alarm and chronometer features, but also a full scientific calculator with over 30 buttons! Three adults laughingly spent hours figuring out how to set the time. What ever happened to those good old *analog* Mickey Mouse watches? ■

JOHN EARGLE

Priority And Control Systems

• Modern communications systems often call for considerable flexibility in routing signals. For instance, a typical paging and announcement system in a transportation terminal may call for wide-area coverage for paging individual passengers, as well as limited-area coverage for announcing boarding information at a particular gate. If we add to this the requirements for overall automatic level control as well as facility for handling automated recorded announcements, it is clear that we have the makings of a very complex system.

On occasion, a sound contractor or consultant will be called upon to design an elementary priority system or a remote control system; these topics are the subject of this month's column.

A SIMPLE SWITCHING SCHEME

Priority systems allow various inputs to take precedence over others. FIGURE 1 shows a simple switching scheme that can handle any number of single inputs arranged in priority order. Paging microphones normally contain manual switches which actuate them. When a given microphone is not in use, the switch shorts it, providing signal continuity around the microphone. When a microphone is actuated, the short is lifted, and the microphone is placed across the line. All microphones of lesser input are shorted and cannot be actuated. The circuit shown in FIGURE 1 can easily be extended to cover more inputs.

The switching arrangement shown in FIGURE 2 provides for a single wide-area page that can take precedence over any number of local pages. Note that a double-pole switch is used. One section actuates the

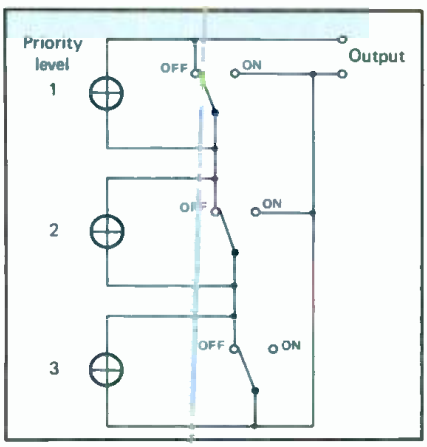


Figure 1. Simple three-level priority switching. Activating an input disables all lower priorities.

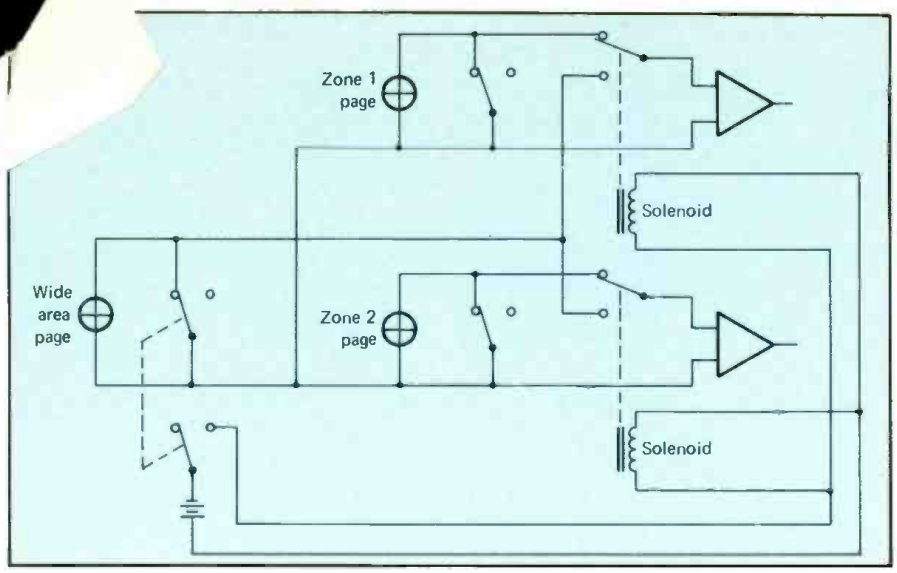


Figure 2. Wide area paging with priority over local zones.

microphone, while the other actuates the distant relays. In practical application, the microphones shown in FIGURES 1 and 2 would be amplified close at hand so that only a high-level signal would be switched.

The circuit shown in FIGURE 3 is unique in that it allows two separate amplifiers, one for paging and the other for background music, to be fed to any number of loudspeakers with only three wires. In the position shown in the figure, the "program" amplifier is actuated, and the line loudspeakers are fed through the center-tapped transformer. The level at the loudspeaker is 6 dB lower than if the loudspeaker-distribution transformer array were placed directly across the amplifier's output. However, there is little loss in the center-tapped autotransformers.

When the "paging" amplifier is engaged, the top and bottom of the autotransformer are shorted together, and the loads are fed through the fairly low resistance parallel path through the two halves of the winding. The feature of this circuit is that it provides added system redundancy through the use of two amplifiers. This circuit diagram was provided by David Klepper and Peter Tappan, both well known and respected acoustical consultants.

SIMPLEXING

The term simplexing refers to a method of transmitting both signal (AC) and control (DC) voltages down the same pair of wires. Transformers with center-tapped windings on one side are required for these applications. The method is related to the now common method of remote

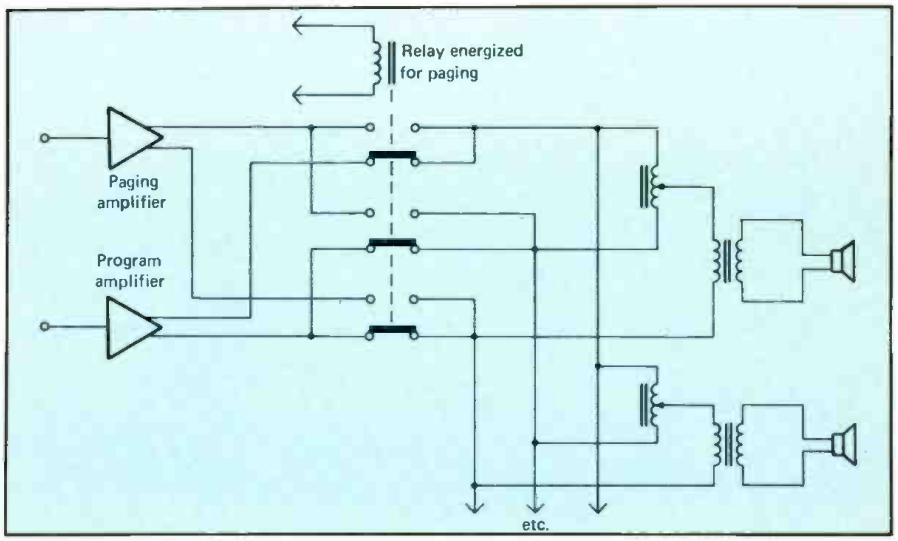


Figure 3. Priority switching at the loudspeakers (circuit courtesy of D. Klepper and P. Tappan).

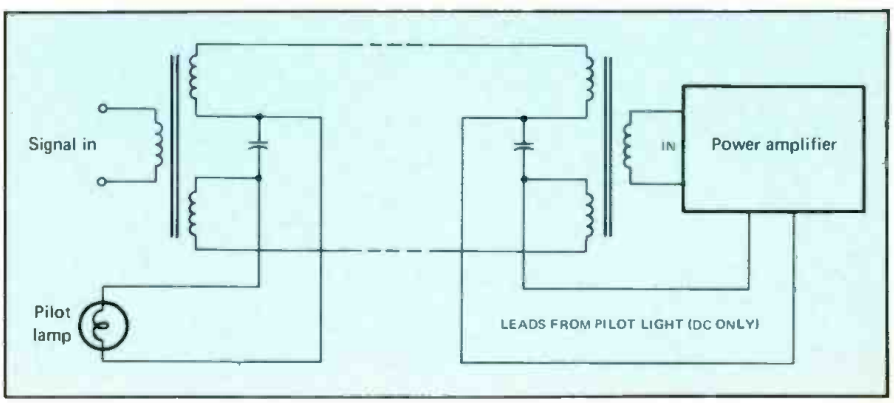


Figure 4. An example of simplexing with remote indication of on-off status.

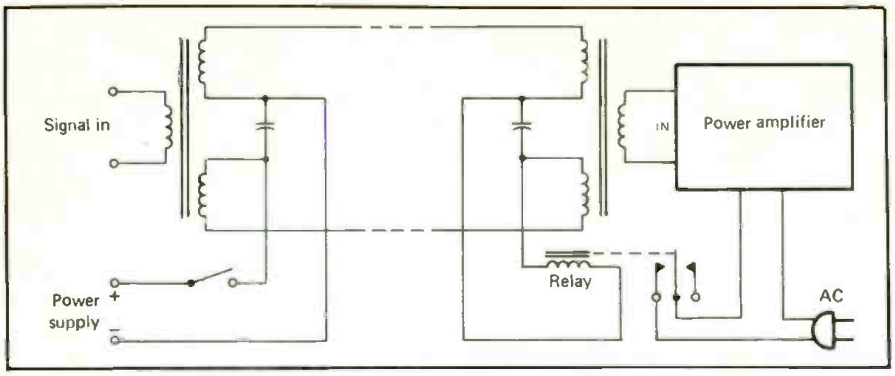


Figure 5. Simplexing with remote turn on-off of a power amplifier.

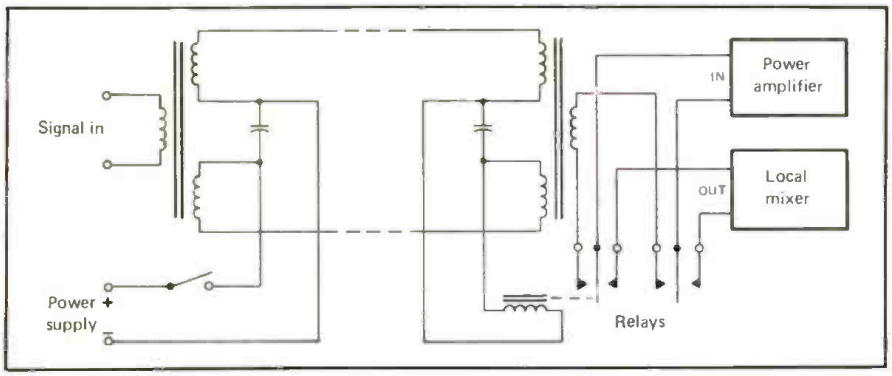


Figure 6. Simplexing—a priority system.

powering of capacitor microphones in today's consoles.

The arrangement shown in FIGURE 4 provides a remote indication of the on-off status of an amplifier located at some distance.

The arrangement shown in FIGURE 5 allows remote turn-on or turn-off of a distant amplifier.

Finally, the arrangement shown in FIGURE 6 allows both signal and priority control voltages to be sent down a single pair of wires.

The quality of audio in these applications depends on the quality of the transformers. And, of course, good transformers of the type shown here are quite expensive. Therefore, we are likely to see such circuits as those used in large communications systems where the emphasis is not on audio quality *per se*, but rather on ruggedness and reliability. ■

REFERENCES:

Altec Technical Letter Number 141, Altec Corporation
Altec Technical Letter Number 222, Altec Corporation
Altec Technical Letter Number 224, Altec Corporation

Changes At db

db MAGAZINE HAS BEEN coming out monthly since its inception in 1967. When we started out we were the only kid on the block, and we soon developed a loyal readership of audio professionals in broadcasting, recording and sound reinforcement/contracting. Over the years, we can happily report that this readership has increased to the point that we can justify the claim that **db** Magazine reaches more professional audio people than any other publication.

Yes, there are other publications now, none of which were around in 1967.

We've been searching for ways to make **db** Magazine bigger and better for you, and we think a way has been found. Effective with the very next issue we will become a bi-monthly with six issues a year, instead of the 12 we have always had.

What does this mean? If you are a subscriber you will still get the number of issues you are supposed to get. We are re-programming our computer so that this transition will run smoothly. If you subscribed for 12 you will get 12.

Each future issue is expected to be larger than those we have published in the past. More pages of editorial content will be included in each issue, and articles will be presented in a more focused manner. The magazine will be divided into three basic sections: broadcast, recording, and sound contracting. Not only the feature articles, but the columns will appear in each issue in a specific section.

At times, of couse, an article will not fit into an exact category. For example, a general purpose article on acoustics would go into the section that most closely approximates the types of acoustics being covered.

Next month, then, you will receive a November/December issue. In that issue, and in the future, we will also be improving the graphic presentation of each page, and there will be more use of color than before.

In short, then, we promise a bigger, better, and (most important) even more useful **db**, **The Sound Engineering** Magazine. **L.Z.**

JIM DOLAN, JR.

Audio For Video Comes Of Age

The expanded role of high-quality audio for video is opening up new possibilities for the recording studio. In future years, the studio itself might have to re-direct its priorities.

THE PROSPECT OF STEREO for television is bringing welcome changes to the communications industry. Many television commercials and programs are already being recorded and broadcast in stereo, television manufacturers are producing sets with sound quality comparable to that of many stereo systems, and television viewers will soon discover a whole new dimension to the medium.

Since its inception, television has concentrated on emphasizing its visual aspects while neglecting the quality of its audio. It remains the only mass broadcast media that does not yet utilize stereo audio. While inferior sound quality is a problem that applies mostly to commercials, TV programs are also affected.

It should be noted that when television was introduced, the technology for high quality audio was not yet available, and when companies finally developed workable systems, the FCC had not yet decided on a standard broadcast format. To proceed without such a format would have spelled disaster to both broadcasters and television manufacturers. The demands of the consumer prevailed, however, and the FCC, after going through a process similar to the one performed when they had to decide on a format for color television (not to mention a great deal of politics), chose the Zenith-dbx process as the standard stereo for video broadcast format.

Jim Dolan, Jr. is the president and studio manager of Streeterville Studios in Chicago.

A NEW ERA FOR VIDEO SOUND

The most obvious benefactor of enhanced television sound is music videos. Viewers will soon be able to watch their favorite musical group or performer on the TV screen and have the sound quality they have become accustomed to when listening to their FM radio or stereo phonograph. But the benefits of stereo television go a good deal further than music video. Programs such as PBS's *Great Performances*, which currently must be simulcast over FM radio, will no longer need to do so.

Programs in which music does not play a dominant role will also be affected. Viewers will notice a higher anticipation when Magnum PI or Remington Steele arrive to save the day, due to the added dimension that stereo sound will be able to create. Better sound effects will enhance the excitement when Knight Rider jumps over a cliff or the Fall Guy is thrown out a window. Advertisers will be pleased to find their commercials having a stronger impact due to the enhancement stereo television will bring to their pitches.

Even programs that use little or no music will benefit. With stereo sound, viewers will be able to hear Johnny Carson out of one speaker and Ed McMahon out of the other, the result being that it will become more exciting as the roar of the crowd and the rally of the organ bring the television viewer even closer to the stands.

To the engineer, the long overdue advent of stereo television should be welcome with open arms. No longer will we have to toil for hours trying to capture details that never made it to the final process. Now when we give our work that extra push it will be noticed on television; indeed, it will be expected. Stereo for television brings new excitement, new challenges, and new responsibilities to audio engineers. It is up to us to maintain the high sound quality that we put into records, films, and radio when doing audio for video. It is also up to us to devise newer and better methods for the recording, mixing, and playback of audio for video. Television programs, especially commercials, are generally made on tight schedules. Stereo television will enhance the demand for high quality sound in television programs and commercials but it will not increase the time in which these programs and commercials are made. It is therefore up to us to develop fast and efficient production techniques. It is also imperative that we strengthen our relationships with our esteemed colleagues in video production. While sound people have always had close ties with sight people, the new demand for high quality audio for video will greaten the interdependence we now share and demand that we work even more closely to develop a premium product.

With all the changes coming into our industry (audio for video, stereo television, digital), our responsibilities as studio people will be keener and more demanding when it comes to making decisions on equipment and processes. Our colleagues in the video and television facilities, who will be expected to facilitate stereo monitoring and precise playback performance in audio, should be able to depend on our experience in those areas just

Figure 1. Neve 8128 48-channel recording console with Necam II in Music I, with equipment room off right.

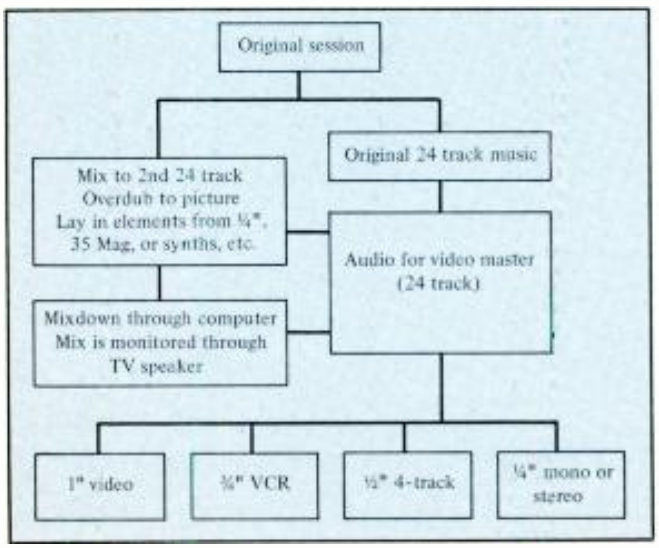
Figure 2. Close-up of operator's computer terminal/ controller of Q-Lock 3.10.

Figure 3. View of the machine room off Music I mixing area with Dual MCI JH-24 24-tracks, and Studer B-67 mono and stereo machines.

as we will have to look to them for video guidance. Neither audio nor video studios are equipped to handle the new technologies alone. Of course both audio and video studios should relish the new challenges that technologies such as stereo television bring, but we should always remember that neither of us have the capabilities to go it alone. We could try, but the resulting drain on finances and manpower would make it unwise. It takes good audio and good video to make a television production effective, and audio studios should be able to work in harmony with video studios just the way sound and sight work together to bring the viewer his or her favorite TV program.

RE-ORIENTING THE STUDIO FOR THE VIDEO AGE

Despite the fact that stereo for television is within our grasp, most recording studios are still not prepared for the demands of audio for video. At times they find themselves flying by the seat of their pants, and end up costing the producer time, quality, and money. At Streeterville Studios in Chicago, we have been recording audio for visuals since we opened in 1967. We have developed rooms, people, and equipment especially for the needs of audio for video. Our remix suite, Music II, is a perfect example of this. It has all the facilities necessary to create top notch audio for video. At this time Streeterville has other rooms in development. Within a year all of our rooms will have audio for video capabilities rather than just the video playback capabilities they currently possess. The rooms will have controllers, synchronizers, etc. All in all, we have come a long way from the 35mm projectors and loop bins that the studio utilized when it opened almost 18 years ago.



Sequence of events from the original session to the final product.

An example of our aggressiveness and eagerness is reflected in our audio finishing process. Here is the process based on a buildup for a commercial: The producer brings in his 24-track music master for mixdown. The 24-track can either be originated at Streeterville, or from an outside facility. The mix is set up as any regular mix would be, but with one change. Instead of optimizing your mix and transferring it to ½-inch 4-track, or ¼-inch mono, the music is optimized and mixed across to another 24-track that maintains the mix's dynamics yet separates it across several discrete channels for later balancing, if necessary. This tape becomes the audio for video master, and is striped with SMPTE Time Code. Once striped, the audio for video 24-track can be sync'ed with a ¾-inch video cassette via our Sony BVU 800, both controlled by the Q-Lock Synchronizer with option 64 software. Now any audio element (i.e., SFX, Announcer, etc.) can be added to the audio for video master while seeing the picture in sync with the music mix. Elements can be laid in from 35 mag bits or ¼-inch tape bits by utilizing the AUTO START and AUTO RECORD functions of the Q-Lock Synchronizer. Announcers as well as musicians' performances, or synthesized effects, can be recorded to picture in sync, and laid straight to the audio for video 24-track.

During the assembly it is possible to reference the audio through a TV set to more accurately address any sonic or placement questions. At this point the producer and engineer have *all* the audio in place and are ready for the mix. The TV monitor is also utilized during the mix for audio monitoring. For this we use our all-time favorite, the Sony Trinitron (original model), which is located in front of the mixing console. With it, we make the final adjustments to the EQ, level, and ambience of the mix. The Harrison Auto-Set I is then used to aid in creating the most effective blend of music, voice, SFX, etc. Once this has been achieved for the producer, the client, and hopefully the engineer, the mix can be transferred via the computer for direct first generation mixes. The mix

can be transferred to any or all of the following formats: 1-inch type C video; ¾-inch videocassette; ½-inch 4-track; ¼-inch 2 track; ¼-inch mono; 35 mag; 16 mag; or even ¼-track or cassette.

The following details the process step-by-step and explains why using this method optimizes Streeterville's audio finishing assignments.

THE AUDIO FOR VIDEO PROCESS

As mentioned, the audio for video process begins with the initial recording which is done on a 24-track machine. After the recording session is completed, the tape, unless all 24 tracks have not been used, is mixed to a second 24-track machine which is identical to the first one in make and model. Both machines are run at 30 ips. By

Figure 4. Close-up of Harrison Auto-Set in control center cabinet.

using identical machines with the same speed and format, we can achieve the highest quality transfer possible in an analog format.

This 24-track system is a vastly superior method to the typical finishing process for a television commercial. As we noted before, the typical process is to use a 3-stripe 35 mag. This is facilitated either by mixing straight to a 35mm 3-stripe machine or mixing into a ½-inch 4-track with 60 Hz/SMPTE which is then transferred to 35mm 3-stripe. The ½-inch 4-track or 3-stripe allow for the separation of three different elements. When you use two 24-tracks, you are not limited and can spread your recording over as many channels as you like. This way you can customize the music and all the audio around its counterparts; i.e., strings on one track, horns on another, sound effects and announcers on the other tracks (10, 20, or more). If you have a soloist, you may want to put him/her on a separate track. The goal, of course, is to eliminate compromise and second guessing. The less compromise and second guessing that has to be done, the more effective the commercial (or television program) can be.

By utilizing this process, if I find that a high horn section is getting in the way of a spoken line, I can deal with just that horn section. This could also apply to less string and more melody or less melody and more string. It allows a more specifically tailored sound. A practical example of the effectiveness of the process is a project Streeterville recently did for a major agency. The sound had been all laid out and the commercial was in its final mixing stage. All of a sudden, the agency's producer discovered that there was a "hole" between the point at which the announcer's voice ended and the music picked up. Because of this process and the availability of open tracks on the project, we were able to add a synthesizer element and thus fill the hole. As a result, the client was able to save time and money in additional recording and mixing sessions.

As noted earlier, the Q-Lock Synchronizer with option 64 software facilitates the laying in of elements via its AUTO START and AUTO RECORD functions. These functions allow a quick and precise buildup of the track, which can go back to its original audio (location recording), or a pre-edited or pre-mixed effect or voice track that the editor has created on 35 mag. What matters here is not only which is best for the project from a quality standpoint but also that of cost-effectiveness. It is very worthwhile to be able to shape the sonic qualities of your effects or your announcer to the rest of the music program while you are building a track. You end up with elements that work more effectively with the music, or vice versa, resulting in a more effective program.

A small but tremendously significant part of the process is the simple monitoring of level equalization. As noted before, during the assembly and final mixing of the audio, the sound is played through a television speaker. Presently speaking, the viewer will be listening to the commercial or program over a similar sound system, not a large stereo. Using a regular television speaker allows us to make our decisions under the most optimized conditions a studio can have and eliminates a good deal of second guessing.

Figure 5. View from producer's desk of Harrison 4032 C transformerless console in Music II.

As the final mixing of an audio for video project is done, a computer is used to record and, if need be, to reproduce the positions of each fader at any time during the mix; the entire process can be reproduced by the automation system. This can become extremely important, especially if you have just completed a 12 hour session, only to find that a word from the soundtrack has to be removed, let's say for legal purposes, and the whole mix has to be redone. A recent case in point: During the production of a product we did for a major beer company, the agency needed approvals from several people and it seemed that each of them had one thing or another they wanted changed. Normally, this can take a toll on an engineer, but because of the automation process, we were able to preserve the

continuity of the mixes and complete the project in half the time. By employing this method, first generation mixes from the audio for video master can be passed across to any format needed for the job. This results in a higher quality product as fewer generations are needed.

THE FINAL MIXDOWN

The final part of Streeterville's audio for video process is mixing the sound down. There are two ways of handling this, either of which can optimize the process, depending on the status of the project and the time of the mix. One way is to mix down color bursted 1-inch video tape. That piece of video tape is pre-striped with audio which puts second or third generation quality on the 1-inch. The 1-inch piece of tape then goes to a video facility where they do the video insert right on the piece of pre-audio striped 1-inch video tape.

The second way would be to lay back the audio right onto the final edited/color-corrected 1-inch video tape. Either one of these processes allow for top quality audio to be put on video tape. (Should the client need a copy of the work for demonstration purposes, we can make him an immediate ¾-inch copy of his product with audio and video in perfect sync.) At Streeterville, we strongly feel that it is better to let us do audio lay-back rather than take our finished soundtrack to a video facility. Video facilities are, of course, capable of doing the final audio transfer, but there are several reasons why it is advisable to let us do it. Most video facilities are not geared for extensive audio work and most of the equipment available at these facilities is not designed for high quality audio.

That about raps up the process. You may have noticed that throughout the entire process, from recording studio to 1-inch mix, only three generations were used. In such cases where only one 24-track tape is needed, we can go from start to finish using only two generations. This is a considerable improvement over the traditional format in which nine to ten generations of tape are used. The result in terms of quality is tremendous, because we can stay closer to the original source of the sound.

Streeterville's ability to handle all the demands of audio for video finalization proved to be crucial in another recent project done for the creators of a new laser disk video game. The client first attempted to have the game's technical audio track done at a facility that was not optimized for video audio post-production. The studio used a more traditional process, and after much expense and frustration the client decided to go elsewhere. The job required recording a 36-piece orchestra, adding over 150 different sound effects, and handling camera dialogue. It was like trying to put together a 30-minute film. Had this project been attempted in a method other than the one we have been discussing, the client would have had to sacrifice some of the audio quality in order to bring the project in on time and on budget. Because they were using a facility that was optimized for this type of work, not only was it done on time, but it was done under budget. More importantly, no one felt as though they had been put through the ringer. This is perhaps the most important dividend of this process; it allows the engineers and other creative people to spend more time being creative and less time worrying about the technical aspects of recording. It

Figure 6. View from the rear of client area of the "Suite."

also allows for experimentation to gain that extra grain of perfection. The result is happier engineers, happier clients, and better audio for video.

A ONE-MAN SHOW

As an engineer, and from an engineer's point of view, the exciting aspect of this process is the fact that it allows the same mixer to follow a project from the down beat of the music straight through to the final layback of the audio. This ensures compatibility of the technical processes from stage to stage. This way we have the opportunity to see our projects through and thus develop a greater sense of pride and responsibility in our work. The process also streamlines the producer's job because he relates his concerns to only one person as opposed to the standard four or five.

As for myself and fellow engineers, it is now up to us to rise to the challenge that stereo television will bring us. Of course, recording in stereo is nothing new, but we will now have to produce high quality sound in, at best, the same time span. I refer, of course, to that old adage, "It's only a commercial. Nobody will hear it anyway." Those times are rapidly disappearing if they are not gone already. I recall that 13 years ago, when a commercial was done, a drum set-up mic'ed with only three microphones (bass drum, snare drum, and overhead) was used. Today, that same drum set would be outfitted with 10 microphones. As mixers, we are all sensitive to the striving that must be done and the constraints that must be overcome for state-of-the-art jingle sessions. Stereo television will increase the need for commercials to be on a par soundwise with records and movies, and thus intensify the situation. In other words, while studios are turning out better commercials today than they were 10 years ago, they are still given a limited amount of time to do so. This is all the more reason for studios to use recording methods that cut down on technical problems and allow the engineers to be more creative and still finish the projects on schedule.

The fact that we can offer total audio service control and continuity from conception through the mix of a project, no matter what the format or medium you are working in, makes this a very exciting time and place for a recording studio of our complement to be in. This situation creates a responsibility that will keep Streeterville at the cutting edge of growth. It should be remembered that audio is just one piece of the television production pie. As the demand for better sound grows, however, it is rapidly becoming a very important piece. ■

A Closer Look At Streeterville Studios

THE FACILITY

Streeterville boasts a variety of support rooms for its studios. A pre-production music and SFX room houses various comprehensive libraries as well as up-to-date talent demo tapes. An extensive post-production room facilitates the making of 16mm or 35mm mag transfer, any format mono or stereo club, and audio cassette, as well as dubbing ¾-inch video cassettes. Also, Streeterville has a comprehensive ¼-inch mass tape duplication service that not only includes high quality dubs but is also responsible for packing and shipping logistics to radio stations. The seemingly unused corners of the facility are not so unused, because Streeterville's hallways, stairwells, and washrooms are all equipped with mic lines for those creative ambiences. Streeterville rounds out its service very comfortably with private lounges, showers, and a kitchen. Following is a detailed rundown of the facilities, including a complete equipment list, studio by studio.

MUSIC I

A state of the art full-service 24/48-track music studio that facilitates 40 musicians. The studio offers a variety of areas and surfaces from live to softer ambiences, plus an isolation booth off the machine room of the control room. Studio package includes a Yamaha 7-feet × 4-feet grand piano and sonar drum set in oak, plus private lounges and showers.

Details

47-ft. × 22-ft. × 14-ft. studio size
25-ft. × 16.5-ft. × 11.5-ft. control room size
Neve 8128/48 channel mixing console equipped with Neacam II computer information
Twin MCI JH-24 multitrack recorders with auto-locator III
(2) Studer B-67 mono and stereo tape machines
Studer B-710 micro-computer cassette deck
Lexicon 224X digital reverb with the "Larc" controller
Urei time align 813 B speakers
Urei 6500 amplifier
Hafler 500 amplifier
BGW 100B amplifier
Crown D150 amplifier
Sony VO 5800 video recorder/playback
BTX 4500 synchronizer
BTX 4600 controller
EMT 140 plate reverb
EMT 240 gold foil reverb
MXR digital delays
Eventide harmonizer
(4) Allison Gain Brains
(4) Allison Kepex's
(2) Urei 1176 LN leveling amplifiers
Orban/Parasound dynamic siblance controller 516 C
Orban parametric equalizer 622B

Pultec EQP-IS equalizer
Urei digital metronome 962
Otari MTR 12 4-track recorder
Panasonic CT-1910 M TV monitor
Additional monitors available on request:
Yamaha, Aurotone, Rogers, and JBL

MUSIC II

A state of the art full service 24-track music studio that facilitates 25 musicians. The studio is treated with hard and soft surfaces for desired ambience, plus a six to eight player isolation booth off the control room. Included with the studio is a Steinway 6-ft. × 4-ft. grand piano and a sonar "signature" drum set. Private lounges, showers and a kitchen are included with the studio package.

Details
37-ft. × 27-ft. × 12-ft. studio size
24-ft. × 17-ft. × 10-ft. control room
Harrison 4032 C transformerless mixing console
MCI JH-24 recorder with auto-locate III
Studer B-67 stereo and mono tape machines
Studer B-710 micro-computer cassette deck
Lexicon 224 x digital reverberation with the "Larc" controller
Urei 813 B time align speakers
Urei 6500 amplifier
Hafler P500 amplifier
(2) DC-150 crown amplifiers
BGW 750 amplifier
EMT 140 plate reverb
EMT 240 gold foil reverb
Eventide digital delay 1745
Lexicon 92 digital delay
(4) Allison gain brains
(4) Allison Kepex's
dbx over easy compressor/limiter
(2) Urei 1176 LN leveling amplifiers
(2) Urei LA-4 compressor/limiters
Orban sibilance controller 516 C
Orban parametric equalizer 622 B
Pultec EQP-IS equalizer
Urei digital metronome 962
Sony VO 5800 video recorder/playback
Panasonic CT 1910 M video monitor
Additional monitors available on request:
Yamaha, Aurotone, Rogers, and JBL

"THE SUITE"

A state of the art finishing studio which excels in automated mixing for records as well as computerized audio for video. The studio has an isolation booth which can accommodate up to 10 musicians for musical sweetening, or for recording talent to pictures. The suite also specializes as a great environment for sophisticated electronic/video sync sessions.

Details
14-ft. × 15-ft. × 12/12-ft. booth size
21-ft. × 22-ft. × 10-ft. control room
Harrison 4032 B mixing console with auto set I computer automation
Twin MCI JH-24 multi-track recorders with dual locators
Studer B-67 mono and stereo tape machines
Studer B-710 micro-computer cassette deck
Lexicon 224X digital reverberation with "Larc" controller
Urei 813 B time align monitors
Urei 6500 amplifier
Hafler 500 amplifier
BGW 600 amplifier
Sony BVU-800 ¾-inch video cassette machine
Sony MCI 110 1-inch layback machine (type C)
Audio Kinetic "Q-Lock" 3.10 synchronizing system
Otari MTR 10 4-track recorder
Sony PVM 1900 TV monitor
Custom Sony Trinitron 19" monitor
BTX-4500 synchronizers
EMT 140 plate reverb
EMT 240 gold foil reverb
Lexicon "Prime Time" digital delay
MXR digital delay
(4) Allison Gain Brains
(4) Allison Kepex's
Orban sibilance controller 516 C
Orban parametric equalizer 622 B
Urei digital metronome 962
(2) Urei 1176 LN leveling amplifiers
(2) White ½ octave equalizers
Pultec EQP-15 equalizer
Urei LA-4 limiter/compressor
Additional monitors available on request:
Yamaha, Aurotone, Rogers, and JBL

PRODUCTION ROOMS

Studios "B" and "C" are customized studios that can facilitate one to five announcers, with mono, stereo, or 4-track Studer tape machines. Both rooms optimize quick and consistent working environments.

All five studios take advantage of a microphone collection that surpasses 100 in count. They include AKG, B&K, Neumann, Sony, Sennheiser, Electro-Voice, Shure, PMC, Beyer. The mics range from brand new to older vintage models.

RECENT PROJECTS

Record—Johnny Winter, James Cotton, Big Twist and the Mellow Fellows, K.K. Taylor, Crystal Wings, Buckinghams

Commercials—United Airlines, Bud Light, McDonalds, Taster's Choice, Sears, Coors, Taco Bell, True Value, RCA

Film/Video—My Roommate, PBS American Playhouse, Johnny Winter's first music video

STEVE MICHELSON

Improved Sound For Filmmakers Using One Pass System

One Pass Film and Video's new process of "electronic cinematography" offers improved audio characteristics for filmmaking. Read on for a novel approach.

A NEW PROCESS of "electronic cinematography" has recently been introduced by the San Francisco based company One Pass Film & Video. The method offers improved audio characteristics when compared to traditional filmmaking. One Pass describes its twin cinematic techniques as FT (film/tape) and FTF (film/tape/film), distinguishing between the role video plays in a film for television and a film destined for theatrical or projection audiences.

Over 150 days of field production experience using the systems have been successfully completed. The projects have included an NBC Movie of the Week, an independent feature film, several national television commercials, and a corporate image film. Through the selective use of videotape and computers, One Pass can substitute some of the costly and time-consuming steps in filmmaking. While FT is designed for a film which has an end use on television, FTF is designed for expedient electronic post-production where video tape is merely an interim "decision-making" format to completing a film.

Both FT and FTF systems share the same field production unit, consisting of a Stereo Nagra, a master time-code module, and a customized digital slate. The rugged field unit records time-code on the guardband-protected FM track of the Stereo Nagra, leaving both channels available for audio and a separate track for pilot tone. The electronic slate is outfitted with high powered LEDs, which optically tag each scene with a series of visible reference points. Likewise, both systems pass through electronic negative synchronization (ENS),

Field Production Unit for the FT System™. Includes a stereo Nagra, Master module, and Electronic Slate.

Steve Michelson is the president of One Pass Inc.

Computerized audio sync'ing takes place in One Pass's Electronic Negative Synchronization Suite.

where first generation Nagra originals are resolved and automatically sync'ed to picture.

Another useful aspect of electronic post-production is the creation of a script and note information summary. During the process of synchronization, One Pass inserts production information into the computerized time-code listing. The notes generally include script and director's comments as well as highlights from the production sound recordist.

Since there is no need for film workprints or sound transfers to mag film for post-production, One Pass can maintain high quality tape sound transfers that are not subject to the signal degradation of additional magnetic film transfers normally required by traditional film methods.

THE FT PROCESS

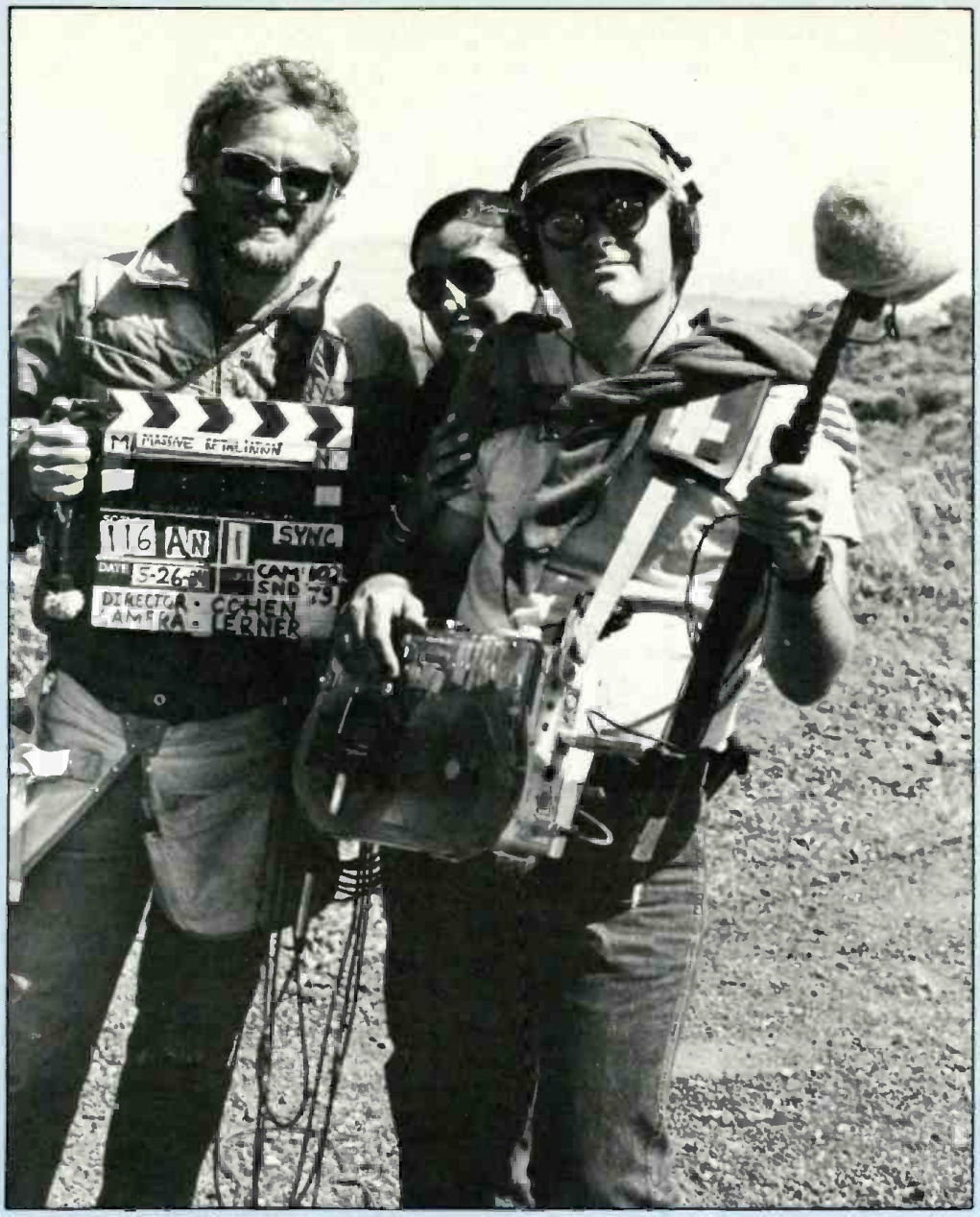
The FT approach to electronic cinematography provides the television viewer with picture transferred directly from a film's negative. The increased latitude, saturation, and resolution of this process, along with the safe handling of negative film by the Rank Cintel Flying Spot Scanner, has become the preferred telecine method in recent years. After more than a year's development with audio manufacturers including Otari, Adams-Smith, and Calaway Engineering, a customized interface has been developed to enable first generation sound transfers to accompany the high quality picture process. Based on industry-accepted specifications for electronic versus magnetic film stock and assuming standard signal processing, One Pass engineers estimate a significant improvement in wow-and-flutter, distortion, and signal-to-noise ratio in the edited television master. Although it is widely recognized that the poor sound quality of television broadcast itself reduces the importance of this improvement, One Pass feels this limitation will be short-lived.

Further, as technology for videocassette and videodisc gain wider acceptance, video sound quality will be required to compare with other high fidelity listener choices. Technical considerations aside, there is a tangible difference in the "presence" of sound transferred via the FT method, which supports the similar vibrance created from the film negative. Several national commercials have already been aired utilizing the FT approach, and the first network broadcast is scheduled for telecast in the fall of 1984. The NBC Movie of the Week, *Partners In Crime*, a Carson Production starring Loni Anderson and Lynda Carter, has been produced utilizing One Pass's new system.

THE FTF PROCESS

While still in the final phase of development, One Pass's FTF process has broad implications for the future of motion picture sound. Unlike FT, where film and sound are transferred directly to a high band video completion format, the FTF process is more of an informational means of conforming film to an edited videotape. Tens of thousands of feet of film, along with accompanying sound rolls, are transferred to videocassette to speed the time-consuming process of editorial decision making. When the inexpensive and efficient video editorial is completed, the film's edge numbers are conformed to the video time-code, thereby reducing a major and costly component of film completion.

The original sound elements can be automatically



Field Production Unit on location for the film "Massive Retaliation." Left, the Electronic Slate; Right, stereo Nagra with Master module.

"checker-boarded" on a 24-channel multitrack recorder for mixing. While there may be certain advantages at present to film sound mixing due to the ease of cutting, slipping, looping, etc., One Pass believes these advantages will be minimized once current electronic editing technology is applied to sound mixing. The primary benefit of improved fidelity from electronic and digital sound recording during post-production mixing will ultimately generate a marked improvement in motion picture sound.

One Pass is currently designing a new audio mixing facility that will incorporate these advances. Certainly there is much to gain in the continual search for improvement in sound quality, yet probably the primary drive in the evolution to electronic film mixing will be economics. With the already high cost of filmmaking, the motivation of producers to switch from the costly and time-consuming process of film sound cutting to electronic editing and mixing will undoubtedly accelerate. ■

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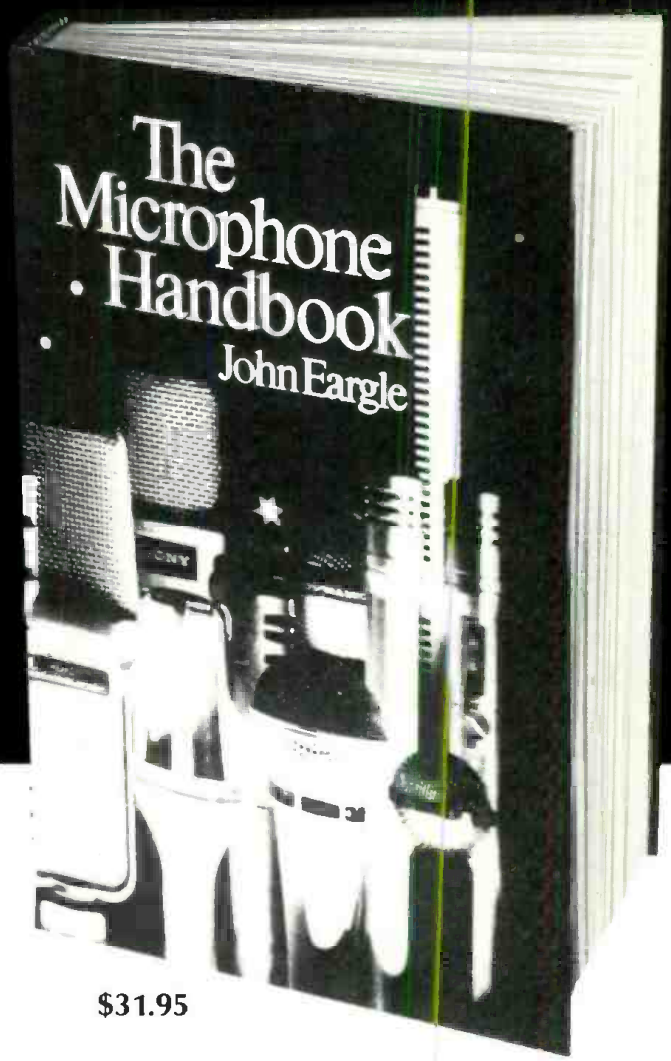

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JIM McCULLAUGH

TAJ Soundworks:
Rock 'n' Roll To Foley... And the Cable Link

The evolution and implementation of innovative ideas which led to this studio's success story are indicative of a "new breed" of recording facilities.

QUESTION: What do a new-breed recording facility, a proprietary new studio cable wiring technology, and the ghostly sound effects of Gizmo from the movie *Gremlins* have in common?

Answer: A formula for success that has put the principals and equipment involved "one step up" in the ever changing matrix of the recording field.

TAJ Soundworks in Los Angeles began its life in 1976 primarily as a rock music recording studio. That was at a time when the music industry was still climbing an upward roller coaster, according to its founder Allan Goodman. The original name: One Step Up.

The music clients were impressive: the Eagles, Fifth Dimension, Steppenwolf, Fleetwood Mac, Diana Ross, Cat Stevens, Bill Withers, Mike Post Productions, Dick Clark Productions, and Olivia Newton-John, among others.

Major national commercial credits included Bank Of America, Levi Strauss, Taco Bell, Honda and Suzuki motorcycles, JBL audio products, Texas Instruments, World Airways, Oldsmobile, Marantz, and others.

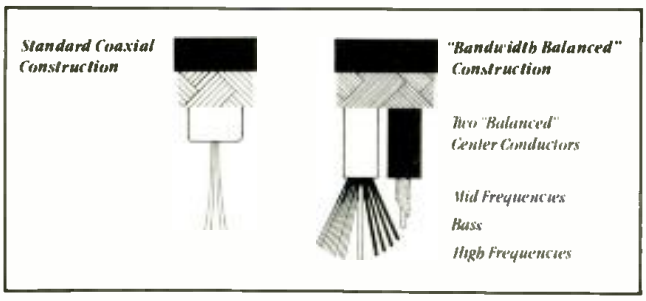

Figure 1. Comparison of standard coaxial cable construction to Monster Cable's "Bandwidth Balanced" cable construction.

Jim McCullogh is the editor of Home Computer *and* Software Merchandising. *He was previously with* Billboard.

BEYOND THE HORIZONS OF ROCK

But as the music industry began to tighten up in the late '70s and early '80s, recording studios themselves were

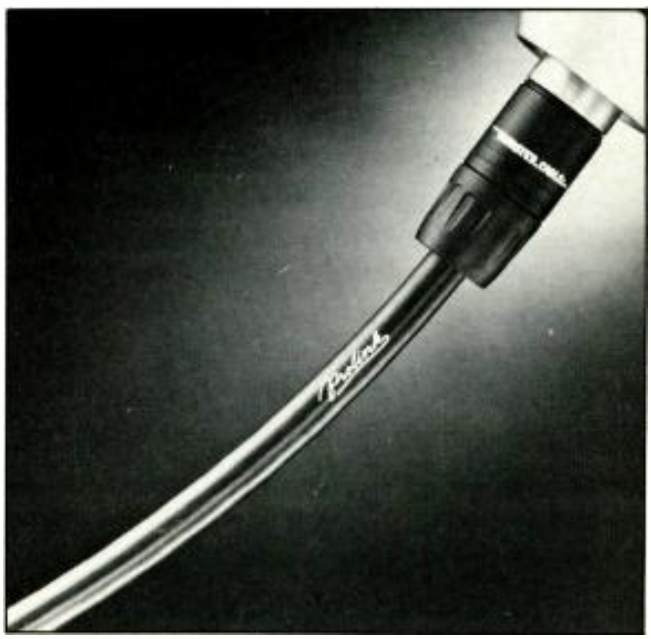

Figure 2. Example of Monster Cable's Prolink™ series professional interconnect cables and connectors.

TAJ Soundworks provided audio post-pro/Foley enhancement for Paramount Films' "Trading Places."

faced with critical marketing and direction decisions. At the time, there must have been some 250 24-track rooms in the city alone. Many featured state-of-the-art equipment, and in some cases the "super rooms" of Kendun, the Village Recorder, and others boasted "ahead-of-the-art" facilities. Those operations that clung adamantly to rock music recording found the going rough. Others diversified by adding film scoring and/or video capabilities. Still others took even bolder new steps...to Foley and ADR (Automatic Dialogue Replacement). Case in point: One Step Up to TAJ Soundworks, which began operations a year ago.

The new TAJ is an interesting combination of people and talents. In fact, the name itself is derived from three of the co-principals: T. E. (Tim) Sadler, one of the industry's foremost sound mixers, Allan Goodman, and John Roesch, one of the film and television industry's quintessential Foley actors. They met two years ago and agreed to refurbish TAJ into a unique facility.

Later, Ed Bannon, a perfectionist studio architect who had been with One Step Up at the beginning, but who migrated for a while to the Bay Area's Tres Virgos Studios as well as Puerto Rico's Crescendo recording studios, came aboard as a partner and now helms the technical side of things.

Foley, named after its progenitor Jack Foley, who did live off-stage sound effects in the 1930s, is the delicate art of recording sound effects such as a person walking, keys tinkling, a knife falling on a sidewalk, or water rushing in perfect sync to picture. Many major film studios find Foley work more cost-effective than recording all sound on location. Moreover, Foley can be done in a controlled environment, and that achieves higher quality.

Good Foley experts and studios have become increasingly inventive, more in demand, and experimentally novel as cinema, itself, has become more innovative.

The expansion of expertise and new co-partners has produced impressive results and credits. Individually and as TAJ, former and recent collective credits include: *E.T., Poltergeist, Tron, The Tempest, Personal Best, On Golden Pond, Raiders Of The Lost Ark, The Black Stallion, Star Trek, The Empire Strikes Back, Legend Of The Lone Ranger, Southern Comfort, The Main Event, Urban Cowboy, 48 Hours, Psycho II, Best Friends, Wolfen, Romancing The Stone, Absence Of Malice, Thief, Swing Shift,* and *Terms Of Endearment.*

Other film projects have included *Sixteen Candles, Footloose, Jaws 3-D, To Be Or Not To Be, Racing With The Moon, Streets Of Fire, Gremlins, Cloak And Dagger, Trading Places, The Outsiders,* and *The River.*

FOLEY MOVES INTO MUSIC VIDEO

TAJ also has the distinction of doing what is believed to be the first Foley work for a major music video short—Michael Jackson's "Thriller."

"Of course," says Goodman, "we had to make a number of changes since the sound requirements for Foley were so much more stringent than they are for music. We had to put in a whole new sub-ceiling which took several months. That cost between $50,000 and $80,000.

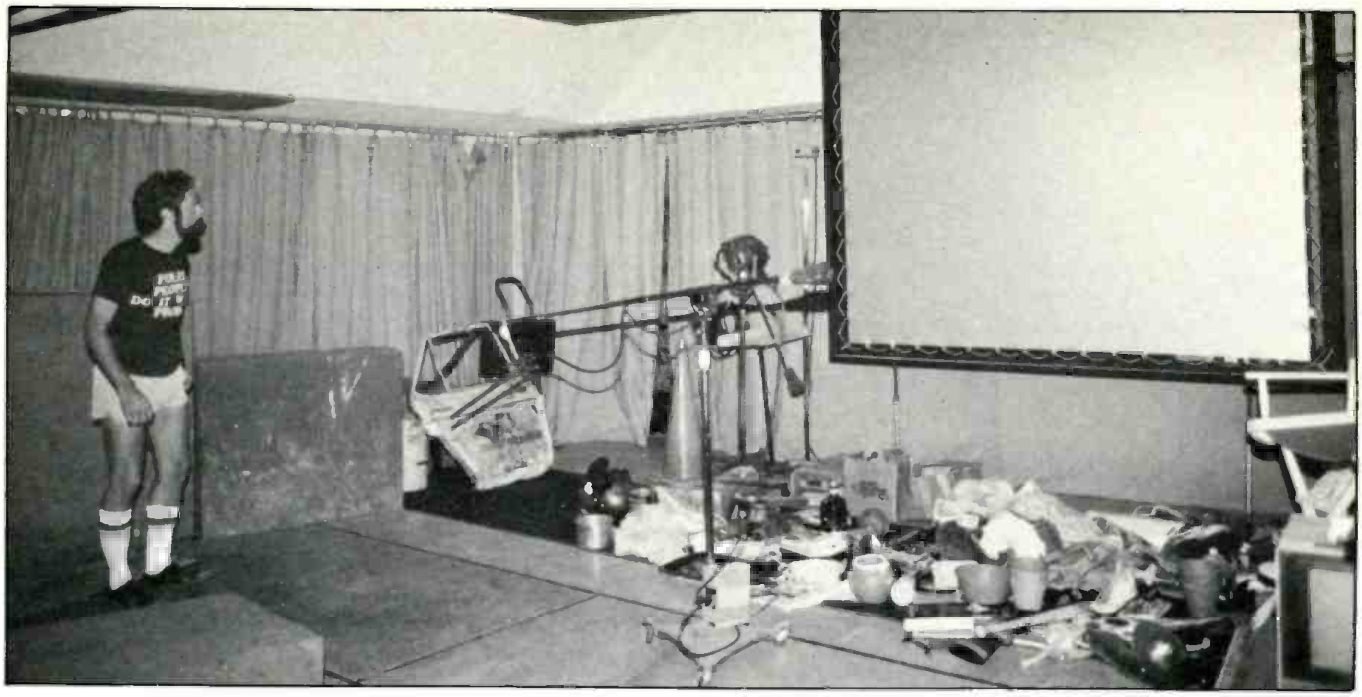

Figure 4. John Roesch checks out assorted props for sound effects in the Foley room.

"The studio floor was customized, as well as the pits. The earth pit is basically a cement chimney, which is filled with 22 cubic yards of earth. It took two dump trucks to fill it up since it goes right down to the ground. There's been a bias against second story Foley stages because when you jump into an earthen pit on the second floor, it doesn't sound right. We built it down to the ground to get it right. The concrete slabs were specially mixed and poured and one is 10-inches thick, which is actually thicker than the walls of some high-rise buildings. It really sounds like a sidewalk."

For *The River*, Universal's new Mel Gibson/Sissy Spacek socio-agricultural epic, Goodman notes, "TAJ faced one of its more interesting challenges. We had to get

a permit from the Department Of Water And Power to hook a fire department hose to a hydrant and erect a 2,500 gallon pool. With the aid of a pump we could fill and drain it.

"We had to create rain, mud, and flood sounds, sand bags falling, and bulldozers falling over. Just incredible stuff. It was normally just beyond the purview of Foley. After that we may have re-defined Foley."

Equipment-wise, TAJ retains its MCI 528 console (although highly modified), MCI multi-tracks and MCI two-tracks, a normal complement of outboard gear (noise gates, DDLs, harmonizers, EMT echo chambers, etc.), and a custom monitoring system in the studio itself, with theatre-like sound. There's also an Audio Kinetics Q-Lock (310-3 model) with software for ADR, a video projector, Sony SL0383 video deck industrial speed ½-inch Beta, JVC video monitors of various sizes, and a single-stripe film audio transfer machine which transfers the cues from the multi-track to a format that the film editors can cut and splice in.

TAJ Soundworks, unlike many other film sound stages, is built on the foundation of a world class recording studio. While now better prepared for the production of audiophile quality for music recordings, TAJ's current retrofit for the film and music video industry enables it to fulfill all the requirements of sound production and post-production for music, narration, and film with the current exception of the final dub.

A MONSTER OF A CABLE

One key element in TAJ's success thus far, believes Ed Bannon, Sadler, and Goodman, has been the recent addition of a proprietary new interconnect cable technology called "Bandwidth Balanced" cables from the San Francisco-based Monster Cable. In fact, according to Noel Lee, Monster Cable president, TAJ is among the first wave of recording studios to use it. They weave in and out of the heart of the studio's electronics, painstakingly installed by perfectionist Bannon.

"If you have a freeway that suddenly goes at a right angle, maybe the first few cars can get through, but then...they pile up," Bannon says. "It's the same thing with good cables and sound. Until I ran into this new wire from Monster Cable, all the others were like freeways with right angles."

He, as well as Sadler and Goodman, recommend the new cable technology and suggest that both existing and newly-built facilities take a serious look at their wiring considerations—whether doing music recording, film scoring, or Foley work in the analog or digital mode. The effects, they feel, are hardly subtle, the benefits cannot be ignored, and it doesn't take a "golden ear" to hear the improvements.

Declares Goodman: "Now it could mean the difference between getting an effect or not. What we realized, slowly, as elements of the system were converted to the wire, was that it forced us to take a look at other elements of the system. Suddenly, we could hear problems that we never heard before, that we never even knew existed. We had to start, for example, from scratch, on the preamp."

"We went through a state of shock for a couple of weeks," adds Bannon. "You come to the painful realization that you've spent so much time and effort on a system but you never had half a chance when you hear

Figure 5. Foley impresario John Roesch (foreground) and chief engineer Tim Sadler (center).

what this wire reveals. And I am talking only about analog recording.

"If we had this wire a long time ago, we wouldn't need digital, in my opinion. We've never heard the best sound we could get from analog."

"The cable actually exposed flaws in other parts of the system. That's how dramatic the difference is," adds Sadler. "We have had sound/film editors say to us, 'What are you doing, because when I turn up my movieola, these people walking around sound like 80-foot behemoths? There must be something wrong.' Our noise floor, the way the signal is laid down, is so radically different, we can get a cleaner signal.

"In rock 'n' roll you will notice the difference, but you won't notice it the same way. I hear the difference by the absence of noise."

Sadler sums up the advantages in two major ways: "First, it makes for a cleaner system...period! There's less noise and it's quieter. Because of that, it reaches out much farther for sound, which means we had to clean up other elements of the system. We had to find a cleaner microphone and preamp. We wanted to find a preamp with more gain because we could now ask more of the system without increasing the noise."

Bannon footnotes: "We had to get a custom preamp from Harvey Rubin and Dave Baskin. I said I wanted 6 to 12 dB more gain with the same noise threshold. It didn't have to be quieter, just more gain without raising my noise. We ended up with 12 dB more gain and 6 dB less noise."

"The other advantage is that the transients are superior. We deal with transients, and if they don't work, it's painfully obvious. In music with lots of instruments, if some of the transients are not working, you can get by. In fact, you might not even know, care, or want them. But when you are doing this type of recording, the sounds only sound one way. If you break a plate and the signal clips, it doesn't sound right. There's no fooling. As a result, transients are critical to us because they are an integral part of the sound. I noticed a marked difference when we put this wire on the new microphone."

According to Monster Cable's Noel Lee, "Most of us think that audio cables are just that, audio cables. But they aren't. Cables do distort sound in ways even the most veteran recording ears may not even be aware of.

"With ordinary cables, all the way from microphone to speaker, the frequency response and phase distortions are very significant. Lack of clarity, poor bass response, vague and imprecise imaging, high frequency brightness, and poor dynamic range are all cable-related problems. And as the industry moves more and more toward digital, these distortions will become more and more obvious and less and less tolerable.

"What we have done is extensive research into the electromagnetic behavior of audio signals, and have come up with what we feel is a new generation of interconnecting cables that solve these types of problems by aligning the signal in both amplitude and phase for a dramatically improved sound."

CABLE PHYSICS

Monster Cable contends that it's all part of what's called "cable physics." As audio signal travels through wire, a series of complex electro-magnetic fields are generated. Thus the audio cable becomes an electromagnetic corridor for current flow. The magnetic fields

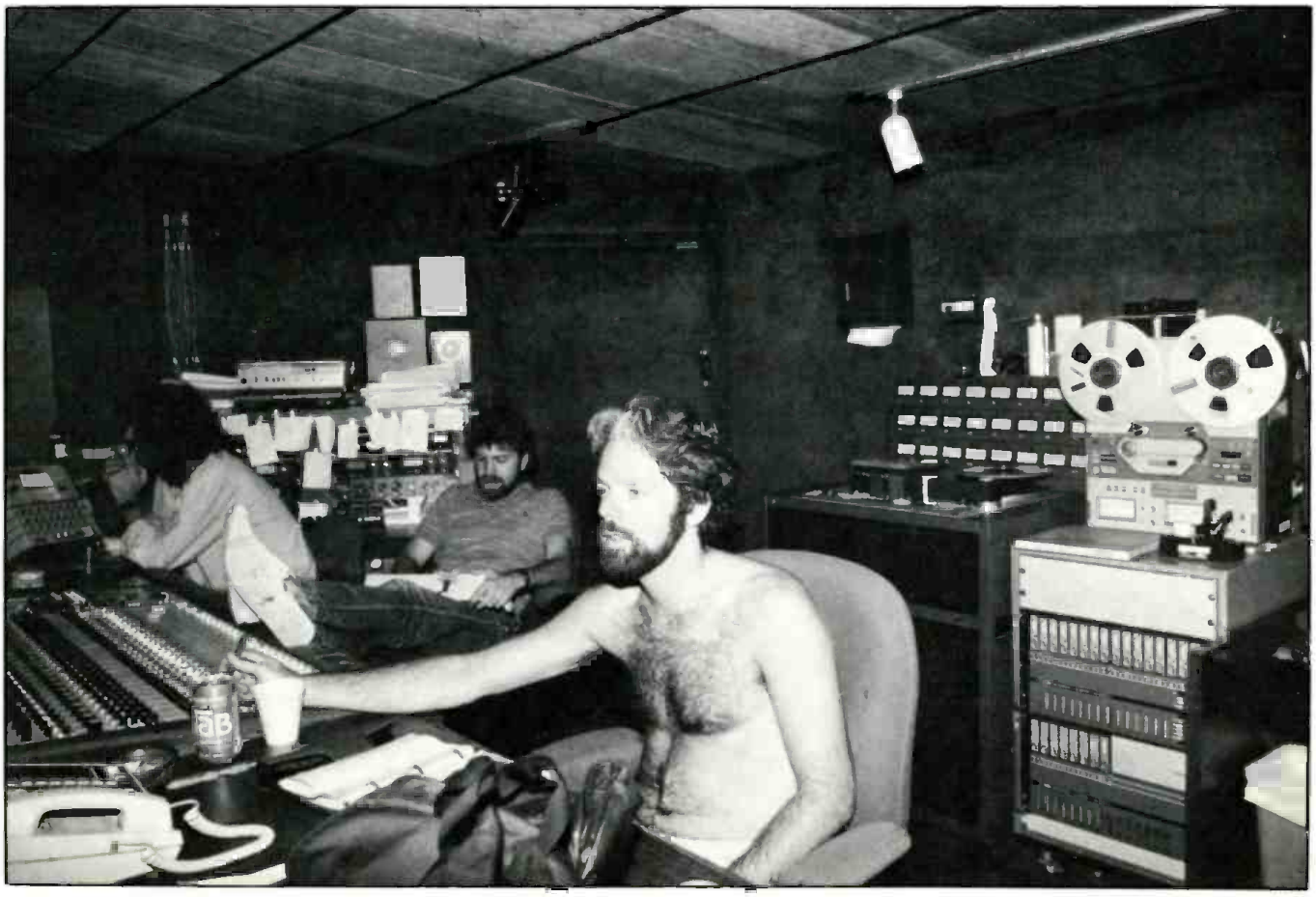

Figure 6. TAJ Soundworks' John Roesch, Tim Sadler, and assistant engineer Greg Orloff in the control room.

vary in intensity depending on the frequency, causing the higher audio frequencies to travel toward the outside of the wire. That produces a higher current flow on the outside of the wire than on the inside, causing the wire to act as a low pass bessel filter. The signal is "spread out" in the wire, which results in frequency dependent current lag at the core of the wire, creating a time-phase misalignment that progressively increases at higher frequencies. This current lag phenomenon causes extraneous internal circuits to be generated, resulting in additional magnetic fields at an angle to the main current flow. The more complex the audio information, the more complex these magnetic fields become. The "time phase" errors smear the audio signal and all of its harmonics causing the delicate interrelationships between frequency and time to become distorted.

This phenomenon makes some cables sound harsh and bright, some deficient in high and low frequency extremes, and, in all cases, removes valuable phase information present in the original signal that gives recordings a sense of space, ambience, and accuracy in instrument reproduction.

The Monster Cable Prolink cable technology, adds Lee, attempts to solve these problems by using multiple gauge "wire networks" specially wound to selectively control the speed and amplitude of the audio frequencies as they travel through wire. Each wire network is individually wound so that all the conductors aid the magnetic field instead of opposing it, as in conventional wire, and permits the transfer of audio signals with greater accuracy, clarity, and dynamic range. Additionally, all Prolink cables are protected with a Duraflex jacket for added flexibility and ruggedness. ■

The Prolink Series

The Prolink Series high resolution studio and professional sound cables are available in three configurations: The Prolink Series 1 "Multi-Phase Aligned" cable incorporates three separate "wire networks" to carry the entire frequency spectrum with superior clarity and definition. The bass frequencies are carried by a single large conductor, aided by four intermediate sized conductors for the midrange, and 350 hair-like strands for the high frequencies.

The Prolink Series 2 "Phase-Aligned" cable, while utilizing the same multiple-gauged "Bandwidth Balanced" technology incorporated into the Series 1, uses dual "wire networks" with a combination of large and small conductors in a special winding configuration that accurately aligns the signals in both amplitude and phase across the most critical parts of the audio spectrum.

The Prolink Series 3 "Phase-Aligned" product uses the "Bandwidth Balanced" technology in an economical winding configuration that delivers a high performance cable at low cost. It utilizes two wire networks, a single conductor for bass, and multiple fine wires for the mid and high frequencies. ■

Our New Subscription Rates

The editorial in this issue mentions that this is the last of our monthly issues. You may be wondering how our new bi-monthly frequency will affect your subscription, and what the new rates will be.

All renewals received through the October issue, and all new subscriptions received before the November issue, will be payable at the current rate. Subscribers will receive either 12, 24, or 36 issues, according to their original agreement. All subscriptions which expire following the October issue will be subject to renewal at our new rates.

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We are sure you will enjoy our new, meatier magazine, and agree that it will be worth waiting for every other month.

JOHN BORWICK

Scenes From Europe

Tenth International Broadcasting Convention

"VIDEO SPECIFICATION: 1125 lines; aspect ratio 5:3; bandwidth 30 MHz," was all I had time to read from the glossy Sony brochure handed to me as I sat down for the demonstration of their HDVS (High Definition Video System) before the tape began to run. I was then totally captivated by some of the best picture (and sound) quality I had ever witnessed. The occasion was the Tenth International Broadcasting Convention, held on September 21-25 in Brighton, a seaside resort 53 miles south of London. And the video specification of Sony's HDVS (developed by Japanese broadcaster NHK) was at the centre of just one of the big standards arguments which dominated the convention.

Broadcasters everywhere are getting excited about the idea of high-definition TV pictures with about twice the resolution (e.g., number of horizontal lines), a wider aspect ratio to make our old 4:3 TV screens more compatible with the wide-screen cinema, and good-quality stereo sound to match. A strong incentive for rethinking and upgrading television quality has been the emergence of cable TV and DBS (Direct Broadcast by Satellite). Sadly, the various committees on whom we must rely to draw up acceptable standards for new formats have been rather slow about it. This Sony solution, which was displayed on a 120-inch screen projection system, has jumped the gun and antagonised the rest of the establishment, though I must say its audio specification was almost as good as the picture: "frequency response 50 to 15,000 Hz +1.5, -3.0 dB; S/N ratio 58 dB; crosstalk -60 dB; wow and flutter 0.1%."

The BBC presented no less than 19 of the convention's 90 papers, and were strongly advocating a rival 1,249-line, 80 Hz, 22 MHz system for European Broadcasting Union adoption.

They were also running excellent 15-minute demonstrations of their development of digital stereo sound for use with terrestrial (i.e., normal) television. Here too,

Figure 1. Neve DSP all digital mixing console.

John Borwick is the European Editor of **db**.

Figure 2. Neve Necam 96 analogue mixing console.

however, there are conflicts with FM analogue stereo systems, and even a rival digital technique. This, as many readers will realise, is the use by Dolby Laboratories of Delta Modulation (DM) in place of PCM. They presented a paper on the topic stressing their system's relatively low cost, ruggedness, tolerance, and efficiency. They also scored a hit by announcing at the convention that a first customer had been found for Dolby DM: the Australian Broadcasting Corporation plans to use it along with B-MAC (Multiplexed Analogue Component) picture transmission via the AUSSAT satellite.

AUDIO CONSOLES

Heated debate continues, of course, in the field of audio consoles, with some people determined to stampede us all into total digital processing while others prefer the easier approach with computer control of desks handling signals still in the analogue mode. Neve managed to attract attention with their developments in both camps. They are naturally proud of the lead they have established in all-digital mixers with their DSP (Digital Signal Processor) console design. The world's first three DSP installations are all in London, and are now said to be alive and well after extended de-bugging sessions on the software. It was the subject of an excellent convention paper and featured in Neve's audio-visual presentation amongst the many mobile recording vehicles, satellite dishes, and other outside features parked all along the seaside promenade in front of the convention hotel.

And yet I found more of a crowd around Neve's latest analogue desk. This was their Necam 96 which, while retaining the motor-driven faders technique of Necam I and II, had made advances in every feature. The faders themselves had a lighter, more positive feel and responded very quickly to the floppy disk stored commands. Up to 96 faders can be controlled (hence 'Necam 96') with instant update at a touch, free grouping, a helpful full-colour display of last and next label, status information, and lists of mixes, events, etc. An intelligent

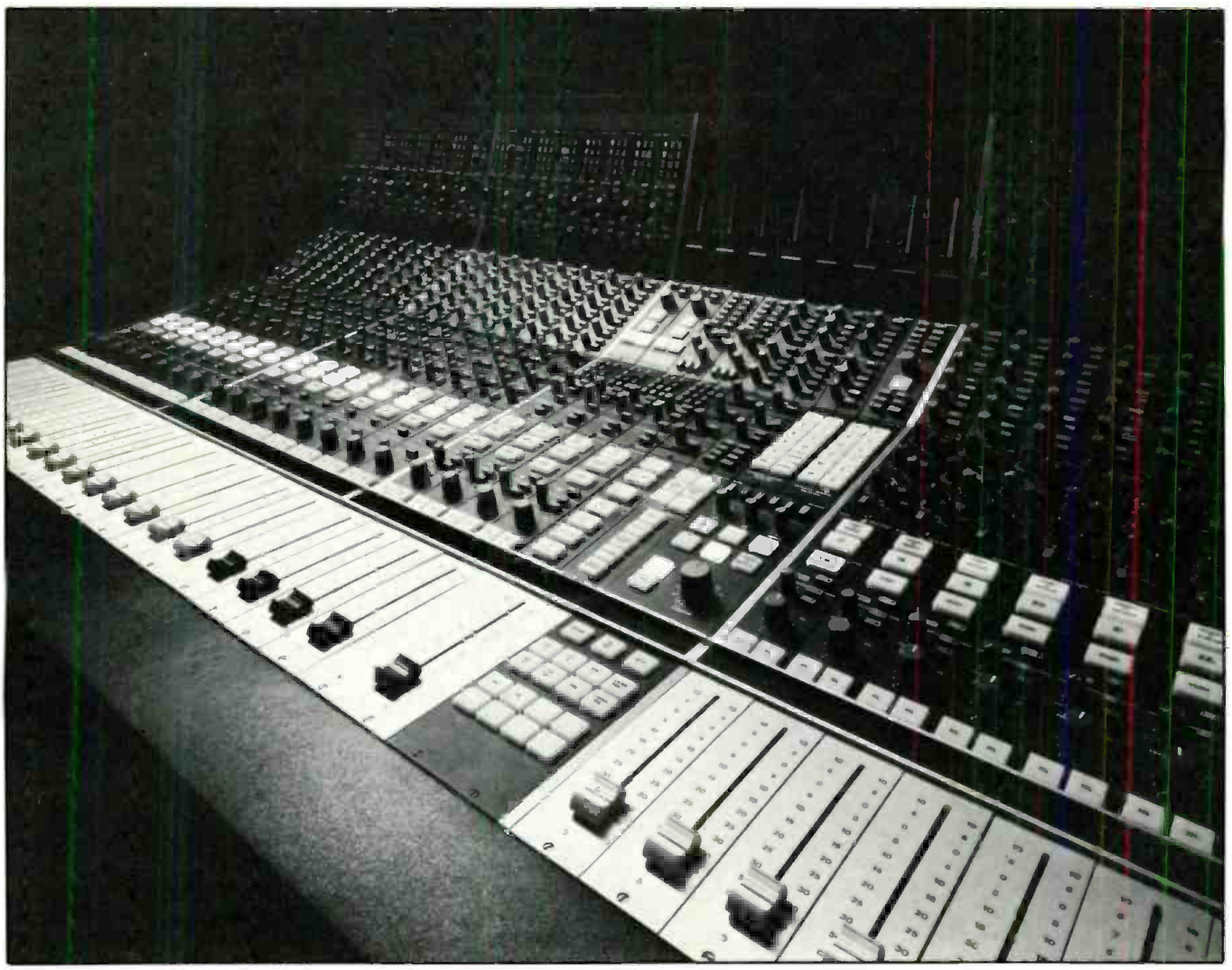


Figure 3. Solid State Logic SL 5000 Audio Production System.

rollback facility makes the tape or film go back as far as the user wants, then replays the updates with the faders always in their correct positions. The engineer can also decide which mixes to keep or merge, and there is a full QWERTY keyboard for names or up to 10 pages of text.

Other console manufacturers were also featuring mixers aimed at the music, film, and video post-production markets. Solid State Logic gave me a complete run-down on their latest SL 5000M Audio Production System receiving its world premiere at "IBC 84." Hybrid chips relying on new thin and thick film technology have reduced the size, weight, and complexity of their modules by a factor of about 10. This has enabled the SSL designers to lay out the audio, logic, and data buses on a vertical and horizontal chequerboard main frame. Between 16 and 56 mono or stereo 40mm wide channel 'positions' form the vertical strips, and these can comprise 4, 5, or 6 horizontal 150mm high 'rows.'

Already, 28 Eurocord 40 x 150mm cassettes (or modules) have been developed, including faders, aux sends, equalisers, compressors/limiters, and expanders/gates, so that custom-tailoring of consoles offers very many options. All switching is electronic, and any cassette can be addressed regardless of its position. Setting up sub-groups, 'mix minus' feeds, or splits such as dialogue, music, and effects, can be performed very quickly, with selective talkback, foldback, and cue sends linked automatically. Being SSL, they have naturally built in computer assistance, and two levels are offered. In the first, known as SSL Instant Reset, the computer will store all switch settings and enable the engineer to reset the console between any number of master and local

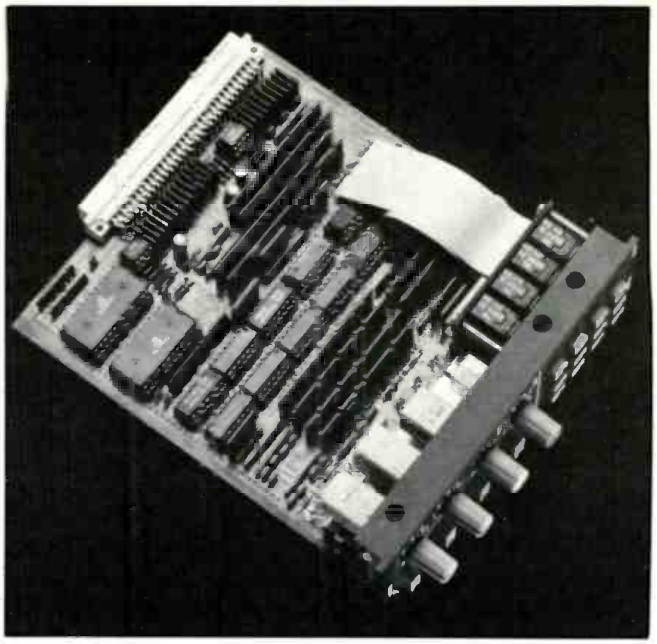


Figure 4. SL 5000 channel module showing use of hybrid components.

configurations. The second level is called SSL Total Recall, and includes the storing and resetting of all variable controls to an accuracy of 0.25 dB. The complete range will be shown at the NAB Convention in March 1985, with deliveries starting in July.

Soundcraft was majoring on a completely new SAC-2

Figure 5. Soundcraft SAC-2 modular broadcast control console.

modular broadcast control console (called the SAC-2000 in the U.S.). This incorporated all the features normally provided as 'add-ons', such as a built-in sequencer for programme cartridges, Music, Spot, tape and turntable remote cueing via a telephone key pad, phase meter; machine and mic timers which reset and start at turn-on; stopwatch clock; Telco inputs, etc. Their TS-24 in-line master recording console also looked good, with optional disk-based automation.

Audio Kinetics, better known for their Q-Lock Synchronisers, introduced a novel Master Mix computer control system to interface with such automation-ready consoles as MCI JH50, Allison 65K, and Melkuist GT800—or almost any console capable of being retrofitted with Master Mix faders. Using a simple keyboard, the system gives computer control of whole mixes, mix-merging, VCA grouping of up to 64 faders, and 'refinement' as the composite of previous and current fader adjustments. Only one audio track is required for SMPTE/EBU time code and a 5.25-inch floppy disk gives 600K bytes of data storage.

Audix also had an ACCESS assignable console, made to look uncluttered by relegation of the floppy disk storage and all signal processing circuitry to a remote rack. The newest Audix Intercom/Talkback system was also floppy disk controlled, carrying up to 60 simultaneous channels to 120 outstation terminals.

ANALOGUE RECORDERS

Though mainly scanning the IBC booths and paper sessions for purely audio developments, I spent some time examining a 1-inch Type C video tape recorder which has been developed jointly by Ampex and Kudelski/Nagra. This VPR-5 is truly portable, weighing only 6.8kg (15 lb.), and could even be carried by the same (strong) man who is wielding the video-camera. Yet it incorporates dual-cue editing, seven built-in audio filters to match scene acoustics, two audio channels plus SMPTE/EBU time code, and 20-minute tape reel (with optional one-hour reel).

The latest version of the classic Nagra IV-S professional portable recorder now has the standard SMPTE/EBU 80 time-code generator/record-playback system. This gives assured synchronisation of sound and picture, even in multiple camera situations, to less than 100 microsecond accuracy. The advantages for editing, dubbing, and post-production work are obvious. Sony also showed a novel Nagra-like portable, the APR-2003, with many of the same facilities—and the possibility of adding a Dolby A noise reduction adaptor.

A sure sign that analogue tape recording still has potential for refinement was given by the first UK showing of the Studer A820. The degree of user control was magnificent, with 40 functions all programmable via the operating-keys. Operator comfort was maximised in the new padded low-profile deck. Selecting Play causes the capstan to start rotating, with a defined ramp, only after the tape has contacted it. Tape strain is further avoided by the capstan motor's gentle acceleration and deceleration characteristics (it can even go into reverse). A shuttle control can wind the tape in either direction at any desired speed, programmed for tape on heads or withdrawn, and gives very accurate edit point location. A very full LCD readout covers internal alignments, menu of options, roll-back with programmable time, spot erase, even half-tone varispeed—the list seems endless. Selection of any function can be ascribed to any preferred push-button (soft key), and a service lock protects the programmed functions against tampering. Studer's new TLS 4000 synchroniser is another product that advances analogue handling capability to new heights.

DIGITAL AUDIO

The DASH (Digital Audio Stationary Head) format has had its critics, mainly amongst manufacturers who felt shut out because they already had viable digital recorders which did not conform. Nevertheless, it appears to be gaining momentum in designs by Studer, Sony, Matsushita, and others. Sony's PCM-3324 24-track DASH recorder is amazingly versatile, with two

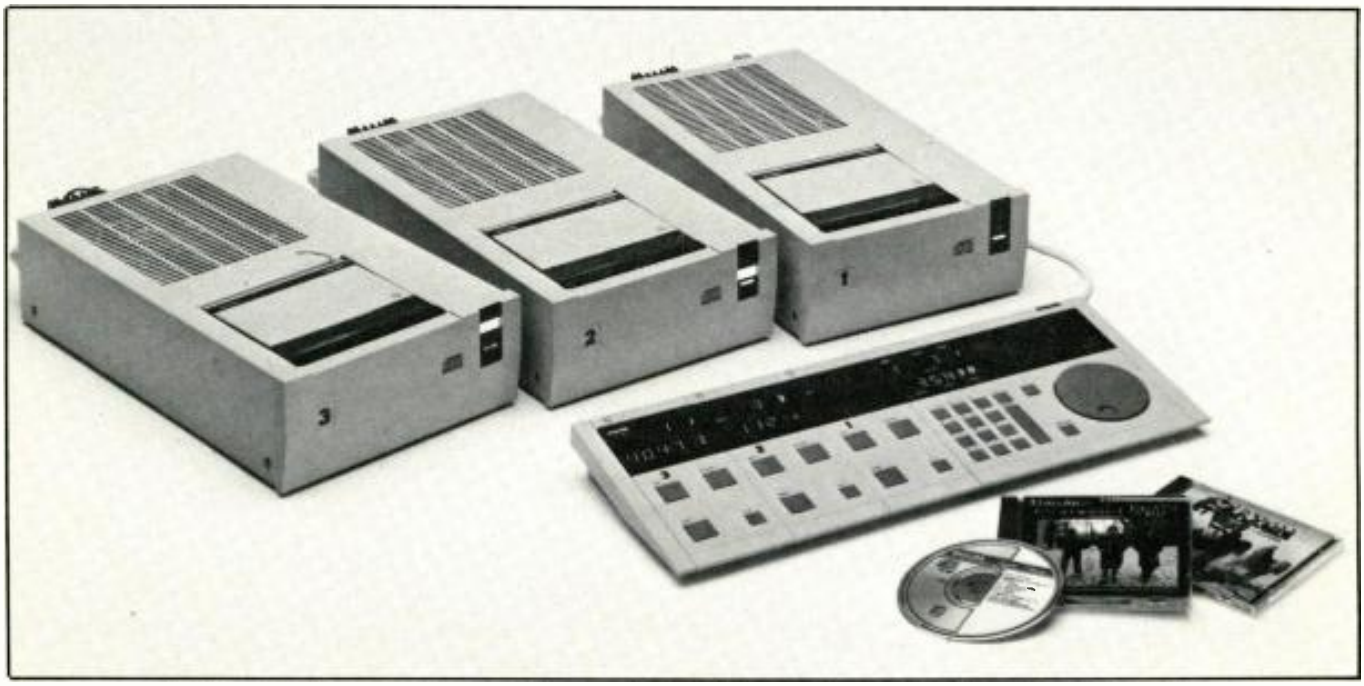


Figure 6. A "triple" configuration of Philips' new LHH 2000 Professional Compact Disc Player System.

Figure 7. Transportable elliptical antenna from GEC McMichael Marconi Studio Systems Division.

analogue channels for monitoring, a control track, and an external data track, as well as 24 audio signal channels. Four channel groupings can be set up, tape-cut or electronic editing is possible, five auto-locate cue points are available, and there are crossfade punch-in and punch-out facilities, variable shuttle, and analogue and digital inputs/outputs. It was demonstrated with a new RM-3310 remote controller, and there was also a neat PCM-3012 2-track DASH recorder.

The longer established combinations of a digital audio processor with some form of video tape storage continues, of course, with the Sony PCM-1610 adopted as the standard mastering system for Compact Disc manufacture. Sony's small portable PCM-F1 and PCM-701 can now be upgraded via boxes from Audio & Design and others to correct for interchannel phase shift, etc. There are even editing aids, such as CLUE (Computer Logging Unit and Editor), which HHB premiered at the Show. As well as editing, it provides automatic logging with autolocation and a printed record of the session if required.

Broadcasters are studying the Compact Disc as a programme source, and the equipment manufacturers have been busy readying hardware to exploit CD's quick-cue virtues. The Philips LHH 2000 system uses a modular microprocessor-controlled unit to drive up to three small CD players (only 9.5 inches wide). Any point on the discs can be accessed in less than 2 seconds to 13.3 millisecond accuracy (one frame), and the LED display identifies Edit, Ready, On-line or On-air modes, with a time countdown to Stop/Cue.

Matsushita was also showing a professional CD player, the Technics SL-P50P, with a numeric key pad for typing exact time-code cues, a search dial (like rocking an analogue disc for quick-start cueing), and automatic search functions. Auto-changers for CD are also beginning to appear, such as the Technics SL-P15 which can jump-cue to any track on any one of a magazine of 50 Compact Discs. They also claim to be able to stack a further four magazines on top, for a total of 250 CDs. (Sony spoke of, but did not show, a 120-CD player.)

The most innovative laser device was Matsushita's TQ-2023F. This is an optical disc recorder which will record either 13.3 minutes of motion colour video or 24,000 still pictures, with two-channel sound onto an 8-inch silver disc. It gives direct read after write (so the buzz-word acronym is 'DRAW') but is non-erasable. Any NTSC composite video source can be used—camera, VCR, or tuner—and access time to any frame on the disc is an incredible 0.5 seconds or less. Variable speed play is extremely versatile, with on-screen display of frame numbers, etc., when required. Audio response is 50 to 20,000 Hz ±3 dB, with noise at -70 dB and distortion at 0.5 percent max.

If you add satellite dish antennas parked in an outside display of mobile vehicle, hi-fi audio on video, cable demonstrations (including optical fibres), electronic 'paint boxes,' and computerized everything, you will see that broadcasting—and all our lives—are on the brink of a technological revolution of vast proportions. This "IBC 84" Convention, sponsored jointly by the IEE, IEEE, IERE, EEA, RTS, and SMPTE, was more exciting than any other I have visited. The papers programme included many forward-looking presentations, all available to attendees in a 418-page bound volume on registration. ■

SUSAN BOREY

Sound On Broadway, Part 2

In the second part of this series, ***db*** *takes a closer look at the applications of sound in Broadway stage productions by taking in a few shows. Come with us as we visit New York's exciting theatrical world.*

CONTINUING OUR COVERAGE of sound on Broadway, db visits three productions and reports on the technicians, equipment, and techniques involved in this challenging facet of audio engineering.

THE REAL THING by Tom Stoppard

Plymouth Theatre

"Doing sound forces you to pay attention all the time," said Barbara Schwartz, who orginally came to New York to be a lighting designer, gravitated into audio, and is now on the brink of designing her first show for an off-Broadway production company. Although *The Real Thing* doesn't require mixing a live orchestra, nearly 50 moves with tape call for split-second timing and careful manipulation. The show utilizes many practicals, objects set up onstage that function mechanically as they would in real life with audio signal supplied by the sound person. Two home stereo systems, a cassette deck, and a radio play integral roles in the action, and Barbara must make their nonchalant manipulation sound completely natural as the actors switch, place tonearms, pound, and drop cassettes while speaking their lines.

Contributing to Barbara's challenging task is her position at the back corner of the theater; her view of the practicals is often blocked by the actors. "They are very good at cooperating with my needs," declared Schwartz. "They have learned to telegraph the position of a tonearm with their elbows, for example." Telephones and doorbells, which are also the province of sound, are run from backstage in this show.

Since a lot of cues start onstage, Schwartz finds it easier to take cues from the actors rather than from the stage manager via headphones. "I'm more in touch with the actors here than with most dramatic plays," she maintained.

Schwartz is given fairly strict instructions by the show's designer, Otts Munderloh, as far as levels go, especially with the practicals. "I'm expected to feel out the fullness of the house, though, and exercise a mixer's discretion over general levels. The actors call the shots," she said, concerning the extent to which she manipulates the dynamics of the vocal reinforcement. "I have to feel applause start up or a laugh die out, though, and work around that."

Susan Borey is a frequent contributor to Modern Recording & Music *magazine.*

Barbara takes about 40 minutes before the audience enters the theater to check all the equipment. Using a tape she is very familiar with, she runs through the 38 speakers in the house which comprise three speaker systems. Twenty-four ADS 3300s, half of them hanging off the lighting truss with the rest mounted under the balcony, provide all the vocal reinforcement except for one scene that takes place in a television studio. For that, a pair of Meyer UBAs carries train noises and the actors' voices through wirelesses, providing a change of imaging which helps differentiate the scene for the audience. In addition, 12 JBL 4212s are used exclusively for musical transitions during blackouts.

Picked primarily for their remote memory, which enables an operator to program load locations in minutes and seconds and call up tape cues at the push of a button, the two Otari MTR 10s garner nothing but praise from Schwartz. "They make life a lot saner for me," she declared. "I have to get to many of the cues very fast, and a lot of them come extremely close together. Also, noise is a prime consideration here and these decks are dead quiet. This show has a wide dynamic range with some very quiet moments. The director, Mike Nichols, has highly attenuated senses, and he can hear the clunk of a transport. You can't just slam home a cart in a tender moment." To further assure a lower noise level, the decks are boxed in. The effects are on two tapes, necessarily run through two decks, because there are places that call for two effects at once. Barbara has three sets of backup tapes. Equalization is used only on the vocals; the sound designer was able to get studio masters for almost all the cues, and they were dbxed.

"THE REAL THING"—EQUIPMENT

HOUSE

Main Speakers—24 ADS 3300; 12 JBL 4612; 2 Meyer UBA

Amps—Crown DC300, DC75; Urei 6300; Altec 9440

Board(s)—Stevenson Interface, custom built 16 in. × 8 × 4 × 2, and 16 in. × 4 × 2 × 1 (for vocals and tapes)

Equalizers—Orban parametric 622B; Altec 1650 1/3 octave graphic

Effects—Klark-Teknik DN 700 digital delay

Microphones—5 AKG 451 with CK1 heads; 3 Sennheiser MKH816P shotguns; 1 AKG 441; 1 Shure SM 58; 2 Cetec Vega 77BD wirelesses

TAPE DECKS

2 Otari MTR 10 with Otari CT 109 auto-locator

SHIRLEY MACLAINE ON BROADWAY

Gershwin Theatre

Although officially a Broadway production, Ms. Maclaine's panoramic solo extravaganza is closer to a Las Vegas-style show in style and technical requirements. A 30-piece orchestra situated onstage is only one of many

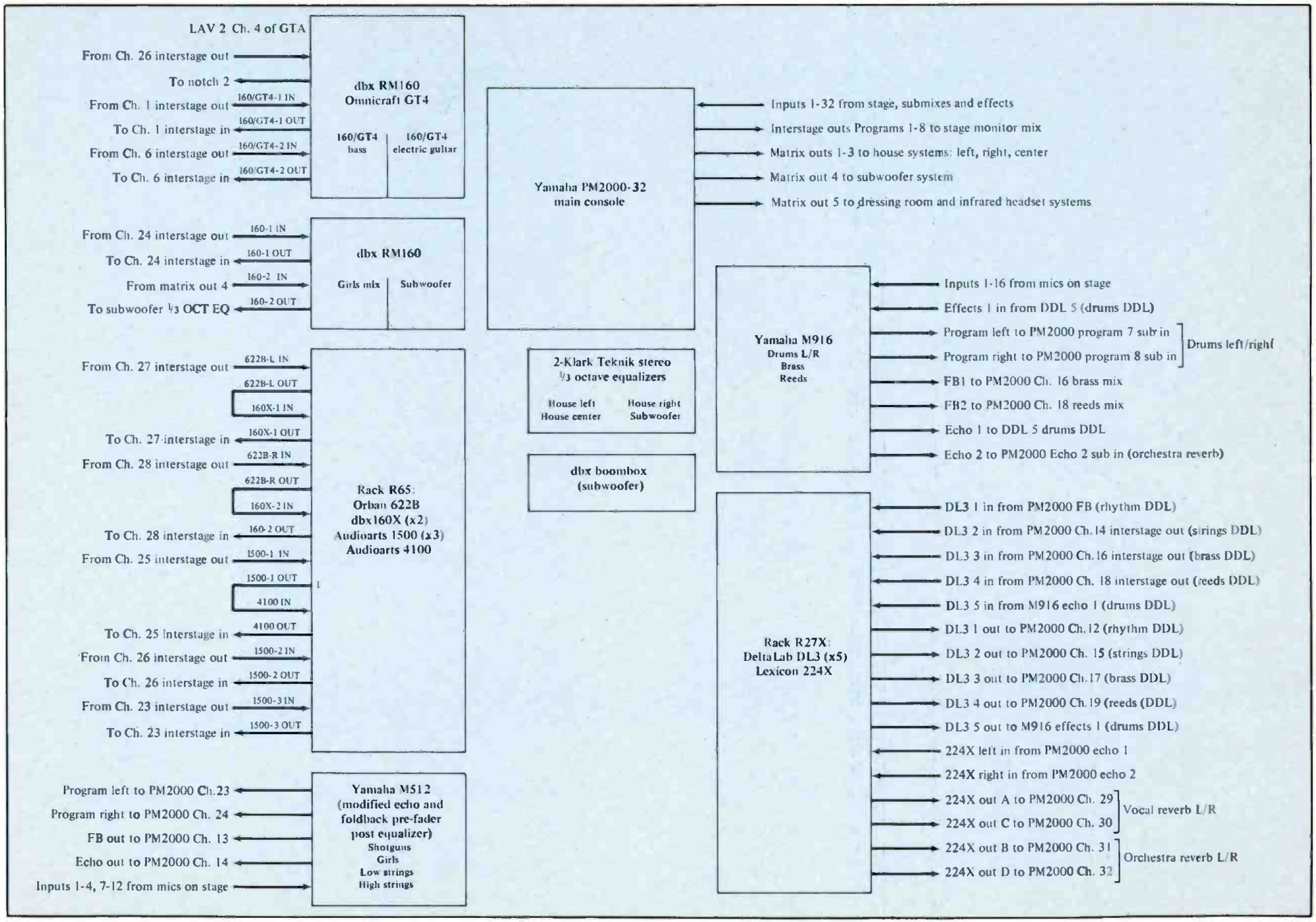


Figure 1. Shirley Maclaine Show house mix.

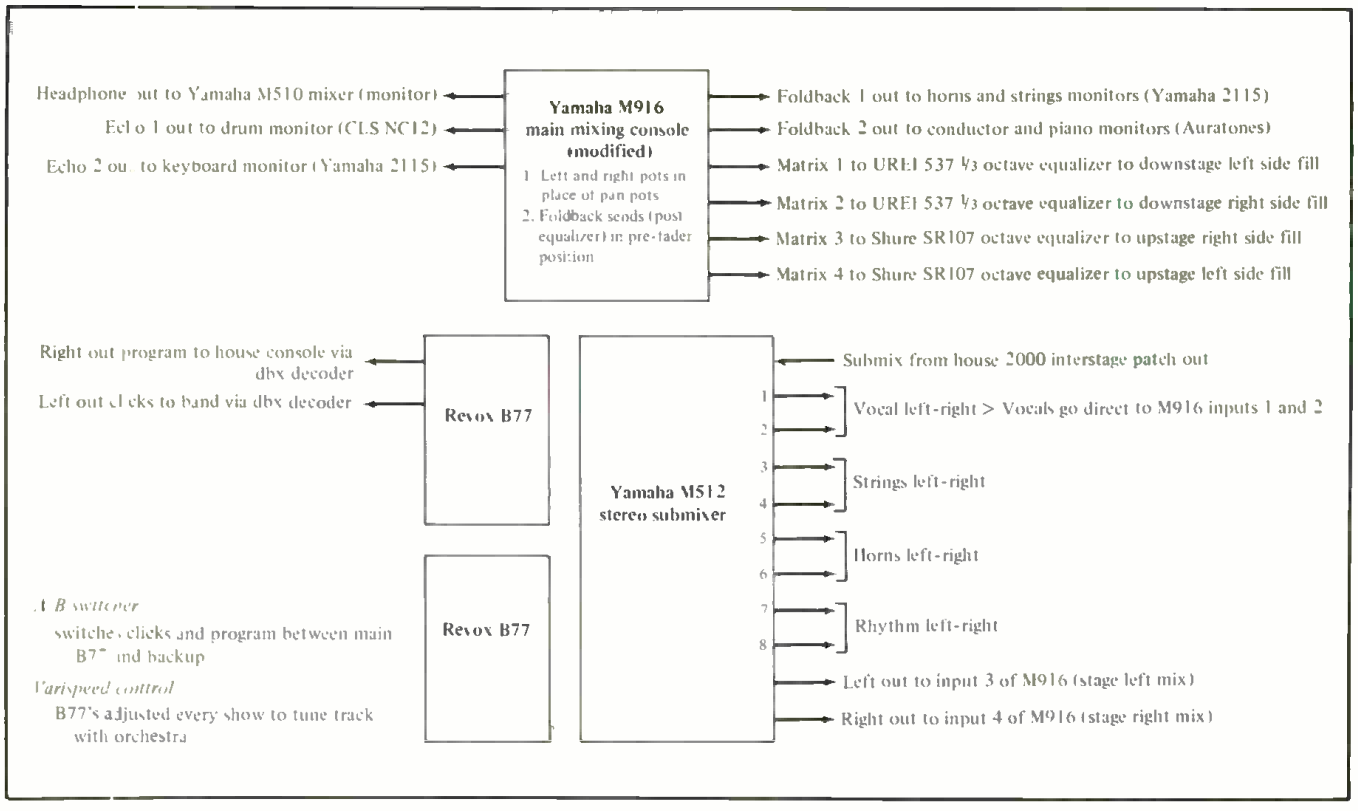


Figure 2. Shirley Maclaine Show monitor mix.

differences in the non-stop performance, which features singing, dramatic interludes, and dancing.

Most Broadway shows are mixed by one engineer; Maclaine's was no exception until Keith Hubbell and Ken Newman took over the extensive responsibilities in August 1983. Now that the increased capabilities afforded by bringing in the team from the L.A.-based A-1 Audio have yielded exemplary results, the sound production for the show has evolved beyond a task one person could effectively handle.

"Ken just has to worry about getting it to sound the best he can in the house," said Hubbell, who engineers the six monitor mixes (two sidefill mixes, orchestra, keyboard player, drummer, and conductor) and runs tape decks from a nook offstage. "The typical Broadway sound system is almost an afterthought; you're not supposed to notice it. In the Vegas and Atlantic City tradition, however, it's a more razzle dazzle, sock it to 'em sound, which is still a lot quieter than rock 'n' roll, decibel-wise."

Part of the dazzling here is achieved through the rare use of a stereo house and monitor mix, an exception to Broadway's typical mono presentation.

"Stereo helps makes things a lot more discernible," said Newman. "Since the orchestra plays such a large role in the show, it gives the ear a chance to make out individual parts. It also permits special effects, like the panning of a whistle from one side of the stage to the other."

The overriding concern with the show lies with accommodating Maclaine's sensitivity and spontaneity, and runs from small details like the specially designed lexan mic clips which allow one-handed attachment and withstand forceful manipulation, to the personalized approach to Shirley's monitor mix.

"What is somewhat unique is that I get the submixes in stereo with reverb from Ken's board, which I can feed into her monitors just as Ken is mixing them in stereo,"

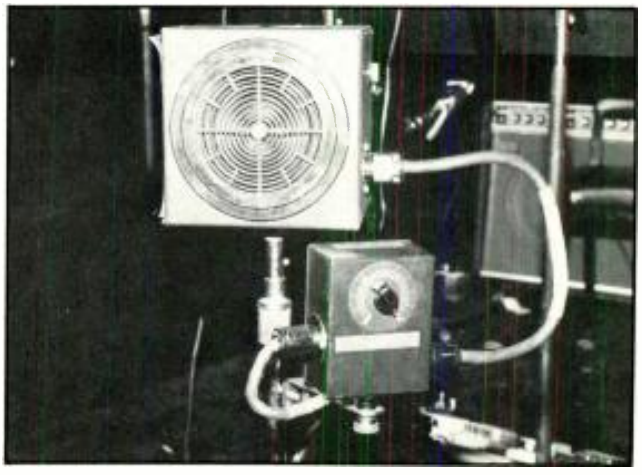


Figure 3. Piano monitor (Shirley Maclaine Show).

relayed Hubbell. It makes the whole thing sound as one, not just monitors and house. This is especially useful in this show because there are some very quiet moments where you basically have the monitors filling in what bounces back from the house. I bring it in mostly in the mid-range, which covers her voice. I EQ them with the low end rolled off except for 63 Hz, the kick drum, which she often likes to have punched, especially when she's dancing. She doesn't like to hear the monitors blasting at her, so I give her more of a full sound that seems to come from all around her. My situation is different than most monitor mixers'. They seem to be fighting the house all the time as they try to get as much level as possible before feedback. With this show, it's so quiet that I have to fight to be subtle."

The effervescent Maclaine wants the opportunity to improvise during her performance; the only way this is possible is with the assurance of consistency in the aural

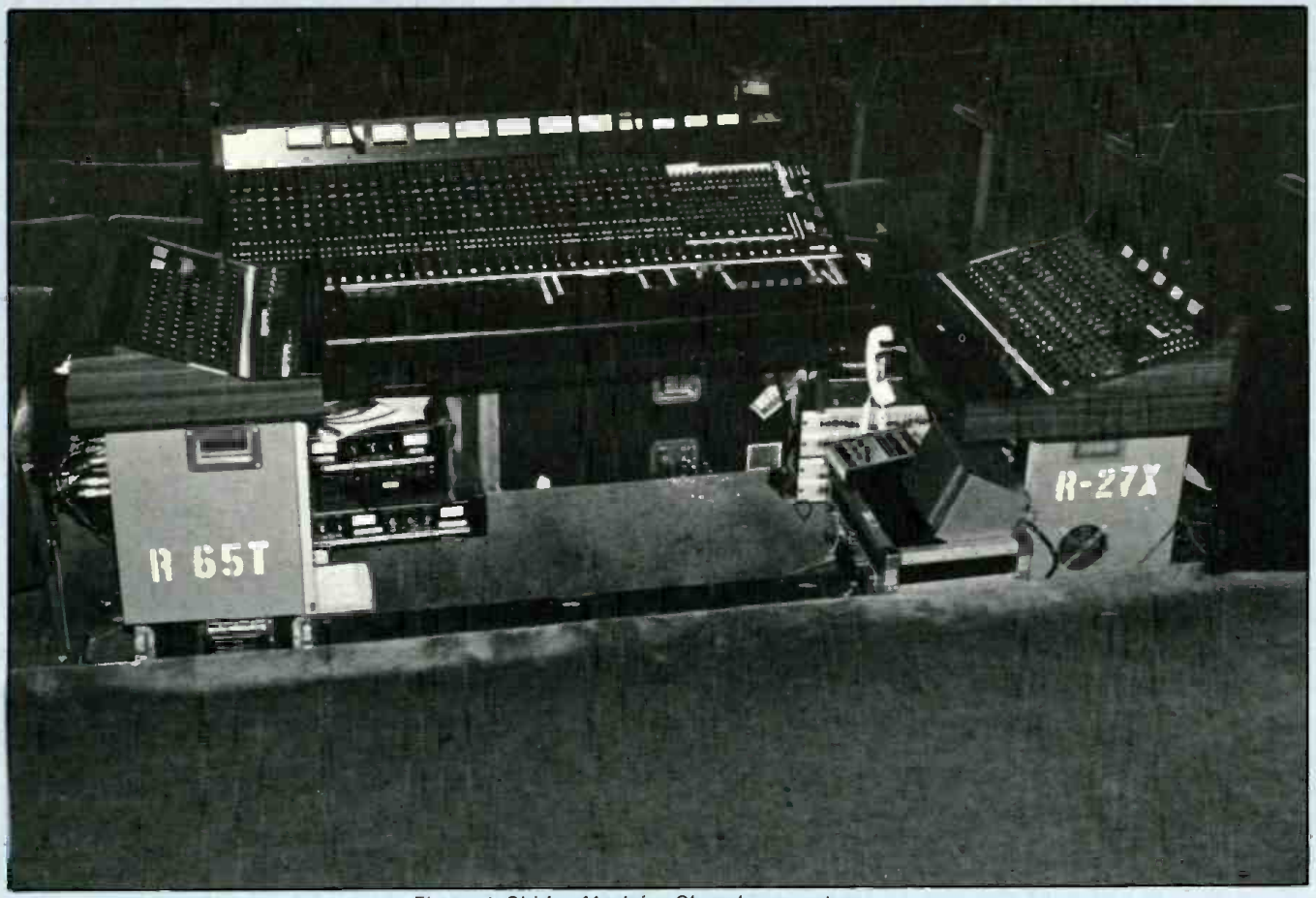


Figure 4. Shirley Maclaine Show house setup: Yamaha PM2000, M918, and M512.

Figure 5. Shirley Maclaine Show click-track system machines.

Figure 6. A-1 Audio custom JBL 4320 cabinet with Gauss 4580 and JBL 2441 compression driver and 2402 tweeter.

backdrop. If there's an oboe solo, for example, she expects it to come up out of the orchestra at the same instant and at the same volume each time. It's Ken's and Keith's challenge to deal with the human fallibilities of the musicians and compensate for their occasional inconsistencies.

A somewhat complex move which combines the precise cooperation of the engineers, orchestra, and performers involves the use of a click-track tape. One side of the tape, which is activated by the conductor who presses a button at Shirley's cue, is the click track; a count of "1, 2, 3, 4" goes to the conductor and percussionists via headphones. The other side of the track is the program. It goes to the audience and consists of appropriately timed fingersnaps of the dancers, which cannot be mic'ed. As the orchestra plays, the fingersnaps coincide with the dancer's pantomime at a precise moment in the score.

Maclaine never leaves the stage during her performance, and the reliability of the equipment which supports her during the non-stop show is of prime importance.

"Shirley doesn't want to think about anything but her performance when she's out there," said Newman, who noted that if a wireless mic's body pack goes out during the show, there's not much he can do about it except go along with Maclaine's hopefully graceful transition to a hand-held mic. "When a failure occurs, whether it's at the technical end, in the orchestra, or some thing that happens with the audience, it can really throw her concentration off."

Fortunately, technical failures have been a rare occurrence; this is due in part to rigorous tests applied before each performance.

"SHIRLEY MACLAINE ON BROADWAY"—EQUIPMENT

HOUSE

Main Speakers—Altec 15-inch woofers, JBL 2441 drivers on JBL 2350 horns, JBL 2420 tweeters (A-1 Audio also brought in a sub-woofer system to enhance the low end with 4 rear-loaded horn cabinets with Cetec Gauss 15-inch speakers)
Amps—house, Yamaha P2200; sub-woofers, BGW 750 with dbx Boom Box
Crossovers—Yamaha F1030
Board(s)—Yamaha PM2000; submixers, Yamaha M916 stereo, Yamaha M512
Equalizers—vocals, Orban 622B with dbx 160X limiter, DeltaLab DL3 digital delay, Lexicon 224X digital reverb, Audio Arts 1500 notch filter, Audio Arts 4100 parametric equalizer on footlight mics

MONITORS

Main sidefill, located downstage—JBL 4320 with high-powered drivers
Upstage sidefill—JBL 4680 columns with 10-inch drivers
Amps—BGW 750
Crossovers—JBL 5234 stereo
Tape Decks—Revox B77
Orchestra—conductor, 2 mic stand-mounted Auratones with volume control, powered by BGW 100, Sennheiser HD44 headphones; percussionist, Auratone; drummer, Community NC12, powered by BGW 100, rest of orchestra, Yamaha 2115, powered by BGW 750

EQUALIZERS

Downstage, Urei 539⅓ octave graphic; upstage, Shure SR 107
Baritone Saxophone—AKG C414
Trombones—Shure SM 58
Trumpets—Beyer 69
Violins—AKG 451
Cello—Shure SM 18

MICROPHONES

Shirley Maclaine—hand held Cetec Vega wireless Dynex II system with Shure SM 87 capsule
Back-Up Vocals—Nady
Piano—Helpinstill 155 through Yamaha PM 180 mixer, also mic'd with AKG 451
Drums—kick, Sennheiser MD421; snare, 2 Shure SM 57s out of phase, high-hat, ECM 30 with Sony lavalier; toms, Senneiser MD 421; overhead, AKG 451
Electric Guitar—speaker mic'ed with Countryman EM 101
Acoustic Guitar—AKG C414EB
Bass—Direct Input
Keyboards—Direct Input
Drum Machines—Direct Inputs
Percussion—tympani and xylophone, Shure SM 81 bells, congas, and vibes, Countryman EM-101; chimes, AKG C451
Saxophones—Beyer 88

"DIGITAL AUDIO IS TRANSFORMING US ALL."

—Giorgio Moroder

"I've heard people say they really wanted to hate digital audio. But, of course, they couldn't. Because nothing sounds as real as digital." So begins Giorgio Moroder, the award-winning composer/producer and owner of one of the world's most extensive Sony digital installations—three 24-track digital recorders and one PCM-1610 mastering system.

"Listening to digital is truly an ear-opening experience. You can't even tell if what you're hearing is a first generation track or a tenth. The fidelity is absolutely incredible."

And these are just a few of the reasons why so many top recording artists and producers, like Moroder, Phil Ramone, Neil Young, Elliot Mazer, Frank Zappa and Nile Rodgers now own or use Sony DASH-standard digital equipment.

"After all," Moroder explains, "I want my studio to be compatible with studios the world over and Sony has set the standard. And, of course, Sony has led this transformation right from the start."

We couldn't have said it better ourselves.

SONY.
The Leader in Digital Audio.

A scene from Giorgio Moroder's rescored version of Fritz Lang's 1926 film classic "Metropolis," which includes the world's first totally digital sound track.

NOISES OFF by Michael Frayn

Brooks Atkinson Theatre

Despite the complex mechanics of the plot, which requires a revolving stage to portray the onstage and backstage antics of a troupe of bumbling British actors, the sound requirements for *Noises Off* are fairly simple. Adapted from a production at Kennedy Center, the show was transplanted to the relatively small and shallow Brooks Atkinson Theatre, which needs very little reinforcement in the house for adequate coverage. In fact, all of the artificial sound in the show is used for effects, but is nevertheless indispensible. During the second act the audience sees the theater set from a backstage vantage point; along with the frantic activity behind the scenes, the actors vainly attempt to carry on with their play, which now takes place upstage, mostly out of the audience's view. However, hearing them fumble their lines is an integral part of the hilarity, so the rear-facing set is mic'ed at six strategic points that the actors play into. Originally staged in Britain, the current production still uses the small Tram microphones that might have been replaced here by Sony lavaliers. These microphones also help differentiate tone between the two sets of dialogue.

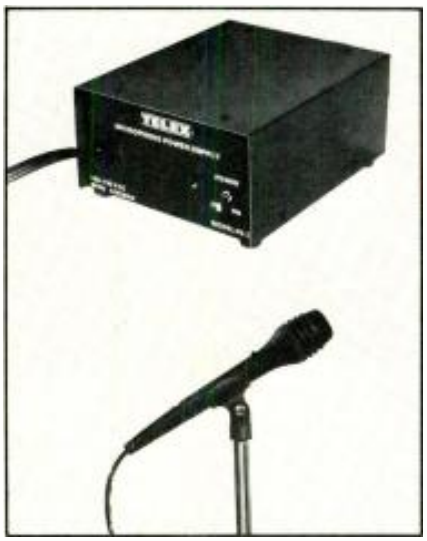
Jim Spradling, the soundman, essentially adheres to levels assigned by the designer, but makes minor adjustments when necessary. Usually handling lights, Jim was thoroughly familiar with the show and it was deemed most convenient for him to step into the audio engineering slot upon the regular mixer's absence. He sees the growing sophistication of audio in the theater as a natural response to technology, not just by designers, but for audiences as well. "With movies and TV being what they are today, people have come to expect a lot more presence and sound in live theater, as well. That's why it's escalating," Spradling said. "Also, with technology increasing, you can do a lot more."

The show uses six tape cues, which Jim runs, and a telephone, which is run by the stage manager, who sees the activity onstage via a video monitor system. ■

"NOISES OFF"—EQUIPMENT

Speakers—5 Altec 9849; 3 JBL 4690
Amps—Crown 2400
Board—Yamaha PM 1000
Equalizers—Urei 562 feedback suppressor; Urei 137⅓ octave graphic
Effects—Lexicon 92 and 93 digital delay
Board monitor—Galaxy Hotspot
Microphones—Tram TR 50 lavalier; AKG 451 footlight; Shure SM 58
Tape deck—Otari ARS 1000

NEW MICROPHONE LINE



• Telex Communications' new series of professional microphones includes the TD11 and TD13, two unidirectional microphones with a newly designed, smaller and lighter dynamic element. These microphones have a wide frequency response with the slight "presence" rise in the mid frequencies that has become very popular with vocalists. Other new models in the line include the TE10, a handheld unidirectional condenser microphone with a completely new shock-mounting technique that suspends the element on three flexible "fingers" inside a double layer of pop-proof material within the mic head, virtually eliminating handling noise. The new PS2, a two-channel portable power supply, is available to provide phantom power if it is not available through the sound system console.
Mfr: Telex Communications, Inc.

Circle 27 on Reader Service Card

WIRELESS HEADPHONE SYSTEM

• Windrose Industries and Communications' new stereo wireless headphone system, available as the IRT 210 or 200, features an infrared headphone receiver with volume controls in a streamline design using a bulb-type non-directional infrared sensor, which allows maximum freedom of movement. This innovative system uses an infrared emitter to send stereo signal to multiple headphones up to 35 feet away. For example, the user can plug the infrared emitter into the stereo headphone output and provide signal for up to 30 wireless receiver headsets.
Mfr: Windrose Industries & Communications

Circle 28 on Reader Service Card

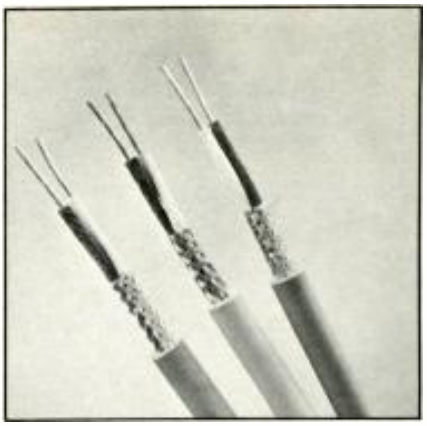
NEW DIGITAL MASTERING TAPE

• Agfa PEM 297 D is a new highly dependable ¼-inch digital audio mastering tape, available on 4,600-foot, 10½-inch reels. The new material features fewer dropouts, lowering the need for error correction and reducing the possibility of program loss. Because of superior slitting, the new tape offers better winding, better handling, and less chance of edge damage. PEM 297 D has the consistency associated with all Agfa products and provides dependable, high performance.
Mfr: Agfa-Gevaert Inc.

Circle 29 on Reader Service Card

MICROPHONE CABLE

• Belden microphone cable is now available with red, yellow, and blue thermoset jackets. Colored microphone cable permits easy identification of individual microphones and enhances their appearance. The color is applied to the jacket without sacrificing the physical or electrical properties of the cable. Belden 8412 has two 20 AWG (26 x 34) stranded, tinned-copper conductors insulated with rubber. There is an overall tinned-copper braid shield providing 85 percent coverage. Nominal capacitance is 36 pF/ft. (capacitance between conductors) and 65 pF/ft. (capacitance between one conductor and remaining conductor connected to shield). Belden 8413 has two 24 AWG (45 by 40) stranded, tinned cadmium bronze miniature conductors insulated with rubber. The cable contains a 100 percent coverage conductive textile wrap shield and a 60 percent coverage tinned copper braid shield. Nominal capacitance is 36 pF/ft. (capacitance between conductors) and 65 pF/ft. (capacitance between one conductor and remaining conductor connected to shield). Belden 8412 is available in 25, 50, 100, 250, 500, and 1000 foot standard lengths; Belden 8413 is available in 15, 25, 50, 100, 250, and 500 foot standard lengths.



Mfr: Belden Electronic Wire and Cable
Price: 8412-1000 ft., $419.40
8413-500 ft., $201.70

Circle 30 on Reader Service Card

UPDATED REVERB

• Ursa Major's new version of the 8X32 digital reverberator, named the 8X32-MkII, has increased flexibility over the original model without sacrificing features or sonic quality and, thanks to improved manufacturing technology, sells for about 30 percent less. The 8X32-MkII is the result of extensive marketing and technological research. While maintaining the specifications and user-friendliness that have made the 8X32 so popular, the 8X32-MkII features four additional user-modifiable reverb programs. The four new programs are: Cask—a short colored program with a maximum decay time of two seconds; Percussion Plate—an uncolored program with an explosive buildup and a maximum decay of four seconds; Chamber—a bright program especially suited for vocal tracks; and Reverse Reverb—a fascinating "backwards reverb" program with a maximum decay time of 20 seconds. These join the original programs of the 8X32—Plate I, Plate II, Hall, and Space—making a total of eight distinct reverb programs. As on the original 8X32, the user has complete control over reverb parameters within each program, including delay and level of first reflections, time and level of initial reverberation, and length of final decay, as well as separate low- and high-frequency decay. All parameters appear on LED displays on the unit's front panel, and all control settings can be stored in the 8X32-MkII's 64 non-volatile memories for instant recall.

Mfr: Ursa Major, Inc.
Price: Rack-mount version, $4,000
Remote-control version, $4,695

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NEW WOOFER SERIES

• Electro-Voice's DLX and new DLW woofers are designed for high-efficiency, high accuracy sound reproduction. The previously introduced DLX low-frequency reproducers are now joined by the DL15W and DL18W very-low-frequency reproducers. All models in the DL series offer a carefully engineered drive system which assures high-efficiency, linear, low-distortion output, and high power capacity. DLX low-frequency reproducers, available in 12-, 15-, and 18-inch diameters, maximize efficiency and provide the low-frequency response extension that is desired for most professional sound applications. Both vented and horn enclosures are appropriate. the DL15W and DL18W exchange some efficiency (about 3 dB) for more extended bass response and greater peak output in the bass region. In the same size enclosure, the DLW provides even more extended bass response. With greater linear cone excursion, the DLW reduces harmonic distortion and produces tight, high-impact bass at frequencies below 40 Hz.

Mfr: Electro-Voice

Price (pro net): DL12X, $192.00
DL15W and DL15X, $205.00
DL18W and DL18X, $320.00

Circle 32 on Reader Service Card

NEW POWER AMPLIFIER

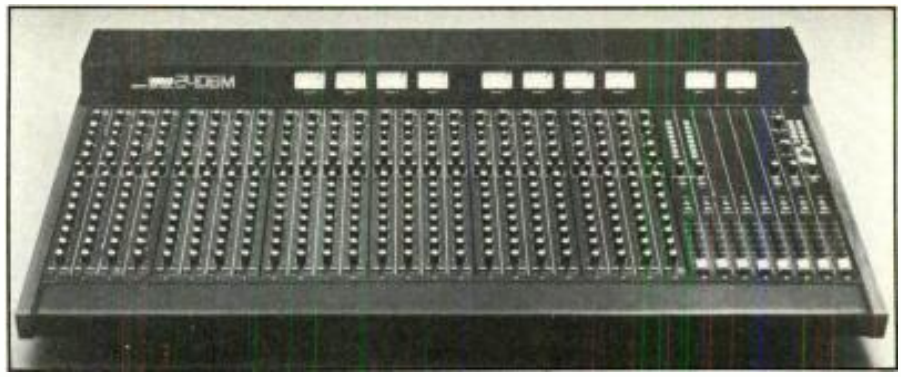
• The Kinergetics KBA-200 high power amplifier offers the sound quality of tube equipment at an affordable price. The amplifier utilizes circuitry to compensate for the hysteresis distortion inherent in mechanical and electronic devices. The patented amplifying circuits used by Kinergetics in the KBA-200 attack the problem of distorted amplification and provide a vast improvement in the accuracy and definition of transient sound. In addition to its unique compensation circuitry, the KBA-200 includes a cascade input to extend the full power bandwidth to the 400,000 Hz range. The KBA-200 delivers 200 watts/channel into 8 ohms with 3 dB of headroom. It is a dual monophonic design with right and left channels totally separate. The KBA-200 joins the KBA-100 amplifier and the KPA-1 preamplifier as the first components designed from the ground up to provide amplification free of annoying hysteresis distortion. Kinergetics plans to add a tuner to its product line early in 1985.

Mfr: Kinergetics, Inc.

Price: $1495

Circle 33 on Reader Service Card

NEW MONITOR CONSOLES



• The latest additions to the new Yamaha MC Series of mixing consoles include the MC1608M and MC2408M monitor mixing consoles, designed with a specific purpose in mind—flexible and efficient on-stage monitoring for performing musicians. The MC monitor consoles are completely compatible with all professional equipment, since all primary inputs and outputs are balanced lines with XLR connectors. The MC1608M has 16 input channels and the MC2408M has 24 input channels. Other than this, the consoles have the same basic features, with 10 meters, eight master outputs, as well as two auxiliary sends and two fully assignable auxiliary returns. The MC1608M and MC2408M are modular in construction, with blocks of four input channels for easy service when necessary. Each input channel features a pad switch and gain control with peak LED, phase reversal switch, three-band equalization with sweep midrange, two post-EQ and pre-fader auxiliary sends, eight rotary master send controls, Channel On/Off and Cue switches with input channel cue priority (explained later). All knobs are color-coded between input and output sections to aid identification in low light situations and make the visual signal flow easy to follow. The Master section includes a 100mm fader, On/Off and Cue switches, and high pass filter for each of the eight master outputs. The two auxiliary returns, with Cue and On/Off switches, are assignable to any or all of the master outputs. The front panel also includes a headphone cue section and assignable talkback section, each with a level control. The low profile meter panel shows output levels for masters 1-8 and auxiliary 1 and 2. One of the unique features of the MC monitor consoles is *input channel cue priority*. When an input channel Cue switch is pressed, the previous master cue is cancelled. While using a monitor console, the engineer will normally monitor one of the eight master outputs. If, for example, a feedback problem arises, the engineer can press one or more input channel cue switches, and the console automatically replace what was previously being monitored with the new channel's program. The back panel has low-impedance, electronically-balanced XLR and high impedance ¼-inch inputs and channel insert in/out connectors for each input channel. Each module of four channels has a +48 phantom power On/Off switch. Each of the eight master outputs has an insert in/out jack, submixer input, and an electronically balanced XLR master output connector. ¼-inch jacks are provided auxiliary 1 and auxiliary 2 in/out, and auxiliary submixer inputs. Despite this wealth of features, the MC monitor mixing consoles and the other MC consoles are surprisingly slim and light.

Mfr: Yamaha International Corp.
Price: MC1608M, $2895
MC2408M, $3995

Circle 34 on Reader Service Card

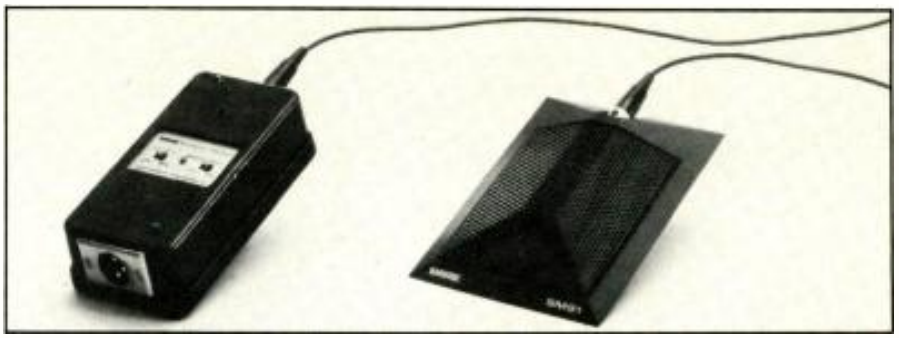
PROGRAMMABLE SIGNAL GENERATOR

• Hewlett Packard's new HP 8656B combines performance, quality, and economy in a programmable signal generator. The unit provides frequency coverage of 100 kHz to 990 MHz, an output range of +13 to -127 dBm with 0.1 dB resolution, flexible AM and FM, 50-watt reverse-power protection, and standard HP-IB programmability (all features of the predecessor HP 8656A). The new model has several added capabilities due to customer request. These include faster throughput, higher reliability, single-sideband radio testing, low-rate digital tone testing, voltage-controlled oscillator simulation, and phase-sensitive device characterization.

Mfr:Hewlett Packard
Price: $6500 (list)

Circle 35 on Reader Service Card

NEW UNIDIRECTIONAL CONDENSER MICROPHONE



• Shure Brothers Inc.'s SM91 Unidirectional Condenser Microphone is a low-profile model designed for surface-mounted applications where a unidirectional pickup pattern is desirable. Like the "pressure zone" microphones in common usage, the Shure SM91 takes advantage of the well-known principle of boundary effect. Because of this principle, placing a Shure SM91 Microphone sufficiently close to a barrier or boundary will cause it to perform with as much as 6 dB higher sensitivity and approximately 3 dB greater rejection of random background noise. According to Shure, the unidirectional SM91 has been designed as a viable alternative to the omnidirectional pressure zone microphones currently in use. Since this is the first unidirectional microphone that utilizes boundary effect, they see it as having several advantages over other pressure zone models on the market. These advantages include minimized low-frequency noise and rumble, less tendency toward feedback, and avoidance of phase cancellation. In addition, the half cardioid (cardioid in the hemisphere above the mounting surface) pickup pattern of a surface-mounted SM91 permits the microphone to operate with much less reverberation and muddiness than omnidirectional surface-mounted models. The unidirectional pattern also allows for effective isolation without the need for physical isolation barriers often used with pressure zone models. At the heart of the SM91 is a new Shure-developed cartridge that provides high output plus a wide, flat frequency response for accurate sound reproduction and excellent off-axis performance. The SM91 is also supplied with a small, sturdy, low distortion, high-clipping-level preamplifier which may be powered either by two standard 9-volt batteries or by an 11 to 52 VDC simplex (phantom) power supply. The preamplifier unit also includes a 12 dB/octave low-frequency cutoff switch for response tailoring, a battery On/Off switch, and a green LED battery condition indicator. Also included is a 25-foot, small-diameter, two-conductor, shielded interconnecting cable with two 3-socket miniature Switchcraft connectors.

Mfr: Shure Brothers
Price: $300.00 net

Circle 36 on Reader Service Card

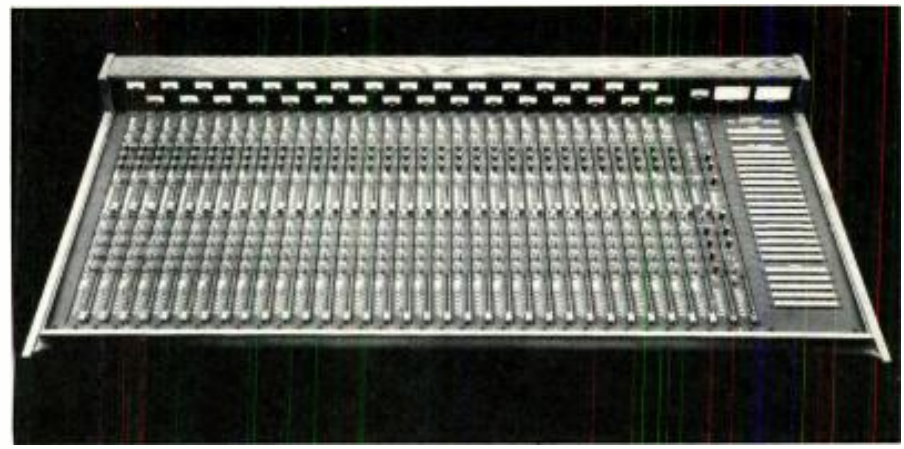
CALL-COUNTING SYSTEM

• JBL's CallCount system allows radio and television broadcasters to instantly measure telephoned audience responses to issues of public concern raised during programming. The system consists of two or more line concentrators (depending on station need), a digital recording device, and a CallCount tabulation/transcription device. CallCount hooks into any basic phone system up to 2000 lines and can handle 15 calls per telephone line per minute. As a question is posed during a broadcast, two phone numbers are provided for callers to voice their opinions and cast their ballots. Through an RS-232 interface, tallies are immediately displayed and continuously updated on screen or can be read from the readouts on the CallCount unit. For stations providing more than two voting options, CallCount is expandable to handle multiple choice questions. The data derived from audience responses with CallCount can also be utilized as a supportive tool for sales and marketing efforts and functions as another revenue-producing advertising medium. System cost is $45,000, which includes factory installation and a one-year warranty. Components are available for immediate delivery.

Mfr: JBL Inc.

Circle 37 on Reader Service Card

NEW I/O RECORDING CONSOLE



• Pulsar Laboratories' new "On Track" I/O Recording Console is a semi-automated console with standard features such as eight VCA subgroups (any channel a master) and eight programmable mutes with a VCA bypass switch. The console also includes a fully balanced patch bay, balanced Inputs/Outputs, balanced Stereo Master Output, P&G faders, PFL/AFL and tape solos, three-band sweep EQ, separate Line/Mic/Tape trims, Slate and Talkback, Calibration Oscillator, Stereo Monitor Sends, Stereo Effects Returns, 32-track monitoring, phantom power, solid oak frame and leg kit, as well as VU meters on all Inputs, Left, Right, and Solo.

Mfr: Pulsar Laboratories, Inc.
Price: $17,000 to $26,000

Circle 38 on Reader Service Card

AUTOMATIC MIC MIXER-PREAMP

• Bogen's new AMM-4 Automatic Microphone Mixer-Preamplifier is designed to provide professional performance at a popular price. The four-channel unit is intended for conference rooms, courtrooms, legislative chambers, churches, convention centers, and everywhere else that multiple microphones are required. Automatic sensing of a signal's presence at a microphone input will activate that channel instantly and smoothly. When no signal is present, the channel automatically deactivates. The AMM-4 permits simultaneous use of many microphones with more usable gain before acoustic feedback, minimal background noise, and no need for an operator. Up to forty microphones can be put to work in one meeting by bridging together up to ten AMM-4s. Four balanced, low-impedance, transformer-isolated microphone inputs are combined on an active mixing bus. This eliminates channel interaction, minimizes residual mixing-bus noise, and provides constant preamplifier gain through constant input sensitivity as channels are added. Each microphone channel has its own volume control and an LED on/off indicator. A master volume control is included, and separate bass and treble controls give up to 13 dB cut or boost at 50 Hz and 15 kHz, respectively. A built-in circuit allows remote control and/or precedence for any microphone, as well as remote control of master volume level. Precedence, in the form of "chairman's override," enables a presiding officer to maintain order during a hectic multiple-microphone session. Alternatively, descending override can be had through a wiring arrangement that assigns status to each microphone. Microphone 1 overrides microphone 2; microphone 2 overrides microphone 3, etc. A recently developed solid-state bar indicator utilizes LEDs to show output level. It is calibrated at 0 dBm and is screwdriver-adjustable to any relative setting from 1V to 6V, full scale. Transformer-isolated, the rated output is +18 dBm into a 600-ohm load with less than 1 percent THD. Frequency response is ±1 dB, 20 Hz to 20 kHz at rated output. Microphone sensitivity is 300V and noise is 75 dB below rated output. Microphone input connections are professional 3-pin female connectors. RCA phono jacks are given for Tape Monitor/To Equalizer, From Equalizer, Tone Control Circuit, AGC Source, AGC Input, and Bridging. A terminal strip serves 600- and 150-ohm outputs. Accessory model CMP/AM is a plug-in compressor board that provides volume compression circuitry. It also prevents possible overdrive. A rear-panel power switch is illuminated, and a self-resetting circuit breaker protects the unit. The AMM-4 is 3¾ inches high and can be rack-mounted with Bogen's RPK-50 bracket kit.

Mfr: Bogen (A Division of Lear Siegler, Inc.)

Circle 39 on Reader Service Card

People, Places

• **Roy Gattinella** has been appointed marketing manager for **Monster Cable**, a leading manufacturer of quality consumer and professional audio accessories. Gattinella's responsibilities include overseeing new product development, creation and implementation of brochures and other promotional materials, planning for all trade shows, and coordinating advertising campaigns. Previously, he was assistant manager of Swarthmore Music Center and cofounder of Studio One Music, both retail pro/musical instrument stores located in Swarthmore and Philadelphia, Pennsylvania, respectively. Most recently, Gattinella served as account representative for Champion International Corporation in San Francisco.

• **Don Cuminale** has joined the electronic maintenance staff as audio maintenance engineer at **VCA Teletronics**, the production, post-production and satellite broadcast division of Video Corporation of America, it was announced by Keith Andoos, Manager of Electronic Maintenance. Having come to VCA Teletronics by Mediasound, Inc., where he served as studio technician since 1972, Cuminale brings to his new post an extensive background in audio maintenance and design. His expertise will be advantageously utilized in serving VCA Teletronics' new audio studio with its ultra-sophisticated equipment and systems.

• Altec Corporation has recently completed the final phase of consolidation of all its operations at Altec's new Oklahoma City headquarters. The relocation of Altec's administration, engineering, marketing, sales, data processing, and literature departments completes the process of relocation begun in 1982. During that year all loudspeaker manufacturing was consolidated in the larger Oklahoma City facility. This operation was then followed in 1983 by the West Coast warehouse, horn and sheet metal fabrication and electronics assembly. The Oklahoma City plant was originally occupied in 1963 by Altec's University Sound Division.

• **Crown International** has announced the addition of **Bill Raventos** as microphone product director. Raventos is responsible for microphone product definition, design input, field evaluations, and working with reps and end users to determine microphone-related needs. For three years Raventos was director of technical services for Ringling Brothers Barnum & Bailey Circus at their Circus World theme park in central Florida. There he was responsible for audio, park and theatre lighting, video, projection systems, RF communications, and telephones.

• **Dolby Laboratories Inc.** has announced a new position title and increased responsibilities for **Tim Prouty**. Prouty's new title is national product manager. In addition to Prouty's current marketing responsibilities, which cover broadcast audio and audio for video, his new responsibilities also include audio products marketed nationally in the music recording industry. After several years experience in the broadcast industry, Prouty joined Dolby Laboratories in April 1979 and assumed responsibilities in the area of broadcast product market development. Since July, 1980, he has held the title of manager, broadcast audio.

• **Steve Krampf** (formerly the manager of research & development) has been appointed to the position of general manager, marketing and sales, of **Otari Corporation**. Krampf's new position includes responsibility for the professional audio, and industrial products divisions of Otari Corporation, as well as the newly-formed subsidiary division of Otari Electric (Japan), Otari Data, Inc. Concurrently, **Tom Sharples** has assumed the overall responsibility of engineering management for Otari's R&D group. **Phil Sun** has been appointed manager, technical services group, a position which makes him responsible for all aspects of service and quality control.

• **Thomas E. Mintner** has been appointed vice president and general manager of **Studer Revox America, Inc.** Mintner has served as director of Studer products at the company since 1982, and he will continue to direct all Studer Division activities in his new position. Concurrent with his own advancement, Mintner announced the promotion of **Doug Beard** to the post of director of technical and marketing services for the Studer Division. Formerly national service manager, Beard will now take a more active role in marketing and key customer relations. Some of Beard's service-related duties will now be handled by **Tom Knox**, newly promoted to service and quality control manager for the Studer Division.

• **William Brin** will serve as national sales manager-audio visual products of **Tandberg of America**, it was announced by Erik Moseby, director of marketing and sales. Previously, Brin was corporate vice president of sales and marketing at International Audio, Arlington Heights, Illinois. He left that position in June to form a national marketing company, **Kar-Pro**, which will continue to operate under his direction. Prior to his joining International Audio, Brin was associated with Pentagon Industries, Chicago, Illinois and Radiant Manufacturing Corporation, Morton Grove, Illinois.

• **Nortronics Company, Inc.**, which manufactures digital magnetic heads and audio, video and computer care products, has announced the addition of **James D. Kuhn** as national sales manager of its consumer products division. Kuhn has a successful 16-year track record in consumer sales. In the last 12 years he has been employed as a regional sales manager for True Temper Corporation and the WD-40 Company. Kuhn will provide leadership, direction and management to the entire sales team, ensure sales goals are met by manufacturer's representative organizations and develop profitable business opportunities through new market, private labeling and special accounts.

• In a move to further strengthen their position in worldwide markets, the organizational structure of Tandberg A/S, Oslo, Norway has been modified, with the division of the parent company into five separate product centers. They are: Tandberg High Fidelity (consumer electronics and tape recorders); Tandberg Professional (reel-to-reel and cassette recorders, monitor tuners and computer-controlled cassette decks for the pro sound/broadcast market); Tandberg Satellite Communications (satellite broadcast/reception systems); Tandberg Educational (language laboratories and related products); Tandberg Production (designing, manufacturing circuitry and finished products for Tandberg and OEM); Tandberg Service. As part of the change in the company's organization, Tandberg of America ("TOA"), headquartered in Armonk, NY, will now be responsible for the marketing of Tandberg high fidelity products worldwide. In addition, TOA will also be responsible for Tandberg professional products and language laboratories in the U.S.

• **Scott Spector** has been named West Coast engineering manager/digital audio for **Sony** Professional Audio Division. Spector will provide comprehensive technical support for Sony digital audio products. Spector previously held the position of technical supervisor with Sony's Video Communications Division. Prior to joining Sony, he was senior support engineering manager for MCA Discovision's manufacturing facility. He has also operated his own music production company. **Richard Lee** has been named national product and systems manager for Sony Professional Audio Division. Lee will be responsible for professional audio product and systems planning and will interface with Sony factories in the U.S. and Japan. He also will conduct marketing research into the needs of both the recording and broadcast industries. Most recently, he was vice president/general manager of Criteria Recording, Miami, Florida, and previously spent eight years in the related fields of acoustics consulting, audio/video/film systems design, maintenance engineering and technical facilities executive management.

• **Altec Lansing** has announced the promotion of **Gayle Campbell** to national sales manager of commercial products. Campbell will also continue to be responsible for Altec's government sales. Formerly western regional sales manager for Altec, Campbell's experience also includes nine years as Altec's Midwestern district manager and six years in commercial sound contracting. Altec Corporation has also announced the appointment of **Robert A. Bushnell** as director of market development for the Altec Lansing sound products division. Bushnell will be responsible for product and market analysis, technical assistance to Altec and its customers, and will contribute to Altec's clinic, seminar, and technical publishing programs. Bushnell has a long background in sound product manufacturing, having worked for Westrex and UREI, and having founded Bushnell Electronics, which build recording and broadcast consoles. He has also worked in design positions for sound contractors, including, most recently, Paramount Sound in Glendale, California, a division of Paramount Pictures Corp. He joins Altec Lansing from Paul Alan Magil & Associates, an audio-visual and security system consulting firm in Costa Mesa, California.

• The SPARS board of directors met July 14th at Regent Sound in New York. The board approved the development of three new programs. SPARS, in cooperation with IMC Communications, will establish a telecommunications network linking SPARS members. The network will provide a quick and efficient means for SPARS studios to exchange information regarding technical problems and solutions, equipment needed or for sale, replacement parts needed, rental needs and other information that will assist in studio operation and management. In support of the network, Digital Entertainment Corporation has provided SPARS with a Leading Edge personal computer and auxiliary equipment and programming.

• **Barry Lee Bindell** has been appointed technical sales representative-Pacific region, for the magnetic tape division of **Agfa-Gevaert, Inc.**, of Teterboro, New Jersey. Bindell had previously been an account representative for the Technics division of Panasonic Company.

• **Charles D. Oesterlein** has been appointed manufacturing director of the Magnetic audio/video products division, with responsibility for manufacturing operations worldwide, 3M has announced. Oesterlein was previously plant manager at Weatherford, Okla., and before that held a similar position at Camarillo, Calif. He is a 19-year veteran of 3M, having joined the company in 1965 as a product development chemist. **Aaron R. Berg** has been appointed professional audio specialist for the Los Angeles area by 3M's Magnetic audio/video products division. He will be responsible for sales of 3M professional audio tape and magnetic film to professional audio recording studios and pro audio dealers in the area. He is located at the 3M sales office in Burbank. For the past three years, Berg was sales representative for the division in the San Diego area. Prior to that, he worked in the recording engineering field, serving as engineer at Rudy Records, Excalibur Studios and City Recording, all in Hollywood, and at Overland Studios in Costa Mesa and IAM Studios in Irvine. Before joining 3M, Berg founded and served as president of the Institute of Sound Recording, a 24-track recording engineering school in San Diego.

. . . & Happenings

An Inventive Idea

• A new organization for inventors has been formed, with headquarters in Sunny Isles, Florida. Appropriately titled The International Inventors Association, the non-profit organization aims to provide betterment for inventors in the educational and communicational fields. Members will receive a newsletter which will inform inventors how to select a patent attorney, market a new invention, build a model for their inventions, avoid rip-offs, etc. Write P.O. Box 36-536, North Grafton, MA 01536.

Vintage Studer Donated To Ampex Museum

• The Ampex Museum of Magnetic Recording has received a vintage Studer C37 recorder as a gift from Studer International of Switzerland. The recorder, restored to almost-new condition, will be on permanent display at the museum in Redwood City, California. Introduced by Studer in 1960, the C37 represented a significant advancement in tape tension control technology. Other innovations included motor-driven tape scissors, an internal incandescent lamp to regulate take-up motor voltage, and cascoded input stage of the reproduce amplifier.

Peter Hammar, consulting curator for the Ampex Museum of Magnetic Recording, with Studer C37 donated by Studer International of Regensdorf, Switzerland.

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FOR SALE: ONE ELECTRO—VOICE SENTRY III SERIES 2 SPEAKER. **Jim DeClerecq, 7163 Glenmore, Lambertville, MI 48144, or call collect (313) 856-7176.**

SENSURROUND LOW FREQUENCY ENCLOSURES with 18" CER-VEG 300 watt die-cast 96 oz. magnet driver. $195.00. **Empirical Sound, 1234 East 26th Street, Cleveland, OH 44114. (216) 241-0668.**

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16 Channel SPECK Professional Recording Console; +4 dBm, custom modifications, 6 band, 3 frequency EQ, spare Input module, 8 Bus, Excellent condition, $2,500 (originally $4K+). **(617) 372-4236.**

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This is our newest multitrack. It is also the most affordable multitrack in Studer history.

For the fourth time since its inception, we've changed the A80VU. We've improved the sonic performance, tape handling, and durability. And we've substantially lowered the price.

Same outside, changes inside. In keeping with the Studer tradition, we made no superfluous cosmetic changes. We're not going to tell you this is an "all new" recorder. It isn't. It is a proven, legendary recorder incorporating several significant improvements.

Uh-oh, Something Is Missing. Yes. The transformers are gone. They've been replaced in the input and output stages with new high performance active balancing circuitry. Other MKIV improvements include a new master bias oscillator, extended record headroom, and a new record and bias driver compatible with all present and future high-bias requirements. Record electronics are now fully compatible with Dolby HX Pro* requirements.

Smoother Shuttling, Hardier Heads. The MKIV's new tape tension control system provides smoother tape handling, while a new extended wear alloy for record and play heads greatly increases head life.

Never Lower. The list price of the A80VU MKIV 24-track is lower than any of its predecessors. And that's in straight dollar figures, without adjusting for inflation. What's more, the A80VU MKIV now has a list price lower than most of its competition.

No Hocus-Pocus. How could we make the A80VU MKIV better and lower the price at the same time? Simple. We make it in Switzerland, and you pay for it in dollars. The favorable exchange rate does the trick. That means you get advanced electronics, Swiss precision, and low price. If you act now. This can't go on forever.

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For more information, call or write: Studer Revox America, 1425 Elm Hill Pike, Nashville, TN 37210; (615) 254-5651.

*Dolby HX Pro is a trademark of Dolby Laboratories.

STUDER REVOX

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