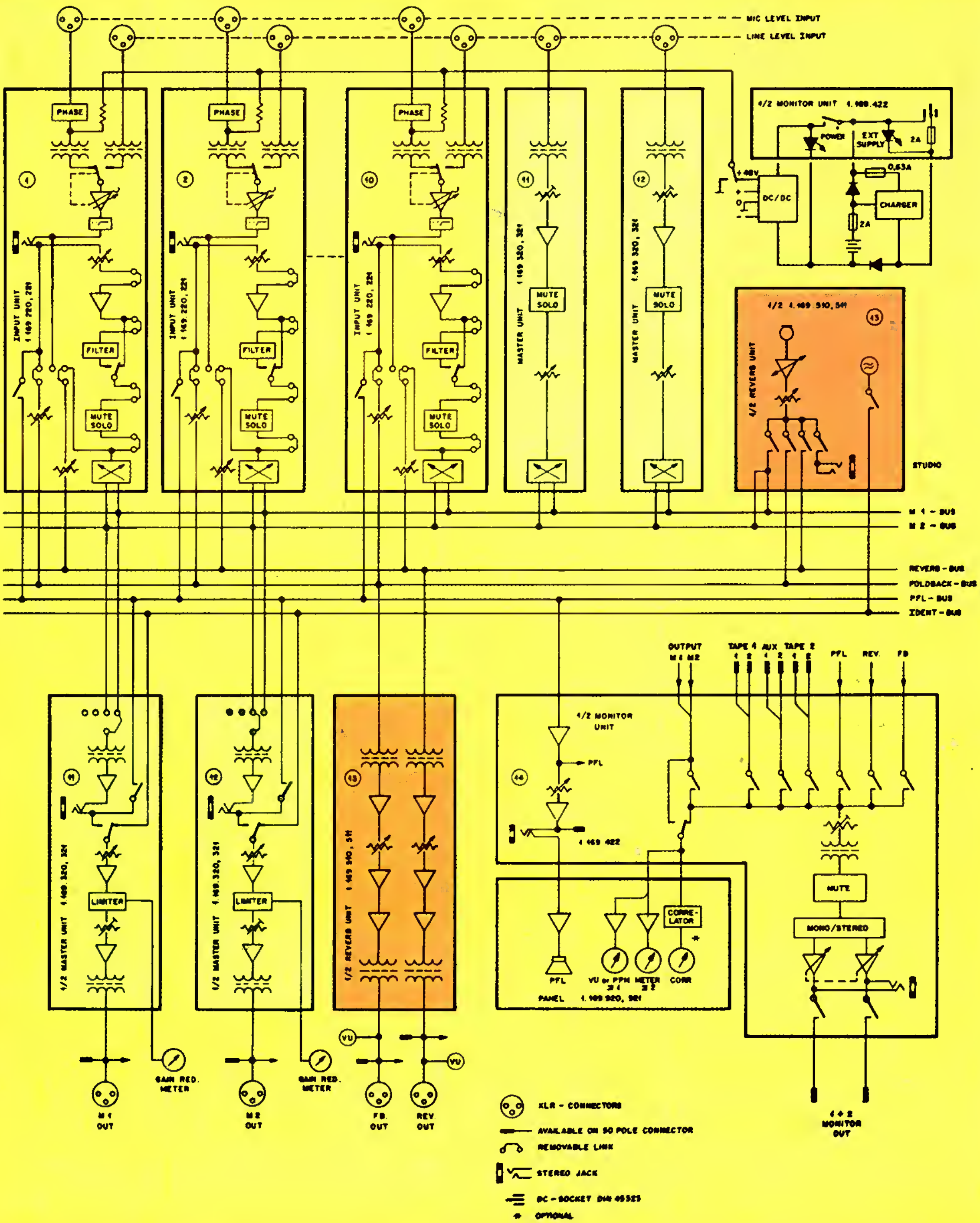
THE SOUND ENGINEERING MAGAZINE

JANUARY 1982
$1.95

db



Publisher
Larry Zide

Editor
John M. Woram

Associate Editor
Mark B. Waldstein

Advertising Production & Layout
Karen Cohn

Classified Advertising
Carol Vitelli

Book Sales
Lydia Calogrides

Circulation Manager
Eloise Beach

Art Director
Bob Laurie

Graphics
K&S Graphics

Typography
Spartan Phototype Co.

sales offices

Roy McDonald Associates, Inc.
Dallas, Texas 75237
First Continental Bank Bldg.
5801 Marvin D. Love Freeway
Suite 303
(214) 941-4461

Denver, Colorado Area
Englewood, Colorado 80112
14 Inverness Dr. East
Bldg. 1—Penthouse
(303) 771-8181

Houston, Texas 77036
6901 Corporate Drive
Suite 210
(713) 988-5005

Los Angeles Area
Glendale, California 91204
424 West Colorado Street
Suite 201
(213) 244-8176

Portland Area
Hillsboro, Oregon 97123
510 South First
P.O. Box 696
(503) 640-2011

San Francisco Area
Emeryville, California 94608
265 Baybridge Office Plaza
5801 Christie Avenue
(415) 653-2122

Sagamore Publishing Co., Inc.
New York
Plainview, NY 11803
1120 Old Country Rd.
(516) 433-6530

db

THE SOUND ENGINEERING MAGAZINE

JANUARY 1982 VOLUME 16, NO. 1

DEPARTMENTS

FEATURES

About The Cover

• This month's cover offers us a colorful look inside a small console. It's Studer's new Model 169-10/2/2.

db, the Sound Engineering Magazine (ISSN 0011-7145) is published monthly by Sagamore Publishing Company, Inc. Telephone (516) 433 6530. db is published for those individuals and firms in professional audio-recording, broadcast, audio-visual, sound reinforcement, consultants, video recording, film sound, etc. Application should be made on the subscription form in the rear of each issue. Subscriptions are $15.00 per year ($28.00 per year outside U.S. Possessions; $16.00 per year Canada) in U.S. funds. Single copies are $1.95 each. Editorial, Publishing and Sales Offices: 1120 Old Country Road, Plainview, New York 11803. Controlled circulation postage paid at Plainview, NY 11803 and an additional mailing office.

db Letters

DIGITAL FILTERING

To The Editor:

Sidney Silver's August article "Digital Filters in Audio Signal Processing" was most apropos and timely. Now that audio signals are recorded, mixed, and controlled digitally, the filtering and shaping process should also be carried out in the same manner to avoid signal degradations.

We have recently augmented our line of adaptive digital filter systems with a programmable transversal digital filter. The filter interfaces directly with a microcomputer, which specifies the filter coefficients. Several types of digital filter structures were discussed by Mr. Silver, and each has its advantages and disadvantages. The principal disadvantage of transversal filters is their computational requirements. (Our new instrument performs over 25 million mathematical operations per second.) The advantages of transversal filters, however, are:

- Accurate control of phase response. Linear phase (constant delay) filters are easily implemented.
- Stability. Transversal filters do not oscillate because no feedback path is present.
- Coefficient insensitivity. Less precision in specifying coefficients is required, as a large number of coefficients (typically over 100) are normally used.
- Finite impulse response and less tendency to ring. This is especially advantageous with percussion effects.
- Ability to implement arbitrary passband shapes, such as equalization curves, in addition to canonical high-pass, bandpass, etc., filters.
- Availability of straightforward procedures for designing transversal filters.

Other filter structures discussed by Mr. Silver have distinct advantages, but I feel that the theoretical understanding, practical hardware availability, and ease in application of the transversal digital filter is best suited for the professional audio community.

Dr. James E. Paul
President
Digital Audio Corporation
Anaheim, California

Index of Advertisers

Coming Next Month

• In February, our subject is Sound Reinforcement. John Eargle returns to our pages with an overview of the subject, Kenneth Bourne discusses the evolution of the wireless mic, and J. Robert Ashley makes us privy to a very interesting conversation on church sound. In addition, we'll have our usual columns and more—all in **db—The Sound Engineering Magazine.**

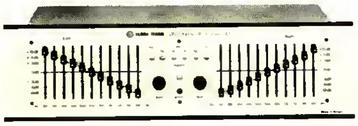
It will discourse most eloquent music.

(William Shakespeare 1564-1616)

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

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

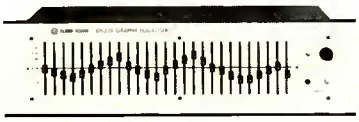
DN22 GRAPHIC EQUALISER



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

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

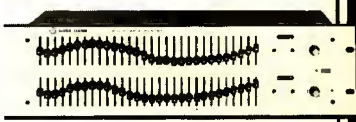
DN27A GRAPHIC EQUALISER



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

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

DN30/30 GRAPHIC EQUALISER



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

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

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

Klark-Teknik Electronics Inc.
262a Eastern Parkway, Farmingdale,
NY 11735, USA. Telephone: (516) 249-3660

Klark-Teknik Research Limited
Coppice Trading Estate, Kidderminster, DY11 7HJ.
England. Telephone: (0562) 741515 Telex: 339821

Omnimedia Corporation Limited
9653 Côte de Liesse/Dorval, Quebec H9P 1A3, Canada.
Telephone: (514) 636 9971

Circle 16 on Reader Service Card

KLARK TEKNIK
sound science

and, SOME DIGITAL FILTER ERROR CORRECTIONS

TO THE EDITOR:

With reference to my article "Digital Filters in Audio Signal Processing" which appeared in the August issue of **db**, I would like to correct several printing errors.

On page 32, the second sentence of the third paragraph should read: "Since feedback is used in its implementation, the output of the filter is effectively derived from the present input values and from past values of the input *and output*.

On the same page, the second sentence of the fifth paragraph should read: "Recursive filters, which will be discussed first, use feedback in such a way that the filtered output is always an explicit function of past and present input *and output* values."

On page 34, the second sentence of the second paragraph should read: "Since there are no feedback elements employed, past *outputs* do not enter into the response."

SIDNEY L. SILVER
United Nations
Telecommunications Sect.

db replies:

Oops! Those outputs were deleted in an effort to clarify the signal path explanation. In the context, we felt there was some semantic confusion, since all *outputs are past inputs, and so past input is almost synonymous with output, and therefore... Oh, never mind. Mr. Silver's letter restores the outputs to their original location.*

AUDIO ON VIDEO (CASSETTE)

TO THE EDITOR:

If some progressive manufacturer would produce an inexpensive interface unit for the home audio market, analog reel-type tape equipment would become obsolete for owners of video cassette machines. The BETA or VHS machine would perform multiple duties: playing prerecorded video cassettes, recording and playing video cassettes from live (camera) and television sources, playing pre-recorded video cassettes, recording and playing video cassettes from live (microphone) and other audio sources. What possibilities! Such a unit would give the videocassette machine a competitive edge for a long time. Its functions would not be duplicated by any other medium of recording/playback and we home users would have all the benefits of superb audio recording on our video cassette equipment. What is still necessary, however, is an interface unit in the $300-$500 range. Can this be done?

R. DENNIS ALEXANDER

db replies:

Sure! Just move the decimal point one place to the right, and turn to Len Feldman's column in this issue.

For every product manufactured, there's always a leader.

And when it comes to Headsets, it's Telex

Write us today, and we'll tell you why.

Quality products for the audio professional

TELEX®

TELEX COMMUNICATIONS, INC.

9600 Aldrich Ave. So., Minneapolis, MN 55420 U.S.A.
Europe: 22, rue de la Légion-d'Honneur, 93200 St. Denis, France

Circle 29 on Reader Service Card

BARRY BLESSER

db Digital Audio

It's Not Where You Are, But How You Got There

• In our previous definition of logic, we divided the world into two classes: combinatorial and sequential. Combinatorial logic can be thought of as a series of simple gates which are combined to form more complex functions. Devices such as OR, NOR, AND, NAND, XOR, NXOR, MASK and INVERT are the basic functions with which large combinatorial arrays can be built. Notice that a full adder of two 16-bit words is actually a combinatorial array with 32 inputs and 16 outputs. Since the state of the output is only dependent upon the present state of the input, the operation is combinatorial. Similarly, a multiplier is also a combining operation. We will speak more of such functions later, since they form the heart of some interesting processing.

We must now turn our attention to the second category, called sequential (or memory, or clocking) logic. Such logic has the reverse property: the output is determined by the *history* of the input, as well as the present input. It remembers. The most obvious example is a flip-flop which has an output which stays independent of the input. Clearly, some mechanism must be provided to change the storage, and this is often referred to as clocking. To use a simple analogy, we may say that the information is stored within a room with a closed door. The object in the room is unaffected by what takes place elsewhere in the house. However, when the door is opened, the object can be removed and replaced with a new object. The object is the information, such as a 1 or a 0, the room is the flip-flop which stores the object, and opening the door performs the clocking function.

For the moment, we will treat the flip-flop as the basic memory element from which all other sequential logic is built. It is not obvious from this definition, but the basic flip-flop itself can be built from combinatorial logic. To do so, we introduce the concept of time in our simple combinatorial logic, for it is not quite true that the output is determined *only* by the present input.

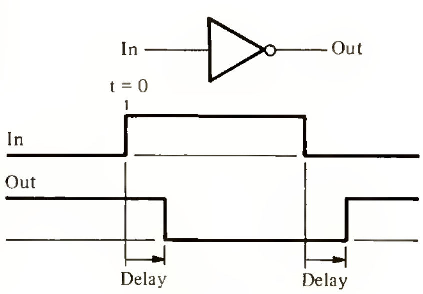


Figure 1.

EVERYTHING TAKES TIME

As our first example, consider the simple inverting gate in Figure 1. When the input is H (high = +5), the output is L (low = 0). Conversely, when the input is L, the output is H. To introduce the concept of time, let's assume that the input goes from L to H at a point called t = 0. The output will respond by going from H to L, but not quite at t = 0. Rather, it takes a certain amount of time for the information to go through all the circuits. The actual time varies greatly, depending on the type of circuits. The very oldest digital RTL (resistor-transistor logic) had delays on the order of a fraction of a microsecond (which is forever by today's standards). The next generation of DTL (diode-transistor logic) was faster by a factor of 10 or more. The current generation of TTL (transistor-transistor logic) takes on the order of 10 nanoseconds (10^{-8} seconds). And finally, ECL (emitter-coupled logic) takes less than 1 nanosecond. In all these cases, the output does not respond instantly to the input.

We can now build a simple type of time-dependent logic as shown in FIGURE 2. Notice that the inputs to the AND gate are always H on one input and L on the other, regardless of the input. The inverter creates the other signal, which is always the reverse of the input. Therefore, one would think that the AND gate output would always be L, since both inputs are never H at the same time. However, consider the case when the input goes from L to H at t = 0. The inverter will not respond immediately, and so for an instant its output is H, even though the input is also H. During this instant, both inputs to the AND gate are H. After the instant goes by, the inverter output goes to its correct state of L. But, before the instant has passed, the AND gate's output goes to H, since both its inputs are H. This situation lasts until the inverter's output reaches its correct state, and then the AND gate will go to its correct state.

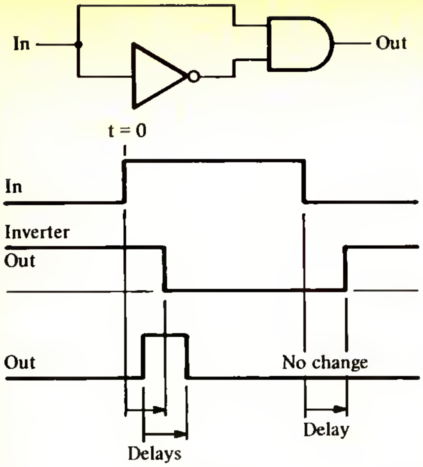


Figure 2.

If we run the input from H to L, there will not be any response from the AND gate, since both its inputs are L for this instant. Since the AND gate has the same output when one or both inputs are L, the output does not show the same kind of temporary H output.

This example is interesting since we have shown that the output from the AND gate becomes H an instant after the input goes from L to H, but does nothing when the input goes from H to L. This is a sequential circuit because the output is a function of the input's sequence. The input being H does not tell us if there will be an output. We need the history information which says that if the input was L and becomes H, the output will respond: history is the state that the input *was*.

THE FLIP-FLOP

We are now ready to build flip-flops, and we will begin by an analysis of the circuit of FIGURE 3, called an RS (reset-set) flip-flop.

With all memory devices, it is difficult to begin an analysis because we cannot know the state of various internal logic levels. We don't know what they are because the device is a memory storage element which can store either a 1 or a 0. By having two possible values, we cannot specify the state. To do an analysis, we first *assume* a value at some point, and then determine the other values. Confirming that it is a storage device requires us to repeat the analysis with the other initial state. When we find that both initial states lead to stable-but-different results, we can say that it can store either state.

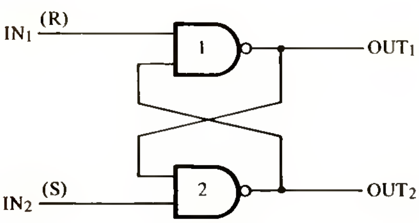


Figure 3.

Begin by assuming that both inputs are H and that OUT_1 is also H. Therefore, both inputs to NAND gate 2 are also H and its output is L. If OUT_2 is L, then NAND gate 1 must have an H output. We initially assumed that OUT_1 was H, and our analysis shows that indeed it was. Now, do the analysis again, but assume that OUT_1 is L instead. Therefore, the NAND gate 2 output will be H. With both inputs to NAND gate 1 being H, its output will be L. Again, our assumption is correct. This circuit can assume that either OUT_1 = H and OUT_2 = L, or, OUT_1 = L and OUT_2 = H.

To continue our analysis, we now change the assumption that both inputs are H. Assume that IN_1 is L, and IN_2 is H. Since one of the inputs to NAND gate 1 is L, its output must be H. NAND gate 2 has both inputs H, so its output is L. In this analysis, we don't need to assume the state of either output. There is only one result, which is not dependent on history. The storage capacity of the circuit is destroyed with IN_1 being L. By symmetry, the same situation is found if IN_2 is L.

Very often, such a circuit is described with a truth table which shows all possible inputs, as in FIGURE 4. The left columns give the four possible input states of H-H, H-L, L-H and L-L. The right columns give the actual outputs for these states. With IN_1 = H and IN_2 = H, OUT_1 = P_1, which is to say that the present output (P_1) is the same as the previous output. When the inputs are IN_1 = L and IN_2 = H, the outputs are defined as H and L. Notice there is no way for both outputs to be L (except for an instant during a transition), and so we call this an "undefined" condition.

In FIGURE 3, the two inputs are called R (reset) and S (set), since they can force the circuit to a given state. When these inputs are both brought back to H, then the circuit can remember that state. This is a very primitive flip-flop, but it forms the basis of more complex circuits. FIGURE 5 shows a modification to the circuit, with the addition of two extra NAND gates (3, 4) to feed the R and S inputs (IN_1 and IN_2) of the previous circuit.

IN_1 (R)	IN_2 (S)	OUT_1	OUT_2
H	L	L	H
L	H	H	L
L	L	H	H
H	H	P_1	P_2

NOTE:
P_1 and P_2 depend upon previous output values.

Figure 4.

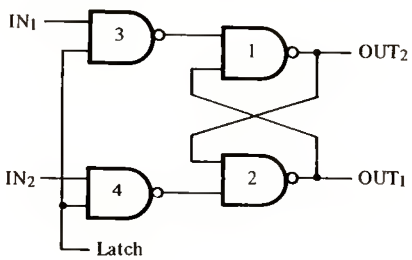


Figure 5.

The only difference between the two circuits is that the act of entering the information, and the information itself, are now separated by an intermediate set of control gates. The control input to these gates, called a latch, allows the information to be entered into the flip-flop only when the latch is H. When the latch is L, changes in input data will not affect the flip-flop outputs. In the previous circuit, IN_1 and IN_2 had to remain H if the flip-flop was to retain its old information. In the new circuit, as long as the latch control is L, IN_1 and IN_2 can be anything at any time, and the flip-flop will remain unchanged. The names of the outputs have been interchanged because of the extra inversion created by the new gates. Ordinary AND gates could have been used conceptually, but the NAND gate is considered the basic element for historic as well as implementation reasons. The latched flip-flop is a higher level flip-flop, and it is an important device in many applications.

In FIGURE 6, we see the next-higher level of complexity. We place two latched flip-flops in tandem, but the latch line to the second flip-flop is inverted. When the control is L, the first latched flip-flop will be in memory mode. Because of the inversion, the second latched flip-flop will be in the respond mode. This is quite simple: the first flip-flop remembers, and the second follows the first.

Now, reverse the state of the control line: the first flip-flop is in respond mode (it follows the inputs), and the second is in memory mode (remembering its input data). In both cases, the final outputs cannot follow the inputs. Either the first is in memory mode with the second responding, or the second is in memory mode, and does not respond to data changes.

Now, consider the case where the control input is in transition from H to L. When it is in the H state, the first flip-flop responds to the input. As the control goes L, the first flip-flop remembers the input state and now the second flip-flop can follow the first. The input information can get to the output at the transition of the control. The reverse transition will not change the output. When the control is L, the first flip-flop is in storage mode. During the transition, the first will allow the data to enter, but the second has just gone into storage mode preventing the new data from passing. This kind of flip-flop is sometimes referred to as a naster-slave flip-flop, since there are two internal flip-flops which act as master and slave. The first is the master, and the second is the slave.

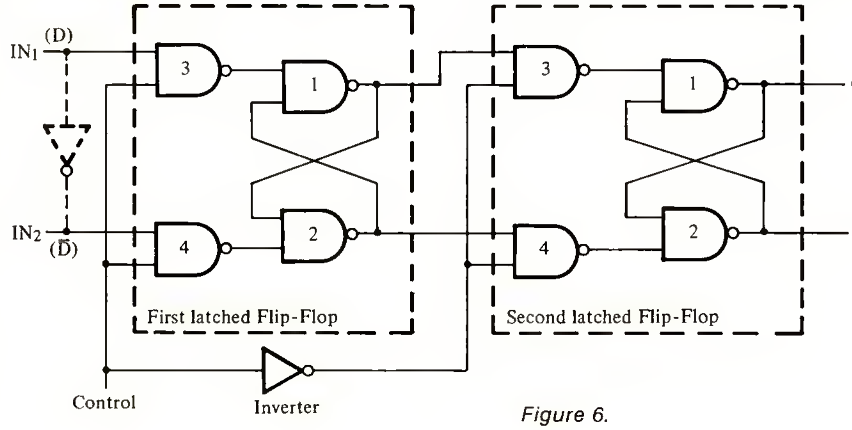


Figure 6.

This is the classical clocked flip-flop, in which the control wire is called a clock line. In our example, the negative clock transitions pass data. The final step is to note that the two inputs are really redundant, since the flip-flop can only be expected to store a 1 or a 0. Since both inputs being H does not change the input drive to the flip-flop, this is a default case. And, both inputs being L is again an undefined state. Therefore, the usual input is a single wire, called a D input, and an internal inverter creates the complement. The dashed-line circuit in FIGURE 6 illustrates this. Thus, $IN_1 = D$, and $IN_2 = \overline{D}$. The outputs are called Q and $\overline{Q}$. We have just built the equivalent of a 7474 D-type flip-flop.

In the next article, we will continue building more complex circuits. However, to review, we should note that the concept of memory storage is actually based on a "non-ideal" property of combinatorial logic: time delay. Without this property, storage could not be built. ■


GREENE, CROWE & COMPANY

Fitting PCM Audio Into The NTSC VCR Format

• While we await the debut of those first compact digital audio discs (present forecasts are late 1982 for Europe and Japan, and early 1983 for us), work in digital recording on tape goes on apace. Just a few weeks ago, I saw the *first* commercially available dedicated PCM tape deck intended for semi-pro and audiophile use. It was produced by the Technics division of Matsushita Electric Company, and is expected to carry a price tag of around $3000. What's interesting about this unit is that it is designed to use *video* tape cassettes of VHS configuration and employs standards which were agreed to nearly two years ago by the entire membership of the EIAJ (Electronic Industries Association of Japan). In other words, Matsushita has taken a digital audio processor plus a VCR, and combined the two devices into a single machine for PCM audio.

Figure 1. The Technics SV-P100 Digital Audio Cassette Recorder.

Such a device (at its projected price) may prove to be an important two-channel mastering deck for smaller studios that can't afford the prices being asked for the more elaborate stationary-head studio-type digital recorders. In order to make an intelligent choice concerning digital mastering hardware, it would be wise to understand how digital information is integrated into what is basically a video recording format.

Those familiar with how a VCR works and with the requirements of the U.S.-type NTSC video signal will appreciate

L	R	L	R	L	R	P error corr.	Q error corr.	CRC
14	14	14	14	14	14	14	14	16
128 bits								

Figure 2. The partial contents of one horizontal line of PCM digital audio information.

the negative aspects of using an *analog video* tape system as the storage medium for digitally-processed audio information. Of course, the VHS/NTSC recording format was originally designed for video, and that means that if you are going to use it as a digital audio storage system you will have to fit those millions of bits which constitute the digital audio code into a *video* signal format. That format includes horizontal sync pulses after every video line, and vertical sync pulses after every video frame. Since there are 30 interleaved frames-per-second and 525 lines-per-frame in the NTSC TV system, you can't record a continuous bit stream onto the video cassette tape. The digital audio information has to be inserted during the active horizontal-line time of the normal video picture format. With an almost infinite number of ways in which this could be done, it was essential that the industry get together on a standard format for PCM/VCR interface and use. Happily, the EIAJ standard describes *only* the makeup of the signal to be recorded, and does not go into such things as VCR type, tape width, speed, etc. Thus, it lends itself as well to U-Matic (professional) use as it does to Beta, VHS or even all-in-one dedictated PCM audio decks such as the one introduced by Technics.

THE NEW TOA RX SERIES SUB-COMPACTS.

THE TRAVELLING CONSOLES THAT WON'T DRIVE YOU TO THE POORHOUSE.

For any demanding high quality professional sound reinforcement requirement, there's now a choice between the real expensive and the embarassingly inadequate: The TOA RX Series.

Starting as low as $1,130.00 for our RX-5A console and all the way up to our 32 by 8 RX-7 Model, you now have the performance, features and flexibility to create a professional sound system that can handle the simple showcase gig or the big concert job. Of course, if you need to record, the necessary features are all there.

Our top-of-the-line Series, the RX-7 is available from 16 in/4 out (pictured below) to 32 in by 8 out. It's modular, and has features too extensive to cover here. When an RX-7 is more than you need, we make three smaller RX Series consoles that will more than get the job done: The RX-5A (8 by 2); The RX-6A (12 by 2); and the RX-5/16A (16 by 2).

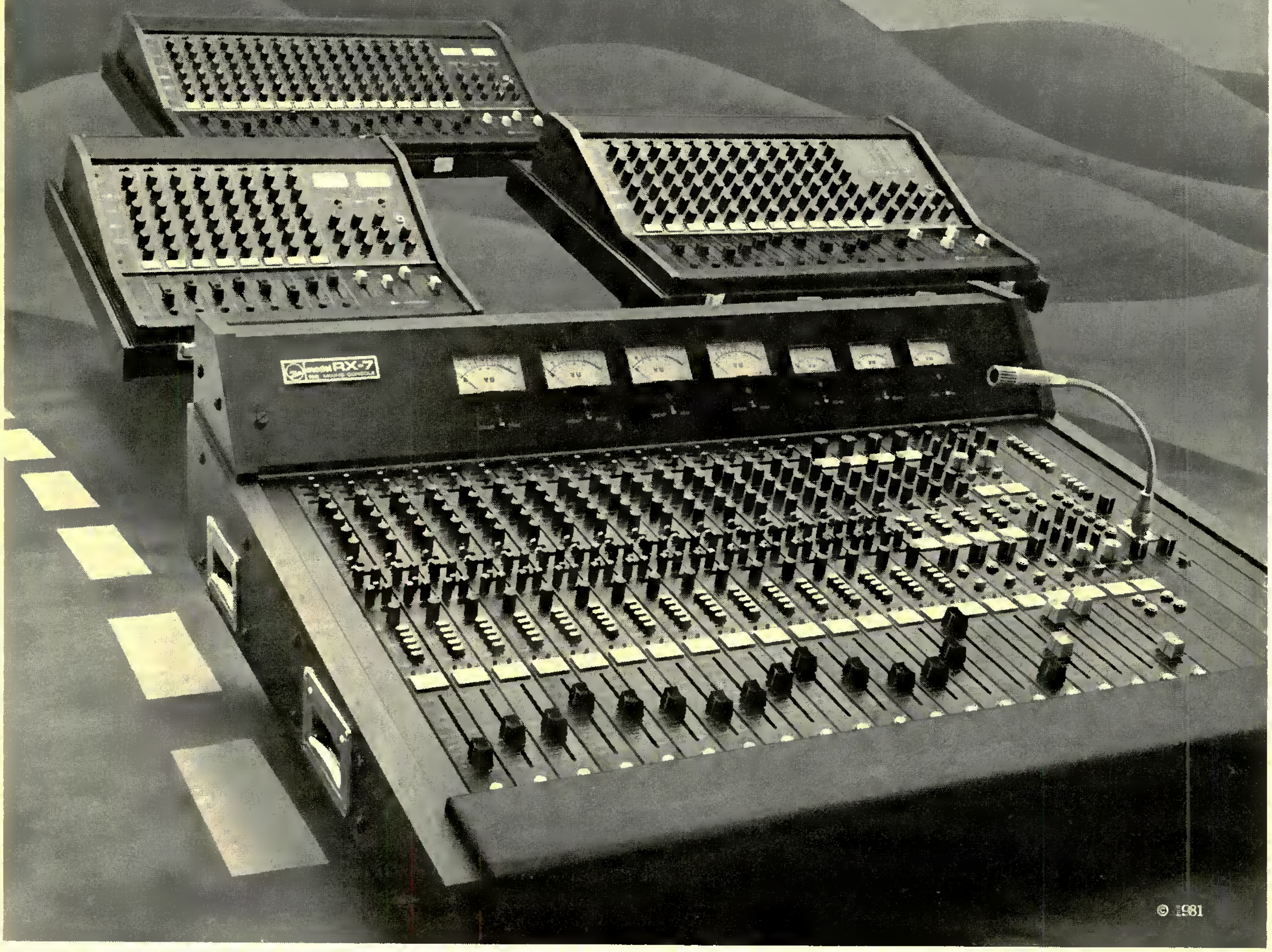
Whatever model you choose in the series, you get a −128 dBm E.I.N. hum and noise figure (20 Hz - 20 kHz). Quiet. THD at +10 dBm is typically less than .08%. Clean. And crosstalk is −60 dB (1kHz, input to output), −70 dB on the RX-7 Series. Professional. All the way, including heavy-duty XLR type connectors on many of the I/O's.

Give us a call at (415) 588-2538. We'll rush out all the spec's and the name of your nearest qualified TOA dealer. We're confident you'll want to sit behind the superb handling of an RX. The consoles that have performance and features to spare —and the serious subcompacts that won't drive you to the poorhouse while you're on your way to the top.

Crafted in Japan. Proven in the States.

TOA Sound & Communications

TOA Electronics, Inc.
1023 Grandview Drive
South San Francisco, CA 94080
(415) 588-2538 Telex: 331-332



© 1981

Circle 28 on Reader Service Card

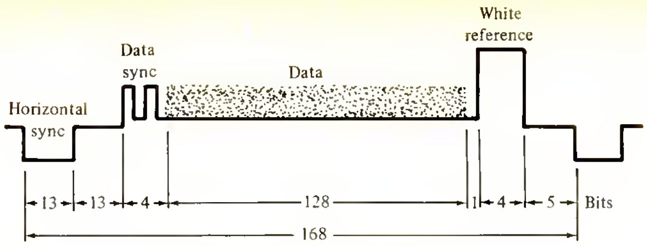


Figure 3. The complete contents of one horizontal line, including standard NTSC video sync components.

THE EIAJ STANDARD FOR PCM AUDIO ON VCR TAPE

The EIAJ Standard is designed for two channels, which means that if you are going to use it for studio mastering, you must figure on a real-time, live mix. Optional pre-emphasis for added noise reduction employs two time constants: 50 and 15 microseconds. Sampling frequency has been fixed at 44.056 kHz. Why such an odd-ball number, when the new digital audio *discs* have settled for 44.1 kHz? Because, while *both* frequencies are more than adequate for recording and reproducing audio frequencies up to 20 kHz (sampling rate must be at least twice the highest frequency to be recorded), 44.056 kHz bears a nice mathematical relationship to the *video* line repetition rate in an NTSC video signal. The EIAJ standard uses 14-bit linear encoding, which means that you can expect a maximum dynamic range of approximately 85 dB from this system.

The total number of bits per second, per this standard, is 2.643 Megabits. This number was determined in part by the need to have enough redundancy for error correction and horizontal blanking that is an inherent part of the standard video signal. FIGURE 2 shows the contents of *one* horizontal line of equivalent video-format signal. It consists of three 14-bit words each from the sampled left- and right-channel audio signals, interleaved as L, R, L, R, etc. This is followed by two words (known as P and Q codes) for error correction, and a 16-bit CRC (cyclic redundancy check) word for error detection. In order to be able to handle long dropouts in the tape, succeeding words of the sampling code are actually interleaved by 16 horizontal lines. That is to say, if the first 14-bit word of the first line of a frame is sample #1 of the left-channel audio signal, then sample #1 of the right-channel audio signal will appear displaced by one word-space, sixteen lines later in the encoded sequence.

FIGURE 3 shows the makeup of a complete horizontal line. There is room for 168 bits, but only 128 bits of data per line are used. The remaining time is used for the required horizontal sync pulse, data sync pulses, and other signal shapes required for the standard NTSC TV signal format around which this standard is based.

The signal format for a single video *field* (there are two fields per frame and 30 frames per second) contains 262.5 horizontal lines of data, as shown in FIGURE 4. Of this number, 245 lines are used for actual digital audio data storage, and an additional line is used for a control-signal block. This control-signal block is made up of 56 bits for indicating the start of the data block in each field,

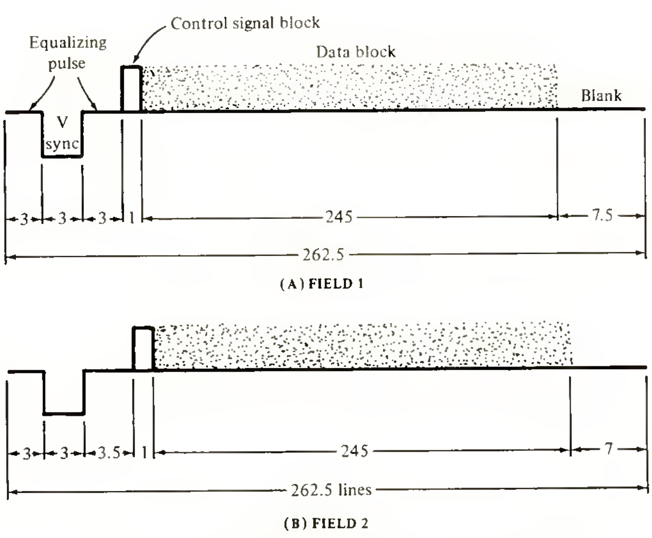


Figure 4. The format for a single video field (two frames) in the EIAJ PCM standard.

Maxell does the job. Without fail.

Maxell audio tapes will never let you down, because we make them to our exacting standards of performance and durability—standards that are much higher than those of the industry in general. At Maxell, we know that professionals who use magnetic recording media on a regular basis demand the advantages found only in superior tape products.

You demand tapes that are responsive to extended frequency and dynamic ranges, with a minimum of noise and the absence of distortion. And what you record, you want to keep intact, for an indefinite period of time, without loss of fidelity caused by shedding, dropouts or print-through. You also need the advantages of a superior cassette with a tape transport system that will smoothly and dependably wind and rewind the tape without stretching, jamming or breaking.

Maxell P/I audio cassettes and professional open reel tape provide you with all these advantages. And many more. For every type of recording, duplicating and editing application, Maxell tape gets the job done—accurately, and without fail. To discover more about Maxell's professional advantages, call one of our Regional Sales Offices: Eastern Office, Moonachie, NJ (201) 440-8020 ■ Midwestern Office, Des Plaines, IL (312) 296-1010 ■ Western Office, Los Gatos, CA (408) 395-1998.

maxell®

PROFESSIONAL INDUSTRIAL PRODUCTS

The Professional Advantage

Maxell Corporation of America
60 Oxford Drive, Moonachie, NJ 07074

Circle 18 on Reader Service Card

14 bits for content identification, 28 bits for address, 14 bits for control (including a possible copy-prohibiting code), identification codes for the P and Q error correction words, a preemphasis identification code and sixteen bits for the cyclic redundancy error detection code. Contents of this entire control line are shown in FIGURE 5.

From time to time during this discussion, we have referred to error correction as it applies to the EIAJ PCM tape standards. Error detection and correction are vital parts of any digital information storage system. Dropouts caused by a tape's coating irregularities or poor contact with a tape head's surface may be insignificant and inaudible when they occur in a conventional analog tape playback system, but losing even a few microseconds of data in a digital data system can seriously alter the numerical code which must ultimately be reconverted to an analog audio signal. For this reason, the error correction system incorporated into the EIAJ Standard Video/PCM format permits highly sophisticated error correction to be incorporated by individual hardware designers. Total correction is possible even if two words in a single line of data are completely lost during record or playback. Furthermore, even if a third error in the same line of data occurs, the system can be so designed that it "fills in" the amplitude of the previous word in its number code, or the average of the preceeding and succeeding words so as to provide smooth continuity of sound during playback.

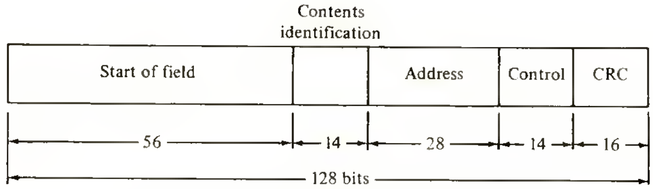


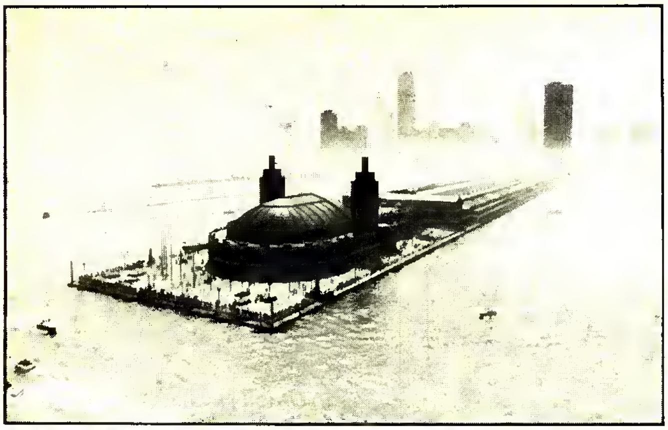
Figure 5. Details of the control line (Control Signal Block) used in the EIAJ PCM standard.

PCM VIA VIDEO HAS ITS PROBLEMS TOO

Certainly, the appearance of dedicated tape decks designed to use video cassettes for storing audio information in digital form is welcome news to those recordists who cannot afford the elaborate multi-channel digital setups, digital editing consoles, and the like now being offered by such companies as 3M and Sony. But don't get the idea that these little machines are going to give you anywhere near the flexibility and versatility of their bigger brothers. For one thing, there is no such thing as "razor-blade" tape editing with video cassettes. And even if you could afford some of the new digital editing equipment now available from several companies, the sampling rates and general digital formats which those editors can accommodate is not compatible with the VCR/PCM EIAJ digital format we have just been describing.

Interestingly, at the recently concluded AES Convention in New York, Studer demonstrated a prototype of a Digital Sampling Frequency Conversion unit which could take a digital audio signal having any of a wide range of sampling frequencies and convert it to another desired sampling frequency rate without degrading the signal or introducing phase shifts in the recovered audio signals. (See John Borwick's feature article in this issue—Ed.) But even such a device does not solve the problems of incompatibility between a VCR/based PCM system and a professional, stationary head digital recording system. Perhaps someday someone will come up with the right hardware or software that will be able to directly interface between EIAJ-PCM/VCR formated digital audio recordings and multi-track, studio type stationary head digital recordings. That is certainly going to be necessary if we hope to produce prerecorded tapes for use on the new home-type PCM/Video cassette tape decks I've been referring to.

Until all that is ironed out, decks such as the one introduced by Technics are likely to remain a curiosity as far as the professional recording engineer is concerned (affluent studios may want to own one just to experiment with digital audio, if they haven't done so up to now because of budget restrictions) and, as always, there will be enough amateur recordists out there who will be willing to spend $3000 to see what digital audio recording is all about. Whether they'll find any analog material worth transcribing on such a superior medium is another question entirely... ■



Bose® passes Fest Test. Again.

For the second year in a row, Bose was chosen as the Official Sound Source for ChicagoFest, the nation's largest outdoor music festival. Over 250 Bose 802 Loudspeakers in sixteen locations were used to bring twelve days of continuous live music to almost a million listeners.

The power and clarity of Bose 802 speakers makes them ideal for use in demanding applications like ChicagoFest. Stacked 802 arrays have the projection and bass response you need for **your** demanding application, without the problems of conventional bass cabinets and horns. And their exceptionally rugged design lets them withstand years of heavy-duty use.

The producers of ChicagoFest have come to rely on the outstanding performance of Bose 802 Loudspeakers. Visit your authorized Bose Professional Products dealer soon, and find out why Bose should be **your** Official Sound Source.

Bose Corporation, Dept. SE
The Mountain
Framingham, MA 01701

Please send me the Bose Professional Products Catalog and your technical data.

Name ______________________
Firm ______________________
Street ______________________
City ______________________
State ________ Zip ______________
Telephone (   ) ______________

BOSE®
Better sound through research.

db Editorial

IN A RECENT PRESS release from Studer/Revox America, Dr. Roger Lagadec writes, "Digital, we are fond of saying, is inherently cheap."

Is inherently *what?* Of course, Dr. Lagadec was looking into the future, and he did acknowledge that for the moment, "The experience of most of us is, of course, that digital is inherently expensive." LSI circuitry will eventually change all that, but not today. He cites the high cost of D/A conversion—a cost which could be driven down by consumer demand for digital disc systems (don't hold your breath).

Another factor in bringing down the price tag will be the introduction of the digital console. With no conversion required between console out and tape in, we could do nicely with a single D/A conversion system, which would be located just in front of the control room monitor system. With careful systems planning, and lots of standardization (again, breath-holding is not recommended) all those digital signal processing goodies could dispense with the A/D and D/A conversions now required at each input and output.

In this month's letters column, Dr. James E. Paul alludes to the signal processing potentials within the digital domain. Equalization without phase shift is something we've often thought would be a nice thing to have available. We often feel that conventional analog equalization comes off sounding, well... equalized. It's as if some sort of vaguely unpleasant side effect was tossed into the mix, along with the eq. Could it be phase shift here that bothers us? And if it does bother us, is this an example of (gasp!) *analog* stress? Anyway, our reservations about equalization-plus-phase shift can be brushed aside once the signal path is all-digital.

But, first we need that all-digital console, if only to keep the prices of things within reason. But, how soon? As of the AES convention late last year, we'd have said it was still quite a long way off. 3M had a tiny prototype (2 in/2 out, as we recall) but weren't saying much about it. (Actually, they weren't saying *anything* about it: It was just an engineering prototype which proved the technology was indeed out there, though not within easy reach.)

How soon? Some time after the show, we received a routine press kit from Rupert Neve, Inc. At least we thought it was routine as we idly rummaged through the announcements of new consoles for recording and broadcast applications. But then we discovered an announcement that the BBC expected Neve to deliver a 48-channel digital console, and to do so before this year is out. The console will feature digital equalization, compression, gain control and mixing, along with variable time delay in each (!) of the 48 channels.

The console specs list sampling rates of 44.1 and 48 kHz as standard. If the board is well-received, this could go a long way towards narrowing—if not settling completely—the sampling rate controversy. How soon then? Maybe by the end of 1982?

Meanwhile, analog is still alive and well. We note the renewed interest in wide-track analog machines, possibly as an alternative to digital. Of course, 24 or more tracks on 4-inch tape will surely appeal to no one, although 2 tracks on half-inch tape has practical possibilities, at least as an interim measure until those digital price tags come down.

Of course, it's not a new idea. We recall seeing some half-inch two-channel Studers in the studios of Radio Moscow more than ten years ago. At the time, while the rest of us were busy unravelling the mysteries of multi-track mono, the Soviets were more interested in coaxing a few more dB onto their stereo recordings.

How times change. MCI recently delivered a fully-loaded multi-track recording van with all the bells-and-whistles to the USSR, while we're all busy discovering Alan Dower Blumlein. Wide-track stereo machines are finding an eager market here in the USA, and someone could probably make a killing operating an East/West equipment exchange service.

Oh well, 1982 should prove to be an interesting year in both the analog and the digital domains. Stay with us, as we try to figure out just what will happen next. It should be interesting.

By the way, many readers have expressed their pleasure/displeasure at the recent 16-page supplement on British Audio, which appeared in our October, 1981 issue. Of course, we're sorry to disappoint those British firms who were left out of the supplement. Therefore, we're working up a few more British audio stories, which should be coming along shortly.

What about the rest of the world? Well, we have another International Audio issue coming in a month or two. Once again, we'll feature stories on what's going on here and there around the world. And once again, we'll get a bunch of letters that say, in effect, "Hey dummy—you forgot about *us!*" So, once again, *we* say, "Hey Peru, Yugoslavia, India, Nigeria—what's going on? We know you're out there! Will we be hearing from you soon? We hope so, but we're beginning to wonder when."

JOHN BORWICK

A Visit to Studer-Revox

Join our European correspondent as he takes us on a tour of the Studer factories and laboratories.

I SUPPOSE THAT somewhere "out there" designers and manufacturers are poised, ready to leap fully-clothed into the digital arena, without prior experience in good old analog. Yet we must surely look to the well-established analog engineers and builders to make most of the decisions—and get most of the mileage—in this accelerating run-up to the digital revolution. How then do these heavily committed people reconcile their involvement with a still viable and essential analog market and the considerable expense in brains and resources needed just to keep up with digital developments—or preferably take a leading part in digital progress?

These were just some of the thoughts running through my mind as I accepted an invitation to join a tour of the Studer-Revox factories and laboratories organized for a party of American journalists. Willi Studer himself, who in his 70th year still actively directs the company activities, welcomed us on the first morning of the tour and answered questions at the end. He is refreshingly forthright about the Studer-Revox attitude to digital audio. He sees the PCM disc as the next inevitable milestone in home music and, by a strange kind of reverse influence, he sees the wide acceptance of such a disc format as having a salutary effect on studio equipment. Once the digital industry as a whole sees a market for supplying digital masters to suit a given consumer format—e.g. the Compact Disc with its 44.1 kHz sampling frequency—they have both the technical incentive to ensure that their installations meet this need, and the financial incentive to get on with the job. Perhaps then we shall see an end to the slightly unreal, and seemingly endless Alice-in-Wonderland tea parties well-meaningly arranged by the AES, APRS and others to discuss the standardization or harmonization of digital recorders and processors.

GETTING IT RIGHT

Dr. Roger Lagadec, who is the Studer-Revox Product Manager for digital audio, pin-pointed some of the areas where the industry simply must get its act together if digital audio is to emerge as the technically superior and economically viable medium it is cracked up to be. Having heard him in action at some of the aforementioned "harmonization" meetings and presenting papers at AES Conventions, I already knew that Roger Lagadec was a persuasive speaker and in total command of his subject.

He sees three main areas of application for digital audio. The first has already been with us some time; namely a digital recorder which simply replaces existing analog tape machines for studio mastering. Since the end-product has been a conventional analog vinyl LP record, there has been no urge to build up an all-digital "system" to include the mixing console and its peripherals. Also, since there has been no great need to exchange master recordings between studios, few practical moves towards an industry standard have been made and at least a half-dozen different, and non-compatible, recorder designs are in use.

Secondly, in broadcasting, the needs have been for robustness, operational flexibility and a reasonable measure of standardization to allow tape exchanges. (This application has perhaps more relevance in Europe than in the USA. PCM has been used by the BBC for 10 years for network distribution, and true digital broadcasting is envisaged in a few years' time.

The third application, only now emerging, is digital recording as a first step in the manufacture of consumer digital discs. Here the basic parameters are dictated by the disc, but the potential market and high technical potential of the consumer product make it highly desirable to work towards a complete digital system so that multitrack production methods all operate in the digital domain.

As long ago as May 1980, Studer signed an agreement with the Sony Corporation on a format for a stationary-head digital recorder and though the wraps are still on at the factories, we were left in no doubt that considerable progress has been made. All the applications just mentioned are covered. On sampling frequency, for example, the format allows for 32 kHz (used by broadcasters), 50.4 kHz (conveniently high to give a Nyquist bandwidth up to 20 kHz without putting too severe constraints on the design and phase response of the necessary analog filters) and 44.1 kHz (specified for the Philips Compact Disc). This last

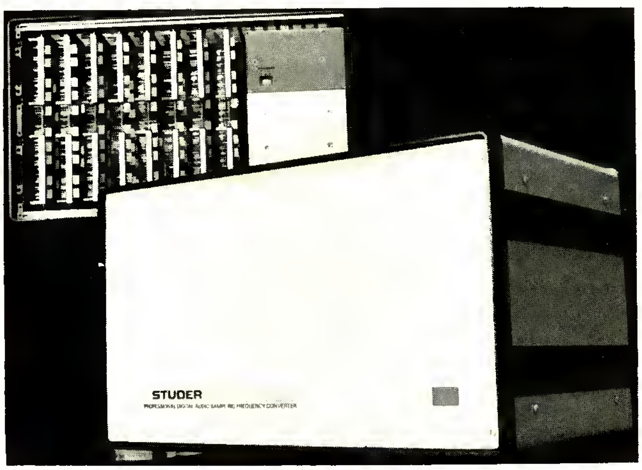


Figure 1. The SFC-16 Digital Audio Sampling Frequency Converter.

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

Does the Revox PR99 remind you of something?

It should.

It's inexpensive to buy, easy to maintain, practical, reliable, made in West Germany, and built to keep on going through years of demanding use.

No gadgets, no frills. Merely a solid machine equipped with everything a professional needs to get the job done efficiently —and with high quality results.

What's more, the PR99 now comes in three versions: all-steel console, transport case with snap-off cover, and rack mount. All three may be ordered with the new monitor panel: speaker, amp, volume control, and track selector.

Other features include: • Die-cast chassis • Precision machined headblock • Balanced line in and out switchable for calibrated or uncalibrated mode • Servo-controlled capstan motor • Edit mode switch • Tape dump • Self-sync • Choice of 3.75/7.5 or 7.5/15 ips • Remote control and vari-speed available.

In short, the PR99 is so versatile, so dependable, and so downright sensible that you could almost call it a "volkscorder."

Except, well...the shape isn't quite right.

STUDER REVOX

Studer Revox America, Inc.
1425 Elm Hill Pike, Nashville, TN 37210 (615) 254-5651
Offices: Los Angeles (213) 780-4234
New York (212) 255-4462
Canada: Studer Revox Canada, Ltd.

Circle 27 on Reader Service Card

reference ties in with Studer's announcement in April 1981 that they have entered into a license agreement with Philips and Sony to begin production of disc players for the Compact Disc.

Additionally, the Studer/Sony digital recorder format incorporates a comprehensive coding scheme which will cope with error correction in the severest drop-out situation and also allow cut-and-splice editing as well as the electronic variety. It can be built for any number of channels from 2 to 48; the two-channel (stereo) version employing standard quarter-inch tape at 7½ ips (19 cm/s) and the 24-channel requiring half-inch tape at 30 ips (76 cm/s).

Figure 3. Studer's entry in the digital arena: 24 audio tracks on half-inch tape at 30 ips.

SAMPLING FREQUENCY CONVERSION

One problem area which Studer have already tackled successfully—and *are* prepared to talk about—is sampling frequency conversion. They have designed and built an all-digital converter offering several useful side-applications in addition to the basic one of interfacing equipment built to different sampling frequency standards.

As anyone who has kept up with digital discussions will know, a plethora of sampling frequencies is currently in use—48 kHz and 50 kHz as well as the 32 kHz, 44.1 kHz and 50.4 kHz already mentioned. Since there is no universal agreement on one of these as standard, there is a very real need for an efficient converter if users of the different systems are ever going to be able to "talk to each other," i.e., exchange programme material. Of course, the simplest method is to go back to analog. The programme to be copied or transmitted is played through its own digital-to-analog converter. Then the analog signal is fed through the usual smoothing, band-limiting and sampling circuits of the second machine and re-digitized.

Obviously, this is far from being an ideal procedure. Despite what the public relations people will tell you, present-day technology cannot achieve this D/A and A/D conversion with all the necessary analog filtering without significant signal degradation due to noise, distortion and phase discrepancies (though Studer have indeed solved the filter design problem; see the paper by Roger Lagadec and colleagues, "High quality analog filters for digital audio" presented at the October 1980 AES Convention, preprint number 1707). The only point in favour of this primitive conversion procedure is that it will work equally well for any random ratio of the two sampling frequencies, and cope with an unsynchronized situation where frequencies may drift.

A second approach has been advocated, relying on high-speed digital filters operating at the lowest common multiple of the two frequencies. The conversion is achieved by interpolation, with most of the samples being discarded. This is technically more elegant, since it avoids any intermediary return to analog. However, it is relatively inflexible in that the given digital filter will work for one frequency ratio only, and is intolerant of frequency drift, so that synchronization is essential.

The new patented Studer converter achieves the best of both worlds. It operates on the basis of the clocks of the two systems being interchanged and so can accept any arbitrary ratios of sampling frequencies, even varying ones, and needs no programming. It is purely digital and has perfect linear phase with ripple less than 0.15 dB over the whole audio band. On receipt of the input signals, it calculates the precise points in time at which interpolation is required with an accuracy of 16 bits. This means a resolution capability in the time domain of 150 picoseconds (0.15μs). The two-channel version has been built to 19-inch rack width with a height of only 11 inches, and includes the digital interfaces needed to connect any two types of digital equipment. Dedicated cards are needed for any given pair of digital equipment types. The input and output connectors on the back panel handle the incoming digital and outgoing digital audio signals, plus word clocks and the bit clocks if required. The front panel simply carries a power on/off switch.

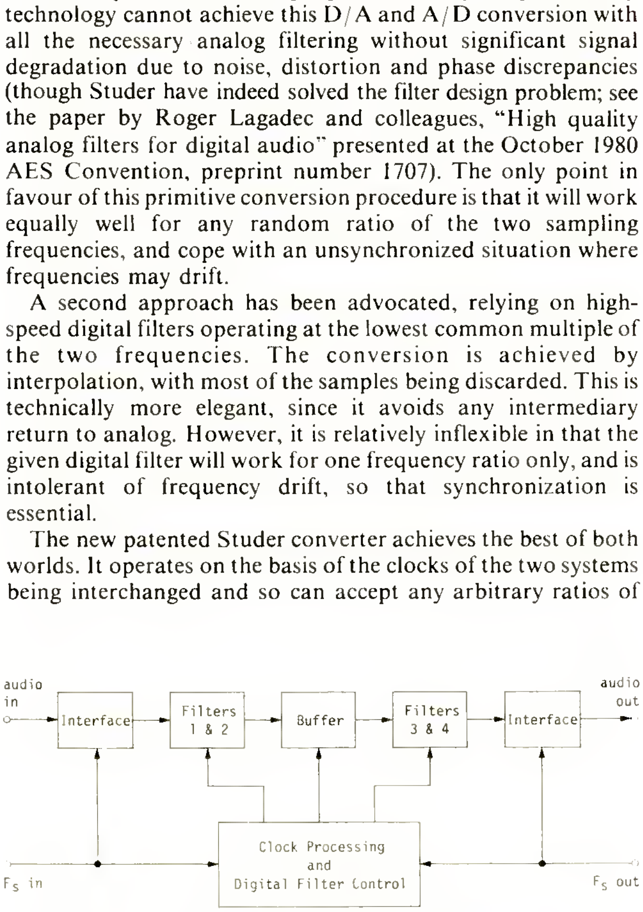


Figure 2. Block Diagram of the SFC-16. The real-time control signals for the operation of the sampling frequency converter are obtained by digital processing of the clock signals, and so it is not necessary to program the converter for any particular ratio of sampling frequencies.

The digital techniques used, not yet revealed, extract the high accuracy timing information from the clock signals. This controls the digital filters, eliminates short-term jitter and effectively ignores drift or other variations in sampling frequency. At the same time, the control circuitry can determine how much band limiting is required (necessary when sampling frequency changes of more than a few percent are being made) and introduce a bank of linear-phase digital filters for the purpose. The same converter hardware can equally perform the operations of increasing or decreasing sampling frequency.

As Roger Lagadec explained it, Studer see at least five distinct applications for their new converter.

1. *Program transfer between digital recorders.* This is only an occasional requirement at present but, should the industry eventually reach agreement on a standard sampling frequency for professional use, the converter will enable all material recorded on other systems to be adapted to the standard without pitch changes, signal degradation or distortion.

2. *Mastering for Compact Disc production.* Interfacing with equipment operating at the proposed 44.1 kHz sampling frequency of the Compact Disc system looks like it will be a widespread requirement, at least from late-1982 onwards. The Studer converter can meet this demand while preserving the full bandwidth of the Compact Disc. It is also universally applicable to all other systems, and is less complex than technicians requiring a dedicated converter for each frequency ratio and application.

3. *Broadcast network distribution.* A broadcast version of the Studer Sampling Frequency Converter is not only compatible with the agreed 32 kHz frequency now used. It also tolerates frequency fluctuations and so relieves the digital recorder of the need for synchronization facilities.

Figure 4. Interior view of a Studer-equipped mobile van.

4. *Pitch control/varispeed.* Attempting to introduce pitch control by varying the tape speed of a digital recorder is impracticable because the sampling frequency is also changed by the same ratio, thus preventing further processing. The Studer converter can accept this changed sampling frequency, restore it to the original value and so permit true digital pitch control as flexibly as in the analog situation.

5. *High-quality error concealment.* All digital audio systems run the risk of drop-outs which are too gross for complete error concealment. The Studer converter incorporates a powerful interpolating filter which can produce better error concealment of detected and even undetected errors than existing digital recorders. In this application, the converter can operate automatically and without supervision, thus saving valuable editing time.

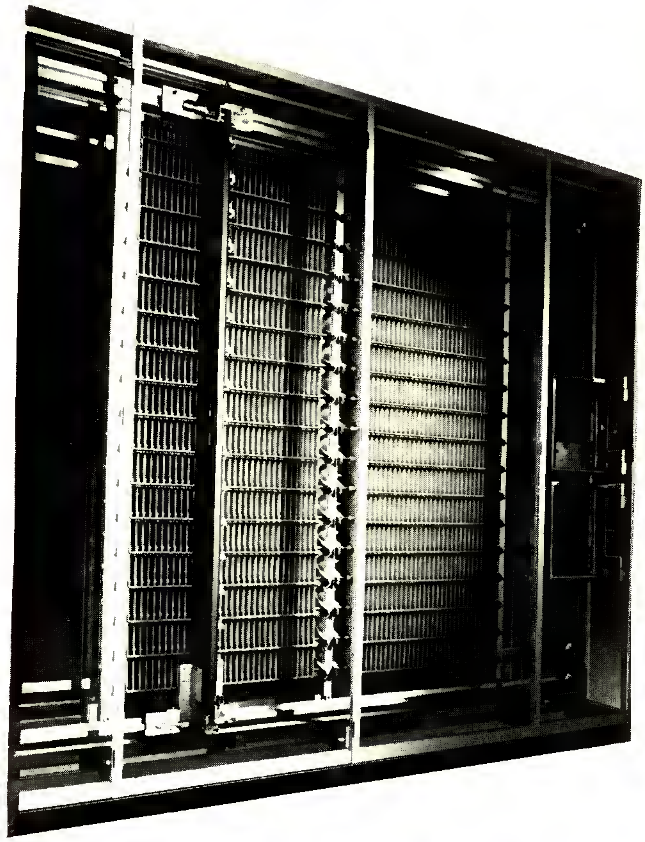
MAKING ANALOG BETTER

While engaged actively in developing digital equipment and techniques to the highest standards, Willi Studer and his team are continuing to apply their high ideals to analog. On tape recorders, for example, they feel that their A80 and A800 series can equal the performance of any analog machine on the market—yet they want to make them better still.

Naturally they demonstrated their version of a "super analog" tape recorder, in the half-inch format also available from Ampex and others. The Studer A80VU two-track mastering machine (suggested list price $10,500) achieves better than 75 dB signal-to-noise, or up to 80 dB dynamic range at 1 kHz and 30 ips speed. It uses the new Studer transformerless line output amplifiers recently offered as replacement plug-in cards for all existing A80 models, (at $150 each). This is actively balanced and fully AC floating, with a low 22 ohms source impedance to operate into long cables or other tricky loads without signal degradation. Amplifier frequency response is 14-50,000 Hz (+0, −1 dB) and THD is less than 0.01% at +24 dBm.

The new A80 version was demonstrated in a mock-up broadcasting studio which served two purposes. First, it was a showcase for all the different tape machine versions, from the A800 24-track down to the B67 and Revox models and including an A80 RC-1 PNVU recorder incorporating a pilot-tone resolver. Second, it underlined the turnkey side of Studer's operation in which they can draw up specifications and undertake complete building of broadcast or recording studios down to the last nut and bolt, and even including transmitters. A centre-point of attraction was the ingenious Studer TLS2000 Remote Control system. This used the SMPTE time code to link all the machines, and could for instance lock two 24-track machines together for 46-track operation. The sync and clock displays also work in fast-wind modes and the accuracy of sync was demonstrated by running two identical tapes in anti-phase to show the effectiveness of signal nulling. The remote Autolocator was a simple keyboard, since all the relevant controls are built into the A800. Apart from all the usual storecue and roll-back functions, this could program loops, for example, and the varispeed could be set to 6-digit accuracy either in percentage speed shift or in semitones. (This did great things for a pop vocal which we transposed by one semitone at a time for a whole octave.)

Studer's range of mixing consoles has always been more familiar in Europe than the USA, but a substantial growth in model types and facilities offered could change that (see article by Thomas Mintner in September 1981 **db** Magazine). Being a European myself, it took me some time to puzzle out what was



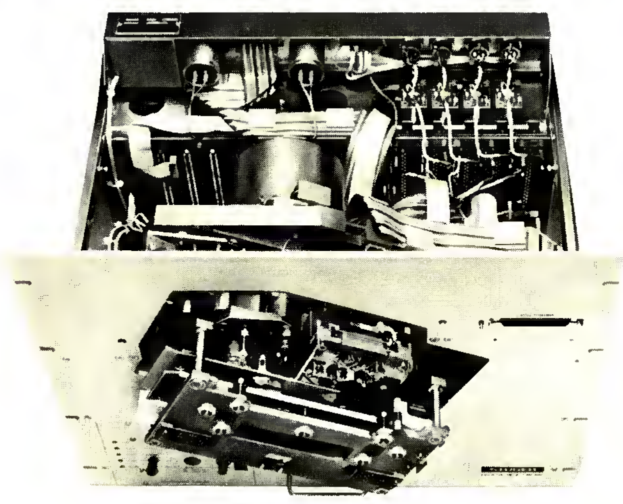


Figure 5. The CAR 3040 cassette Library System, and CAD 3010 Cassette deck.

GO AHEAD. STOMP YOUR FEET!

You've got an ATM Instrument Microphone System.

You're on stage to make music, not noise. But most microphones will respond to everything that hits them. Including noise coming through the mike stand. Except these new ATM microphone systems. Because each of these specially-designed instrument mikes includes a *very* effective shock mount and a windscreen.

Even if you're on a "bouncy" stage, you needn't tiptoe when an ATM microphone system is at work. Distracting noises are reduced... not amplified. Including floor resonances from speakers nearby. Or the clunks when you raise or lower the mike. All the audience hears is your chops.

But a great microphone system is not just a shock mount or a piece of foam. At the heart of our systems are three superb studio-quality microphones: a unidirectional dynamic, a unidirectional condenser and an omni condenser. Road tough? Of course. But with response specially tailored with uncanny accuracy for instrument reproduction.

With these ATM microphones a trumpet is bright, not strident. Trombone is dark but not murky. Reeds are full but not thick. And drums are crisp and clean, not fuzzy or thumpy. For two important reasons.

First, frequency response is smooth and peak-free and extends well beyond the limits of your instrument. So the balance between overtones and fundamental isn't distorted. And one part of your range isn't favored over another.

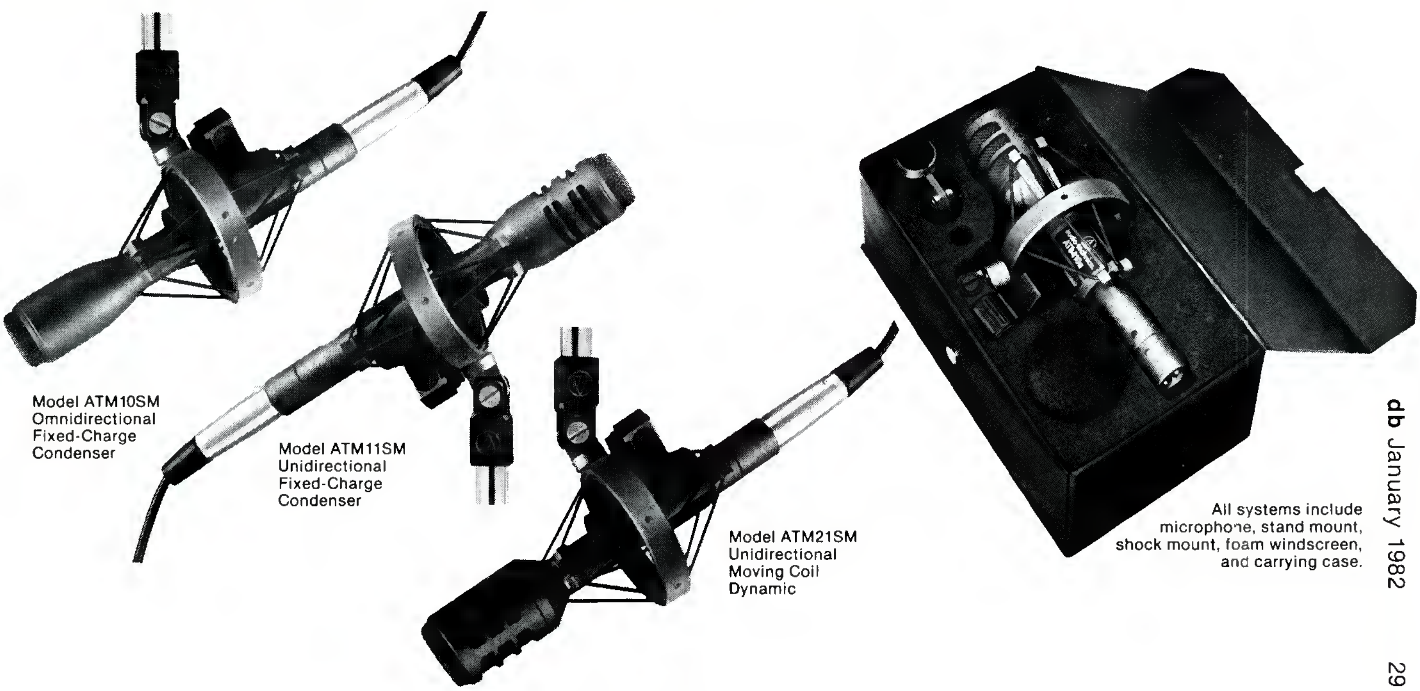
Second, and equally important is our wide dynamic range... designed to capture and amplify all of yours. It's almost impossible to overblow our ATM dynamic, for instance. And our electrets will handle up to 130 dB with ease. So your *fff* crescendo won't come out just *ff*.

Great sound and no distractions. The best possible way to start your sound system working *for* you. ATM Instrument Microphone Systems are waiting for you at leading pro music dealers everywhere. Kick up your heels!

AUDIO-TECHNICA U.S., INC., 1221 Commerce Drive, Dept. 11BD, Stow, Ohio 44224. In Canada: Audio Specialists, Inc., Montreal, P.Q.

audio-technica

Great sound right from the start!



Model ATM10SM Omnidirectional Fixed-Charge Condenser

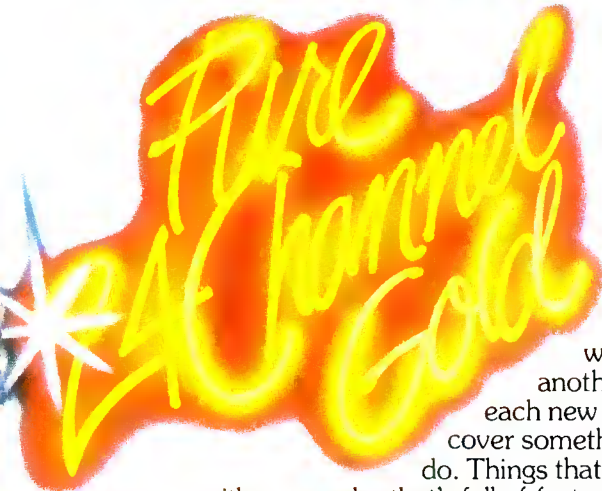
Model ATM11SM Unidirectional Fixed-Charge Condenser

Model ATM21SM Unidirectional Moving Coil Dynamic

All systems include microphone, stand mount, shock mount, foam windscreen, and carrying case.

Circle 21 on Reader Service Card

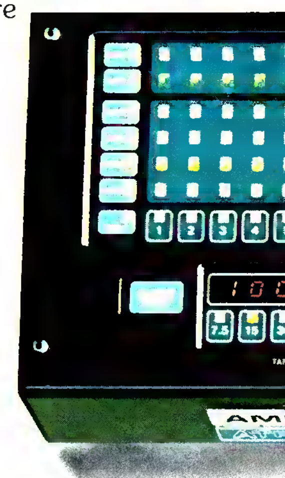
Introducing a present



Once you go through a recording session with the new ATR-124 24-channel recorder by Ampex, you'll want to go through another. Because with each new session you'll discover something new you can do. Things that you can only do with a recorder that's full of features of the future.

ATR-124 gives you the unheard of: Time on your hands.
Which means you can use that time to give clients more of what they're paying for—your creative skills. With the ATR-124 microprocessor-based control system, you can pre-program what you want to do ahead of time so you won't waste studio time setting things up. When their time starts, you're ready to record by touching a single recall button.

ATR-124 also lets you duplicate a technique you may have used earlier in the session without having to rethink what you did. Just touch the memory button and it'll all come back to you. ATR-124 lets you rehearse what you've got in mind, without recording it, to make sure what you've got in mind is right. Tape can be manipulated faster which means you'll get the sound you want sooner. And the chance to try something "a little different." All because of the speed and accuracy that ATR-124 puts at your fingertips.

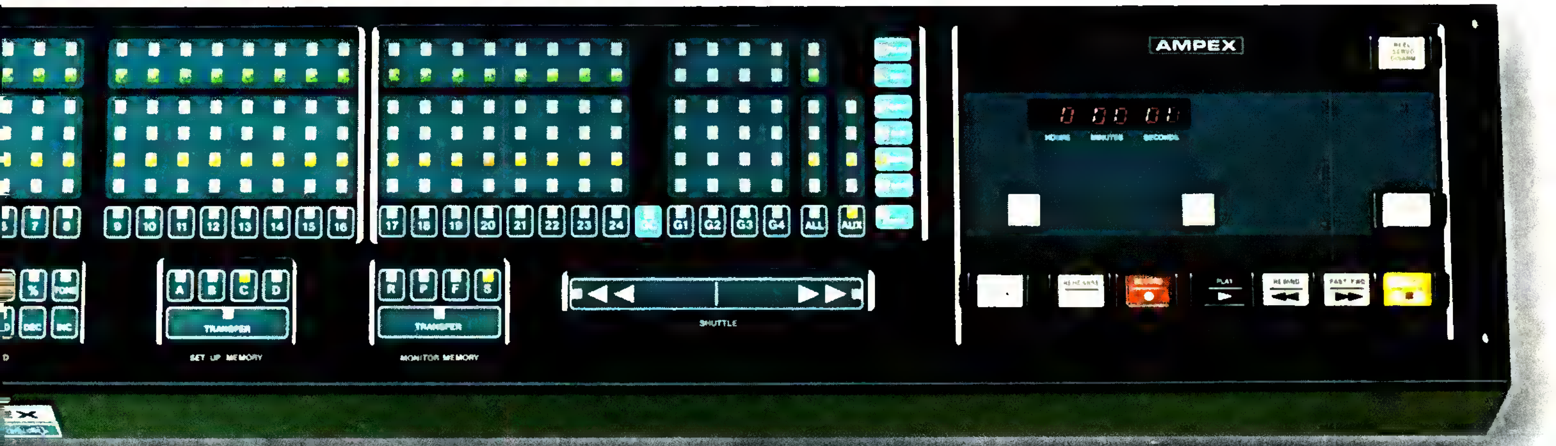


ATR-124 doesn't take away your creativity, it adds to it.
The less time spent setting up, correcting, and redoing, the more time spent creating. And when you add features that help you create to the ones that help you save time, you've got one very potent piece of audio machinery. Take the control panel for instance. It's like nothing you've ever seen. Pushpads linked to a microprocessor give you a new level of creative flexibility. Program a setup, then change it. Then change it back, all with a single fingertip.

A repeatable, variable speed oscillator for pitch correction and special effects is built in. In addition

from the future: ATR-124.

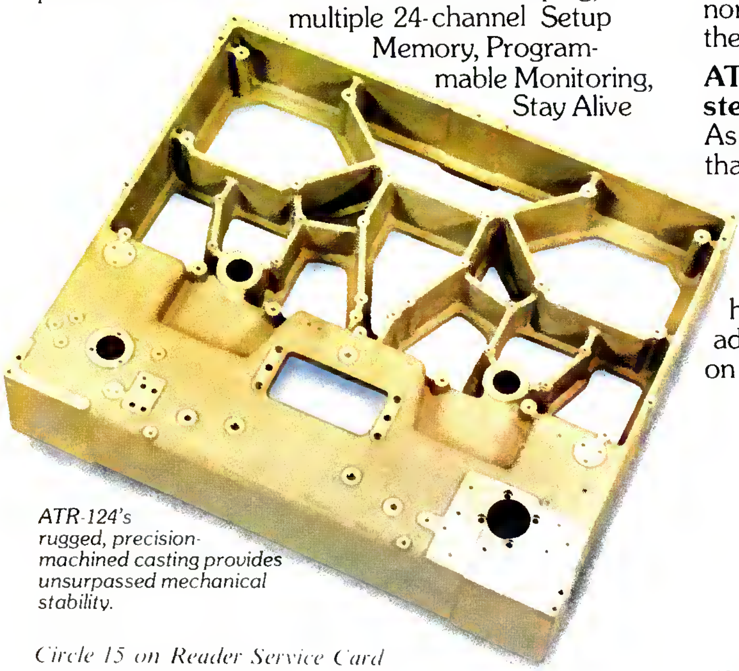
to the standard output, there is an optional auxiliary output with each channel that enhances flexibility. So don't think that ATR-124 is going to Memory, and Record Mode diagnostics. The point is this: If you like the ATR-100, you're going to love working with the ATR-124.



ATR-124's Control Panel. Speed and accuracy at your fingertips

replace anything that you do. On the contrary, it's going to improve the skills you have, if not help you develop some new ones.

ATR-124 picks up where ATR-100 leaves off.
It's only natural that the people who brought you the ATR-100 should be the ones to bring you something better. ATR-124 offers you 24 channels instead of 4. You also get many new and exclusive features. The kind that have set Ampex apart from the crowd for the last 30 years. Features like balanced, transformerless inputs and outputs; a patented flux gate record head; 16" reel capability; input and output signal bus for setup alignment; membrane switch setup panel; fingertip-operated shuttle speed control; and microprocessor-based synthesized Varispeed -50% to +200% in .1% steps or in ¼ tone steps. ATR-124 also features microprocessor-based control of Channel Grouping, multiple 24-channel Setup Memory, Programmable Monitoring, Stay Alive


ATR-124's rugged, precision-machined casting provides unsurpassed mechanical stability.

ATR-124 options.
As impressive as the ATR-124 itself.
With the addition of a built-in Multi-Point Search-To-Cue (MPSTC), you can rehearse edits and control five tape-time actuated events and be compatible with SMPTE time code. Separately controlled auxiliary output amplifiers with each channel provide simultaneous monitoring of normal and sync playback as well as all other monitoring modes. A roll-around remote control unit can also be added to the ATR-124 which contains all control features normally found on the main unit.

ATR-124's Multi-Point Search-To-Cue (MPSTC). Provides 100 cue locations.

ATR-124. Your next step is to experience it firsthand.
As you scan the points we've covered, remember that you're scanning just a small portion of ATR-124's story. We haven't even begun to discuss the accessibility of key components for easy servicing and minimal downtime, or the features we've built in to give you greatly improved tape handling. To find out more, write to us at the address shown below. We'll send you a brochure on ATR-124, our latest audio effort.

ATR-124. Pure 24-Channel Gold From Ampex.

Circle 15 on Reader Service Card

AMPEX

Listen to the future

Audio Video Systems Division
401 Broadway, Redwood City, California 94063 415/367-2011

unusual in the Studer demonstration studio; then I finally got it. Slap in the middle was a Harrison mixing console. (Studer are agents for Harrison in Switzerland and naturally include Harrison desks in their range of turnkey complete installations.) To my eyes, this perfectly handsome heap of multi-coloured LEDs and knobs on a black leather-look panel with walnut side-panels seemed garishly out of place in a roomful of low-key, European-style equipment made almost entirely of burnished steel and soft grey enameling. Visual styling is clearly a matter of quite different tastes on either side of the Atlantic; and a Studer A800 might stick out as equally unusual in a typical USA studio environment.

CASSETTE AUTOMATION FOR BROADCASTERS

It was the turn of my American companions to look perplexed when we were shown a huge new system being built for South German Radio (SDR) Stuttgart. This was the first version of Studer's CAMOS 3000 Cassette Automation Modular System to be sold, though other orders are in the pipeline. The sheer size of the project was staggering—just the cassette player stack was going to cost $80,000, and the whole Studer package, at about a half-million dollars, was only part of a modernization scheme which the Stuttgart radio station had contracted Siemens to install. Such large networked radio stations, though commonplace in Europe, are again in marked contrast to the American preference for small local stations. Nevertheless, while expecting to do most CAMOS 3000 business in Europe, Studer had carried out an extensive market survey in the USA and felt that the main networks might indeed be interested. They calculated that, out of 9,000 US local radio stations, 7,900 broadcast commercials. The total income from commercials last year was $4 x 10^9$, of which $1 x 10^9$ came from networked broadcasts. Therefore, Studer argues, there would seem to be enough dollars around to invest in one or more CAMOS 3000 systems which provide automated, foolproof, programming of commercials—or other material.

Very briefly, the system consists of a library store of cassettes which can be accessed by computer, with command keyboards and visual display units in the studio and production rooms. The desired sequence of commercials or other programme items for a whole day's broadcasting, or any other time span, can be keyed in advance. The relevant cassettes are automatically extracted from the store (it is fundamental to the system that cassettes are never touched by human hands) and carried by conveyer to the four-deck player stack. At all times the disc-jockey, and indeed everyone concerned, has access to a complete display of the program order and all relevant data, so that incorrect sequencing is impossible.

The system is based on the professional Unisette (BASF registered trademark) which resembles the Japanese Elcaset and is about the size of a VHS videocassette. It uses ¼-inch tape at 3¾ ips and the rugged Studer CAD 3010/11 deck gives wow and flutter of 0.03 percent with a frequency response of 30-18,000 Hz ±2 dB. Start and stop times are 200 and 40 milliseconds respectively, and the winding speed is 4.75 metres (187 inches) per second. An SMPTE cue track gives speedy search and cue operation with 100 ms accuracy.

Each CAR 3040 library store module has capacity for up to 1,024 cassettes. The CAPS 3030 player stack has four decks and two storage belts holding up to 43 sequenced cassettes in each. All play and record modes are controlled by a minicomputer, and complete logging and statistical data of the station's daily operations are entered automatically. The modular construction allows each installation to be custom designed, and special versions for program building or commercials-only are available.

Touring the Studer-Revox factories (there are two in Switzerland and four across the border in West Germany) has been an illuminating experience. Practically all parts are made in-house and only the very latest machine tools, computer-controlled component-inserting robots and automated quality-test cycles are used. Suddenly, when each tape recorder or mixing console gets to the end of the line, automation ceases. Then a single skilled engineer takes a whole day to check the final assembly and technical specification of each individual unit—even though the record-card with each item shows that every component part has already undergone careful testing and run-in cycles. Whether it is analog or digital, Studer-Revox seem assured of a special place in the audio world now and in the future. ■

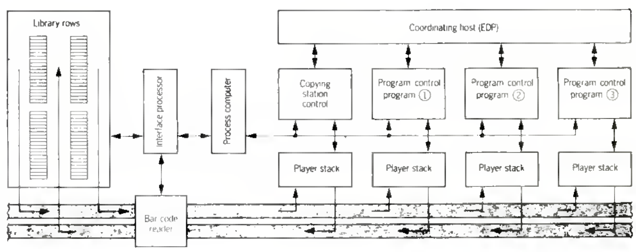


Figure 6. Block diagram of a completely-automated Program Execution System, as installed at South German Radio in Stuttgart.

MUCH MORE FLEXIBILITY.

Presenting Yamaha speaker components. Some have been available only in Yamaha-designed enclosures. The rest are brand new. All are designed to deliver outstanding performance, reliability and durability.

And now, with this full range of Yamaha speaker components to choose from, you have the flexibility to design a sound system that meets your specific needs.

Low frequency drivers: JA3882 & JA3881 The new JA3882 low frequency driver is specifically designed for high-efficiency sound systems. Its very high sensitivity level enables it to produce high SPL with less amplifier power. With large edgewound copper voice coil, durable cone assembly and stiff compliance, the JA3882 produces clean, powerful low frequencies.

☐ 15" diameter cone
☐ 102dB SPL/1 meter/ 1 watt sensitivity
☐ 30Hz to 4kHz Frequency Range (maximum recommended crossover, 800Hz)

The new JA3881 is an excellent choice for use in stage monitors, keyboard speaker systems, or for 2-way systems. Its flexible suspension and lightweight edgewound aluminum coil give a usable frequency response of 40Hz to 5kHz.

☐ 15" diameter cone
☐ 97dB SPL/1 meter/1watt sensitivity

High frequency compression driver: JA6681B With high sensitivity and high power handling capacity, the JA6681B compression driver makes an excellent mid to high frequency reproducer for use in 2- or 3-way full-range, high-level sound reinforcement or monitor speaker systems. Its unique S-shaped beryllium/copper diaphragm suspension system and precision construction ensure long life.

☐ 145dB SPL reference sensitivity, 1 watt, using 1 inch plane wave tube
☐ 108dB SPL at 1 meter, 1 watt (using Yamaha H1230 horn)
☐ Nominal 800Hz-12kHz, usable down to 500Hz

Combination high frequency horn & driver: JA4280B/H1400 This high frequency reproducer's versatility enables it to be used as the mid and high frequency reproducer in a full-range stage monitor, keyboard monitor, or general sound reinforcement system.

☐ 90° H x 40° V dispersion
☐ 106dB SPL at 1 meter, 1 watt
☐ Aluminum horn with damping
☐ Nominal 1,500Hz-16kHz, usable down to 800Hz.

Compression tweeter: JA4281B This new tweeter is a high-sensitivity, integral horn/driver unit designed to handle the uppermost portion of the frequency spectrum. It is an excellent super-tweeter for use in 3-way or 4-way full-range, high-level sound reinforcement systems. Its superb on/off axis response and absence of diaphragm resonances also make it a fine choice for studio monitor systems.

☐ 120° dispersion pattern at 10kHz
☐ 108dB SPL at 1 meter, 1 watt
☐ Nominal 7kHz-20kHz

High frequency radial horn and throat adaptors: H1230, AD3500 & AD3502 The new H1230 aluminum radial horn is designed to provide controlled dispersion (90° H x 40° V) of high frequencies in high-level, wide-range systems. The AD3500 throat adaptor is used to couple the horn to the JA6681B driver. Use the AD3502 throat adaptor to connect two drivers for greater output.

That's the lineup of Yamaha speaker components that leave the system design up to you. So now, with Yamaha mixers, power amps, signal processors, and separate speaker components, you've got the total flexibility you always wanted. From the people who know what music sounds like – Yamaha.

For complete information, write: P.O. Box 6600, Buena Park, CA 90622. (In Canada, write: 135 Milner Ave., Scarb. Ont. M1S 3R1)

Because you're serious.

YAMAHA

Circle 26 on Reader Service Card

DAVID HOLMES

Sigma's New Console
or Sphere Meets Sigma

The following is a description of what evolved from a series of design meetings between Sigma Sound and Sphere Electronics.

Sigma Sound Studios' association with Don McLaughlin began some fourteen years ago when we purchased our first console, a sixteen in, eight out custom Electrodyne board. The solid state technology employed made it the forerunner of what was to become today's modern audio recording consoles. Twice sold and constantly updated, it is still giving clean, reliable service today. So, it was no surprise that when Studio 1 was to be totally rebuilt, the console we would choose would be one designed and built by Don McLaughlin. His company, Sphere, is one of the few remaining audio equipment manufacturers still offering to build consoles of custom design.

Studio 1 was built in 1968 and approximately every 2 years since that time Sigma has opened another multi-track recording studio. We now operate six 24/48 studios. Our object with each room has been to make it better than the last. Studio 1, being our first, is special to us; so it made sense that everything done in its rebirth should be special. We wanted a console capable of superior sound quality, efficient to use, and containing all of the latest practical features. We decided we wanted a 48 input system with custom equalizers that featured 4 bands, variable Q, and low and high pass filters. We wanted a completely new constant update automation system that was SMPTE time based.

Joe Tarsia has some pretty good ideas about what it takes to operate a successful business in the increasingly competitive studio industry. He's logged fifteen years as a very successful studio owner, worked on many albums as engineer or producer or both, and had a stint as president of SPARS.

When he began planning the renovation of Sigma's Studio 1, there was perhaps one variable that, along with sound quality, stood above all others: versatility. The new room and equipment had to be able to serve jingle clients in the morning, SMPTE-based video scoring in the afternoon, and R&B all night—and all this with minimum changeover time.

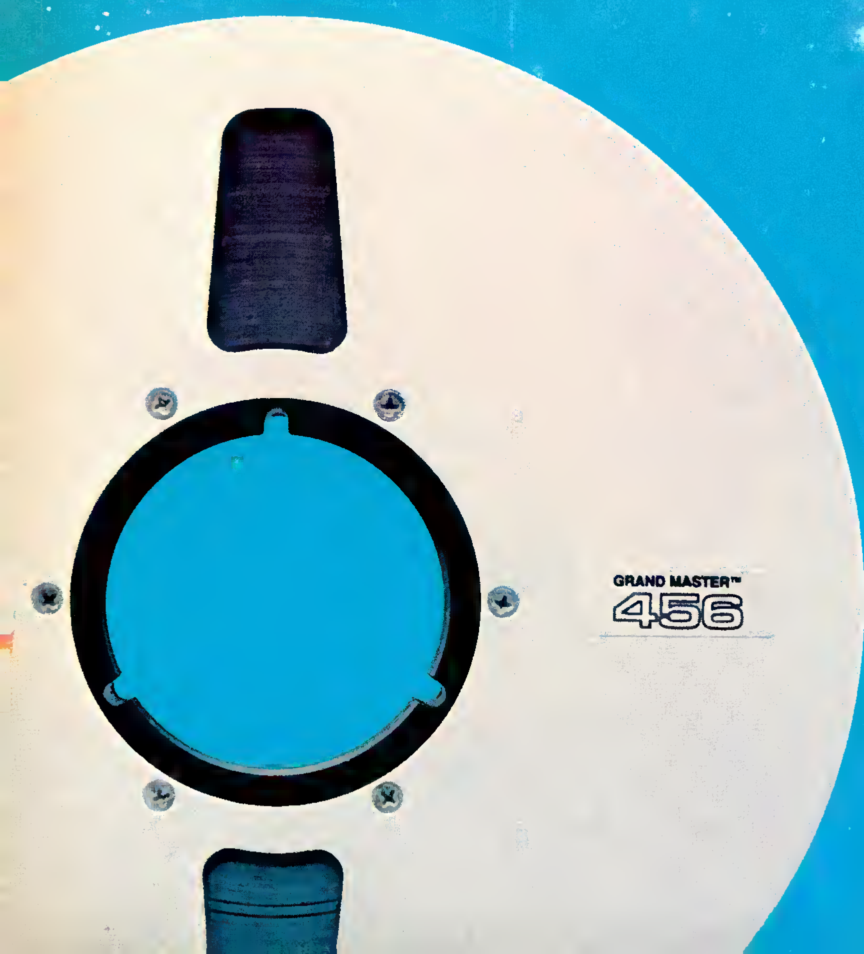
Figure 1. Sigma's new Sphere console, prior to the installation of the Travis faders.

REPEATABILITY

One other variable figured almost as close to the top of Tarsia's list and provided us with the greatest challenge: exact console repeatability. Why? For the most part, a studio with as much diverse business as Sigma cannot offer "lock-out" bookings. Therefore, if a mix doesn't quite "happen," the engineers are obliged to keep painstaking and time-consuming records of all console settings. Later on, when the group returns to carry on where they left off, much time is spent getting back to square one.

David Holmes is the sales and marketing manager for Sphere Electronics.

AIM HIGH FOR MORE HITS. DEPEND ON AMPEX TAPE.



3 OUT OF 4 RECORDING STUDIOS DO.

Ampex professional tapes are used to master more hit albums than all other brands combined. Moreover, they are used by 3 out of 4 studios in America. Impressive facts. But, so are the reasons.

In just 7 short years, our Ampex Grand Master 456 Professional Recording Tape has become the unquestioned industry leader. It has a wider dynamic range than any other professional recording tape. It's bias compatible, so you won't have to waste valuable studio time adjusting bias. And it's a "hot" tape—the kind today's professionals demand.

Naturally, Ampex 456 has all the other characteristics you'd expect from a professional recording tape. Like the highest possible signal-to-noise ratio and a saturation capability that's the best in the business. It also has the industry's lowest distortion, unwavering physical stability, high durability, and the ability to perform perfectly under all conditions.

If you still can't decide which tape to use for your next session, here's a simple test. Ask 4 studios. Ask 40. Odds are they'll recommend Ampex.

AMPEX

REFLECTIONS OF REALITY. AND BEYOND.

Ampex Corporation, Magnetic Tape Division, 401 Broadway, Redwood City, CA 94063 415/367-4463

Circle 33 on Reader Service Card

This problem also exists, albeit to a lesser extent, during tracking and overdub sessions. The week(s) spent tracking would flow much smoother if the engineer didn't have to get the drum kit sound from scratch each day. And, if the monitor mix changes could occur as fast as tape is loaded, overdub sessions could concentrate more on musical performance than logistics.

As Sigma looked to the future, they saw that the solution to these problems would become even more important as they diversified their client roster to overcome the effects of a slumping record industry.

One of the reasons Sigma chose Sphere was our willingness to do custom work. However, I must admit that if we had known then what we know now, we probably would have said, "Pass." But in retrospect, I'm extremely pleased we didn't. Sigma provided many of the ideas, much of the inspiration, and some of the money that propelled Sphere smack into the middle of the pro audio digital decade. And now, with the excellent reception our Travis digital fader and attenuator got at the last AES show, we hope we can "lead the way" in digitalizing consoles. (More on this later.)

Level repeatability was already reasonably in hand at Sigma, since all of their rooms were equipped with Allison (Valley People) 65K automation. There was much initial discussion about an SMPTE-based constant-update automation system, but all finally agreed to solve first things first. In any event, the modular console format of our Type C console allowed us to provide 65K VCA automation that would continue Sigma's present room intercompatibility until the time when our dual floppy disk system was ready to retrofit the console in Studio 1, and in a number of other Sigma rooms as well.

We all agreed that the first point of attack was the equalizer. Repeatability problems were most severe—and most critical—in this console component. Furthermore, Joe Tarsia had some definite feelings about the eq format he wanted. It was to be a four-knob *true* parametric, with high/low pass filters and variable Q; that's not an easy format from which to gain accurate repeatability. The design that came out of our first meetings was for a dented parametric that would be the easiest way to solve the problem. (If anyone wants a fine, detented parametric just let us know; all the engineering is done.) So much for easy.

By this time Sphere was well underway toward completing the first Sigma console—note I said "first." As discussions continued between Sphere and Sigma, it became increasingly clear that the one we were building wasn't the one for Studio 1. Fortunately, two very timely events occurred. Sigma construction on Studio 1 was delayed for a number of reasons and months. This gave us ample time to begin afresh and to do the job right. As a bonus, we found a ready buyer for the just-about-completed console, who scored a 48-input type C frame for the price of a 40-input version. Everybody was relieved and we looked forward to greater accomplishments.

The second time around was a whole new ballgame. Our discussions so far had concentrated on equalizer repeatability. After one long late night meetings, Tarsia was driving Don McLaughlin (Sphere owner) to the Philadelphia airport and the conversation reportedly went something like:

Tarsia: "...as long as we're going to the trouble to do the eq, how about the rest of the console?"

McLaughlin: "Huh?...oh sure...we'll look into it, Joe." (I think he may have been drinking a bit.) In any event, we shortly hit upon a way to sample the type C that provided a sort of console check-out as a by-product. The way some of the "other guys" do it is to hang an extra pot or switch lug on each point to be sampled. By contrast, we generate an audio signal that passes through the very pot or switch and that signal is converted to digital data and stored. We call our system STATUS DATALOG.

NUTS AND BOLTS

Now that we had the format, we started on the nuts and bolts. We left behind the notion of a detented parametric since it was no longer needed for repeatability. The sampling scheme could look at a continuously-variable pot as easily as anything else, so that's how we made it. Sigma also liked the idea of having a single eq level knob that may be *either* boost *or* cut, so that when the knob is rotated full-left, the engineer knows he is at "O." You can't get that exact "O" with a continuously-variable pot, even with a detent in it—electronics drift a bit from day to day. So now we had a four-knob parametric with continuously-variable controls on frequency select, level, Q, high-pass and low-pass filters, with shelving choice at the top and bottom and phase reverse, all of which were able to be sampled and the information stored.

We applied the same sampling logic to the rest of the console controls. Each knob, switch and pan on the whole input strip (including the cue, echo and effects sends) is sampled and stored. So is the entire master panel and each echo and effects return, level and pan; the whole enchilada. Once this data has been loaded into computer memory, it can be stored on any tape or on a floppy.

To recall this information, and to reset the console, the stored data is reloaded into computer memory and a sequence is begun, starting with the masters and echo returns, and continuing through each input.

There are two ways to align the console to the returning data. The Datalog write-in strip is an 8-digit alphanumeric display above each fader unit. A console-mounted keyboard allows entering track assignments (kick, snare, hat, etc.). As each point on the input is reached by the computer, this display will read out what it is and the status relative to incoming data; for instance, "CUE A LOW." As soon as the null point is reached, the computer indexes to the next point. Old and new levels can also be visualized on two VU or bargraph meters. A provision is made to let the computer index through each point or to manually direct the computer to the section desired.

The console mounted with Status Datalog was on display at the AES-NY in November 1980 and then delivered to Sigma. While there have been the inevitable programming errors to debug, the experience has been very positive and we are quite pleased. We are even more excited about the sister Datalog system for fader level control that is just now into production.

Quite frankly, when we started work on the dual disk drive, SMPTE time code-based-automation system that Sigma wanted on their consoles, we had no idea it would lead to the rather revolutionary system we now have ready to go. In their New York studios, Sigma has three custom consoles that are mounted with a fader that was rather novel a few years ago. Its features include a continuous belt that is pushed along by the engineer's finger, and an LED cursor that indicates level. Sigma wanted something similar, but without the belt.

Numerous ideas were explored and abandoned. Then John Hall, a Sphere design engineer, introduced McLaughlin to

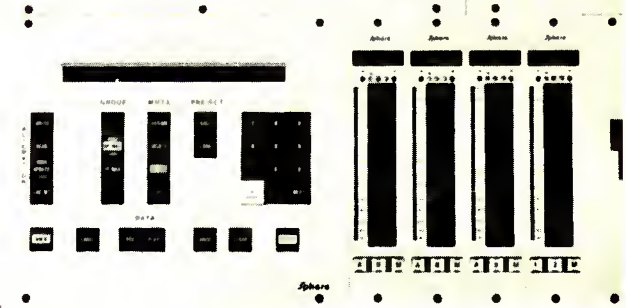
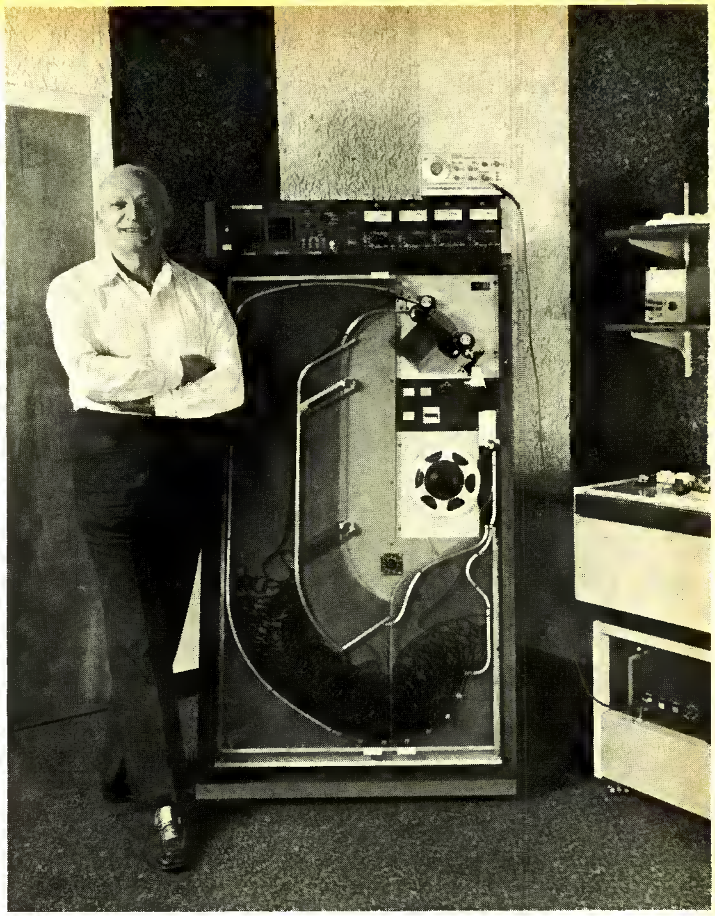


Figure 2. The Datalog automation controls and four of the Travis faders.

With An Otari Duplicating System, You Can Duplicate The Success Of These Duplicators.

When it comes time to make the decision to expand your business capabilities in high-speed duplicating, it's also time to seriously consider the OTARI DP-7000 High-Speed system. It's an endless loop, 64:1/32:1 system that's specifically engineered for the tough environments and continuous-duty operation that's demanded in the professional facility.

Here's two testimonials from DP-7000 users that tell this story best:

Frank Gspann, Sound Arts Co., New Jersey:

"We're a recording studio and duplicating company. In the studio we've worked with a pair of Otari 5050's for over four years and they have literally been Workhorses. So, we decided to invest in the DP-7000 High-Speed system when we wanted to expand our cassette duplication operation to a high-speed, high-output line. We duplicate a variety of materials from spoken word to computer program cassettes, which are much more demanding than music cassettes. The quality has been so consistently high that we violate all the rules and use the DP-7000 for some of our short runs because we know the product will be better. In this competitive business, you've got to sell a better product or you won't make it. The Otari DP-7000 let's us do that... and profitably."

Leonard Gross, Custom Duplication Inc., Los Angeles:

"Custom has been in the high-speed duplication business for a number of years and we had to add another line. In this business reliability and performance are number one,... but, in today's economy, price is a close second. When we checked around, we discovered that our money could buy two OTARI systems for close to the cost of a single system from the nearest competitor. The Otari is already putting-out almost half our total volume of four to five million cassettes a year...and that includes everything from spoken word Bibles to game and program cassettes for one of the largest game/computer manufacturers."

Let Otari's over 15 years of experience as a world leader in tape duplication systems go to work for you as it has for Frank and Leonard. Find out how an Otari DP-7000 can be a profitable, economical addition to your company. Call Michael Pappas at OTARI (415) 592-8311. He'll tell you how you can duplicate the success of the duplicators.

Please send me more information on the Otari DP-7000.
Name
Address
City
State
Zip
Phone
Mail to: Otari
2 Davis Dr.
Belmont, CA
94002

OTARI®
The New Workhorse

Otari Corporation
Industrial Products Division
2 Davis Drive Belmont, CA 94002
(415) 592-8311 Telex: 910-376-4890

Circle 23 on Reader Service Card

local studio owner Larry Travis, who was looking for an "honest consolemaker" to show his digital fader prototype. It included many of the features we were looking for in a very clever package. The timing was uncanny. The Travis Fader had found a home.

At the same time, McLaughlin was becoming disenchanted with VCA technology. One thing led to another, and the Travis Fader and Digital Attenuator were married to Datalog Automation, producing the first commercially-available system that places the analog signal wholly under direct digital control.

BYE BYE VCA, HELLO TRAVIS FADER

Goodbye VCA! These devices have been handy enough over the last few years but the active circuitry introduces noise, distortion and a certain coloration to the sound quality. They have always been viewed by Sphere as a "weak link" in the electronic environment. Furthermore, manipulating VCAs takes up a lot of computer space and power. McLaughlin felt there must be a better way. The success of the digital fader and attenuator has proved to those who have seen and heard it that indeed there is.

The Travis Fader is a digital encoding device that makes use of infra-red light bridge technology. The output of the fader is an 8-bit level word that routes to the attenuator and to the Datalog computer, if fitted.

A vertical row of LEDs alongside each fader trough tracks level and is analogous to "knob position."

There is a 6500-series microcomputer on board that handles a basic block of four faders. This division of duties between the fader's computer and the Datalog computer contributes to system simplicity, overall design elegance, and makes the Datalog cost-efficient. Another major reason for dividing up computer duties is reliability. If the main Datalog computer goes down, the faders continue to operate: you may lose automation, but you can continue to mix.

The fader computer recognizes only finger *movement*, and not where it is positioned in the infra-red field. You don't have to grab even an imaginary knob; anywhere in the field will do. After a few moments with the Travis Fader you probably won't miss the old knob at all. Each fader also includes two preset level memories (A & B) and a fader Mute or Solo depending on which logic is selected on the master panel.

The attenuator is a resistive ladder, CMOS switching device that does *not* convert analog to digital. In fact, the analog circuit is passive so far as the attenuator is concerned. Output is buffered by a high slew rate IC.

The hardest problem to confront McLaughlin was, of course, getting rid of noise created when the device is changing levels. While not wishing to seem coy, we do want to keep ahead of our competition and so will refrain from discussing the solution here. Suffice to say it was a difficult, time consuming problem.

The attenuator switches level as fine as .375 dB and there are 222 discrete steps between +15 and −68.2 dB. The attenuator "OFF" is the last step and is better than −100 dB.

Datalog fader automation is a multi-level, building block approach, partly in response to Sigma's wish to maintain multiple-room compatibility, and partly because it makes a lot of sense. Datalog Level I is the basic communications computer that ties all the fader blocks together and allows unlimited, freely-assigned fader grouping. Also part of the basic package is a series 80 master console level preset, similar to the individual fader preset A & B. This allows setting up a mix as a series of presets that the engineer may sequence as the multitrack is played. There are also two mute groups that are freely assigned.

Keep in mind that Datalog automation can retrofit *any* console, Sphere or not, new or old. If the retrofit buyer already has a 65K programmer, Level I will interface. In fact, as an option, Sphere will provide a program that prints level data on a multi-track channel and is similar to the 65K format—still the least expensive way to achieve real-time fader automation.

Datalog Level II builds on hardware and software already in place and is an SMPTE time code, dual-floppy disk "constant update" system that includes "butt splice" mix editing among many other features.

Figure 3. The fader computer recognizes only finger movement, and not where it is positioned in the infra-red field.

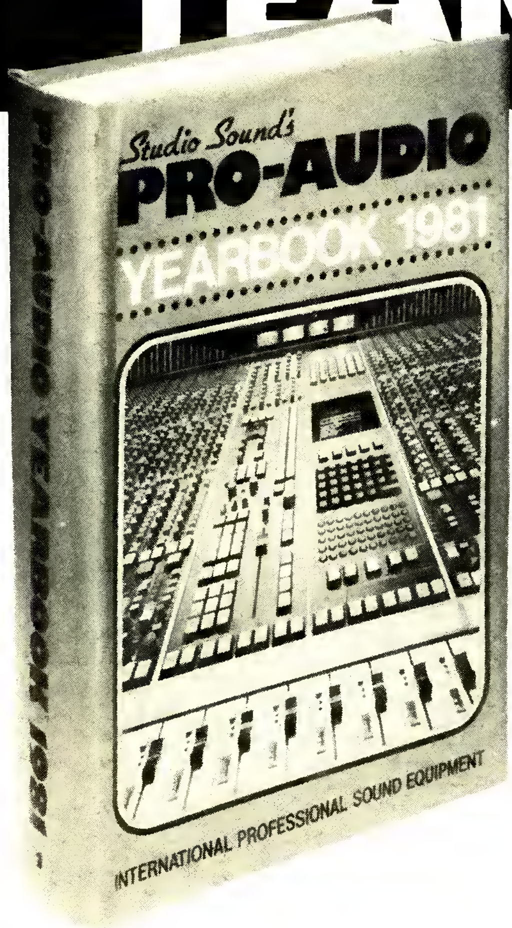
At this writing, Level III is a conceptual framework for a custom-designed computer system that expands on Level II to include intermix editing (plucking the drum kit from one mix, rhythm from another and so on). At Level III, the computer also keeps logistic and fiscal records, such as musicians' names, phone numbers, W-2 info, etc., and can provide instant invoicing at the end of a session.

A very interesting aspect of this approach, to Sigma, is the ability to be able to service clients with 65K type automation, whether it is coming in from a Sigma room in New York or from anywhere else. Again, versatility and client compatibility will mean more business for Sigma's rooms.

If, at times, the effort to design a complicated, computer-oriented new product seems somewhat disjointed, then strike a blow for peace and tranquility. But then again, it may be that there is a better end result when people of good will argue and compromise their way to consensus. We at Sphere wish to thank and congratulate Joe Tarsia and the rest of the Sigma family for their absolute professionalism and sometimes for just understanding. We look forward to the next challenge. ■

INTRODUCING THE PRO-AUDIO YEARBOOK

Special Offer to db readers



As a busy professional engaged in the fast moving world of sound engineering, you are obviously aware of the importance of reliable, up-to-the minute information relating to this growing industry.

Just think then how useful and informative you would find a book containing 7,000 separate entries with over 600 photographs, in excess of 2,500 complete addresses, almost 1,000 pro-audio dealers and distributors, PLUS manufacturers, prices and specifications in over 70 countries worldwide.

YOU NEED STUDIO SOUND'S PRO-AUDIO YEARBOOK!

With this essential book you will be able to find out at a glance who makes what, what it costs, where and how to obtain any product or service relative to the recording and sound broadcasting industries throughout the world.

Publication of the new edition is not now due until the summer of 1982, so we are making our remaining stocks of 1981 editions available at a special price of $35 including delivery.

A LINK HOUSE PUBLICATION

Please fill in the coupon and return with remittance to: Sagamore Publishing Co, 1120 Old Country Road, Plainview, New York 11803, USA.
(In NY State, add applicable sales tax)

PRO-AUDIO YEARBOOK

VISA MasterCard

☐ I enclose cheque for --

☐ Please charge my Mastercard/Visa credit card number:

☐☐☐☐ ☐☐☐☐ ☐☐☐☐ ☐☐☐☐

Name

Company

Address

Town/City

Code/State

Country

Phone

Signed

BRUCE LARSON

Flux-Frequency Measurements: A Means for Verifying Alignment Tape Accuracy

With the demand for precision in recording systems comes the need for calibration methods and equipment that meet or exceed those systems in accuracy.

THE WORLD OF AUDIO magnetic recording has become an extremely precise science in recent years. The result is a demand for the utmost precision from manufacturers of the intricate components within the recording systems, and from the composite systems themselves. To maintain a high degree of precision within a recording system, calibration methods and equipment must be at least as accurate as—ideally, should exceed the accuracies of—the system itself.

This article will focus on the accurate verification measurements of an alignment tape used for calibrating a tape recording system. What we'll really be verifying is the conformance of an alignment tape to a defined standard for pre- and post-equalization of frequency-versus-level. The defined standard will be the NAB 7.5 ips broadcast cartridge standard, employing upper and lower time constants of 50 microseconds and infinity, respectively. The new NAB cartridge standard specifies a constant fluxivity (record current) recording characteristic below 400 Hz, and a new NAB reel-to-reel standard now being drawn up will be the same. However, all reel-to-reel machines in use, and most cartridge machines, were designed around the old recording equalization, which specified a *rising* low frequency fluxivity; typically +3 dB at 50 Hz. The 50 microsecond time constant relates to a 3 dB roll-off frequency of 3180 Hz, while the infinity time constant translates to a frequency of zero Hz, indicating a fixed low frequency fluxivity for the alignment tape.

SHORT CIRCUIT FLUX LEVEL

The recorded short-circuit flux level is defined as the total flux level of the recording, commonly measured in nanowebers-per-meter (nWb/m). The NAB standard for determining flux level-versus-frequency response requires us to: 1) optimize the record and playback signal-to-noise ratios; 2) adjust the bias level to achieve a satisfactory setting between minimum third-harmonic distortion and minimum high-frequency self-magnetization (or, adjust bias for optimum high-frequency response); 3) consider the system(s) in which the particular standard will be applied. FIGURE 1 plots the curve of short-circuit flux level-versus-frequency for the NAB standard 50/∞ microsecond time constant.

It should be clarified here that the short-circuit flux level does not necessarily relate to any voltage or current signal level applied to the recording transducer. It is strictly the arbitrary flux level dictated by the defined standard as capable of being retrieved from the tape by the reproduce head. The common practice for recording a precision alignment tape is to employ a calibrated reproducing transducer and an equalized reproducing amplifier, calibrated to the prescribed frequency response standard. Then, using a nominal full-track recording transducer, its optimum bias level is set for the particular type of tape used. The reference fluxivity level at the reference frequency

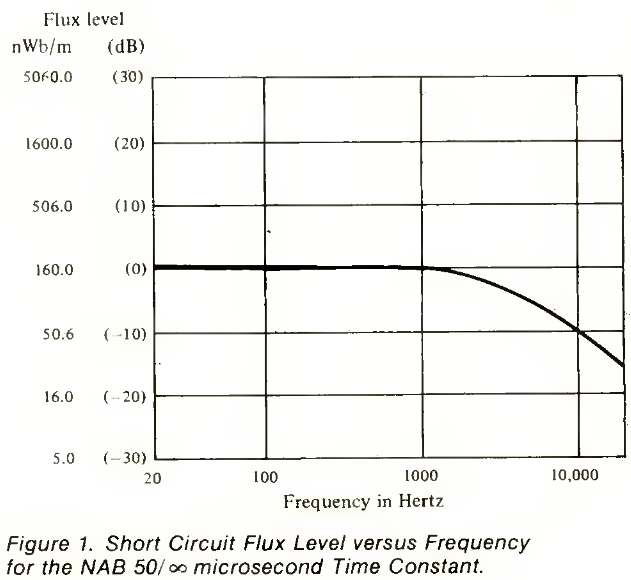


Figure 1. Short Circuit Flux Level versus Frequency for the NAB 50/∞ microsecond Time Constant.

Bruce Larson is chief engineer, Recorder Care Division, Nortronics, Inc.