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Recording engineer producer

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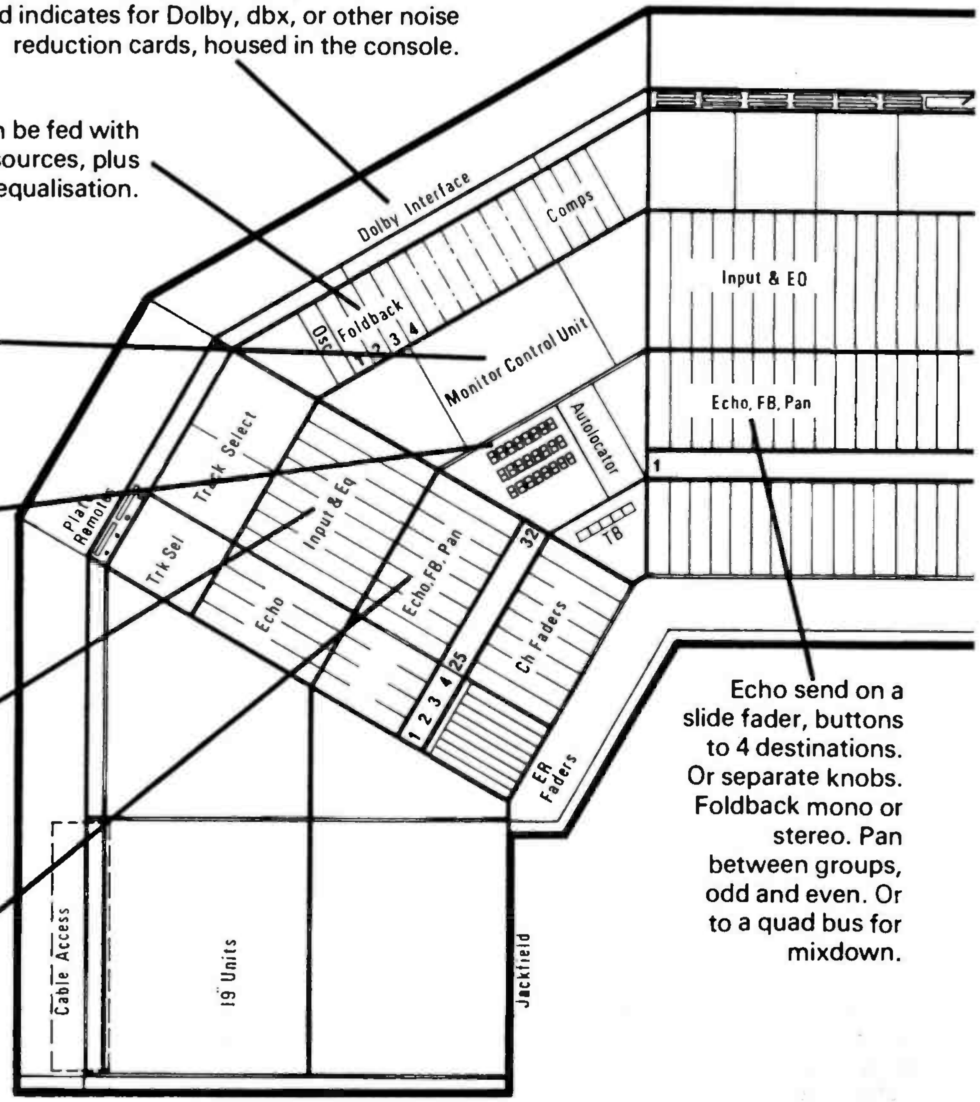

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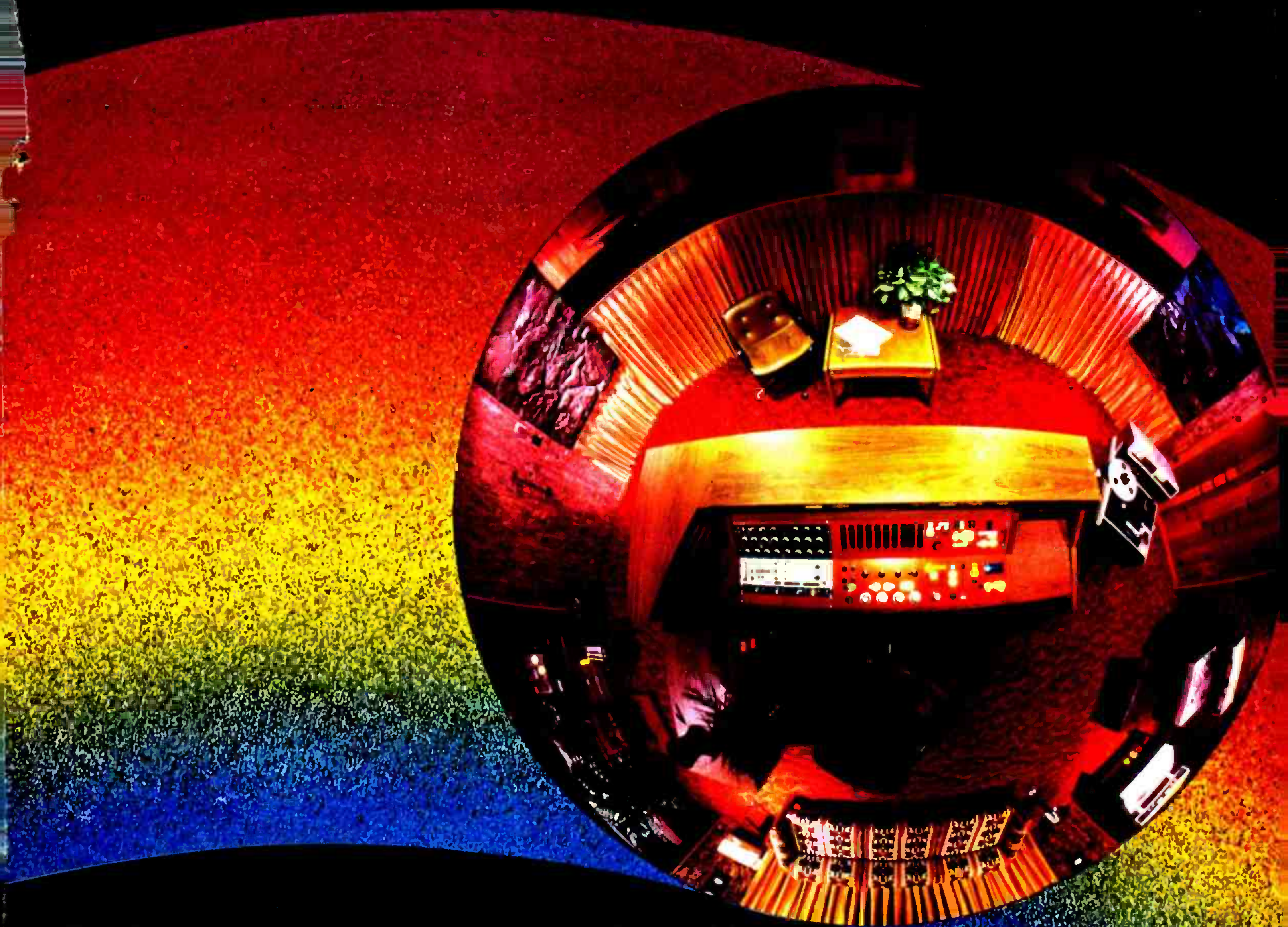

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VOLUME 8 — NUMBER 2

—CONTENTS—

THE GRAMMYS by Howard Cummings

RECOGNITION 1976
The BEST ENGINEERED RECORDING
— the GRAMMY winner,
— the nominees . . . page 23

— other classifications of GRAMMY winners, listing engineers, producers and studios . . . page 40

an interview and album analysis:
engineer of the year AL SCHMITT, *"BREEZIN'"* . . . page 25

HOW GRAMMYS ARE WON! . . . the selection . . . nomination . . . voting process . . . page 42

CONTROL ROOM DESIGN for the SMALL STUDIO
by Abadon/Sun's Woody Smith & Galen Carol . . . page 51

RECORDING THE ELECTRIC BASS by Paul Laurence . . . page 60
Dee Robb . . . continued on page 62
Leonard Kovner . . . continued on page 66
Ron Malo . . . continued on page 70

maintenance . . . A BETTER WAY TO ADJUST AZIMUTH
by Neil Hopper, of Sound City, Van Nuys, CA . . . page 77

POWER . . . How Much Is Enough by Chris Foreman . . . page 81

page — 8 *Letters & Late News*
page — 8 *AUTOMATION "DITHER"*
letters from Dave Harrison (Harrison Systems, Nashville), Paul Buff (Allison Research, Nashville), reply from Tom Hay (MCI, Ft. Lauderdale).
page — 98 *New Products*
page — 106 *Classified*

THIS ISSUE OF R-e/p IS SPONSORED BY THE FOLLOWING LIST OF ADVERTISERS

A & R Record 113
Abadon/Sun 85
Allison Research 20-21
Altec 11
Amber Electronics 36
Ampex 27
Appalachia Sound 106
Audio Concepts 22
Audio Consultants 93
Audio Distributors 74
Audio Industries 38
Audiotechniques . . 110/113
Auditronics 43
Auratone 28
BGW Systems 82-83
Bruer, Rudy 75
Capitol Magnetic 29
Cetec Audio 105
College Rec. Arts 109
Commco 62
Comm. Light & Sound . . 101
Crown Int'l. 95
dbx 89
Duncan Electronics 45
Eastlake Audio 4-5
El Tech 103
Everything Audio . . Cvr-2-3
Express Sound 12
Gregg Audio 104/109
Hammond Industries 35
Harrison Systems . 114/Cvr-3
Holzer Audio 112
Inovonics 74
Interface Electronics 57
Lexicon 19/112
MCI 78-79
Magnetic Recovery 48
Magnetic Ref. Labs 52
Marshall Electronics 73
MicMix Audio 39/111
Mincom - 3M 76
Neve 67
Orban/Parasound 18/37
Otari 69
Panasonic/Technics 87
Peavey Electronics 91
Quad-8 Electronics 71
Quantum Audio 16
Recording Supply 110
Saki Magnetics 96
S.A.E. 80
Scully/Dictaphone 33
Shure 41/Cvr-4
Sierra Audio 4-5
Soundcraft 99
Soundcraftsman 102
Sound Workshop . . 49/56/98
Spectra Sonics 7/100
Sphere Electronics 15
Stanton Magnetics 8
Studer of America 97
Sugarloaf View, Inc. 24
TAPCO 34
TEAC 53
Telex 31
Trident Audio 47
Uni-Sync 55
Urei 17/113
Westlake . . . 58-59/61/63/65
White Instrument 44
Windt Audio 112
Yamaha Int'l. 50

Letters and Late News

from: WINN L. ROSCH
CLEVELAND, OHIO

Paul Laurence's article on Panning in the December 1976 issue, seemed a good effort on bringing a difficult-to-handle subject under control, but I believe he left out psychoacoustical principles that are both important and necessary to understand, especially regarding the *size* of a panned image.

In stereophonic recording or transmission system, relative levels between the two channels do influence the placement of a sound source; to a more minor degree, stereo localization is determined by the relative phases of the two similar signals. When considering the "size" of the image produced by a stereophonic system, this phase difference is of primary importance.

Mr. Laurence only hinted at this fact when he indicated that "spillover" and reverberation can increase the size of an image. It should be noted that in each of the above cases, amounts of information of varying phasing is distributed in the two stereo channels.

Perhaps the best method of creating a genuinely "big" stereo image is to use stereo mikes or more than one channel for the various mikes used for a given source when making a multitrack recording. For instance, to have a string ensemble that is more than a mere point of sound, we can dredge up the age old practice of two channel stereo (remember when there only were two channels?) and have a beautiful image that stretches from speaker-to-speaker.

Another way a big, fat image can be produced is with a stereo synthesizer like the Orban/Parasound. Just run one channel of noise into it, and a phase and spectrum shifted pair of images fall out filling a full stereo panorama. And it all combines down to a mono sound that is identical to the initial mono input – perfect for AM radio mono mixes.

The addition of a phase-shift pot to augment a simple pan pot on a console would give the engineer full command over not only the size but the placement of the images in stereo mixdown. I am not aware of any equipment on which this facility is currently available.

reply from: PAUL LAURENCE
Hollywood, California

Okay. Believe it or not, I've been waiting for a response like yours for a long time. To the best of my knowledge, I have sort of been holding down the fort *in this area (trying to relate the musical idea to the techniques and technology of audio and all the way through the aesthetics), which can give one a certain insecure feeling. And everyone I've heard from has said, "It's great somebody is finally addressing themselves to this long-overlooked aspect", and, "far out", etc. Which, of course, gives me no indication of how I'm really doing.*

A statement of where I'm coming from *is perhaps in order here. The germ kernel I guess of my veiwpoint is the asethetics –* the what does it sound like? – *and from there I work back into the hardware. Where it seems most of the people who contribute to professional audio publications seem to be rooted in acoustics and electronics, my background is as a musician and producer, aided and abetted, like you, with a Psychology degree. I very much want to know how what I've done interfaces with these two areas – what might be termed* empirical *(as oppsed to* functional*) acoustics and electronics – and will not be happy until I've got it totally squared away.*

As long as you seem to have the inclination, why don't we take the opportunity and go whole-hog on this? Just off the top of my head, there are three things you can do: First of all, I would like a point-by-point analysis of my last article (actually three small articles), adding as much as you can. You could also no doubt provide a good working bibliography on the subject of psychoacoustics or other writings in this general field. And by all means, write your article. As you've noted, it is a difficult-to-handle subject. Mind-fryer *is the expression that always occurs to me.*

Yes, let's get on with this. And how about you other R-e/p *readers out there? Any ideas or suggestions? How do you conceive of it? Any literature that you know of? Let's hear from you!*

AUTOMATION "DITHER"

from: DAVE HARRISON
President
HARRISON SYSTEMS
NASHVILLE, TENNESSEE

It is not customary for Harrison Systems, Inc. to respond to editorial material supplied to periodicals by our competitors. Articles of that nature usually confine their inaccuacies to an over enthusiasm and over optimism as to the performance of a new or proposed piece of equipment.

The article supplied by MCI, Inc. and published in the February 1977 issue of R-e/p requires that we make an exception to our usual policy, in that it contains serious errors in both its appraisal of existing automation systems and the performance of those existing systems.

As approximately 50% of all working automated consoles in existence today are Harrison Consoles mated to Allison

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AUTOMATION "DITHER"
continued . . .

65K programmers, we must assume that most of the erroneous allegations were directed at us.

Rather than respond to all of the inaccuracies presented, I will confine myself to those which would, if left unanswered, lead to serious misunderstanding and confusion about the Harrison/Allison automation combination.

To quote the MCI article: (pages 63 and 65), "In the Analog Updating method, two full scans are required to complete and Update . . . Also, the accuracy of the system is severly limited since the digital data is being processed by analog circuitry."

Harrison response: The Harrison/Allison combination uses an analog update method. Contrary to the above verbage, both the Digital to Analog, and the Analog to Digital conversion for a specific function is accomplished on the same I/O scan and is not subject to the scan delay suggested by MCI.

As to the accuracy of the analog system, the useful range (upper 48dB) of the faders in the Harrison Console are encoded in 1/2 dB steps. It is my understanding (from their representatives) that MCI's resolution is also 1/2dB steps. I fail to see a loss of accuracy here.

MCI: (page 63), "If sequential scanning can be made at a rate close to the limits of human perception, a tape dropout will not be noticed, since the loss of a single data word can be ignored until the next scan. A drop-out would cause an error of less than one-tenth of a second in a progressive level adjustment."

Harrison response: MCI has indirectly suggested that the total scan time of their programmer is on the order of 80ms, and that a dropout of one scan period and therefore an additional delay of 80ms would not be noticed as it is "close to the limits of human perception."

The emperical work done at Harrision Systems, Allison Research, and other organizations indicate that the limits of human perception as it relates to control delays in audio are on the order of 30ms. In essence, when the level of a function is changed more than 30ms after it was originally programmed to change, there is a high probability that the delay will become noticeable and bothersome to the mixer, producer, and musicians.

MCI: (page 63), "The Recording Method: Two methods are in current usage: (a) Signal digitalization and *data packing*, or (b) Sequential scanning. Obviously, in a system containing a very large number of automated functions, data packing would be necessary. However, data packing has several inherent drawbacks, not the least of which is the loss of a great deal of reliability and usually has a limiting factor in the number of 'updates' it can recognize within a given time frame. A tape dropout can cause a major error during a mix, since a significant piece of data not repeated would be lost."

Harrison response: The data formatting system used in the Allison 65K programmer does not fit neatly into one of the two possible methods mentioned by MCI. The Allison programmer in fact uses both methods simultaneously. Sequential information is interleaved on a 1 to 3 basis with priority information to get all of the benefits of both systems while nullifying virtually all of the disadvantages.

The two major arguments voiced by MCI against priority (data packing) systems are:

1. Error due to dropouts
2. Limited update response

In the case of either dropouts or many updates occuring at precisely the same instant, the Allison 65K simply reverts to a sequential scan mode of operation at the worst case data rate.

MCI elsewhere states (page 63, column 3) that their programmer always works "at the worst case data rate." I would agree.

I appreciate the opportunity afforded by R-e/p to make this limited response. If further details or response is desired by anyone, they are invited to call me directly at (615) 834-1184, or write me at Harrison Systems.

---

from: PAUL C. BUFF
President
ALLISON RESEARCH INC.
NASHVILLE, TENNESSEE

After reading the article describing the MCI Automated Mixdown System, in the February 1977 issue, I feel I must comment, and in certain respects, take issue with the philosophy of the proposed MCI system.

The first point is one of definition, and regards the advertising of "Automation Ready" consoles. By my definition, the term indicates a console which contains all the internal circuitry required for automation, not simply VCAs and un-connected switches. An example of this definition can be found in the Harrison console, which may be automated by the simple expedient of plugging a programmer into the console connector provided for this purpose. The console itself contains, in addition to the VCAs, all automation signal processing circuitry required, and said consoles are fully tested, in automated form, before leaving the factory.

The current MCI advertisement, which appears in the same issue as the article in question, states "MCI JH500 series consoles require *no modification* to accept the new automation", yet the photographs included in the article indicate, unless my eyes deceive me, a great deal of modification.

CAPACITY REQUIRED

MCI states that their market survey indicates "24 to 40 automated functions are needed to implement modern mixdown methods." Obviously the market they surveyed was a different one than my company serves. The 256 function capacity offered by our first generation programmers proved insufficient to our customer demands, within the first year after its introduction in 1972. Our second generation programmers are good for 8,192 analog funtions, or 65,536 digital bit functions.

Systems utilizing the equivalent of several hundred analog functions can be, and presently are being configured to be cost competitive with, and operationally less complex than their non-automated counterparts.

This is particularly true since the advent of programmable equalizers and other complex functions, and promises to be even more true as users demand systems where many mixes may be stored on one data track, and programmable editing between these mixes made possible.

DATA PACKING—
PRIORITY ENCODING

MCI goes on to claim a number of defects in what they term "data packing" programmers, into which catagory I assume they place our 65K series priority encoding programmers.

First they claim a loss of reliability. Wrong! A well designed priority encoding system is infinitely *more* reliable than a sequential scanner, by nature of the fact that each word, or function, in the code carries with it digitally encoded information describing which console channel, or sub-system belongs to the information. Thus, each word may be instantly and individually verified, as the validity of both its data and console address, before it is released as valid data. Sequential scanning systems run the risk of the decoder losing track of which information belongs to which channel, since a dropout can (and does) destroy the decoder's ability to *keep in step* with the encoded sequence. This possibility is usually anticipated by deleting one or more scans of data while the decoder finds its place, in the event of the most minute dropout.

The manifestations of this type of error detection can be disasterous when the tape is punctuated by a multiplicity of sub-millisecond duration dropouts, of the sort which frequently show up on well worn master tapes. Under such conditions, the sequential scanning system can be rendered completely inoperative for long periods of time, while the word addressed system simply deletes tiny fragments of data corresponding to the actual lengths of the dropouts.

Contrary to MCI's allegations, fragmentary data deletions in a well designed

AUTOMATION "DITHER"

continued . . .

priority encoding system are not lost, and do not cause audible errors to the mix. This would be the case only if the designer of such a system encoded *changes only*, and failed to provide redundant circulation of static parameters.

Our 65K equipment, in addition to providing fast priority access to changing parameters, also provides redundant recirculation of non-changing parameters, at a rate not far removed from the sequential data rate described by MCI. Thus if an important change should be deleted by a tape defect, it will be around again a few milliseconds later.

The likelihood of a change being deleted, however, is several orders of magnitude less than in a sequential scanning system.

DATA OVERLOAD

I quote the MCI article: "Automation systems which use data packing (and usually have insufficient storage) often overload when it is necessary to make a large number of changes at the same instant."

In rebuttal, I challenge the entire MCI engineering staff to leave their drawing boards and go out into the field to a console equipped with an ALLISON 65K programmer, Then I suggest that they all huddle around the console, take one control in each hand, and, on command from their leader, change it. When the tape is replayed, they will, lo and behold, find that the system did not overload but, in fact, responded probably a little bit quicker than their own 64 function system!

Why? Because a well designed priority encoding system does not, contrary to MCI, have a defined point of *overload*. When subjected to a continual stream of changing parameters, it effectively takes on the characteristics of a sequential scanning system, while maintaining at least an order of magnitude better dropout immunity.

THE VCA CUSHION

In all multiplexed automation systems, an analog movement of a control (say a fader) comes back from the decoder as a stepwise approximation of the original motion. The resolution of these steps is directly related to the speed of the control movement vs. the number of times per second the control is sampled. Based on this, allow me to set up a hypothethical situation, using both the proposed MCI programmer and the Allison 65K programmer.

According to the article, the MCI programmer would, in 64 function format, sequentially sample the faders at the rate of 1.2 msec. per fader, or one sample from a given fader each 76.9 msec. (13 samples per second).

The 65K programmer, on the other hand, will service a moving fader approximately 235 times every second, even though the system may be processing hundreds, or even thousands of funtions.

Now, let us assume that the operator moves the fader from top to bottom in a time span of one second. The MCI system (based on my analysis of the information given) will remember this change as 13 steps of around 8 dB each.

The 65K programmer will remember the movement as 235 steps of under ½ dB each. In both cases, the steps must be integrated, or smoothed out, so as not to be audible in the mix.

In the case of 13 steps of 8 dB per second, the integration time required is on the order of ¼ to ½ second, if the stepping effect is to be rendered inaudible on pure tone instruments. Unless a fairly powerful computer is employed to introduce this integration only when mixing maneuvers demand it, it must be built into the system full time. The net result is that the fader must operate sluggishly and be incapable of rapid movements, or else it must introduce an audible 13 Hz staircase modulation, under certain mixing conditions.

In the 65K example, however, the introduction of a mere 20 to 30 msec. of integration will completely remove any trace of tone modulation, without imposing any restraint on the rapidity of fader movements.

I must point out that the above situation is not simply a laboratory hypothesis, but is a field proven reality, which may be verified by a short conversation with any engineer who has had field experience with sequential scanning automation systems.

My own recording studio experience indicates that serious trade-offs begin to emerge when the sampling rate for a moving fader falls below 30 samples per second. This realization, indeed, was one of the prime considerations which affected our decision to abandon our first generation sequential scanning programmers in favor of a more sophisticated approach.

ACCUMULATED DELAYS

MCI claims that their system produces accumulated delays, which they term *bounce delay*, of about 1.2 msec. per encoding pass. Since I am not versed in the internal workings of their programmer, I will not directly dispute this claim. I am, however, somewhat puzzled as to just how such a feat is possible, in a real world system, when one considers the physics involved.

Allow me to attempt to assemble a system, based on the information given.

We are given a sequentially scanned frame consisting of 64 words of approximately 12 bits each, together with some unique data pattern which is used to locate the beginning of each scan. The stated word length is approximately 1.2 msec., which indicates a scan time of around 77 msec., plus the start code identification time required. The decoding system is required to validate each word, as received, without the aid of a buffer memory or frame validation system. It is also required, in the event of a dropout, to abort any further decoding action until the start of a new scan, where it can reestablish its counting sequence.

So far, there's no big problem. What we've described is essentially the same technology as the Allison first generation system, although the MCI scan time is some 2.4 times slower.

Now comes the puzzle. If we are to achieve the stated 1.2 msec. "bounce delay", the encoder and decoder must be at all times precisely synchronized, and offset by exactly one word, or 1.2 msec. This, of necessity, indicates that, during an Update pass, the encoder must be controlled by, or locked to, the decoder. This is entirely feasible if we assume the decoder is an absolute source of stable clocking, and exhibits zero speed variation.

In the real world, however, the decoder is not a stable source of clocking, since it can be punctuated by long periods of inactivity due to tape errors, and is modulated by the flutter, wow and speed variations of the storage medium.

These factors indicate to me that hard synchronization is not practical if realistic speed variation specifications are to be allowed. Instead, a flywheel controlled loose form of sync is required in order to bridge the gaps and frequency differentials involved.

In order to insure a "Decode before Encode" relationship in a loosely locked system, the offset between decoder and encoder must be increased beyond one words duration. The amount of increased offset is dependent on the amount of speed variation to be tolerated, as well as the permitted inactivity time of the decoder. Increasing the offset, of course, indicates an increased updating delay.

To further complicate matters, let us assume that the data track to be bounced is not a continuous flow, but is an assemblage of "punched-in" data. (Data punches are a field reality, and are used to avoid "going to the top of the song" when making spot corrections to the mix.)

Now where are we? The encoder is merrily following the decoder along, and the decoder decides to alter its sequential relationship (due to a data punch). The encoder has to either accelerate or slowdown (within the limits of its flywheel) in order to re-establish sync, or it must leave a hole in the code and wait for a new start code to be received. In either case, the resultant error would appear to become a permanent part of the code and, I would think, would be compound-

AUTOMATION "DITHER"

continued . . .

ed by further update passes.

Since I assume that MCI would not attempt to market a system which was incapable of coping with data punches or splices, I will continue under the assumption that they have found some ingenious method of handling this dilemma. I would even concede the possibility that, through another feat of engineering skill, that they have indeed accomplished a 1.2 msec. "bounce delay" within the framework of an acceptable speed variation tolerance and tape defect rate.

[For the record, the Allison 65K programmer has a speed variation tolerance of just under 2 octaves, a data access time of 3.2 msec. and an accumulated delay factor of 4.8 msec. (reduction to zero may be accomplished with external storage mechanism). It employs completely non-synchronized encode/decode cycles. The recovery from any form of data discontinuity, be it caused by dropouts, splices or data punches, is under 3.2 msec.

These relationships are constant for any number of functions employed, with the exception of the access time, which increases during times of multiple instantaneous changes, at the rate of 3.2 msec. per simultaneous word change. A word, in 65K terminology, is 16 bits of data, and is thus equivalent to two analog functions.]

MCI's information indicates, for a 64 function programmer, a constant access time of about 77 msec., together with a 1.2 msec. accumulated delay factor. They claim that 100 update passes will result in a delay of under 1/8 second (125 msec.).

By my calculations, the net delay for 100 passes would equal (100 x 1.2 msec. + 77 msec.) or 197 msec. (about 1/5 second).

Looking at a more realistic 6 pass mix, the author's analysis of the MCI system will define a delay of (5 x 1.2 msec. + 77 msec.), or 83 msec., with the additional probability of a ¼ to ½ second control voltage integration requirement.

The 65K programmer, under the same conditions, but with essentially unlimited function capacity, will undergo a delay of (5 x 4.8 msec. + 3.2 msec.) or 27.2 msec., with no significant requirement for control voltage integration.

Even if the 65K mix were undergoing the unlikely situation where 10 functions were simultaneously and constantly changing throughout the mix, the total delay would be (5 x 4.8 msec.) + (5 x 3.2 msec.) or a total of 40 msec.

The implications of a system which always runs at its worst case data rate (as is the case with the MCI proposition) are clearly evident.

OPERATIONAL PSYCHOLOGY

MCI has taken the editorial standpoint that the automation system should be made as *transparent* to the user as possible. They have gone to considerable lengths in this direction in what they term *Automatic Nulling*. While the concepts involved represent a commendable effort from a technological standpoint, I personally disagree with the psychology, from an operational viewpoint.

Firstly, my contention is that past automation systems are already too transparent. If man and machine are to communicate harmoniously, man must have the ability to interrogate the machine, and know what's going on in its little binary brain. Being the real creative force, man should be free to make all the decisions, good or bad, if he so chooses, just as he does in a non-automated situation. The machine, on the other hand, should do what its operator tells it to, and should, in no way block its operator from deciding what is right and what is wrong. It should assist man in arriving at the right decisions, but it should never, or certainly not by design, force man to lose out in a confrontation over the aesthetic value of a desired effect.

I will back up this rather ambiguous psychology with some hard mixing realities:

In "old fashioned" automation systems, UPDATING (adding or subtracting levels from a prior program) is generally accomplished with the assistance of a physical INDEX POINT on the fader. If the opeator wishes to transfer from READ MODE to UPDATE MODE *without an abrupt level change*, he first places the fader on this mark, selects UPDATE and then moves the fader to achieve the desired modification. He may then, if desired, return smoothly to READ MODE by re-positioning the fader to the INDEX POINT and selecting READ MODE.

In order to achieve certain effects in the mix, the operator may sometimes choose to purposely introduce an *abrupt*, but pre-determined level change by selecting UPDATE with the fader at some point other than the INDEX POINT. Although the system designer may not approve of this maneuver, the producer may love it, because it works!

MCI's *Automatic Nulling* system sets some pretty concrete rules governing this sort of activity. The system states: "If you want to UPDATE, you'll do it smoothly, or I won't let you do it at all". — "If you want to go from READ to WRITE, you'll do it my way, when I'm ready. I won't tell you where I am, you'll have to find me with your fader. When you cross my threshold, I'll put myself in WRITE, but it will have to be smooth because my designers don't like abrupt changes (like emphasizing drum licks and horn shots)".

CONCLUSION

Audio automation is an exciting field, and promises to define a new level of technical excellence within our industry. It is also an area of limited, but real practical experience, an essential ingredient in the success of any endeavor. When entering such a field, a new participant can ill afford the luxury of ignoring the realities of industry experience in favor of the promotion of laboratory hypotheses, particularly when those hypotheses have been proven inadequate in field service.

---

reply from: TOM HAY, Chief Engineer
M C I
FT. LAUDERDALE, FLA.

We would like to thank the editor of *R-e/p* for publishing our original article on Automation. It certainly seems to have brought a lot of *facts* out of the Automation woodwork.

MCI's original article was not directed toward any particular manufacturer. However, we feel that we must respond directly to a few of the particularly confusing and inaccurate statements in the letters from Mr. Harrison and Mr. Buff.

Mr. Buff attempts to refute our *Automation Ready* claim by pointing to the pictures of boards plugged onto the bottom of our console *(please refer to Figures 3 and 4 of the article, on page 64 of the February, 1977 issue of R-e/p).* These are Automation boards plugged directly on to *pre-existing connectors* which are built into every JH-500 console. We have chosen this approach for two reasons. First, that we may build our Automation within the Console frame – not in a remote rack at the end of a cable. Thus eliminating the requirement for additional rack space in the studio and the extra cable to interconnect the Processor to the Console. Second, so that a studio buying a non-automated console is not forced to pay for useless electronics and cables.

Concerning the accuracy of the *Analog/Digital* system; Mr. Harrison states: "... the useful range (upper 48 dB) of the faders in the Harrison Console are encoded in ½ dB steps." We have also been told by Mr. Roundtree of Allison Research that – while the A to D is 8 bits, only the most significant 7 bits are used due to digital "dither" of the least significant bit. Let's analyze how this compares to the MCI system.

If only 7 bits are actually recoverable, the true number of preserved steps the fader can be divided into is 2^7 or 128. This may be the reason Mr. Harrison only discussed the resolution of the top 48 dB. Assuming they have 100 dB of range total (as with MCI) the bottom 52 dB must be divided into 32 steps of about 1.5 dB each.

The MCI system divides the entire 100 dB range into .4 dB steps – 20% better in resolution than the best portion of the Harrison/Allison system. MCI is able to take advantage of full 8 bit Analog to

AUTOMATION "DITHER"

continued . . .

Digital resolution without "dither" problems because we use a voltage range of -1v to +12v, not 0 to 5v. By using a greater dynamic control voltage we effectively increase noise and interference rejection. By using the greater voltage swing we have made the .4 dB window a bigger voltage change — thus making the *window* an A to D must stabilize on, two and a half times bigger. In audio terms, our control voltage signal to noise is 8 dB better.

The digital hardware of the MCI system is also different from the Allison system. The basic *thinking device* of the MCI system is a microprocessor instead of the hard-wired logic processor used by the Allison system. We would like to take this opportunity to further explain the internal workings of our digital system.

The MCI system is composed of several functional blocks (see Figure 1). One of the most important attributes of the system is that each of these blocks performs its functions completely separate from the others. The internal workings of the system are completely asynchronous. No functional block has to wait for the correct clock cycle to start doing its job. No pre-synchronized clock is needed.

The time consuming process of acquiring and returning data to and from the tape machine is taken over by an Asynchronous Communications Interface

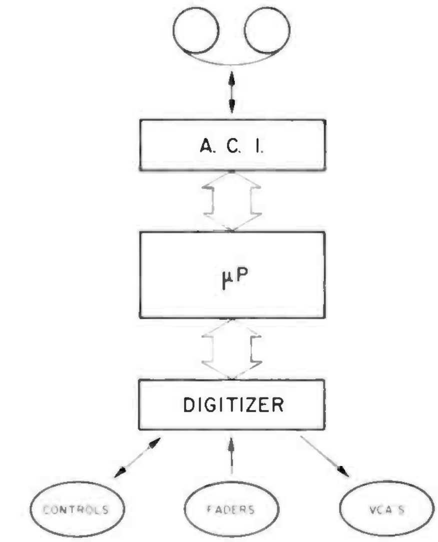


FIGURE 1

(ACI). This functional block presents the Asynchronous serial data from the tape machine to the microprocessor in parallel form with parity already checked. When data is being sent to the tape machine, the ACI performs all of these functions in reverse.

The Digitizer contains the A to D and D to A functions in addition to routing data to and from the console. The conversion functions in this block work fast enough that the D to A converter is always waiting to receive data on command and the A to D converter generates the new data for updates and rewrites at a rate which assures that the Processor will never have to wait for information.

The microprocessor is the Central Control Block of the system and is capable of making 400,000 decisions per second. The CPU receives data from the ACI, picks up the new fader values, computes a new updated value, sends a new VCA control *word* to the Digitizer block and a data word to the ACI to be recorded, in approximately 300 microseconds.

The recorded data and the update data are both handled asynchronously, allowing the modifications to the recorded data to be handled digitally by the microprocessor. For this reason the system does *not* have to send the previously recorded data all the way to the fader electronics to be updated by analog methods before being re-recorded. The asynchronous architecture of the MCI system results in a *bounce delay* of 1.2 msec. per pass, regardless of the number of faders being changed.

The design of our system and the use of *phase encoding* for data storage makes our system immune to tape machine speed variations. Data being sent to storage is *phase encoded* to create a self-clocking code. In Phase Encoding, bits are not represented by *high* or *low* levels, but by positive-going or negative-going transitions. In this system each relevant value

for additional information circle number 10

to be processed is represented by a *character* consisting of a number of digital bits. Each *character* is a separate entity and does not need to be rigidly related in time to the preceding and the following *charcters.*

The data access time of 3.2 msec. claimed for the Allison system is deceiving. This figure is the access time of a *single* moving fader. As more faders are moving the total mix, this figure derates by 3.2 msec. for *each additional moving fader.* This time becomes serious when there have been a number of update passes on a particular mix. Several faders are normally updated on one pass, then more faders are updated on subsequent passes. As the total mix progresses, a large number of faders may be changing at the same time. Thus, in the Allison system, the access times and the bounce delay times can become significant accumulatable errors.

The MCI system has an average scan time of 44 msec. This *first time only Encoding Error* is independent of the number of updates, and independent of the number of functions which change at any time in the mix. This scan time is *not* comparable to the Allison access time — which is variable, depending on the number of changing functions, but never less than 4.8 msec.

With the MCI system, the accumulated delays are always a quanta of 1.2 msec. Unlike the MCI system, the Allison system is a nebulously changing value depending on the complexity of the mix.

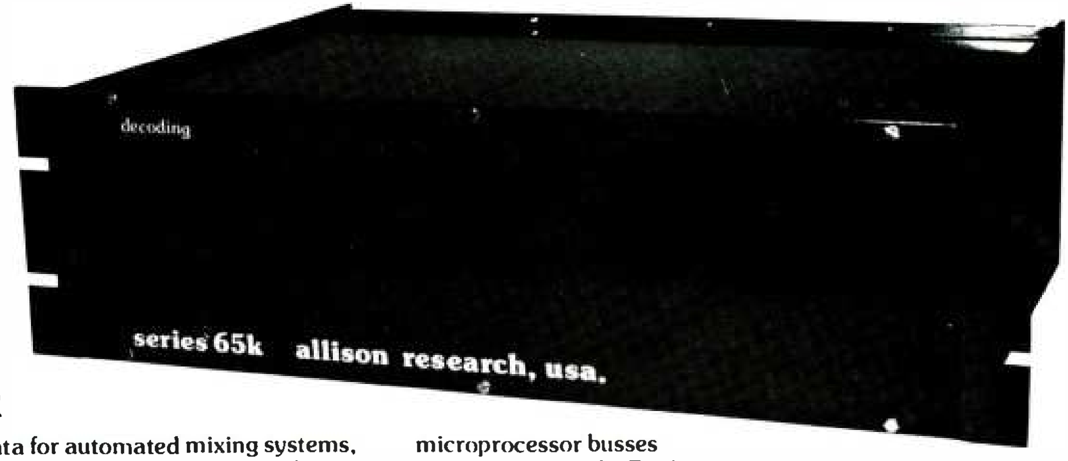
One of the problems encountered in the day-to-day use of Automated mixdown systems is the fear of incurring an esthetically damaging accumulation of delay times. This fear will cause the operator to limit the number of update passes so that the total delay (due to the number of passes) will not become noticable. When this fear is present, it damages the basic purpose of having automated mixdown systems in the studio. If the mixing engineer cannot feel free to use as many passes as he would like to polish off a mix, then half of the value of automation is lost.

We agree with Mr. Buff that having a system which works is not enough—the system must be responsive to the operators requirements. In our article of February, '77, we discussed only the unique operator features of our system. We agree that for some mix requirements the old manual methods permitting jumps in level are necessary. Close examination of Figure 6 in the February article shows that the Automatic Nulling features can be cancelled on command. For normal operation, however, automatic nulling capability makes automation a more instinctive type of man/machine interaction than the use of nulling lights. Automatic nulling also

continued on page 113 . . .

SUCH A DEAL!

If you presently own an Allison Research first generation programmer and want to upgrade to the 65K standard, we'll give you 30% off the tab. (Limited offer – not available through dealers)



THE 65K PROGRAMMER

When it comes to storing data for automated mixing systems, there is but one standard. The Allison Research 65K Programmer. In case you're wondering just what it takes, performance wise, to be top banana in the audio programming field, here's our secret:

Capacity: 65,536 bits, 8192 analog functions, or any desired combination of analog and digital functions. Enough capacity for tomorrow's needs as well as today's. No compromise, no excuses.

Speed: 3.2msec access time from console control to data track. At least 10 times faster than any other system we know of, existing or proposed. Accumulated delay of 4.8msec per update pass can be reduced to zero with external data storage. Exclusive Allison priority encoding technique holds access time and accumulated delays constant, regardless of the number of encoded functions.

Speed Variation: The 65K can tolerate nearly 2 octaves of speed variation in the storage medium. Turn the reels by hand – it'll work.

Storage Medium: You name it. Master tape, synchronized audio tape, digital tape, cartridge, cassette, disk – the 65K is adaptable to any storage medium.

Interface: Whatever suits you. 16 bit digital data buss, wire per function analog, multiplexed analog etc. The 65K easily adapts to any combination of these interfaces, as well as to microprocessor busses and data terminals. By the way, you pay no premium in processing speed with our analog interface system. We'll still get you on tape in 3.2msec.

Code Reliability: Individual word addressing plus 4 levels of data verification on each 16 bit word assures *absolute validity* of decoded information. After 14 months of gruesome testing, we were unable to produce a single data error by introducing noise, crosstalk, speed variations, punch-ins, splices, phase shifts and level variations. No glitches – no excuses.

System Reliability: The 65K is a *simple machine.* All super reliable C-MOS circuitry. Virtually no heat dissipation. Most I.C.s are socketed. All boards are plug-in replaceable. The 65K has proven reliability. Ask the people who own them.

Track Record: More 65K programmers are in use, worldwide, than all other working systems combined (including our own first generation 256 series)
Total customer complaints: Zero
Total field failures: One opamp
Customer feedback: Splendid

Cost: Ah yes, the bottom line.

1024 bit digital (65K-D1-1K)	*$3100
32 function analog (65K-A1-32) (Includes 512 digital bits)	$4286

THE GREAT EQUALIZER

Here's your chance to witness tomorrow's technology, TODAY!

Give us less than one square foot of console control surface, and we'll give you up to 64 channels of fully programmable equalization. As a matter of fact, we'll throw in 4 random access static presets of each equalizer, (expandable to 64).

Now, of course, when Allison designs an equalizer, you can rest assured it ain't no run of the mill device. Through its powerful 32 bit digital control, THE GREAT EQUALIZER boasts a total of over 2 BILLION possible E.Q. curves, per channel. Besides 24 overlapping E.Q. frequencies with peaking or shelving characteristics, you get 14 frequencies of 18 dB/octave high and low cutoff filters, and phase reverse switching.

Regardless of whether you set it by hand or the 65K Programmer sets it from memory, you may visually verify the *exact* E.Q. setting of each channel, instantly. You may also instinctively update or override any programmed E.Q. setting without any nulling or head scratching.

Through the aid of a peripheral display device, currently in the design stage, you may even opt to view a 1/3 octave real time plot of any equalizer in the system, *without interrupting the mix.* Although each plot accurately covers the full 20 – 20KHZ spectrum, it may be generated in under 500msec, thanks to a proprietory Allison frequency synthesis scheme.

Now if that's not enough to tickle your fancy, here's a few

*Prices subject to change – generous O.E.M. discounts available.

Honoring . . .
the ENGINEERS . . .
their STUDIOS . . .
the PRODUCERS . . . for the parts they played in making the recordings which won 1976's GRAMMYS . . .

RECOGNITION 1976

BEST ENGINEERED RECORDING *(Non-Classical)*

Engineers — **AL SCHMITT**, assisted by Don Henderson
Producer — Tommy Lipuma
Studio — Capitol Records, Hollywood
Disc-Mastering — Doug Sax, The Mastering Lab, Hollywood
— *BREEZIN' - George Benson*

BEST ENGINEERED RECORDING *(Classical)*

Engineers — **BUD GRAHAM**
RAY MOORE
MILT CHERIN
Producer — Andrew Kazdin
Studio — A — Piano Player Track - Cleveland
B — Columbia Jazz Orchestra, Columbia's 30th Street Studios, New York City
Disc-Mastering — Cass (Casper) F. Mondello, CBS, New York City
— *GERSHWIN: RHAPSODY IN BLUE — George Gershwin (1925 Piano Roll) and Michael Tilson Thomas conducting the Columbia Jazz Band.*

BEST ENGINEERED ALBUM NOMINATIONS

Engineer — **Jay Lewis**
Producer — Gary Wright
Studio — Sound Labs, Hollywood
Stronghold Studios, North Hollywood
Disc-Mastering — Mike Reese and Doug Sax, The Mastering Lab, Hollywood
— *DREAM WEAVER, Gary Wright*

Engineers — **Alan Parsons,** assisted by Tom Trefethen
Producer — Alan Parsons
Studio — Mama-Jo's, North Hollywood
EMI, London
Disc-Mastering — Allen Zentz, Allen Zentz Mastering, Hollywood
— *SOMEWHERE I'VE TRAVELED, Ambrosia*

Engineer — **Ron Hitchcock**
Producers — Lincoln Mayorga, Doug Sax
Studio — The Wylie Chapel, Hollywood
Lathe Operator — Mike Reese, The Mastering Lab, Hollywood
— *THE KING JAMES VERSION, Harry James Big Band*

Engineers — **Alan Parsons,** assisted by Pat Stapley and Chris Blair
Producer — Alan Parsons
Studio — EMI & Kingsway Hall, London
Disc-Mastering — Doug Sax and Mike Reese, The Mastering Lab, Hollywood
— *TALES OF MYSTERY AND IMAGINATION—THE ALAN PARSONS PROJECT Alan Parsons*

Compiled by Howard Cummings

* *Addendum; see page 109*

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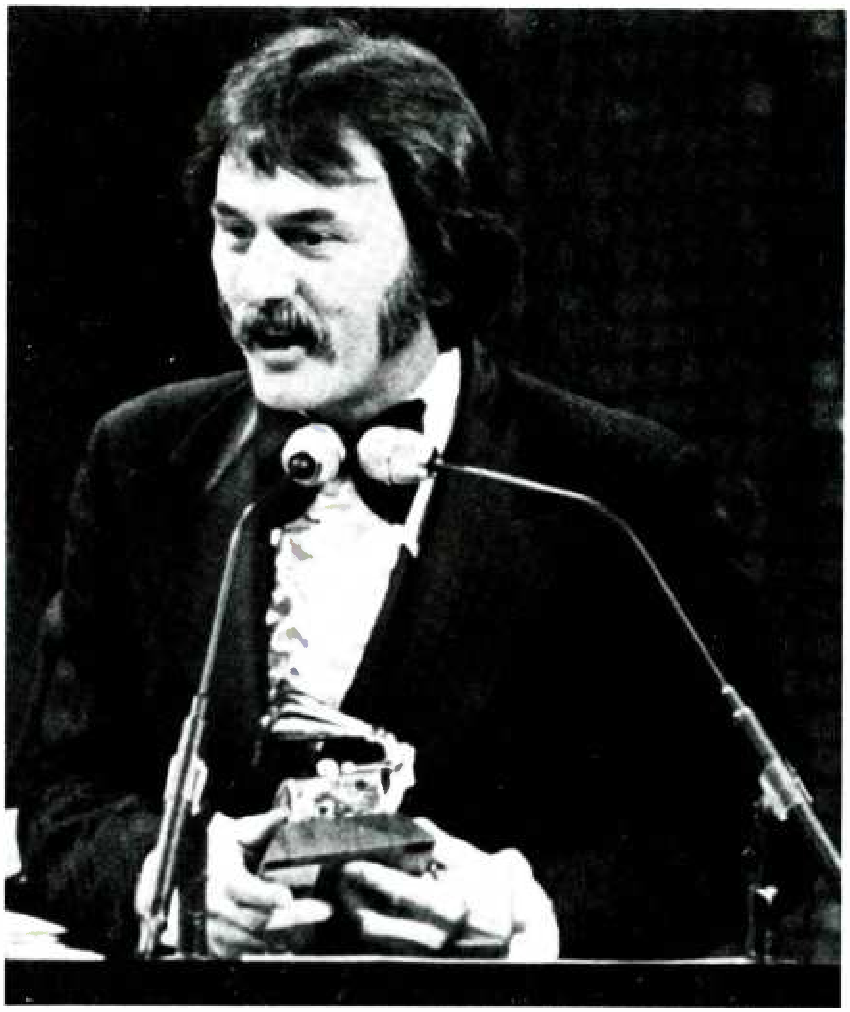


engineer/mixer of the year GRAMMY WINNER AL SCHMITT

an interview and an album analysis by Howard Cummings

BREEZIN' WITH AL SCHMITT . . .

1976 was the year for George Benson. After years of being on other record labels, George finally realized that "magic formula" with the team of engineer Al Schmitt — producer Tommy Lipuma, and shot to the top of the jazz and pop charts, even copping a hit single out of the association.

Al Schmitt's career began in 1950 when he was persuaded by his uncle to try the recording field instead of printing. As a result he started work with Apex Recording in New York City alongside chief engineer Tom Dowd. "It was love at first sight," says Al. "I liked it so much I'd stay from 9 a.m. until midnight every night."

Since much of the work of Atlantic, Prestige, National and Mercury Records was done there, he had a chance to work with many of the top R&B groups and songs of the 50's including the Clover's DON'T YOU KNOW I LOVE YOU SO.

AL SCHMITT: We did that in half an hour (DON'T YOU KNOW I LOVE YOU SO). Herb Abramson (Atlantic Records staff) was the producer and it started out as a demo. I think it was the first hit record I ever cut.

Our board had six inputs with one out. The tape machine they had was a Brush Sound Mirror and *then* they got an Ampex. We had people bring in recorded wire and that became a pain when we had to edit it. We had some old Altec speaker, and the funny thing was, some studios would have it mounted on the left side of your ear and some would have it mounted on the right side for your right ear. We'd have ear tests to see how developed our ears were getting — one side over the other.

HOWARD CUMMINGS: Did you walk around with your head tilted? Why didn't they put the speaker in front of the board?

AL SCHMITT: There was actually no room in front of the board for the speaker – the studio glass literally being only inches away from the rear of the board.

Next I moved on to Nola Recording (NYC), a little demo studio. Then Fulton Recording had a job-opening – again I was with Tom Dowd.

HOWARD CUMMINGS: How did you get to California?

AL SCHMITT: I had worked at a place called Coastal Recording, which bought-out Fulton, and I had done some sessions with jazz producer Dick Bock (Pacific Jazz Records) in New York. During the time I had also been working with the Drifters, Clyde McPhatter, the Modern Jazz Quartet, and Connie Francis, and it was at this point that Dick suggested I move to the West Coast. I said OK, if he could find me a job out there because I didn't want to go out on spec. So when I came out I loved it. The scene was great, the weather was great. So I started out here at Radio Recorders and worked with Elvis on his first post-army LP. G.I. BLUES. Elvis was one of the first to start the "marathon session" – the 12-hour days where food was sent in, etc. He was great, very polite and a real gentleman.

Then I started at RCA studios in 1959 at their then-new Sunset & Vine location in Hollywood. The rooms were great. There must have been seven studios over there. I used to see Groucho (Marx) every day and got some of his one-liners as he would go by on the way to his studio.

Around 1964 I wanted to become a producer. I felt that being an engineer for as long as I had, and with as many producers who didn't know anything about making records – and were getting all the credit – I wanted to do it. Steve Sholes (the man who signed Elvis, Chet Atkins, and Eddie Arnold to RCA) and I talked it over. And I told him I had been offered another job as chief engineer with a lot more money, *but*, if I can get into A&R, I'd stay. So I took a pay cut and went into A&R. He encouraged me a lot and backed me on some of my avant-garde ideas.

HOWARD CUMMINGS: Like what?

AL SCHMITT: We did a thing with Paul Horn and Lalo Schifrin called THE JAZZ SUITE ON THE MASS TEXTS[1] – the Catholic mass – and that got some people upset ('65). We got Father O'Conner in New York to do the liner notes after sending him some tapes, but some guy on the (RCA) board who was friends with the Pope and very big in the Catholic Church, got very upset.

HC: Because of this jazz-religion fusion?

AS: Exactly. But Father O'Conner thought it was great and he even gave us the title! It was the first album of that kind done and was very spontaneous – done 'live' – and Lalo did some nice arrangements, including some good sax-voice things that just happened – there was nothing written out.

HC: Did you use a lot of reverberation to simulate that church ambience?

AS: Certain kinds of things, yeah. The album had an incredible cover – I think it won three Grammy Awards*, but was not especially successful in terms of sales.

But when I got into A&R, I didn't touch a board for about five years – didn't do any engineering – and what was funny, the first year I was out of it, I got a Grammy in engineering for the year before on HATARI.[2]

HC: Was there such a thing as points in those days (royalty percentage) for staff A&R?

AS: You could make like ¼ of a point – it was almost impossible to make money with the RCA structure and deductions. I think the biggest bonus I got was like $2,500, and I had had a couple of hit singles and albums. It was crazy. For example, the guy that produced THE SOUND OF MUSIC – which sold in the millions – never got a dime (in royalties) because he had the title of "Manager of the West Coast", and once you had a "title", you were off the percentage thing.

HC: How were the HATARI sessions with Henry Mancini?

AS: The HATARI things were all done at one time. I think there were 60 men and Hank had gone to Africa for the motion picture and to listen to some of the African music, etc., and he brought back all these instruments.

It was the first time I ever saw a Kalimba (African finger piano). He also had a bass in the studio – it was a large piece of wood hollowed out and it was the *way* you hit it and *how* you hit it and it gave this *whoomph*. It was really a difficult instrument to record with this big fat sound.

He also brought back some of the long dried beans – they were huge – about 4 or 5 feet tall, and you'd shake them and the pods would rattle. I think we did the album in three sessions – all done to mono and 2-track.

HC: Simultaneously – for mono and stereo release?

AS: Yes, and then we edited. We had like six bass flutes, two upright basses, French horns, brass, strings, all the percussion going on . . . I think there were four guys who doubled on everything.

I set up about four or five mikes and got an overall balance on those and then we kind of adjusted, on the run-downs, where the guys should move – to move back or play with softer mallets on vibes, for example, to balance the whole thing. So they would do the balancing of dyna-

* *Best Original Jazz composition. Best Album Cover (1965).*

TOMMY LIPUMA on AL SCHMITT and BREEZIN'

Al is ready when the music is ready. I'm too busy listening to the music, and don't want to have to worry about things like equalization, etc., which are things that *also* must be taken care of. He's able to get a balance within 10 to 15 minutes, while the musicians are still hot, and while they feel good.

Al is a very *musical* guy; he's got sensitivity to the things that are important. Anything that he treats with EQ is very subtle, it's all mike technique with him, very natural sounding. You're not being hyped by EQ, limiters, etc., when you listen to it. He doesn't overemphasize anything when he's recording – if it's used, it's used only if it's needed.

When we recorded George, the atmosphere was very relaxed. It was a team effort. A lot of care went into the record; a combination of the band, the mixing and the songs. I try to *cast* songs for the artist, and I cast them for albums – not singles. I make albums and then sequence them so there is a good balance from the start of the LP to the end – – like on BREEZIN'.

Al Schmitt, Tommy Lipuma and Don Henderson.
Photo: Howard Cummings

mics among themselves instead of my relying a lot on faders.

The way Hank wrote, the whole sound was more of a blending – it was hard to pick out individual instruments.

HC: Was this in the large studios at RCA?
AS: Right. It had about a 30' ceiling, 40' wide and about 60-65' long with the parquet floors – just beautiful rooms. It took awhile to learn the room but once you had it down, you could do anything – and I used to change set-ups all the time just to experiment with the room. No one would ever say to me, "Don't try anything." It was just like we really had it together. If you knew the room at all, some sounds were great, no matter where you set things up. But it was always good – it was never bad. I've had as many as 110 musicians in at one time.

I tend to think that only a bad engineer can mess up a good room. I don't think a good engineer is going to get great results out of a bad room either, but I think he's got at least a good chance of getting something decent.

HC: Well, my philosophy is that people can overcome equipment.
AS: Exactly, I agree, and that's why I do a lot of things at Capitol now because most of my training, most of my background, was in large rooms in New York and at the Annex at Radio Recorders (Hollywood).

HC: When you say "the Annex" – is that Thorne Norgar's place?
AS: Right, part of Radio Recorders – and down the street from that was the old RCA soundstage.

HC: Because that stereo stuff that Thorne did with Elvis in the early 60's was just so great.
AS: Oh, I know what you mean.

HC: How about the equipment in those days?
AS: At RCA they had 604 speakers. When they went to 3-track they had three speakers, when they went to 4-track, there were four speakers. We had separate monitoring for cues. The way the board was set up was: four faders and a sub-master, four faders and a sub-master, four and a sub-master, and four and a sub-master. So there were 16-in.

HC: Was this a Westrex or an RCA design?
AS: An RCA design. The EQ was "high" and "low". If we did anything, we used to bring in Pultecs and Langs. But if I need a lot of EQ, I go out and check the room or change the mike instead of trying to save it with EQ. If I'm adding more than 6 dB to anything, there's something wrong *for me* and I better go outside and change it. I really don't like to go more than 2-3 dB on anything.

HC: What sort of mikes might have predominated on HATARI – a lot of (RCA) 77's?
AS: No, I was into (tube) 47's, 67's, (Sony) C37's – as many condensers as I could. I used a lot of 49's and AKG's. On celli, RCA had developed a mike, the 10001, and it looked like a (RCA) 44 on one side and the other side looked like a (RCA) 77 with that round back, which was the dead side. It was a fantastic mike for low strings using that ribbon cardioid.

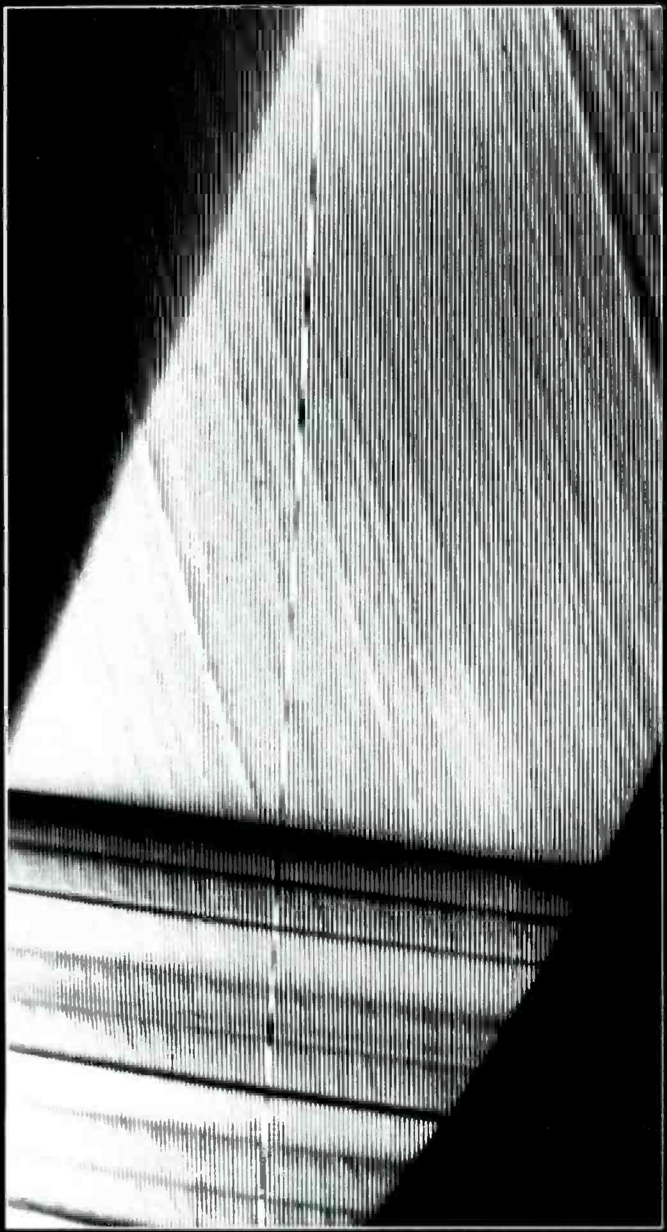
But I used to experiment with microphones all the time. RCA had a microphone called the BK 5, then the BK 10 – it was like two BK 5's together which I used with Bing Crosby once.

In those days, if you were using a 77 on the vocals, we'd take a pencil, put it across the mike and tape it at the bottom so the pencil was off the mike and the "wind" of the vocalist would be diffused after hitting the pencil to prevent vocal "pops". We also used cheesecloth at times.

HC: I even thought of – if someone doesn't know how to work a mike properly – of wrapping pipe cleaners around with the ends sticking out and telling them to keep that distance away from it.
AS: Right, exactly. I always used to tell people to put their fingers to their nose and stay away at least that far. It's a little different today in recording. Stage acts are usually on top of the mike all the time – there are very few people that have really good mike technique today.

HC: That's another thing about Elvis – he knows how to work a mike.
AS: Elvis was great. Rosemary Clooney was one of the all-time greats. You put a mike on her, get a level, and you're all set. I *never* had a microphone problem with her. She'd lean in for the low notes – everything was "even" and done in good taste – she really had it down.

Those were the things where you got great experience working with full orchestras, good singers, and good arrangers. The writing never got in the way. It's different today – things tend to get "overbusy". A lot of arrangers now have the philosophy that, "Well, look, let me stick it all in. It's always easier to take it out than it is to add things." So a lot of guys tend to over-write things and, of course, it tends to put an engineer into fits. By the time you get things cleaned up or the producer gets things cut out, a lot of time gets used. In those days they wrote exactly the way they felt; with the "fills", the things under the vocals were "cushions" and there wasn't a lot of open brass on the vocals. It was a lot easier to do "live" sessions in those days also, whereas today you do the rhythm section first, then they'll overdub the singer, then the flutes, then the strings. I've seen sessions where



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they overdub the low strings, then the violins so there wouldn't be any leakage with that. Again it's OK because it's the end result that counts. There are a lot of good-sounding records out there. Just for me, personally, I prefer it the other way. I like leakage, I like that "open" . . .

HC: Interaction.
AS: Yeah.

HC: How did you get involved with the Airplane?
AS: At RCA I was the first person to see the Jefferson Airplane — from the A&R staff anyway. I went up to San Francisco and saw them at the Matrix (ballroom). But I really liked them — everyone was excited about them. I told RCA they were worth the advance money.

Then they came down to (RCA) studio C to record. Their first LP had bells and all kinds of stuff and it was terrible — I couldn't believe that a group that sounded like they did on stage in San Francisco sounded like this on record. It was a big disappointment *to me.*

HC: It's ironic that since you signed them or had a say in signing them — that you didn't produce.
AS: I wasn't really interested in producing them at the time. I was doing a lot of other things and was basically interested in jazz things at that time.

HC: You produced some of their LP's starting in '67. I tended to favor BAXTER'S[3] *over CROWN OF CREATION.*[4]
AS: BAXTER'S took a long time. I liked VOLUNTEERS ('69) an awful lot. We spent over five months on BAXTER'S though. It was a whole new experience for me — of experimenting with things. We tried on one song to rewind a tape in seven seconds to make it fit into an ending — we tried that for three days and never got that right. (Song-POONEIL)

HC: Maybe I could set up the scene. Rick Jarrard had produced their previous LP, SURREALISTIC PILLOW . . .
AS: They hated it. Although it had two hits in it, they hated it.

HC: It had a lot of reverberation — super reverb.
AS: That's it. They said it wasn't them — it sounded like it was done in a tunnel, and they didn't care how many hits were in it, they said they didn't want to work with Rick again.

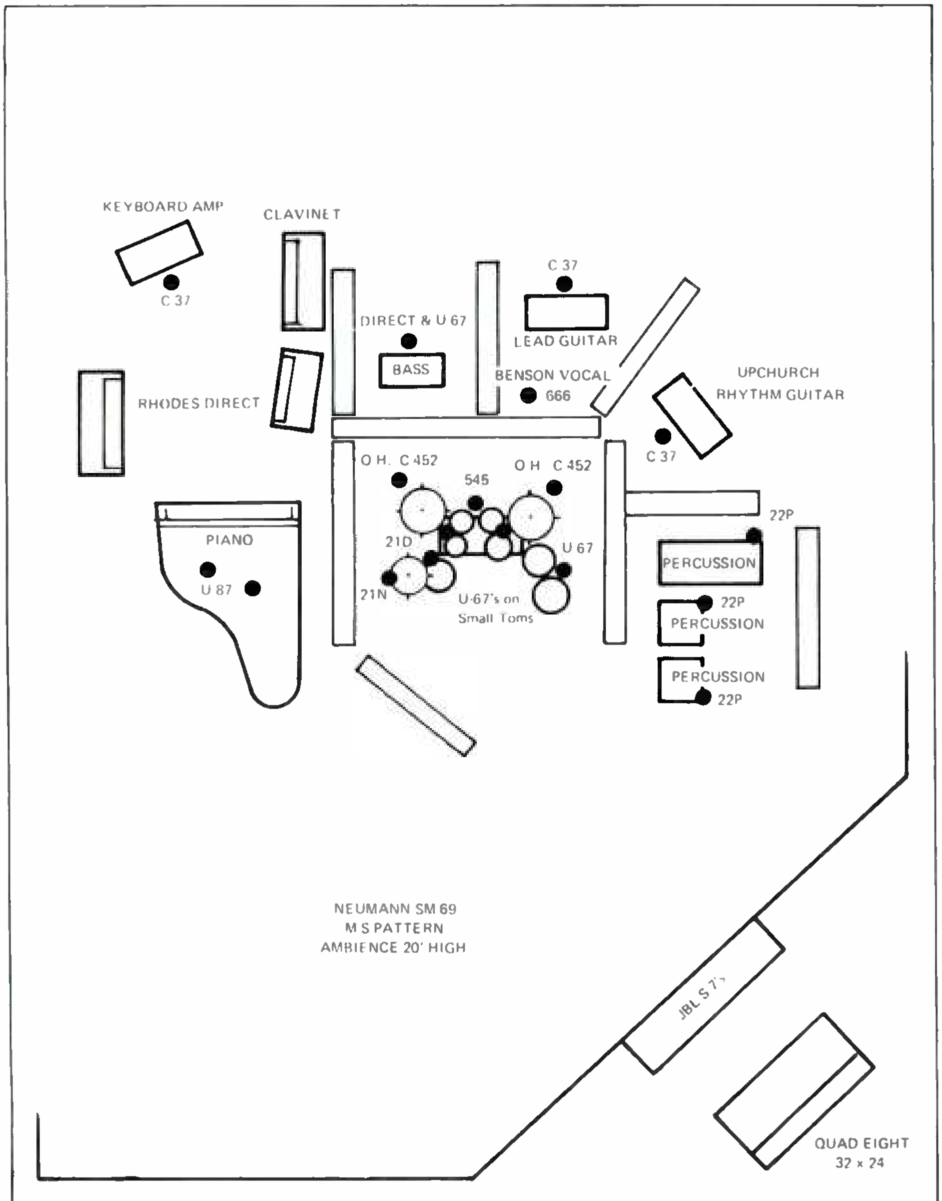
So RCA asked them if they'd like to work with me and they said, "OK, we'll give it a shot."

HC: This was the album that really set them up . . . it was the SERGEANT PEPPER era. Everyone had heard what the Beatles had done on PEPPER — the Doors, the Airplane. Now when they came in to you, did they say, "This is what PEPPER sounds like, this is what the Beatles are doing, we want the same thing."?
AS: No. The important thing was that I allowed them to make their music — that was the discussion. They didn't know anything about recording. It was their third album, I didn't know a lot about them, and I learned an awful lot in working with them as far as patience.

And some of the things they did were done for shock value — not because they actually believed in those things — but because they wanted to become noticed and looked at.

Each person would have his own song and would act as co-producer with me. But after awhile I was putting in *so* many hours. With Eddie Fisher in the afternoons and singles and LP's for eight or nine acts and the Airplane from nighttime until early morning, it all became too much. So I called the A&R head of RCA in New York and told him about the long hours and he said, "What do you mean long hours? Truck drivers do it." So I said, "Then go get yourself a truck driver," and handed in my resignation. This was around CROWN OF CREATION ('68). So I went independent. When the Airplane found out, they wanted *me* to continue working with them since there was no one else they wanted to work with at RCA. It became better for me mentally and financially, and they continued to ask me to work with them right on into Hot Tuna. So we had a good rapport and relationship.

HC: Did you have a say in cutting the disc-masters?



BREEZIN' LP with George Benson
Engineer: Al Schmitt at Capitol Records, Hollywood
30 ips, 24 track, 3dB elevated level, Ampex 407 (1 mil base)

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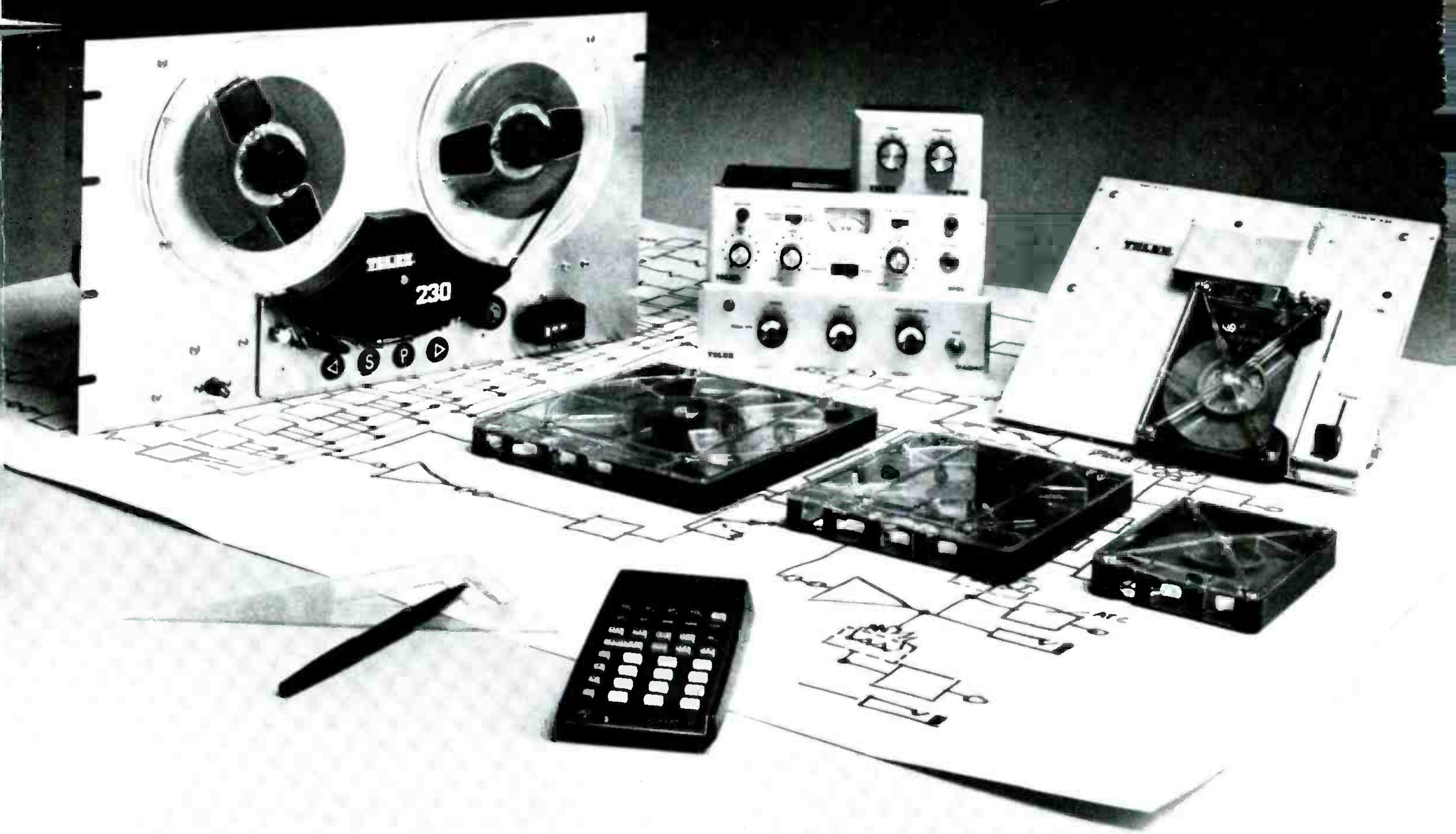

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AS: I had to approve the level. We had to submit the "right" tape and then RCA just ran it.

When I was at RCA, if you submitted a tape to be mastered, they cut it flat. If it needed more high-end, *you* either had to go back and make a copy of that tape and add the high-end to it, or re-mix it.

HC: They wouldn't add it in the channel?
AS: No, sir. They just wouldn't because they had to cut all over and they wanted the tape to be perfect. That way there would never be any complaints.

HC: I guess you kind of scooped everyone on this King Kong craze by using part of the 1933 movie in their live *LP, BLESS ITS POINTED LITTLE HEAD.*
AS: Right. They were showing the end of the film before the Airplane came on, so we had this audience reaction to all this and we decided to use the line from the film, "It wasn't the airplane – it was beauty that killed the beast." (Chuckles) So I thought that was good and left it in.

That was another good thing about our relationship. If I had an idea, they'd be willing to try it. If they had an idea, I'd be willing to try it. Nobody was ever against anything. We all really worked hard, and obviously they've got something going for them because they've been around 10-12 years.

HC: Maybe I could name some tracks from BAXTER'S and you could explain the involvement: TWO HEADS –
AS: Grace wrote that. It was great. The idea to do the double vocal thing was mine. We did it as an overdub, both done at different times. She hadn't heard the other while she was doing one so it was unplanned. It seemed to work. We thought it was going to be a smash single, but it was released as a 'B'-side. The first single I put out with them was POONEIL.

HC: Another one I liked – that was another song I was going to mention: THE BALLAD OF YOU AND ME AND POONEIL.
AS: That was one of my favorites, mostly written by Paul (Kantner) who was very underrated as a contributor of song ideas, like MARTHA[3] for instance. I love that.

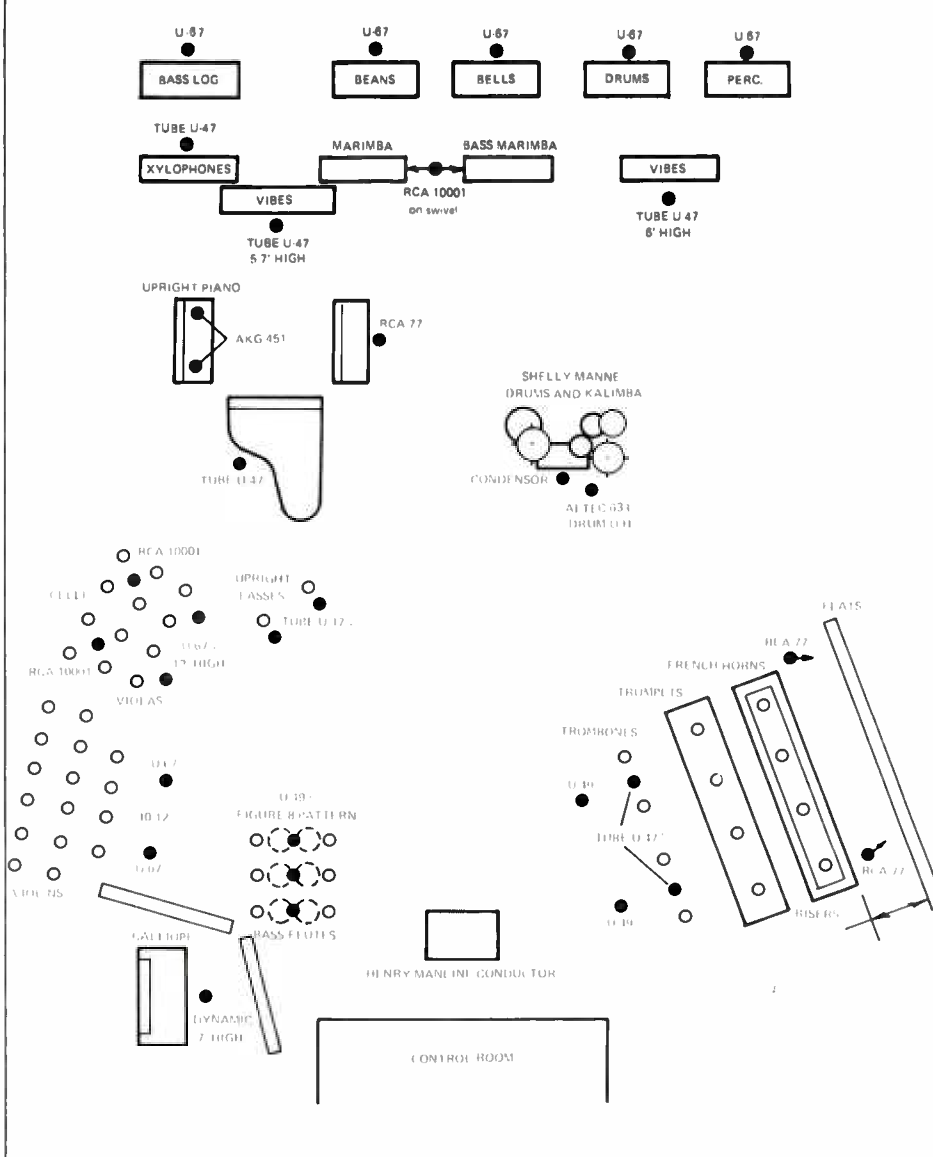
HC: Another song I was going to mention. Those are my favorite three.
AS: When we mixed POONEIL, there were so many changes and mixes and vocal balances, we had to mix it in six or seven different sections. On the vocals we used U67's.

HC: How about VOLUNTEERS?
AS: When we went to Wally Heider's in San Francisco, we had a few problems on VOLUNTEERS. Jack Casady (bassist) wanted that big, big sound. We only had a little amp in the studio, but Heider's had this room they weren't finished constructing. So we took his big bass amps and put them in the other room so he could turn them up to 10 and wouldn't bother anyone else. Another thing was that since Heider's was so new at the time – we were one of the first groups to go in there – we had to take a lot of time experimenting with the room.

The Airplane *did* teach me a lot of patience like I say. I was always a nervous person. I hate to waste money, I hate to waste time, "C'mon, c'mon, let's get it done." It's a good virtue, but sometimes that's not *always* the way. You have to look at their side of it – if they weren't in the mood to perform – cause sometimes rushing it took us ten times longer.

HC: Did you find the same things when you worked with Neil Young for ON THE BEACH?
AS: Neil was more marathon sessions. We'd start around 7 p.m. and work until 7 a.m. We'd have a lot of people like Steve Stills and Graham Nash come down and we'd have to stop what we were doing and string up other multi-tracks to let the guys hear what was going on. So I asked Neil to let me make some rough ¼" mixes for convenience. He liked *those* mixes so much that that was what later came out on the LP!

HC: Were they dry?
AS: Dry, 15 ips, no EQ, and I later talked to Neil about it for an hour at my birthday party and almost begged him to let me re-mix them. "Just let me have a day."



HATARI · Al Schmitt
RCA Studio A, Sunset and Vine, Hollywood

He said no, he liked them the way they were. And my argument was that, "You can always use those but I think I can make them sound better and more state-of-the-art." But there was nothing in the world I could do to change his mind. I even think the (lacquer) masters were cut flat off the tape.

Another thing, we set the orchestra up – orchestra, it was three guys – in like a living room with dim lights so we could barely see. Rusty Kershaw (Doug's brother), who is a *magnificent* guitar player, was playing while drinking and had had a few. The thing was going along really nice and all of a sudden we'd come to the acoustic guitar solo, and it would disappear! I'd be cranking up the pot and getting room noise and I'd look out into the studio through those dim lights and couldn't see what was wrong. Finally I snuck out into the studio and found that as he would come forward to play, he would push the mike aside . . .

HC: (laughter)
AS: he did that to me two or three times – just push it out of the way and play. They had to carry him home – he was so out of it.

Most of the good material we got was in the first few hours. After we ate, it was just . . . just . . .

HC: downhill from there.
AS: But Neil was great, he kept it together.

HC: I was always curious about the Streisand sessions you and Tommy were involved in for THE WAY WE WERE. The three top songs for me off the LP were THE WAY WE WERE, SOMETHING SO RIGHT, and ALL IN LOVE IS FAIR. I understand there were two different versions of THE WAY – the film soundtrack version and the LP version.
AS: We did four things I think, of which only three went into the album. That was also all *live* – there were no overdubs. First of all I went and found a U49, which I knew was the microphone she really liked, so when she saw that she was really comfortable.

We got an earphone mix quickly on the session and then she wanted echo in the earphones and we couldn't do it – the way it was set up. We just could not.

HC: Not even any "slap"?
AS: I think that would have bothered her more . . . she was looking for an echo effect to fill out the cans. Another thing that upset her occurred in the first run-down. I always record the first run-down cause it might be good – you never know. While I was adjusting the faders, the maintenance man at United-Western was trying to adjust the limiter on her and as he was getting it set, she hit some peaks that were just killers and the limiter just sshoomp . . . it was 30 dB and you could hear it suck. So she came into the control

room and wanted to hear it and I wasn't nuts about playing it back because I knew those things were there. I hit the playback and she knew right away: "You're using a limiter on me? It's terrible", and this and that. So I think my comment to her was, "Look, you don't always do it right the first time. Give us a shot, too."

HC: How did she take that?
AS: She took it not badly — she didn't get crazy or anything . . .

HC: Maybe because you stood up to her and she accepted it on that (gestures) profession-to-profession basis.
AS: Right. So then she went in and I think on the fourth take we played back again. When she came in and heard the fourth take after we had it together she said, "The sound of my voice is magnificent. It's just beautiful — I love it — it's great." She was very pleased and happy.

Most guys wouldn't record the first take but me and Tommy have worked so long together, we automatically do it. We were there about 15 hours straight with all those musicians also.

HC: With about 40 people?
AS: 40, 42, 44. Some of those cuts in the album were done in the 1960's, too.

HC: In the can all that time?
AS: Yeah, But when we went in to mix, Tommy told them the only way he would be involved was to mix it all — 60's cuts with the new stuff we did — so it would all sound like it was done at one time, which they gave us the OK to do. Some of the stuff was done on 8-track, some of it wasn't marked, we had to figure out which vocal tracks to use. Then Barbra and Jon (Peters) came down to listen and they loved it — the only thing was that we used a different ending on THE WAY WE WERE for the single.

HC: Then there are three versions: the film soundtrack, LP track, and single track.
AS: Right. The single version and the LP versions are different in that the endings are different. There were no marks, no one had left us any information on any of these things to pick the right endings. So when she heard it on the radio two months after it was out, we got a phone call from her, but there was nothing we could do at that point.

She has an incredible memory. When we worked on BUTTERFLY, she'd call me on the phone and sing endings to me. She'd say, "I did this on take one, I did this on take four." Great memory when it comes to what she's done musically. And doesn't sing badly — ever.

HC: That's what impressed me about those cuts I mentioned on the LP: the selection of material, the execution, and the technical end — cause I was never much of a Streisand fan before that.
AS: Well, one of the violin players who I respect a great deal said, "Look, God just came down and kissed her on the throat. It's like a bell." Her intonation is just incredible, but she's a little intimidating. I'd have to think twice if she asked me to work with her again.

HC: How did you gravitate back into engineering after five years — being away from the board and into A&R?
AS: Tommy Lipuma — we've been friends for ages — asked me to come in and overdub a couple of things on Dave Mason (ALONE TOGETHER LP) which Bruce Botnick had originally started. I said OK. Then he asked me if I would *mix* it, and I said to him, "Gee, Tom, I haven't done any mixing in so long and I don't know if I could do it any justice." He said, "Try it. I think you could do it." And I didn't want to embarrass myself and embarrass him by saying it's not working — but I ended up saying OK again. Fortunately it came out fantastic — I was just thrilled with it.

HC: Would you favor that one over BREEZIN' as far as mixing?
AS: Ah . . . different time, different place, different studio. The difference was, BREEZIN' was done in 9 days — start to finish. Three days to record it, three days to mix it, and three days for acetates and that sort of stuff. It was quick, really quick.

We did eight things, six of which went into the album, but out of the eight things, five were the first take. The vocal on THIS MASQUERADE was 'take one'

— that was it. We went on and did extra takes and the sound got better on some of the things, *but* we never got that "feeling" again — so we went back and used that first take.

HC: So by the time the technical end was there, the feeling was gone?

AS: Right. People don't buy records for the technical end. Certain buffs do — but most people buy for the feeling of the record — how it makes them feel. That's what's important. I've seen engineers say, "I've gotta do one more, I've gotta do one more, I've gotta do one more cause this wasn't right, that wasn't right." Hey, the hell with it. If the record feels good, if it sounds right, if the producer and everybody's happy — that's it. You accept having something not sound exactly the way you would want it to sound. You have to give up a little of that ego and say, "That's not important to the record." I've seen a lot of producers and engineers go past the "feeling" time and time again and wear the thing down.

HC: How did the George Benson - Tommy Lipuma — Al Schmitt association come about?

AS: Tommy and I were in a cab in San Francisco going to record some Dan Hicks I think, and all of a sudden we see this marquee that says "George Benson". He's one of my favorites and one of Tommy's favorites, so we went in and caught his act. That was the first time I heard him sing — it knocked us out. At that time he was signed to CTI, so when he signed to Warners Tommy was going to produce and Tommy asked me if I'd like to work on it and I said, "Are you kidding? Sure, I'd love to. It'd be a labor of love." And I knew it would be one of those things where it would all come off at once. So it was nice. We had great musicians: Ralph McDonald played on it, Harvey Mason (drums), Jorge Dalto (keyboards), Ronnie Foster on keyboards, Stanley Banks (bass), and one of my favorites — Phillip Upchurch (rhythm guitar) — who set some of the moods and got a lot of the feeling going — he's great, fantastic.

HC: How did you go about screening studios?

AS: Well, we'd been working a lot at Capitol with that nice echo chamber and Tommy likes it there. They seem to give us almost anything we want in equipment, it's a nice-size room and we could fit the people in it.

As far as the lay-out, we had the drums and then everybody on risers around them, so everybody was so close — they could almost reach out and grab everybody's hand. Benson was facing the drummer, the piano was wide-open, the bass-player was facing the drummer. So everybody could hear well — nobody really needed earphones. Although we did have some, most of the time they didn't use them.

HC: How did you like 67's for the bass?

AS: The U67's are fantastic. They're a much better mike than the 87. But I use the 87's on a lot of things. They sound better when they're open all the way around (pickup pattern) than when they're only in the cardioid position. It gives you a chance to get part of the room. I use it in omni when I do an acoustic guitar or a horn.

The other ones I like a lot are the AKG C12, and the (Neumann) SM69 stereo mike in the M-S pattern for overdubbing strings.

HC: You talked about the sessions themselves — the general "feel". Were there any problems?

AS: Little problems with the bass.

HC: Is this acoustic or electric bass now?

AS: This is electric. Also a lot of engineers came down. As a matter of fact, Hank Cicalo came down, and once people started to hear what was going on, it was kind of an underground thing that "Benson was in town recording and it was really happening" and sometimes at the end of sessions there would be 20 people coming down to sit in the studio while we were playing-back.

HC: So the only minor things were bass things.

AS: Well, another thing happened. I put up a mike because I *thought* we were going to overdub his voice. So I stuck up this cheap mike, a (EV) 666 and then we realized we had this thing (performance) that was great.

So on his next album (IN FLIGHT), I wanted to use a better mike and he (Benson) wouldn't let me — even though it might have sounded a little better to me. He's a very superstitious person and in fact he wanted the same studio, the same engineer, the same personnel, the same microphones. The important thing is as long as he's comfortable. So I stuck it (666) back up and it was fine. He had that little thing of wanting to duplicate . . .

HC: Success.

AS: . . . his success, of course. In other words, "Why change now?" Even though I knew an 87 or a 67 or whatever would give me a better sound on his voice.

HC: Was the original intent to do the vocals as a "scratch-track"?

AS: See, I didn't know at first and when I put a 666 up there, I *assumed* we'd probably do the vocals over because I thought they would be playing some hard things and they're sitting right in the middle with no baffles around him at all and everybody was so close. I thought we'd have a problem but we didn't. As I said, the first take was it, I didn't even have a chance to change microphones. We did a couple more right after that but they were so excited and so happy. Maybe I was fortunate I used the 666 because it kept a lot of the leakage out.

HC: What about EQ and compression while you were recording on the 666?

AS: I didn't use any EQ at all and I used about 2 dB of limiting — as a matter of fact, I don't even think we had the limiter in while we recorded. I think we limited when we mixed. But there are spots on the record where it *sounds* like he's being limited, but he's not. It's the 666 and he moves his head a lot. When he moves out of range, it sounds like it's being "pulled". But I say again, the important thing is that it feels good and it's comfortable.

I never thought I'd win a Grammy for it, to be honest with you, but I wasn't ashamed of it. I thought it sounded awfully nice.

HC: Did you expect it to get nominated?

AS: No I didn't . . . those things are so much a matter of luck at times. There are a lot of good-sounding albums that shouldn't necessarily have to be popular to win a Grammy for best-engineering. The good-sounding albums that never get brought to the attention of the people who are on the (Grammys) board just seem to go by the wayside. One in particular was the Dave Mason album (ALONE TOGETHER), which I thought was a phenomenal album. But in those days they were afraid to give it to anything that was over-dubbed. Over-dubbing wasn't considered engineering.

HC: I noticed you used (Ampex) 407 which is 1 mil.
AS: Yeah, on BREEZIN' and on IN FLIGHT also. With George, when they get burnin', they just keep playing. The song may run 6½ minutes one time and you'll do take 2 and *that* may run 11 minutes. So once they got cookin', no one wanted to stop them and rather than take a chance and run out of tape, we used the 1 mil and it worked out fine.

HC: How did you handle the strings?
AS: This is called, "Get your shit together" (laughs). We sent 7½ ips of ref tapes over to (conducter) Claus Ogerman after we were done so he could do the arrangements. So we decided that because they had such great string players in Munich, we would record them there. Well, we had this studio booked, I can't remember the name of it, and we went over the arrangements with Claus. Then myself and Tommy flew to Paris for a couple of days while Claus worked out the arrangements. The next day we were going to record at 7 p.m. and we had 39-40 players who were going to show up. So I brought the four 2" reels I had been guarding with my life, carrying them around, and we got to the studio and it's incredible. All the equipment is fantastic, everything was so modern. They had their playback panel in a drawer — you pull the drawer out. If you even wanted to EQ your playback, you could — it was magnificent. So I looked over at the tape machine and it's a Telefunken, and all of a sudden my mind does a little skip. The only other time I had an experience with a Telefunken was when I was doing some mastering and the machine only ran 7½-15, while *I* had a 30 ips tape resulting in the guy having to bring another machine in. So now we asked *these* people, "Is it 30 ips?" and they said, "Oh, no."

HC: So you had 30 ips-2 inch.
AS: We had 30 ips-2 inch and all they could do was 15, and here we had 40 musicians coming up. So Claus says the only other studio is Musicland. We called them and Giorgio (owner) is there working on his stuff. Fortunately he didn't have an outside client. So we practically threw him out of his own studio — what it boiled down to was we wouldn't let him say 'no'.

Now we get in this room, and when I tell you the ceiling — you stand on your tippy toes and out-stretch your hand — you could touch this ceiling. I had three flutes in a vocal booth because it's the only place I could put them. I couldn't put *anybody* in any conventional set up. I had people bending around corners, everyone was so close together, there weren't enough music stands, not enough lights, there were no earphones. The only person with earphones was Claus. He conducted the whole thing, which is a tribute to him. I thought the session was a disaster but I went into the room and he said, "What do you think?" I said, "Look, we're going to have to pay for the musicians anyway. I don't think it's going to come off but let's try it anyway – we've got nothing to lose."

HC: Now what sort of music was Giorgio doing?
AS: He's the guy that does the Silver Convention-Donna Summer stuff.

HC: Symphonic disco.
AS: Yeah, but all little six string things, four violins. But the room was really small.

HC: Do they have only the one studio there?
AS: Only the one. Also the equipment was kind of strange – out-dated. There were two meters on the board. If I wanted to see what the violins were doing, I had to throw a switch to #1 to read what

track one was doing, #2 to read what's on track two –

HC: 16-rotary position?
AS: Right. You couldn't see them all at a glance.

HC: Two meters and take your pick.
AS: That's it. Also the musicians had worked all day playing on a symphony, then they came in to do this. By 1 a.m. we had four things done – there were supposed to be six – and they couldn't work anymore. They were ready to fall down. We couldn't get the studio the next day. There was nowhere to go, so we finally decided to call London to see if we could get in there.

We got in there on a Sunday morning. We did two long things and we did two of the (LP) introductions. It was a *big* beautiful hall (CTS: soundstage) with Tannoy speakers, plenty of room, plenty of facilities with Neve and Studers, and great help.

HC: Was it a problem getting in on a weekend?
AS: No, it was amazing, and there they pay the musicians at the end of the session. We had great musicians. The contractor is called "the fixer". They also put up these little tiny speakers in front of each guy.

HC: Like five-inchers for foldback?

AS: Yeah, and we adjusted those so each guy could hear exactly what was going on. We didn't get much leakage and it worked out OK. So we were in London for about 24 hours.

People talk about Grammy award records. Well, sure I would have liked to have done the strings at Capitol, but this was something we had no control over – it just happened that way. When we were at Captiol with the strings (IN FLIGHT) and I had the room and space, I could use the M-S stereo mike to get that spread effect so that the strings would *appear* to be positioned left but there was a lot of bleeding and leakage to the right side and the same thing with the low strings to eliminate that violins left, violas right. Just to eliminate that total isolation.

HC: It seems strange to me when I hear it with that total isolation.
AS: I hate it! But the room for the BREEZIN' strings was so small I had to bleed it in the mix with echo, etc.

HC: It seems like you assigned all of your strings to one of the stereo buss' on some things. (BREEZIN')
AS: On some things. What I tried to do was spread them as much as I could: violins on one side, violas and celli on the other, flutes kinda' centered and spread.

But in the rooms we worked in, there was no room to use stereo microphones. There was no space to put a mike up to get room ambience, all we got was the sound off the instrument. The mikes couldn't have been far from the instrument — 2-2½' cause that's all we could do. We just made-do with it.

Again I say, the album has sold 2 million copies, it won a Grammy, it was record of the year . . .

HC: And netted . . . netted *a lot of money with only nine days of recording expenses.*

AS: The expense was nothing — not even 70 hours. We mixed it in no time at all — about two days.

But George got rich, he deserves it. He's only been another 20-year overnight sensation, struggling for years and years.

DISCOGRAPHY REFERENCE — AL SCHMITT
[*Grammy Wins] [(P) As Producer]

	TITLE	*STUDIO*	*ARTIST*	*U.S. LABEL & DATE*
1.	JAZZ SUITE ON THE MASS TEXTS	B	Lalo Schifrin	RCA 1965
2.	*HATARI	B	Henry Mancini	RCA 1962
3.	AFTER BATHING AT BAXTER'S (P)	B	The Jefferson Airplane	RCA 1967
4.	CROWN OF CREATION (P)	B	The Jefferson Airplane	RCA 1968
5.	THE WAY WE WERE	H,A	Barbara Streisand	Columbia 1973
6.	ON THE BEACH (P)	E	Neil Young	Warners 1974
7.	*BREEZIN'	D,G	George Benson	Warners 1976
8.	IN FLIGHT	D	George Benson	Warners 1977

STUDIOS:
A - HOLLYWOOD SOUND, Hollywood
B - RCA, Hollywood
C - WALLY HEIDER, San Francisco
D - CAPITOL, Hollywood
E - SUNSET SOUND, Hollywood
F - WALLY HEIDER, Hollywood
G - CTS, London & MUSICLAND, Munich
H - UNITED/WESTERN, Hollywood

RECOGNITION 1976

RECORD OF THE YEAR
Engineers — **AL SCHMITT**, assisted by Don Henderson
Producer — Tommy Lipuma
Studio — Capitol Records, Hollywood
Disc-Mastering — Doug Sax, Mastering Lab, Hollywood
— *THIS MASQUERADE - George Benson*

ALBUM OF THE YEAR
Engineers — **GARY OLAZABEL & JOHN FISCHBACH**, assisted by Dave Henson
Producer — Stevie Wonder
Studio — Crystal, Hollywood
The Hit Factory, New York City
Record Plant, Sausalito, California
Disc-Mastering — Andrew Berliner & Jeff Sanders, Crystal, Hollywood
— *SONGS IN THE KEY OF LIFE*
Stevie Wonder

PRODUCER OF THE YEAR – STEVIE WONDER
SONGS IN THE KEY OF LIFE

POP VOCAL – FEMALE
Engineers — **Val Garay**, assisted by Greg Ladanyi
Producer — Peter Asher
Studio — Sound Factory, Hollywood
Disc-Mastering — Bernie Grundman, A&M, Hollywood
— *HASTEN DOWN THE WIND - Linda Ronstadt*

POP VOCAL – MALE
Engineers — **Gary Olazabel & John Fischbach**, assisted by Dave Henson
Producer — Stevie Wonder
Studio — Crystal, Hollywood
The Hit Factory, New York City
Record Plant, Sausalito, California
Disc-Mastering — Andrew Berliner & Jeff Sanders, Crystal, Hollywood
— *SONGS IN THE KEY OF LIFE*
Stevie Wonder

POP VOCAL – DUO, GROUP OR CHORUS
Engineers — **Wayne Tarnowski**, assisted by Tom Likes (Armin Steiner)
Producer — James William Guercio
Studio — Caribou Ranch, Nederland, Colorado
United-Western, Hollywood
Disc-Mastering — Doug Sax, Mastering Lab, Hollywood
— *IF YOU LEAVE ME NOW - Chicago*

POP INSTRUMENTAL
Engineers — **Al Schmitt**, assisted by Don Henderson
Producer — Tommy Lipuma
Studio — Capitol Records, Hollywood
Disc-Mastering — Doug Sax, Mastering Lab, Hollywood
— *BREEZIN' - George Benson (album)*

R&B VOCAL - FEMALE
Engineers — **Paul Serrano & Steve Hodge**
Producers — Chuck Jackson & Marvin Yancy
Studio — PS Recording, Chicago
Westlake Audio, Los Angeles
Disc-Mastering — Wally Traugott, Capitol, Hollywood
— *SOPHISTICATED LADY (SHE'S A DIFFERENT LADY) - Natalie Cole*

R&B VOCAL - MALE
Engineers — **Gary Olazabel & John Fischbach**, assisted by Dave Henson
Producer — Stevie Wonder
Studio — The Hit Factory, New York City
Crystal, Hollywood
Record Plant, Sausalito, California
Disc-Mastering — Andrew Berliner & Jeff Sanders, Crystal, Hollywood
— *I WISH - Stevie Wonder (track)*

R&B VOCAL – DUO, GROUP OR CHORUS
Engineers — **Jim Vitti, Reggie Dozier, & Ellis "Pete" Bishop**
Producer — Don Davis
Studio — United Sound, Detroit
ABC Records, Los Angeles
Disc-Mastering — Phil Cross, ABC Records, Los Angeles
— *YOU DON'T HAVE TO BE A STAR (TO BE IN MY SHOW) - Marilyn McCoo, Billy Davis, Jr.*

COUNTRY VOCAL – FEMALE
Engineers — **Brian Ahern, Stuart Taylor, Bradley Hartman, Miles Wilkinson, and Rudolf Hill**
Producer — Brian Ahern
Studio — The Enactron Truck (remote)
Disc-Mastering — Rudolf Hill, Amigo Studios, North Hollywood
— *ELITE HOTEL - Emmylou Harris (album)*

COUNTRY VOCAL – MALE
Engineers — **Bill Harris, Al Pachucki, Chuck Seitz**
Producers — Tom Collins & Jack D. Johnson
Studio — RCA, Nashville
Disc-Mastering — Randy Kling, RCA, Nashville
— *(I'M A) STAND BY MY WOMAN MAN*
Ronnie Milsap (single)

COUNTRY VOCAL – DUO OR GROUP
Engineer — **Barry "Byrd" Burton**
Producer — Barry "Byrd" Burton
Studio — Sam Phillips Recording, Memphis
Disc-Mastering — Larry Nix, Ardent, Memphis
— *THE END IS NOT IN SIGHT (THE COWBOY TUNE) - Amazing Rhythm Aces*

COUNTRY INSTRUMENTAL
Engineers — **Bill Vandervort**, assisted by Mike Shockley and Ray Butts
Producer — Chet Atkins
Studio — RCA, Nashville
Disc-Mastering — Randy Kling, RCA, Nashville
— *CHESTER & LESTER - Chet Atkins, Les Paul (album)*

JAZZ SOLOIST
Engineer — **Bob Simpson**
Producer — Norman Granz
Studio — RCA, New York City
Disc-Mastering — Richard Simpson, RCA, Hollywood
— *BASIE & ZOOT - Count Basie (album)*

JAZZ GROUP
Engineers — **Bernie Kirsh**, assisted by Michael Frondelli
Producer — Chick Corea
Studio — Electric Lady, New York City
Disc-Mastering — Bob Ludwig, Sterling, New York City
— *THE LEPRECHAUN - Chick Corea*

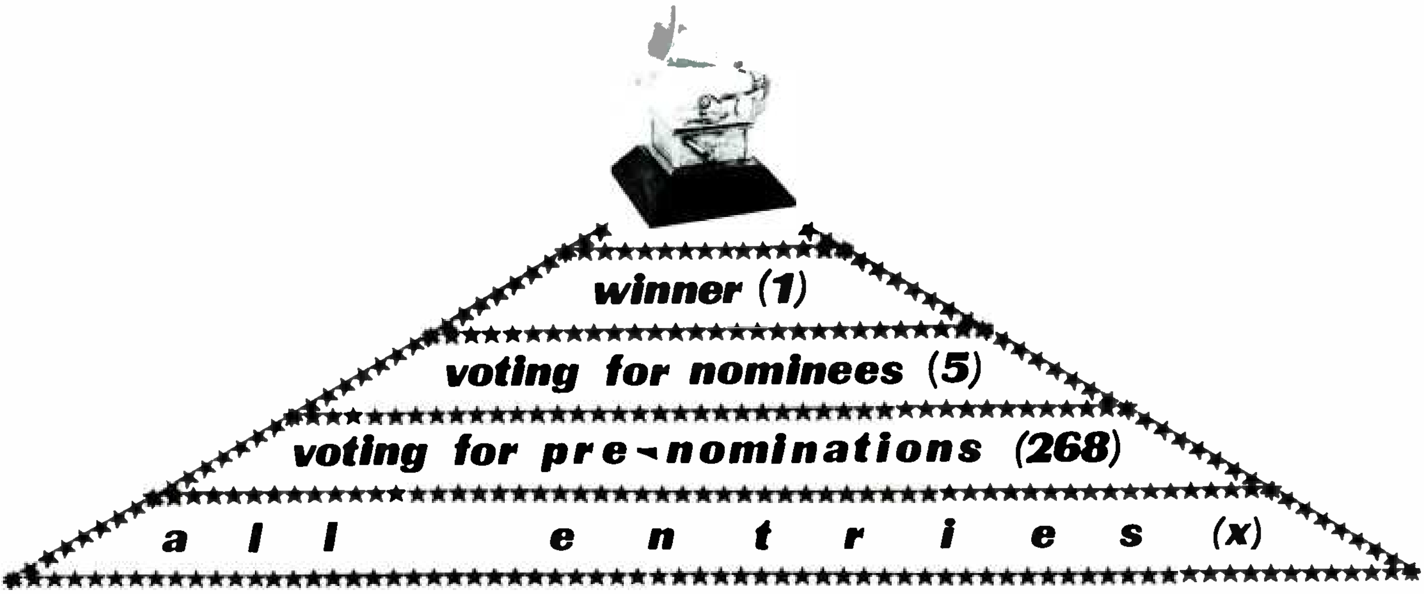
JAZZ BIG BAND
Producer — Duke Ellington (1959, 1971, 1972)
Disc-Mastering — Richard Simpson, RCA, Hollywood
— *THE ELLINGTON SUITES*
Duke Ellington

Compiled by Howard Cummings

* *Addendum; see page 109*

the selection—nominating—voting process—

How GRAMMYS are Won!



The recording team, after multiple mike set-ups, "takes", delays, upsets, revisions, mixes, and test pressings, finally realizes the day when their creation is borne from the recording studio to disc, and finally — into the *world*.

Critical success and sales may follow — and if they should happen to be so lucky, that *attention* that will result in acclaim from one's peers — a GRAMMY nomination.

But there may be questions in the minds of some of those who transform that recorded sound into feeling and emotion — and even in the minds of those who follow the process closely. What are the workings of the GRAMMY procedure? Who votes and how are the final winners determined? And, specifically: How are the engineering entries handled? As a result of these questions, this article will attempt to explain that process. Most of the discussion is devoted to the GRAMMYS in general, and the non-classical engineering category in specific.

First of all, some of the ground rules should be layed out.

NARAS (The National Academy of Recording Arts and Science), is made up of seven chapters situated in the cities of Los Angeles, San Francisco, Chicago, Memphis, Nashville, Atlanta, and New York City. Each chapter is composed of active members who have participated creatively in the field of recording in some way, shape or form, whether it be in art design, arranging, engineering, etc., and *these* are the people who vote. Record company chiefs (unless they are in a creative capacity), marketing directors, promotion men, and the record companies themselves are *not* allowed to vote. Only the individuals who have contributed creatively to the minimum number of recordings receive ballots with which to vote.

Figure 1

GENERAL CATEGORIES:

1 - Record of the Year
2 - Album of the Year
3 - Song of the Year
4 - Best New Artist

SPECIALIZED CATEGORIES:

1 - Pop, Rock, Folk
2 - Rhythm and Blues
3 - Country
4 - Inspirational and Gospel
5 - Ethnic, Traditional, and Latin
6 - Children's, Comedy, Spoken Word, Documentary, and Drama
7 - Composing
8 - Classical Music

CRAFT CATEGORIES:

1 - Arranging
2 - Engineering
3 - Album Covers
4 - Album Notes
5 - Producer of the Year
6 - All Jazz Performances

Feeling that NARAS members should only vote in those categories with which they are familiar, the National Trustees set a limit on the number of creative fields in which a member may vote. In round-one of the voting, the round that determines the ultimate finalists, members can vote in no more than five out of eight Specialized fields. (See Figure 1) In the final round, a member can vote in seven of the fourteen areas (Specialized and Craft combined). Each member can also vote in all four of the General categories: those of "Record of the Year", "Album of the Year", "Song of the Year", and "Best New Artist".

THE VOTING STAGES

Pre-Nominations List: This is the list that is the largest. The National office sends entry forms to members and record companies on which they list those entries (in various categories) which they feel are worthy of consideration in the first half of the eligibility period.* Record companies are invited to submit entries at this point** because they are less likely to overlook recordings that members may miss. Screening committees (critics, reviewers, trade press, record company members, and experts in various musical fields) then refer to these entries to make sure nothing gets left out. These entries (sometimes totaling 5,000 in all three cat-

**The Eligibility Year of 1976 ran from October 16, 1975 to September 30, 1976. In 1977, this will be changed from October 1, 1976 to September 30, 1977 to give Craft committees more time in screening/selection.*

***A record company does not enter everything it has released during the year, but they do provide 7 copies (one per chapter) of everything entered.*

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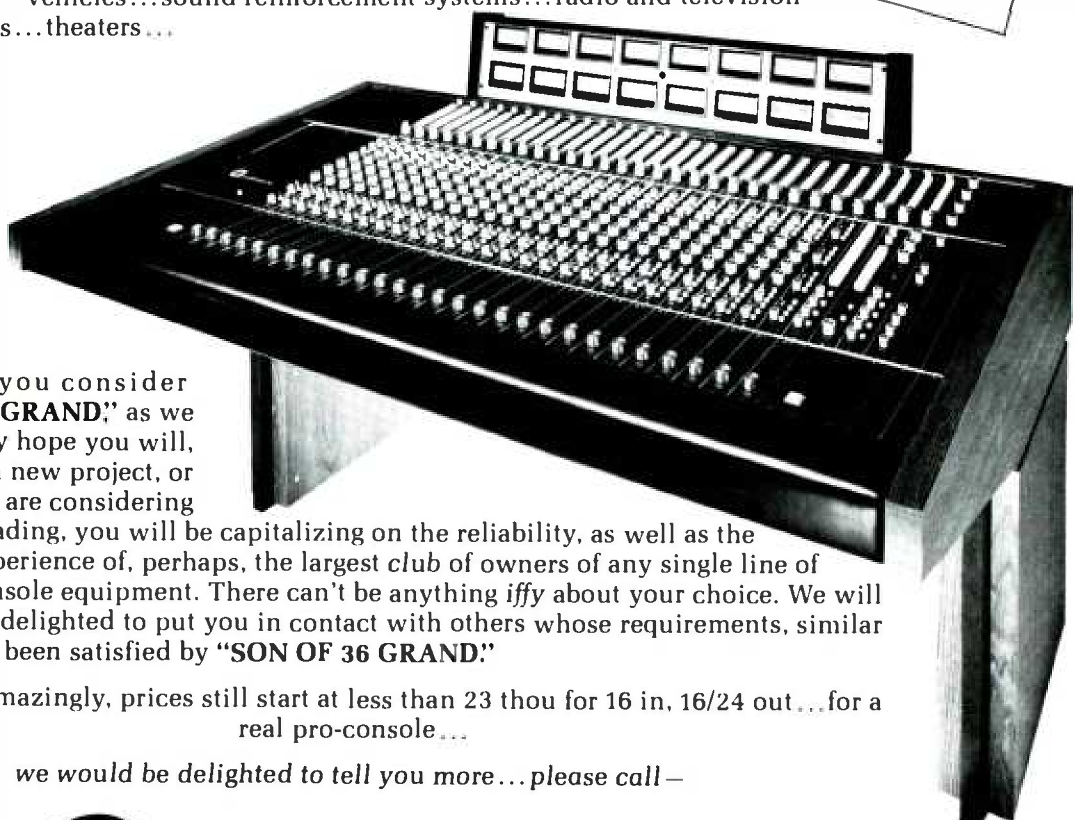
...both of these ads appeared a few short years ago...when we were delivering serial numbers 18 and 38. Now, we are delivering numbers 105 and 106...and the claims made in these ads have certainly borne the test of time! But, we would like to have you think of it this way: These deliveries are additional units of *improvement, update, experience, refinement,* and *change.* Yes, a lot of this is the result of updated component technology. (When we find a better way to make "SON" we immediately make those changes.) More important though, "SON OF 36 GRAND" is still very much the *product of perpetual* progress talked about in those early ads ...mostly because of the great wealth of feedback from so many of the installations in the U.S. and abroad where - "SONS" are continuing to *pay the rent,* day after day, night after night, after night...in recording studios...remote vehicles...sound reinforcement systems...radio and television studios...theaters...



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egories) are then reviewed and added to the Pre-Nominations List. In the event of cross-over records, a committee decides by majority vote where each belongs. This national meeting takes place because of the overlaps that may exist, for example, between jazz, rhythm & blues, rock, and even classical music. The voting in this stage was completed for this year's awards on November 1.

After the Pre-Nominations List has been completed, copies are mailed to each of the Academy's 4,000 active members. On the list are those entries in the General and Specialized categories (but not in the Craft categories). Here the members are allowed to vote for as many as five selections in *each* of the four General categories and in five of the eight Specialized fields. These ballots are then mailed to the independent accounting firm of Haskins & Sells in Los Angeles.

THE CRAFT NOMINATIONS

Entries in the Craft fields are *not* submitted to the general membership in the first round of voting. These nominations are handled by five-person committees in each category in each chapter. *These* are the people who vote on the various entries in their particular craft. For example, the arranging category is made up of arrangers in that chapter, the album packaging committee is made up of art directors, designers, photographers, and illustrators, and the engineering categories* are composed of the engineers in that particular chapter. *These are the people who sift through the hundreds of initial entries in their respective craft categories* to nominate the recordings he or she feels are most artistically creative and worthy of GRAMMY Award.

JUDGING CRITERIA FOR THE ENGINEERING CRAFT

According to the Academy, the basic criterion for the judging should be the finished product – the recording itself. Recognition of creativity in engineering is also encouraged. Other factors such as overdubbing, the number of tracks used or the location or mode of recording become secondary. Members are asked to consider the overall mix, the quality of the sound and the trueness of its reproduction – all the factors a recording engineer tries to achieve in the recording process. What it is said to come down to, is *what is in the grooves.*

The quality of the pressing should of itself have no bearing on a member's judgment of the engineering that went into the recording. The Academy states that engineers should not be jeopardized because of inept factory quality control.

Except where one or more tracks are specifically indicated, any evaluation of a long playing record should be made on the basis of the *total record* itself.

Engineering credits should *not* be the committees concern. Whether one or two or even six engineers were needed to create the recording is not important. What is important is *the finished product.* The National Trustees are the ones who determine who is entitled to receive A GRAMMY, and this they do after consultation with the recording companies and the recording's chief engineers.

Throughout both rounds of judging each judge should function solely on his own as a member of a national committee composed of members from all NARAS chapters. He votes as an individual representing his craft rather than his chapter.

THE JUDGING PROCEEDURE

Round I: After listening to all submitted records each judge lists his top 10 choices in order of preference on a ballot supplied by the Academy. Afterwards, the Academy in conjunction with Haskins & Sells, tabulates all ballots on the basis of 10 points for the first choice down to one point for the 10th choice. The Academy then supplies to each committee, alphabatized lists of the top 10 choices of the *other* committees, inter-mixed, so that no committee votes on its own first round of nominations, except for those selections that have *also* been nominated by another chapter.

Round II: After listening to those selections nominated by the other chapters, each judge lists in order of preference, his five top choices on the second round ballot supplied by the Academy. The five final nominations are then determined on a basis of five points for the first choice down to one point for fifth choice. The five top selections become the finalists in this category, to be voted on by the gereral membership.

All records are retained after the first round, even those *not* selected by the local committee, in order to have them on hand for possible use in the second round of voting.

Some committees have found that the judging can become more objective if the identities of the artist, engineer, and record label are not divulged prior to the playing of an entry.

We now proceed into industry member comments. Responding are:

Fred Catero, of the Automatt, San Francisco. Producer, engineer, and President of the San Francisco chapter of NARAS. Grammy nominee in 1969.

Jay Cooper, Los Angeles entertainment attorney, NARAS National President.

Ray Moore, Classical recording engineer at Columbia Records, New York City. Grammy winner: 1970, 1973, 1975, 1976.

George Simon, Special consultant to NARAS, New York City. Author, producer.

*Classical and Non-Classical

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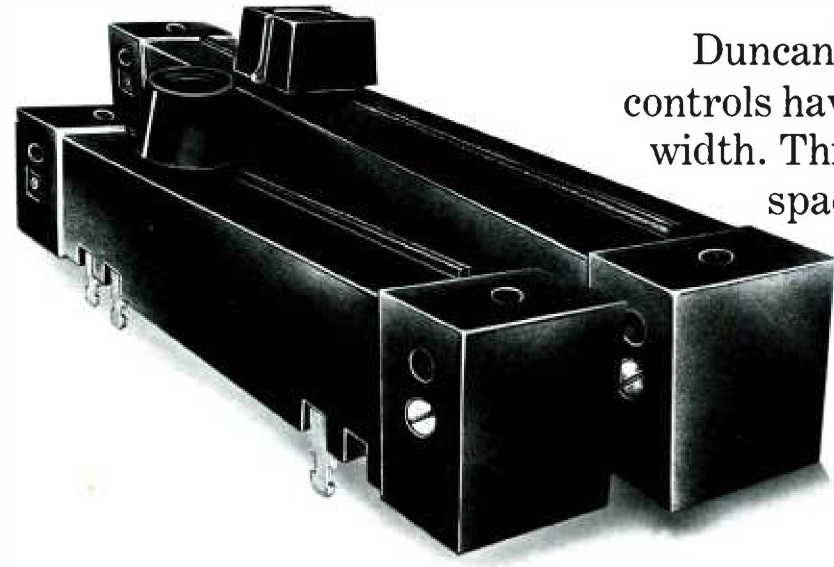

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Questions posed to the record companies: Of what importance are the GRAMMYS to you and what sort of support do you give your artists once they have been nominated?

Responding are: A&M Records, Hollywood; Capitol Records, Hollywood; and Warner Brothers Records, Burbank.

BOB MERLIS: Publicity Director, Warner Brothers Records – Burbank

We are nothing but exultant about the number of nominations we have received this year as opposed to previous years. As far as support we give on behalf of the artists once they're nominated, there's no real campaigning. We're not going out and breaking arms. If they win, it'll be on their own merits. I don't think you'll see any label-campaigning like the Academy Awards where page upon page of ads in *Variety* and *Reporter* occurs.

I think the best thing to happen to George Benson is he's got a hot new abum at the same time as he's nominated from his BREEZIN' album, which keeps his name in the news. So he's doing his best kind of electioneering, which is the musical kind.

The award is also important as far as album sales – there's a little more mileage because of the exposure the award affords the artist – both the work involved and future material. A case in point would be (Simon & Garfunkel's) BRIDGE OVER TROUBLED WATER.

A spokesperson from Capitol Records Hollywood:

Obviously, they're important because they increase sales. As far as support, we send out press releases to the trade if they're nominated. If they win, we put special stickers on the albums saying it's a GRAMMY winner and send out a news release to that effect to bring it to people's attention.

We get behind our artists if they're nominated, but it's not going to hurt if they're not. If an artist has talent, he'll make it anyway – a GRAMMY won't make or break him.

Fred Catero: In a way, I disagree with the (voting) process because a lot of the times the general membership is not sensitive to the fine points of the particular craft. They just vote on the "vibes" they get from it or the popularity it has achieved without any actual knowledge of what goes into creating the ultimate winner. But at least the one that wins is one of the five finalists that the craft committee has chosen. It's not one of these things that's chosen out of the blue.

What engineering criteria do you use in evaluating GRAMMY nominations?

Ray Moore: How the thing was recorded – 'live' vs. an overdub situation. It's not the last word. Over the last three years I may give a little extra "something" to a performance recorded that way – as opposed to an overdub situation. That's only a frame of reference. Sometimes engineers will give themselves away with some splices, or a change in ambience and if it gets to be a little too quiet in a 'live' recording, you become skeptical. Given enough time, you can make things become what they're not – obviously – but being in the business long enough, I know what can be done and what can't be done. At the Academy, they do not like to make any awareness of that – if it's in the grooves, it's in the grooves – which *I* think is an old wives' tale. I think that was fine 10 years ago, but knowing what you can do now, you have to think about *how* it was done. That's my opinion, not theirs.

Certainly if the album itself has got a lot of different *kinds* of ensemble – whether overdub or 'live' – and they've handled it well in my opinion, I add a little extra weight to that, too. When there are four instruments, and I have to compare that to where I have a lot more, I sort of down-play the four instruments* unless they're *extremely* good. I figure it takes a little more challenge to be able to handle more than four instruments. Like ECM (jazz record label) has had some things over the past three years which I thought were fantastic and they never got past the original nomination.

Another subjective point: I try to temper it with the fact of how long the album is. If I had two selections, and I liked them equally well, I try to look for something that I can really find to cut the other one down or go with the one I think is better. I also try to take the length of a side into consideration, too. The *loudness* of a record should not play that much of a part. So, if I'm overwhelmed by the *sound*, I try to analyze if I *like* the sound.

I'm not that country oriented or "Hollywood-country", but last year I chose RHINESTONE COWBOY (Glen Campbell) because it was cleanly done. For what the engineer was doing, the instrumentation he worked with, and everything else considered, it was a damn good job. I thought it was clean, and it was a lot of instruments, not just a few – and that went nowhere. So I *can* be objective. There was also a Stan Kenton thing that I was very partial to, and that didn't even fit in the top 10. So it's things like that that bug me after awhile.

Sometimes I try to *avoid* knowing who the engineer is, but I've noticed lately I have a tendency to look for these people I think highly of to see what they're now doing.

** Author's Note: Does this explain the lack of ECM nominations over the years?*

We had around 268 nominations (in engineering) this year including albums, singles, and LP tracks. This is what we had to go through.

Fred Catero: I'm a "purist" and in a way I have to fight that (being a purist). If I buy a record of a 'live' concert, I want everything in there: mistakes, distortion, audience noises – everything. If that's the way it happened, that's the way I want to hear it. When I first started evaluating records, if I heard a harpsichord very loud above a rock ensemble, instantly it offended me because I felt it was so unnatural. There's no such thing as a harpsichord being able to cut over a set of drums or electric guitar amp. But then I re-evaluated. I said, "They're not *trying* to re-create a true sound. They are using the medium to do what the ear would never hear in reality." Based on that, in the past I have voted for records which were not well-engineered at all, but which contained so much engineering. Last year I voted for a two-record set which sounded bad from a *sound* point of view, but I voted on it because all the other albums were a one (mike) set-up thing. But this record was a *mammoth* undertaking, every cut was a different piece of engineering. You *know* these mothers must have spent a year in the studio – every cut had engineering contributions. The music and sound were lousy, but so varied and had so much work in them. I went against my purist feeling in that case. So I go on engineering involvement, too.

George Simon: We vote on what is on the record – the finished product, but some people may be too concerned with the creative *production* instead of *reproduction*. These are two considerations. I think it's a great engineering feat to be able to reproduce something beautifully.

What equipment does your area use for evaluation?

Jay Cooper: Some people use their personal equipment. A few of the chapters have a listening session that people are welcome to come to, and that's late in the game. But there's no control as to what people listen *on* or listen *to*. It's up to them to determine on their personal equipment or equipment at the studio or whatever – there's no way to do it. At a recording studio "listening", people come in to catch up on things they may have forgotten over the year. But they do not take a vote there and then walk out.

Moore: I've been involved three separate years and it's been done three different ways. I don't think any one of the ways is better than anything else. The first year we went to Electric Lady (NYC), then to A&R (NYC). The second time was at the NARAS office in New York with KLH 33

speakers and a Garrard turntable. At other times I've evaluated at work or at home. But I don't think it's all that critical to have the most outlandish equipment — as long as you maintain consistency from disc-to-disc.

I remember one year being knocked out by a particular recording and then taking it to another place for a listen, and feeling, "Why the hell did I think it was so great?", and when I checked into it, I found out the copy was from a different batch — a different stamper copy. So it was an influencing factor I had to take into consideration.

Catero: We may rent, from some hi-fi shop, some good stereo gear. But mostly each member will take the discs home for evaluation on his own system. The records are not special copies, in fact, some of them are even DJ copies with the holes punched in the label.* An interesting note here is that the packaging committee (liner notes, photos, etc.) get all the jackets without the records and now they'll start getting the records too in order to see if the package is *suitable* for the records — a valid point.

Each committee member is supposed to go through every record and every note for proper evaluation. Let me say, I cannot do this. It turns out that most of the people that are super-qualified are also super-successful and therefore very busy. They can't take three months to sit and listen to 300 records. So we have to spot-check them. You *do* acquire a skill after awhile and you're able to tell after a minute or so, if you should continue listening. But in all honesty, I listen to *every* one for at least a minute and weed them out from there down to 100 or 150 and concentrate on those for repeated listenings.

Is there anything a Producer/Engineer can do to strive for a GRAMMY award? Any special "formula"?

Catero: Yes. And this is the worst thing in the world for me to say, but it's the way I feel and I think it's true, and part of the way things are set up. We're trying to change it but there's nothing we can do. The only way I honestly feel an Engineer/Producer can win an award in the GRAMMY system, the way it is set up today, is to be lucky enough to have himself a hit, to work for a top label, to work for a top *act*, i.e. Stevie Wonder, Elton John, etc., then he'll win. Otherwise the *most* he can do is become nominated — which is not the worst thing in the world.

Look back on almost every nomination; almost every winner has been on a major label, or a record that got super exposure, or it's a well-known artist. And this is very common — it's not anything against the GRAMMYS because the Oscars work in the same way, the Obies — any of the other awards shows where the mass membership votes. Nobody is going to vote for something they've never heard, and it's unfortunate because most of the membership doesn't have the time or the inclination or whatever to hear these records and that leaves the radio for exposure. If it's had enough exposure when the ballots come up and the membership sees the five finalists, they say, "Oh, yes, I remember that was a big record last year — that was outta sight. Let's vote for that." We've had meeting upon meeting to try to change it to make sure an artist wins because they deserve it, and not because they're popular.

Simon: In a technical sense, he would have to know what "sound" appeals to each member — but I don't see how that could be done.

Cooper: I feel there's nothing you can do to become nominated. You don't know what people are going to vote for. You get some big surprises. For instance this year George Benson has become nominated — that to me is a surprise — not because he's not talented or great, but because he doesn't have the "popular appeal" that some of these other people have like Chicago, Paul Simon, etc., — and yet he's right up there among them.

Moore: My previous comments on the final product would hold. If you've got a group that over-dubs a lot and has different types of instruments I already said *that* was quite a challenge for a Producer — to get the sound of the instruments, regardless of how he does it — and then enmesh them into the others to get an overall "feel".

What about record company support/influence to promote a nominee winning?

Simon: I did a survey in New York and found out that no record company had more than 3% of the membership. If it's "support" in a bad sense, I've talked to record company heads to make sure it gets discouraged.

Cooper: That question has been raised from time-to-time. I don't see how it's possible. Here's the structure: There's 4,000 members of the Academy. Nobody has the (membership) list except the (L.A.) office itself. That list is never given out for any purpose whatsoever. The people (membership) are spread throughout the entire country, they're not necessarily in any one place or any one company's headquarters. No company knows (their own membership) unless they go around within the company to each member. The ballots are sent to the people's homes. There is no particular way that they can mail (promotional)

** Seven copies each (one for each chapter) of every LP, single, liner notes, and record covers are secured.*

literature to people, there's no way they can find out who the list is or what they are. No one company has enough members within its own company to sway the ballots in any one way or another. So it's difficult to see how it could be done.

There's just no way for the record companies to know who they are to reach them. There are members in Nashville, Memphis, all over Georgia, San Francisco, Seattle, Portland, Minneapolis—all over the country. How are they going to reach all these people?

Catero: I worked for CBS for 9 years, and they *did* come around and say, "I hope you're going to vote for some of our stuff." Nobody ever told me, "Look, this is the one we're pushing—you better vote for it." Nobody ever told me, "Hey, Fred, remember which side your bread is buttered on. We expect you to vote for this." But they did sort of *imply*, "Don't forget—take care of your own." This is 8 or 9 years ago remember. And I never did vote that way. But as far as the record companies exerting power that way, I don't think so. I really don't think so. I voted for Columbia product only if I believed in it.

I think the reason it appears the big labels have power is because so many of the records that have been nominated are on the major four labels. They have more exposure — they get stuff played on the radio and have tremendous publicity campaigns. That's why they're big and I think they exert more power that way. By exposing their records so much, people know the records are out.* If a little record company had the promotion dollars and could get one of their records played on a radio station day and night — and make it a turntable hit — it would stand a good chance of becoming a nominee.

Moore: I have not been aware of it if it does occur. Certainly no one has come to me saying, "You'd really better vote for this one because it's ours." I think the people here know me well enough, that they know if they did that, they'd have a good chance of not getting a vote.

As far as Columbia's concerned, they *do* make a list over the years, mainly as a guide to see *what* we've had out this year. But nobody is going to tell me to vote for any particular thing — because I know I'd be annoyed enough to complain about it.

**Author's note: Based on Fred's comments, I could almost pick the engineering winner for the year.*

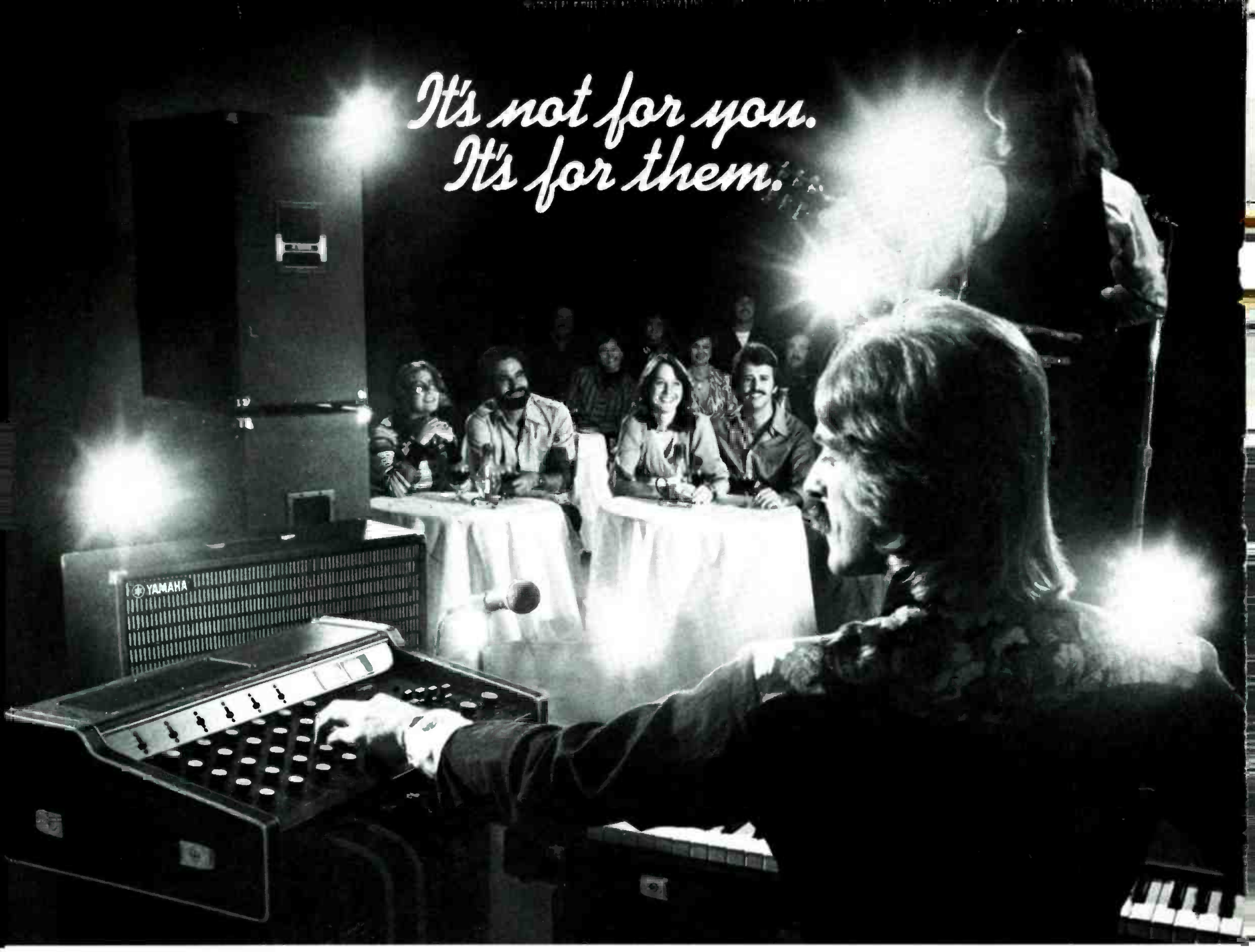
Fred Catero on the GRAMMYS in general:

NARAS is set up to acknowledge and present awards to people who have achieved *excellence and meaningful contribution* in their particular field of endeavor in the recording industry. We are not in business, if you want to call it business, to promote sales — though that's what the record company gets a chance to do when it's over. We are not there to stroke our friends. We are not there to do anything but show appreciation and acknowledgment to those who have done this.

To my knowledge, no one, since I've been in NARAS, has come to me and said, "Look, can you convince your committee to vote this way?" Never ever, ever, and I'm a life member — I've been in NARAS for some 17 years. However, when I ask people how they voted, they say, "Oh, yeah, I voted for my friends — for the records I know."

A lot of people still feel it's a bunch of bullshit — that it's just to perpetuate the record companies and serve self-interests, and the people vote only for the stuff that *they* engineer. I *rarely* do (vote for himself). For me it's a priviledge — that my opinion weighs.

Anybody who has been involved in the manufacture of a record — and thinks that NARAS is just bull — *join*, cast your vote and get involved and we can fight those selfish people who vote for their friends and themselves. Fight for the artform. We get enough people fighting for principle, I guarantee you, the major record companies won't stand a chance with *all* their exposure. A person who really cares will find those records that are great. □□□



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CONTROL ROOM DESIGN for the SMALL STUDIO

BY: WOODY SMITH & GALEN CAROL

We at Abadon/Sun decided to write this article because we felt a large number of small studio owners required more information to properly design and build their first studio. What we've tried to do is to simplify things as much as possible and provide the builder with the design fundamentals. Since most small control rooms are rectangular in shape we'll detail the design considerations for a room of that shape. Our primary goal is to eliminate severe resonant modes in the room and establish the low-frequency response of the room through the proper selection of room dimensions.

RESONANT FREQUENCIES

A room with parallel surfaces and no acoustic treatment will exhibit resonant modes between opposite surfaces. That is, we will have resonances created between the two side walls, the front and rear walls and the floor and ceiling surfaces. When selecting the various room dimensions the objective is to avoid common resonances between any of the room modes to avoid build-up of sound at the resonant frequencies. If a common resonance is present, it will probably result in an increase in volume of that one frequency producing a very boomy sound (for low-frequency resonances). Since the low frequencies are the main souce of difficulty in room design (and are the hardest to correct) this article will concentrate on them.

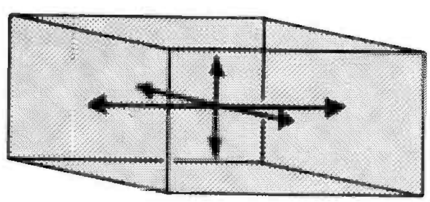

Figure 1. ROOM RESONANT MODES

The frequencies at which resonance occurs are determined by the distance between the two walls under consideration. The formula for the resonant frequencies is:

$$f_n = \frac{565(n)}{d}, \quad n = (1, 2, 3, \ldots)$$

where d is the room dimension (in feet) and f_n is the resonant frequency (in Hz. or cycles/second). The resonances will occur at multiples of the fundamental frequency f(1). For that reason we use the multiplier (n). For example, two walls separated by 10 feet will produce resonances at 56.5 Hz., 113.0 Hz., 169.5 Hz., etc. By this method the resonances occuring in the room can be readily calculated. Considering a room 10' x 15' x 20':

d =	10'	15'	20'
f(1)	56.5	37.6	28.3
f(2)	113.0	75.3	56.5
f(3)	169.5	113.0	84.7
f(4)	---	---	113.0

Our conclusion from these calculations is that 10' x 15' x 20' is a very bad choice of room dimensions. The reason is that we have a resonant frequency (113 Hz.) common to all three room dimensions. This would produce a very bad room resonance everytime we encountered a 113 Hz. signal which would probably occur fairly frequently.

Changing our choice of dimensions to 10' x 14' x 22' and applying the same equations we find these resonant frequencies:

d =	10'	14'	22'
f(1)	56.5	40.4	25.6
f(2)	113.0	80.7	51.4
f(3)	169.5	121.0	77.0
f(4)	---	161.0	102.7
f(5)	---	---	128.5

As can be seen there is no common resonant point between the dimensions selected this time. One limitation we will impose on your choice of room dimensions is that the ratio of dimensions should lie within the limits of the graph given in Fig. 2. For example, a room measuring 10' x 11' x 18' would have a ratio of dimensions of 1:1.1:1.8 which would not be acceptable. A room with dimension ratio of 1:1.4:2.2 is within the acceptable range.

[NOTE: Some of the ratios within the limits of the graph may produce undesirable additive resonances. For this reason you must check them thoroughly by calculating the resonant frequencies before committing yourself to a selection.]

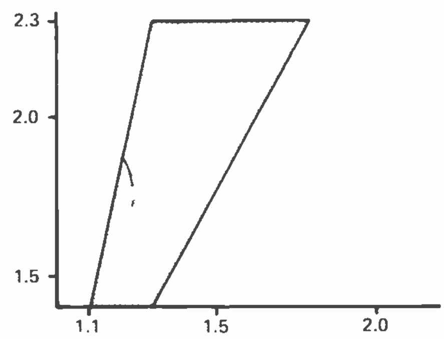

Figure 2. RANGE OF ACCEPTABLE RATIOS

Another calculation which will enter into your choice for dimensions is the diagonal dimension of the room. For a room to reproduce low-frequencies well, there has to be a sufficiently long dimension to allow the low frequency waves to propagate themselves. The equations are:

$$f_0 = \frac{565}{d_d}$$, where d_d is the room diagonal (in feet).

The room diagonal can be found from the equation:

$$d_d = \sqrt{(\text{length})^2 + (\text{width})^2 + (\text{height})^2}$$

From these equations you should be able to determine the lowest frequency that can be propagated in the room. The diagonal lengths required for some sample frequencies to propagate are given as:

f_0 =		d_d =	
	30 Hz.		18.8 Ft.
	35		16.1
	40		14.1
	45		12.5

Looking at another example, a room 9' x 11' x 14' is suggested. The ratio of dimensions is 1:1.2:1.6 which is acceptable according to our graph of acceptable ratios. The lowest frequency will be:

$$d_d = \sqrt{81 + 121 + 196} = 19.9 \text{ ft.}$$

$$f_0 = \frac{565}{d_d} = 28 \text{ Hz.}$$

for additional information circle number 36

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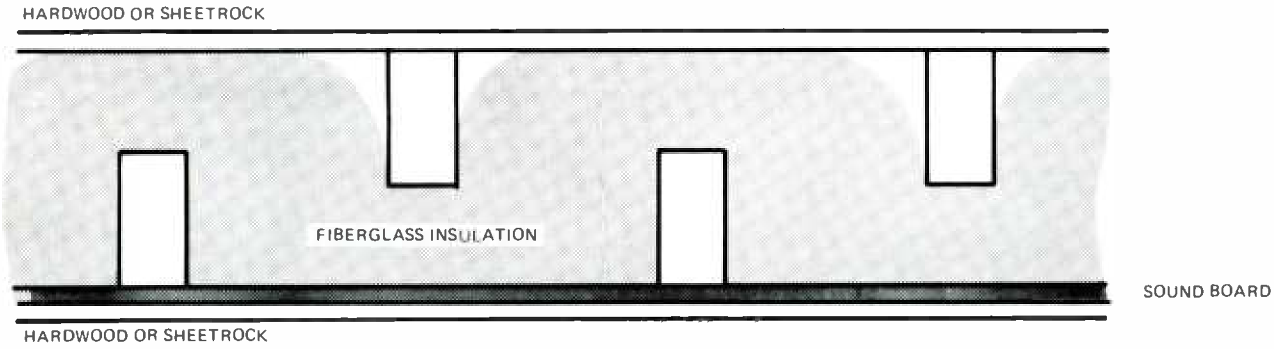


Figure 3. DOUBLE WALL CONSTRUCTION DETAIL

Room resonant frequencies are:

f(n)	Length:14'	Width:11'	Height:9'
f(1)	40.4	51.4	62.7
f(2)	80.7	102.7	125.5
f(3)	121.1	154.1	188.3
f(4)	161.4	205.5	251.1
f(5)	201.8	256.8	313.8

From these calculations we can see that this is an acceptable choice of dimensions.

CONSTRUCTION DETAILS

Once the room dimensions have been selected we can begin looking into the finishing of the walls, ceiling and floor. Sound travels by both acoustic and mechanical means. That is, it not only travels through the air but also through solid objects. So, to minimize the leakage of sound between rooms we must provide both acoustic and mechanical isolation. Maximum mechnical isolation is best achieved when the inner and outer walls are independent; as in the use of double-wall construction. This type of construction uses separate, staggered studs for the inner and outer wall surfaces with fiberglass insulation in between. (Fig. 3.) Also, care should be taken to insure that no holes exist between the two rooms, such as through adjacent electrical outlets or mic lines run through the walls, since an air-tight seal between the rooms is required for maximum isolation from air-borne sound.

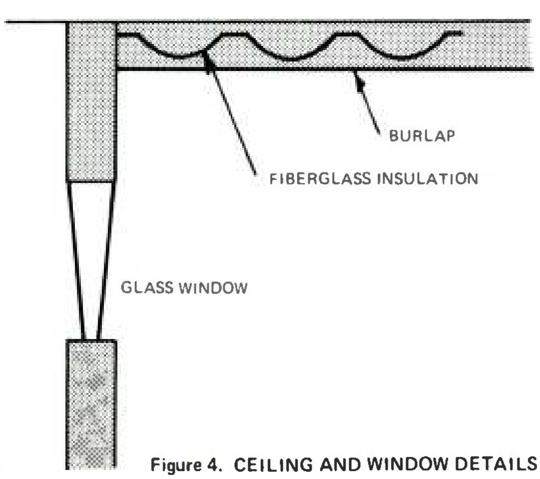


Figure 4. CEILING AND WINDOW DETAILS

Walls other than those between the studio and control room may be of single-wall construction but should be packed with insulation.

As to wall surfaces, the heavier a wall covering is, the lower its' resonant frequency. Normally we'll want the resonant frequency to be as low as possible. Sheetrock will resonate at a lower frequency than thin paneling because of its increased weight. We can also reduce sound transmission and lower the resonant frequency of a wall by the use of a highly damped material such as sound attenuation board. For example, by attaching thin wood paneling to sound attenuation board added weight and thus, damping will be achieved. This provides the twin advantages of an attractive surface that exhibits good acoustic characteristics.

Typically, windows should be of double-pane construction with the two panes at angles to each other to eliminate internal resonances. All seals around the window should, of course, be air-tight. (Fig. 4.)

Experiments have led to the belief that the ceiling should be padded with 4" thick fiberglass insulation folded slightly to produce a corrugated surface. This should then be covered with burlap or other similar material. The effect is to deaden the ceiling and reduce floor-to-ceiling resonances.

To retain a proper stereo image in the control room the need to have a room symmetric about its center (front to rear) axis has been well proven. That is, the side walls should be mirror images of each other. If they are not, the acoustic characteristics of the two walls may be different. Let's say one wall has a hard flat surface and the other wall is made very dead. The hard surfaced wall would produce lots of reflections back to the console making that side seem both bright and loud. The deadened wall would do just the opposite by absorbing the sound hitting it, resulting in a dull sound at reduced level. It's obvious from this that the hard surfaced wall and the dead wall both have shortcomings. We prefer to use side walls which are irregular in shape and texture to minimize resonances between the two walls and also to break up any reflections from the walls. We will try to avoid using deadening materials on the two walls because we do want them to retain a bright sound. This type of side wall construction can be seen in Fig. 4, for which we've used cedar shingles to form a pattern on the wall. The wall also has a slight curvature to it to further reduce resonances between the two side walls. The surface formed by the shingles is acoustically hard but very irregular so that reflections produced by the wall are diffused. This will further enhance the stereo image produced by the monitors through de-centralization of the sound source.

An additional prerequisite for proper design is that the

console must be centered between the walls, and the monitors should be placed at equal distances from the side walls.

The criteria for monitor placement has been outlined in Fig. 5. From your position while mixing, the angle between the monitors shouldn't exceed 90° and their axes should cross about one foot in front of you. If possible you should be about two-thirds of the distance back from the monitors to the rear wall.

The choice of monitor loudspeakers is practically unlimited. The monitors ultimately selected should be as reliable as possible and be capable of high sound pressure levels. The reasoning behind this is that in mastering use a monitor is fed a fairly high continuous level of uncompressed sound. Allowing 10 dB of headroom for peaks in the music, a 95 dB SPL signal may reach 105 dB SPL on transients. This is not very dramatic to the ear, but is a drastic change to the speaker. 10 dB translates to ten times power, so if we were using 20 watts continuous power for 95 dB SPL the speakers would require 200 watts peak input for 105 dB SPL. A speaker requiring only 8 watts for the same 95 dB SPL would require 80 watts peak. It is advisable to allow at least 10 dB above your continuous monitor level for peaks and you should select a monitor capable of handling repeated inputs at 10 dB above that level without failure or distortion. Obviously, the power amplifier should be chosen after you choose the speaker unless you plan on buying a big amplifier. We normally install Crown D-150A's, with JBL 4311's or Klipsch monitors and Crown DC-300A's with everything else.

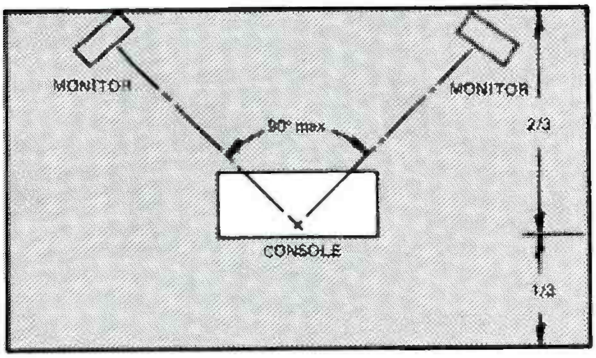


Figure 5. MONITOR/CONSOLE PLACEMENT

Placement of the monitor speaker in a corner of the room will reinforce the bass output of the monitor as will placing it near the floor or ceiling of the room. Placement in the center of a wall will provide the least amount of reinforcement. If you want to demonstrate this effect simply place a speaker on the floor, run pink noise (or FM inter-station noise) through the speaker and slowly lift it off the floor. You will notice a marked change in the sound of the speaker.

We recommend that you allow some time to try out several different monitor placements in your control room before committing to a certain spot. However, the relationship between monitor and console placement as outlined in Fig. 5 should still apply regardless of your final choice for the monitor placement.

At Abadon/Sun we consider several things to be important in the early stages of design concerning the asthetics of the control room. We center our designs on providing the mixing engineer a comfortable place to work. This includes providing comfortable surroundings and eliminating unnecessary distractions. For that reason we like to place all the recorders and accessories behind the mixer, or to his side so that when he is mixing there are just the artists performing and the console readily visible to him. This excludes, of course, such things as limiters which must be adjacent to the console but should be below eye level. Proper lighting of the control room is also

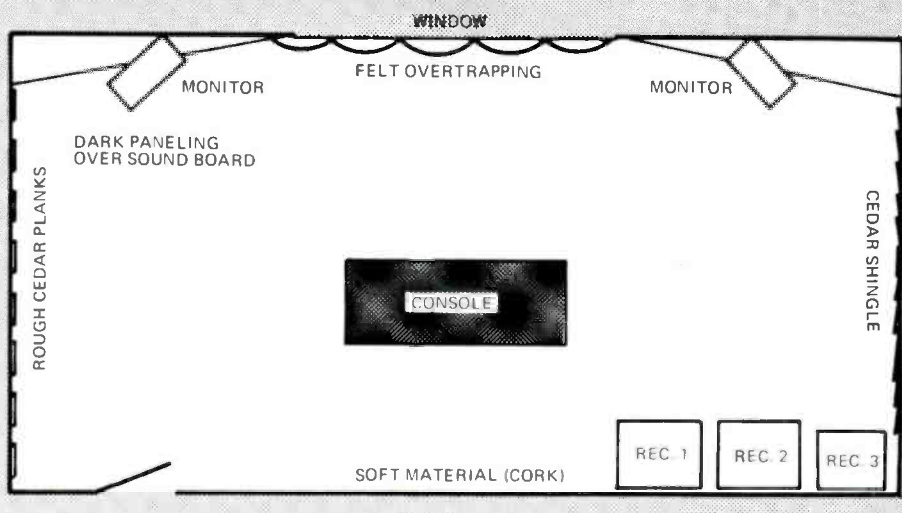
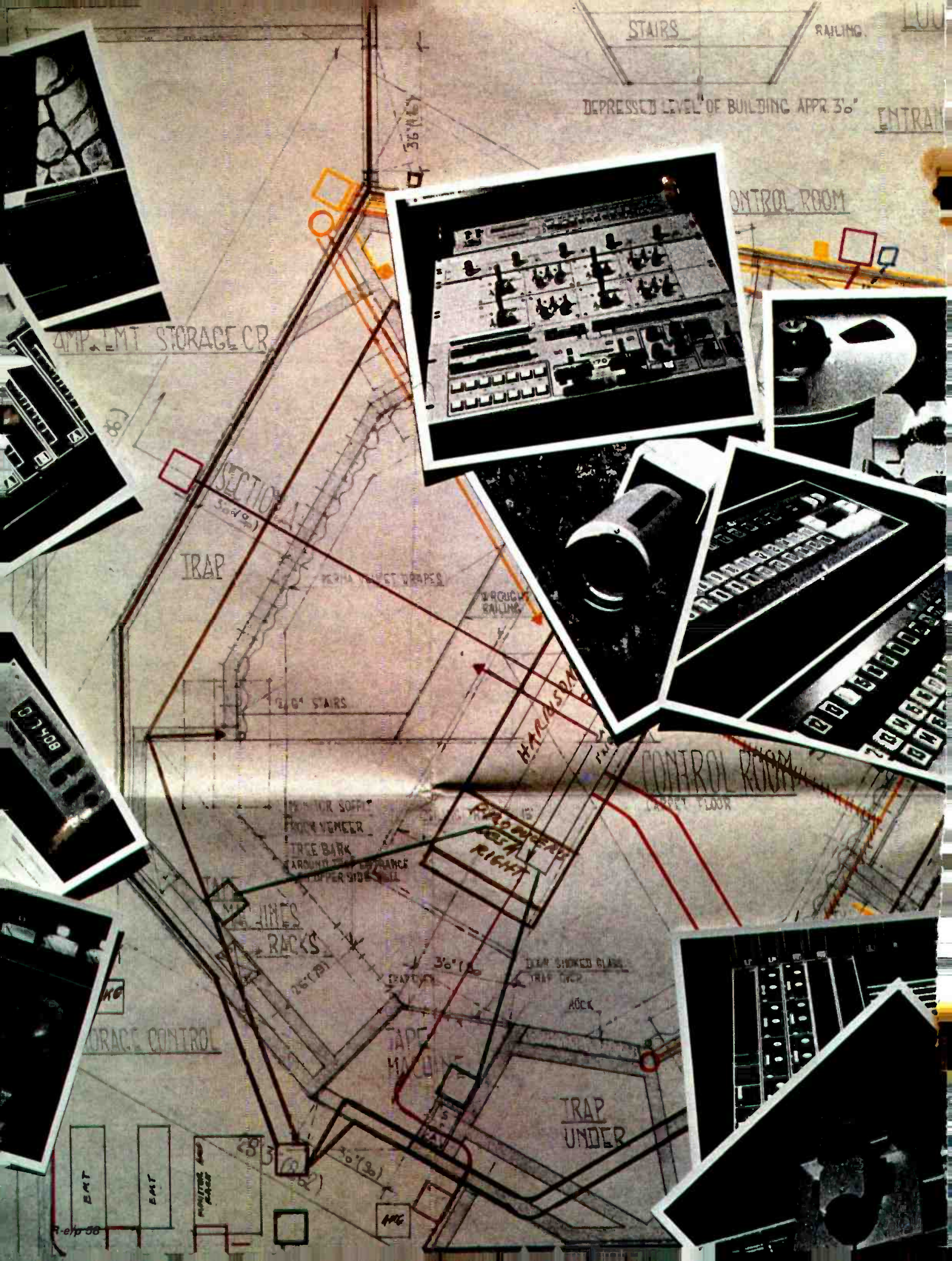


Figure 6. CONTROL ROOM OVERVIEW

important. We normally light the area of the console with multiple low intensity flood lights on overhead tracks and soft light the recorders with additional track lighting. This eliminates quite a bit of the distraction that arises from all the moving meters and lights on equipment other than the console. The point behind all this is that the mixing engineer's attention is concentrated on the work at hand which in turn leads to a better product. As a further extension of the same ideas, we recommend the use of dark colors and wood surfaces in the control room. For example, try using sections of dark walnut paneling and sections of padded felt in dark blue, black or The effect is very pleasing.

You should now have all the basics required to construct a good control room without a lot of guess-work. The 9' x 11' x 14' room discussed earlier has been recently built and seems to work very well.

There is another part to this article concerning the design and construction of the studio which will become available shortly. We would like to invite your comments about this article. Please remember that it is based on a small control room situation and as such is merely one design approach, however, it is a proven approach which provides excellent results in a minimum of space. Please direct your responses to:
Abadon/Sun, Inc., P. O. Box 6520, San Antonio, TX 78209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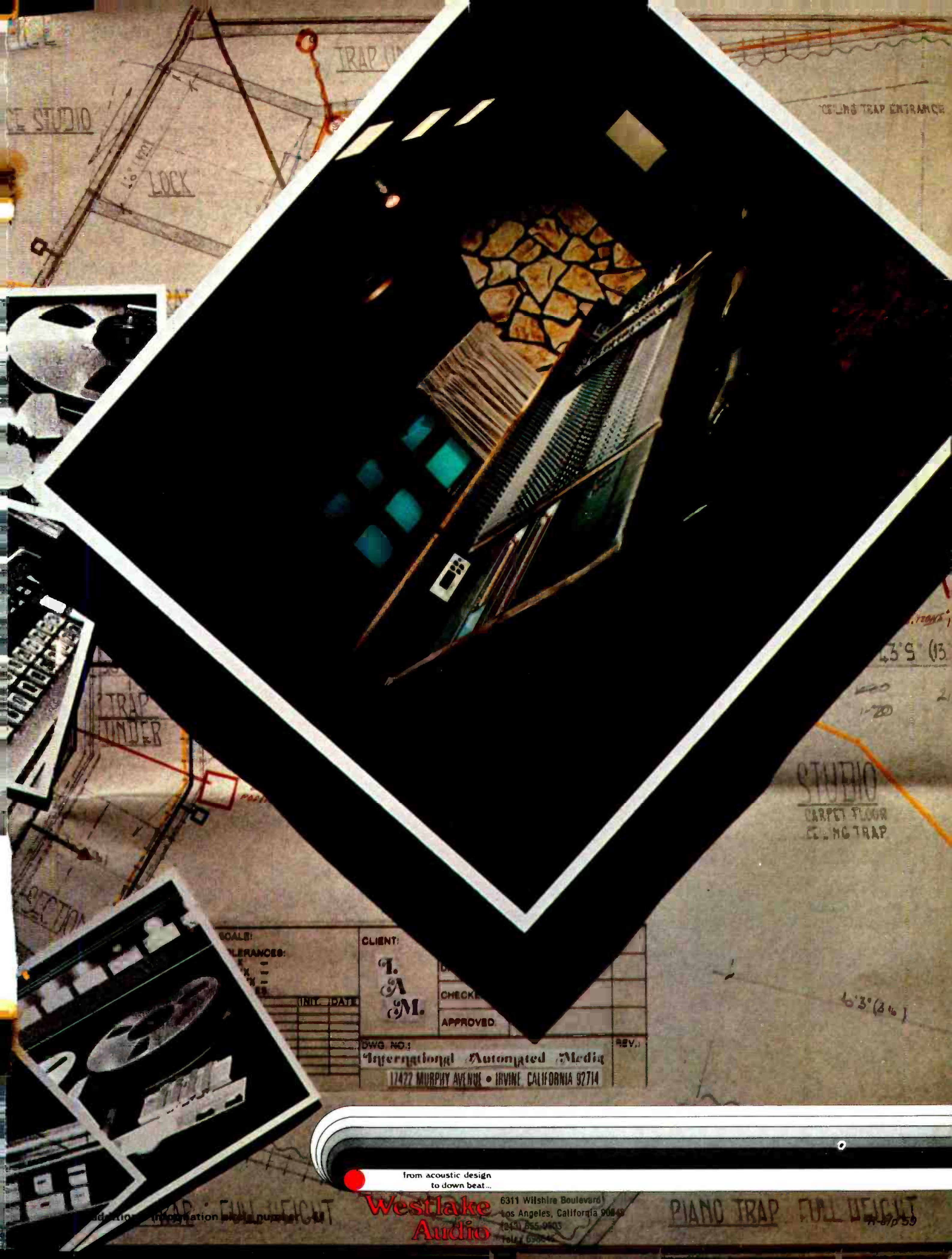

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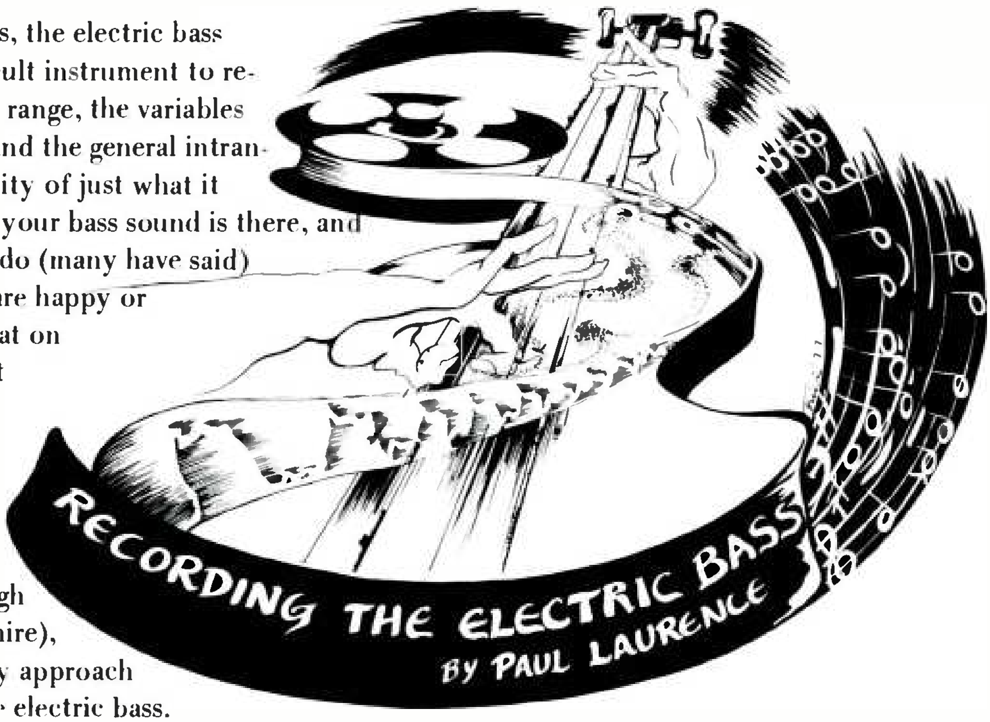
Apart from drums, the electric bass is perhaps the most difficult instrument to record, because of its tonal range, the variables in the particular player, and the general intransigence and unpredictability of just what it sounds like. Some nights your bass sound is there, and others it's not, having to do (many have said) with whether the spirits are happy or not. Most would agree that on those nights when it's not there, there isn't a whole lot you can do about it. Cherokee's Dee Robb, Davlen's Leonard Kovner, and independent engineer Ron Malo (though he works a lot at Devonshire), talked about the way they approach and execute recording the electric bass.



DEE ROBB

Dee Robb was born in Ann Arbor, Michigan in 1943. While still in his teens, he, two brothers, and a cousin formed a band, known unsurprisingly as the Robbs. At age 17 he had his first release, a single on Chess called "The Prom", penned by Del Shannon, which featured Dee as the artist/vocalist. They were Del's band at the time (Dee played guitar), and would back a number of the day's top pop artists, like Bobby Vinton, Brian Hyland, and the then-reigning Gene Pitney.

After four or five singles for Chess, Dee and the other Robbs went to RCA. And, amazingly, Dee had been able to grab two years of college at Marquette University in Milwaukee. For two years during the middle '60's, the Robbs were regulars on Dick Clark's "Where the Action Is" TV show, and had a few moderately successful records on Mercury.

Dee "stuck" (as they say) as a staff producer for Mercury, and later formed his own independent production company, producing records mostly for the Atlantic label, but also United Artists and Columbia. Dee laughs when he remembers these days and these records. "They were mostly silly things, like Wolfman Jack . . .".

From there it was into engineering, recording artists like Little Richard, Rick Nelson, Dr. John, and Richie Havens. He had co-founded Cherokee Recording Studio, which was located originally on a ranch in Chatsworth, California. However, there were some problems with the location (not the least of which being unamenable neighbors) and so they had to move. In January of 1976, Cherokee opened its doors at its present location, at 751 N. Fairfax, in West Hollywood. Since that time, Cherokee has become one of the really "happening" studios, and Robb one of the hottest engineers in town, working with artists like Rod Stewart, Art Garfunkel, Cat Stevens, Donovan, the Manhattan Transfer, and Joan Baez.

. . . continued on page 62

LEONARD KOVNER

Leonard Kovner is a fairly remarkable guy. He started playing guitar at 7, studying under four of Los Angeles' finest teachers in Stanley Black, Jay Lacey, Ted Green, and the venerable Ernie Ball himself. By 12 he was immersed in creative home recording, having fashioned a primitive home studio where he taped space noises via a microwave receiver, original comedy monologues, and all manner of sounds and sound effects. And none of this prevented him from seeing "every science fiction movie ever made" and amassing a formidable collection of monster magazines. During this time he was doing "a great deal of listening" as well, to artists like the Everly Brothers, Duane Eddy, Phil Spector, Ricky Nelson, Del Shannon, Dick Dale ("to listen to him have his Dual Showman on 10 with his reverb unit was like really a . . . you know, moving experience"), and even country artists like Roy Clark and Chet Atkins.

By 1964 he had made it to disc. Only 14 at the time, his mother drove him to the studio, where he and his band cut a single for Capitol Records, made possible, Leonard thinks, because somebody big at Capitol was one of the band member's Little League coach. Three years later, he began playing sessions as a guitarist. And shortly after that, he started engineering, doing much work at American Recording with Richie Podolor (whom he credits as a major influence recording technique-wise), and later producing.

In the fall of 1972, frustrated with working in not-so-good studios and on product that was going nowhere, he decided to build his own studio. He spent a full year in preparation, amassing as much information on the subject as he could. He took a tour of the United States and Europe to see personally all the great studios. He didn't just "visit" – he took along some of his own tapes . . . booked time . . . ran some mixes . . . and took it all in at over 100 different studios, some being Sunset

. . . continued on page 66

RON MALO

Ron Malo, like many, became an engineer "by accident", through answering an ad for a job with the Northwest Sound Company in Detroit that had a position for a "radio repairman, no experience necessary". While still in high school, he began recording band concerts and making – literally – his own records. He graduated to building quite a few little recording operations in the Detroit area, and at the same time doing remote radio broadcasts for station WJLB. During this period he recorded the Royal Tones and Johnny and the Hurricanes, two of the earliest-ever rock bands.

He met Motown founding father Berry Gordy at the Flame Show Bar, a "black and tan club", ultimately building the original Hitsville for him at the site of a former photo studio on West Grand Avenue. Around Christmas of 1959, he left Detroit to become head engineer at Chess Records in Chicago at a salary of $150 per week.

Malo remained at Chess for over a decade, during which he had at least something to do with almost every record released by the label and personally engineered a huge disproportion of the classic first-generation rhythm & blues records. Some of the artists he worked with: Howlin' Wolf, Bo Diddley, Muddy Waters, Chuck Berry, Etta James, Little Walter, Willie Dixon, one of the Sonny Boy Williamsons, Otis Rush, Buddy Guy, and Jimmy Witherspoon.

These records were the very foundation of the British R&B boom in the 60's, and so when those artists came to America, they wanted nothing more than to come to Ter-Mar (the Chess studio's real name) and work with the "legendary blues engineer" himself. The Yardbirds cut one of their early albums with him, and the Rolling Stones cut 21 tracks at Chess in just two sessions, appearing on their second through fifth U.S. LPs. They were just knocked out with Ron, the studio, the equipment, the sound – everything.

. . . continued on page 70

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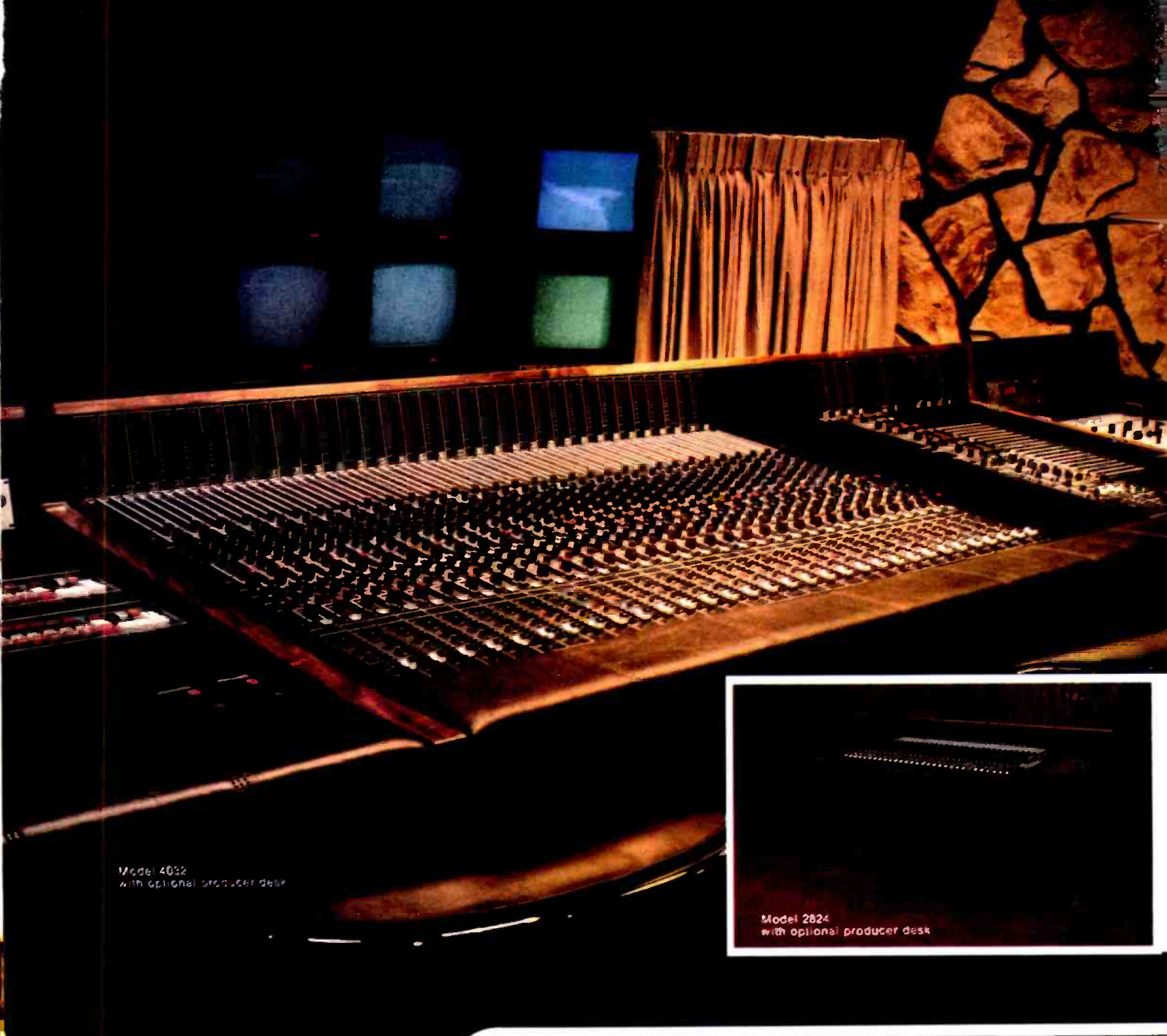

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Dee Robb

continued from page 60 . . .

Approach to Bass

Absolutely, I think it's the least consistent instrument. However, I often think that it's the player. 'Cause I have had situations where we have taken people who are open-minded and willing to work with you, and we've wanted to punch a bass line in that we'd done the day before or something, and boy — I mean it doesn't even sound close! And we'll end up by saying, "Okay now, it sounds like you're playing a little harder than you were — ease off a little. Okay, that's better, *now* that had a little more edge on it the other day . . .". Finally all of a sudden, boing!, you get what you had the other day. And it's real, real . . . It's fine points, it's a very critical instrument.

Favorite Bass Sounds

Probably two of my all-time favorites are Ace "How Long Can This Keep Goin' On" and "Funky Broadway". But that depends on what you're going after. I mean, I *like* those two . . . generally I like rock & roll and I like R&B, probably my more favorite. The Doobie Brothers have a great-sounding bass.

I liked a lot of the bass sounds on Rod's album. I liked some of the bass things on Artie Garfunkel's album — there were some nice things on that. On Rod's I liked . . . "Georgie", and "Tradewinds" I like very much. On Artie's I like "Breakaway", and . . . Oh, Christ, the first song on side one, I never could say the name of it — the Stevie Wonder song. It is "I Believe When I Fall In Love".

The Player

I'll tell you why *I* think it's one of the harder things to record. 'Cause I don't think there's hardly anybody around that can *play* the damn thing, is really what it's down to.

Dynamics and touch. I mean, there's lots of guys that can play great feel and notes and on stage it's incredible, but when you get into a studio, a bass is one of the most sensitive things, and touch is so all-important. Lee Sklar doesn't necessarily *always* play the greatest part that could be played, but his touch is so great that you don't even have to put a limiter on him, you don't have to do anything. Just shove a mike in front of the amp and a direct box and, boom!, you're off to the races. And there's a few guys like that, but *very* few, and I mean even some of the greatest bass players with the greatest names . . . I have had a *hell* of a time recording. And it's not the room — you know, you're going direct — it's not the equipment . . . There's no variables, everything's the same, it's the guy. And it's funny — I have had situations where I get in a lot of hassles with bass players. Where I will ask a good bass player — I'm not gonna mention any names, but people that are well known — I'll say, "Look, you know, could you be a little steadier with your dynamics, a little more constant? Can you play it a little brighter — can we get a little more edge on the thing? And they'll really take it as an insult — "Look, man, I've made millions of records . . .", you know. Okay, well so have I, but we're here to make . . . you know, to do the best thing we can for *this* record.

Two good bass players can pick up the same instrument and it will sound totally different! The same bass player, without touching his controls, can run the gamut from pure mush to . . . just biting pop, simply by changing his touch. That's why bass is such a critical instrument to record. You have to be *so* consistent on the thing. You know, once a player is into a place — as far as his touch — he really has to stay, and work within a very tight set of parameters to keep from varying the sound enormously. Which is consequently why about 90% of the time you end up limiting basses and compressing and things like that. Duck is one of those few people that I don't have to limit. His touch in general is *very* consistent, and once he locks into what you want, why he can stay on it all night long.

There are times when, for a bass to get through, you *need* that "smack" that a pick gives it. And it's a different type of thing than popping it with your fingers. Popping it with your fingers, you're using a little of the slap on the fret, you're sometimes getting the slap of the string on the pickup. With a *pick*, you can play lightly and delicately and still put a "smack" up on the top of it that is a very definite, defined thing. And as I say, there are times when that's not attractive if you're doing a very open ballad and you want a nice round kind of bass. But if you're doing something where you have a full track for instance, and you have a lot of mid-low program, sometimes the only way to get the bass through with any definition is by *using* a pick and having that tight kind of control. Another thing in a situation like that that I do quite regularly is to place a little piece of foam underneath the bridge of the bass, just touching the strings, and then you have to work with it until you get it just right so that the strings decay a little quicker. Again, this would be a program where you *have* a lot of low end. You don't want that low end roaring around down there with, say, a grand piano that's very fat and some acoustic guitars that are very fat — that type of thing — so sometimes you will just mute the strings a little bit, and use a pick, and it will give a very nice, defined bass sound, without getting it sharp and nasty-sounding and . . . You know, "funky" — without that kind of thing. It shortens the decay. It's like just putting the damper slightly on a piano or something. Or, it's like putting one strip of tape on a tom-tom, on a drumkit. Sometimes there's just too much of that roaring around down there, and *especially* true if you have to limit your bass player, and as I say, most of the time you really do.

Equipment

Depends on the player again. And it depends on what you're going after. I have had *very* satisfactory results, because we were going after a sort of effect like with a Rickenbacker. There's nothing else that sounds like it — it's got that string-y sound and for an effect it's great. Some people can play a Fender Jazz Bass and it


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has all the punch and sock . . . But overall, a Precision — a Fender Precision — is probably the best instrument, no doubt about that, for overall bass-playing.

Almost every bass amp that I have ever recorded I have found gets mushy and washy when you really crank the volume up. We have two Ampeg B15's which are my favorite bass amps in the world, and our maintenance partner keeps them very, very quiet, and so that's what I use most of the time. And I find that at a very low volume, the amp has a tremendous amount of punch — a lot more pop than the direct box does. The direct is a little . . . more "present" a signal, but it doesn't have the bite on the top end, so the combination of the two usually is very attractive.

I really think of Roto-Sounds as an effect. Most of the session bass players, if they have Roto-Sounds, they usually have 'em on one instrument and they use that as an effect, and then they'll have a nice good old set of Dead Fenders or whatever.

Miking

I usually use a 421 on the bass. Sometimes an RE-15, but usually a 421 — the same thing I use on a kick drum. It's a great-sounding . . . *I* think it's a great-sounding mike for bass.

Direct

I don't necessarily like all direct, because there's tonality that, if the amp is good, you get that you just don't get with a direct box. And a lot of times you get a lot more punch with an amp, and high end and snap and everything. I like a bass to be . . . Well, again, depending on the material — if we're talking about a ballad like "Yesterday", then that's something else. But if you're talking about the average medium- to up-tempo rock & roll/R&B song — you know, that ballpark . . . I basically like a bass to be fairly tight. I hate any ambience on a bass — it destroys a bass — so I like it to be tight, I close-mike it real tight, usually turn the amp down pretty low. It usually sounds better with the amp down low unless again we're going for a distortion effect or something like that. One thing — I'm surprised at how many people I talk to about this and don't realize it — is that 90% of the time when you're recording a bass miked and direct, they're out of phase with each other. Reason being that the speaker takes time to move, and consequently it's *enough* time that it just throws a direct and a miked bass out of phase. And so I always check it — I never just do it as a matter of course — but I usually end up flopping the phase on one or the other, it doesn't make any difference.

I would say about 50% of the time I use both — a combination of direct and amp — and about 25% of the time it's direct and 25 it's amp. It depends on the instrument, the way the guy's playing, how good the amp sounds — you know, a lot of variables.

Limiting

Because of the general inconsistency of most bass players, bass is an instrument that you tend to limit. And the other reason that you tend to limit it is because there's no transients to a bass, and so you're not losing anything so . . . it's easy to limit a bass and keep it more consistent. The problem *is* that the sound varies, depending on how hard the guy's hitting the limiter. So you can put a guy into the very same limiter, and you can set the controls exactly where you had 'em, and it's going to be different, depending on how hard he nails that limiter. And so consequently, quite often punching *basses* in is a real hassle, trying to match it up.

I limit the bass most of the time. I usually use an 1176 on the bass, and I never pull it more than -3, and I set my attack a little slower than midway, and I set the release fast. I set the release real fast, because the detrimental thing with a limiter is that it will hold the tones up, and it will sound mushy, it makes it sound mushy. Especially on an up-tempo thing. Like the most extreme example is a Stevie Wonder/R&B real fast kind of bass. Boy, you gotta be real careful with limiting on those, because that kind of approach totally depends on the separation of notes, and if you start . . . You know, it starts to get boomy, you start to run 'em together a little bit, why you lose that.

Equalization

There's a quality in the bass I call pop. When I'm talking about the *percussion* of a bass . . . I like basses to be percussive. I know guys who can come in and play with their fingers . . . One of them being Duck Dunn from Muscle Shoals. That son-of-a-gun can come in and he can play with his fingers and it's got more bite or pop than 90% of the guys playing with a pick. I don't know what he does. I've no idea what he does, I watch him, I try to figure it out — you know, if I could figure it out I could *tell* people! I don't know what it is. But he can do it and there's a couple other guys that can do it.

I generally like a percussive, popping bass. And, if it's percussive, I always find that it works better with the kick drum — if they're playing tight together. Of course, the kick drum, if they're playing tight, adds to the percussiveness of the bass.

The percussive quality is in the upper mids, like you were talking about. Actually about 3k's a little high in most cases — usually about 2k. Right around in there is where I really find that percussiveness that I want to hear. That's where I usually find it, *but* as I say, I am usually not successful in boosting at that region — I'm usually more successful in *cutting* at another region. As a matter of fact, I do that most of the time. I boost EQ very little, and very seldom. I usually go for the opposite approach — I usually find out where there's too much of something and I cut it. For several reasons: One is because you're eliminating equalizer distortion, which may be nil in a good console like the Tridents. You know, you can hardly hear it *but* . . . everything you do multiplies, and you equalize it when you record it, you equalize it when you mix it, and it builds. And so consequently I prefer to cut things. In general I do not like low midrange. There's hardly anything that I like low midrange on except in some cases vocals, because like 500 Hz is a lot of times where you will get the "warmth". Or 400 — somewhere in there — where you get the warmth to a voice. Very few voices are effective like down at 100, because it starts to get mushy.

Well, I'll tell you, I have found something very interesting. I have found that first of all, a bass is the most unsatisfactory instrument to EQ that there is. If it doesn't sound good, there's *nothing* you can do to the damn thing — generally — to make it sound good. However, I stumbled onto something not too long ago, and that was I found that for about 90% of the electric basses I record, I can roll out low-mids, in a fairly sharp curve, at about 250 Hz, and it does amazing things. It takes away distortion that you would swear was on the top end, it takes away a lot of the mush. The amount that you roll out is totally dependent on the program. What it does is, it seems you would swear that you were *pouring* like 3k on. You'd swear you had *tons* of it on! All of a sudden that pop and the strings start to come up, you'll hear a nice round low end . . . It's been the single most satisfactory thing that I can do with an electric bass. Where I use it heavily — and I seem to do a lot of those projects for some reason — is when I'm mixing other people's material, and the bass is on the tape and it's all . . . You know, there's no option of working with the player.

Track Allocation

Unless I'm asked to do something else, I usually record the bass on two tracks, so that we have a choice down the road somewhere. Also, because of the fact that there are no transients to a bass, it's an easy instrument to bounce over to another track. And I record it direct and miked.

We've always been very pleased with the combination of direct and amped. The only thing you have to be careful of direct and amped is a lot of the time they'll be out of phase, it's . . . Certain frequencies get cancelled and lost, others don't, and so actually that's another thing that you can play with, depending on the type of material that you're doing. By leaving the two signals out of phase, you get a very round, elastic sound — toppy


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and bottom, bottom end. Loads of bottom end, and very little midrange. Like if you're doing real quiet, open things, it's very nice. Possibly "Tradewinds". I did it — I can't remember, to tell you the truth, but I might have done that on "Tradewinds". There certainly are going to be places on Joan Baez's album that I did it.

One Particular Player (Duck Dunn)

Well, first of all, the obvious thing is that he plays the right things. He's not a busy bass player, he works *totally* conscious with the drummer . . . He works with the drummer, he treats the bass not as a lead instrument, but as an integral part of the basic rhythm section. He really strives to make it fit in and make it *mean* something to what you're doing. I mean, he analyzes the material and then he'll get into it and if he doesn't like what he's doing he'll stop for a minute, listen, then try something else. Till he gets what he wants. Technically, the obvious thing is that he's consistent. He's incredibly consistent — all the notes are basically the same intensity. In general, bass is not an instrument that *should* be played with a lot of dynamics — it's like a foot drum. I mean, if the entire *track* has a lot of dynamics, then that's something else. But not the way you would emphasize things with a rhythm guitar, or you'd emphasize kicks and pushes and stuff. The bass, it's like the kick drum — it's gotta be there all the time. And a lot of people don't realize how important this is.

The other thing is Duck'll do anything you want. He's just as happy to play with a pick as with his fingers — he doesn't care. A lot of bass players . . . I mean, I have literally gotten into nasty, nasty situations with bass players over playing with a pick! I've had it to the point where, "Look. We're *paying* you, and if you want to do the session, *you* play with a pick. That's what we want". And that's terrible. I hate that — I *hate* to hassle with musicians. I mean, that is so detrimental it's the worst thing that can happen in a project. I'd rather have the tape machine blow up than have a hassle with a musician on a track.

So Duck has all those things covered. And the *other* thing that's great about Duck is he is *incredibly* versatile. Duck, boy, he really runs the gamut — I mean, he can play just about anything! The guy is really good.

He has two basses. He has a Precision and a Fender Jazz Bass, and most of the time he uses the Precision. And that's all — no amp. It's usually up to the discretion of the engineer or the producer which one you want him to use, depending on the sound you want — you know, a Precision is a little punchier, funkier sounding bass, and a Jazz Bass is a little cleaner, "rounder"-sounding instrument. And so he'll use generally whichever one you ask him.

He never has brought an amp in with me. Now I don't know whether he has one or he doesn't, but he never has. We have a studio amp. A lot of that customized stuff has been given to us by companies, and then — some of them — we've customized a little bit and some of them we haven't, but the main companies are Ampeg, and Camco drums and Roland Synthesizers and effects. They have given us a whole array of synthesizers which are very handy, and have been used on almost every project we've done in there. Premier has given us tympani and we have orchestra chimes and we have a full drum set also.

photo: Andrew Sackheim

Leonard Kovner

continued from page 60 . . .

Sound, Amigo, Sound Labs, Producers' Workshop, Westlake, Caribou, Criteria, the Record Plant, Electric Lady, A&R, Trident, Olympic, A.I.R., Island, Abbey Road, the Chateau, Strawberry, and Sarm.

In April of 1975 it was unveiled to the world — his baby. Leonard researched it, he designed it, he raised the money, he owns it, and he runs it today: Three years in the making, Davlen Sound Studios on Lankershim Boulevard in North Hollywood. And it's quite a nice studio, too.

His credits as an engineer include Quincy Jones, Van Dyke Parks, Les Dudek, Sarah Vaughn, and Boz Scaggs on his recent "Silk Degrees" LP.

Approach to Bass

"The first thing you have to consider is what it's playing musically. I mean what its part is within the track. Is it a part to be featured, is it a part that's back, is it a disco track, is it a ballad . . .? The bass sound depends a lot upon the drum-kit, and how it's going to combine with it. It depends on the material. There's a tune on the new Fleetwood Mac album where the bass is like really important to the whole arrangement — it's the loudest instrument. It's like a really well-recorded bass, sounds really nice. I think it's important, before you record an instrument, to know what *musically* it's going to play, note-wise — get an idea of what the *arrangement* of it is going to be so that you don't just always get the same sound on it that you always do. That seems to be a very common thing in Hollywood. You get a whole set of *sounds* — you get a drum sound, you get a bass sound, you get a guitar sound — long before you've ever even heard the song itself. And that's sort of . . . an ass-backwards way of doing it. It's like you're going to make the *song* fit into the *sound*, or are you going to listen to the song, and derive the character of the instruments from the mood of the song? I kinda like to work that way, I get concepts for drums and bass and things long before I even come in and record them — how I want them to sound — and if I can get them to *sound* what I imagined in my head, then I'm way ahead of the game. Otherwise, I've defeated my purpose.

Quite often you come in and get a really good sound, and it just doesn't fit the track. For example, I'm working with a strange producer, and material that I haven't *heard*, I'm expected to get a whole *sound* happening, and I don't even know what the song is! I may mike the drums for a gigantic, powerful drum sound — like all punch — then I find that it's a ballad or it's all stick-and-rim and high hat through the whole thing. You know, with an occasional "toop" of the tom-toms or the kick drum. And the bass is a very unimportant part of the tune. If the bass and drums are playing a contrapuntal part to let's say the piano and the guitar, and they're not getting in the way of the song *vocally*, then maybe they should have a different kind of sound. So I like to hear what it's gonna be first. I don't usually get that chance, but I'm trying to eliminate that. You know, by not taking outside work anymore! So I hear a song first, and then I get a sound to fit the character. You don't want to go in and have it sounding like . . . Led Zeppelin when it's supposed to be a ballad, and you don't want to have a gigantic drum sound like Supertramp if it's going to be a rock thing. *You've got* to sort of characterize the sound of the instrument to fit the song. Unless you're really into stylizing yourself or something.

I like a bass that's clear. A bass that's full, yet clear — that every note is audible within it, and is recorded in an audible sort of way. That in the mix, it remains audible throughout the *entire* mix, and never once is ever *in*audible, no matter where it plays. It has integrity, apart from the kick drum, or maybe it is part of the kick drum — you know, maybe it's joined together — or maybe it is an integral part *of* a piano part . . . But just so that it has integrity, and if it's supposed to be combined, *how* it's combined.

Favorite Bass Sounds

"Silly Love Songs" to me is an excellent bass sound — I love the sound of that bass. It's clean and clear . . . Of course, that's the bass-and-drum *mix*. I mean, it's like all bass and drum. But it's a good bass sound — it certainly sounds good on

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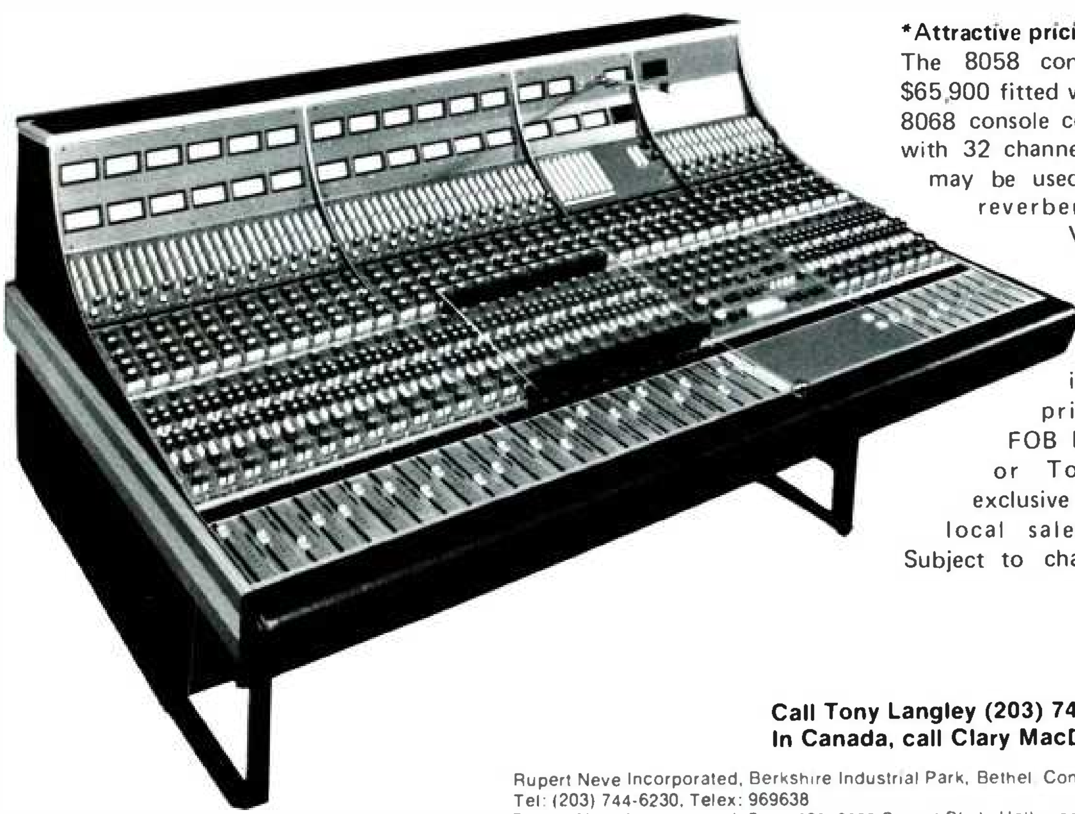

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the radio, it sounds good in my studio.

Silver Convention's, that "Fly Robin Fly". When I play that in the control room, it *really* sounds good, okay. The basic tracks for Boz Scaggs' album were done here — the album wasn't mixed . . . the bass on that is pretty good-sounding. It comes through across the radio pretty well. I would have liked to have heard a little more bottom on it, but . . .

I loved McCartney, his bass sound. Usually I like his bass sound — they're the loudest thing on the album usually. Very loud, yes, all the way to "Abbey Road" in which it is almost all bass.

The Player

Some bass players are really easy to record — an example David Hungate, Reinie Press . . . These people have good-sounding basses to begin with, they have good technique, they play evenly, the sound is just generally there to begin with. I've got some bass players who come in whose axes *sound* bad — they're muddy, and you have to do radical things like roll *all* of the bottom off, and filter the bass and compress the bass to try to make some sort of . . . semblance of solidity. That's the biggest problem.

Pick players tend to be cleaner — much cleaner players. Much more dynamic. It depends on the player, it depends on the part. The other night we had an excellent player in — an unknown kid — but he played with his fingers, and had such great right-hand technique that I swore he was playing with a pick, at first, and then realized that he wasn't. It was that clean.

The Equipment

Roto-Sound strings are usually awful, in here. The one that all the kids use to sound like Yes, they're usually awful. The roundwounds are what all the kids use, the flatwound medium-gauge are fairly good — La Bella makes a good one, Fender I guess still makes a good one. I think I have Roto-Sound flatwound mediums on my Ricky now. On the Precision there's a flatwound . . . same thing, medium flatwounds. They don't have that Roto-Sound twang.

Little Ampeg amps are nice. Little amps turned down real low are real good. Now in my home studio, I record the bass . . . If I'm recording like a phased bass, I happen to use a Dyna Comp, a Rickenbacker with heavy-gauge flatwound Roto-Sounds, no pick, and I use a phaser sometimes. Okay, and I run it through a little Music Man amp and I like it at a very, very low volume — 2½. *Very* low volume. In fact, if you don't have the headphones on, it's very disappointing the way it sounds in the room, but coming back off the tape, it sounds very big and punchy and huge.

Miking

If I can *not* record the bass with a mike, I do try. I've had very little luck with miking the bass. If I do have a choice, I use a U47 tube mike. That seems to work very well. It depends on the amp and the player. If you're using those little Ampeg amps, sometimes it's great, you can get away with like . . . a Sennheiser 441 or 421, depending on if you want it not-so-bright or bright. Depends. Sometimes a U87 or a U67 will work, or a 251 Telefunken.

Now I have some guys who come in who like to do a whole number with a lot of effects. It really depends on the player. And it depends on the amp. Really depends on the sound of the instrument, the sound of amplifier. If he's gonna play light and soft . . . A lot of bass players like Chuck Rainey come in and they sit and turn their bass . . . Even though they're direct, they have the bass amp behind them and it's turned up just a *little* bit, so they get to hear it, the sound of it. Most usually I take it completely direct, if I can.

Direct

We're using really good direct boxes to start. We're using Larry Comara's Fat Box. It is a transformerless direct box, which uses a discrete amplifier to achieve the impedance match. We also have another direct box, which is the same sort of thing as the Fat Box. It is a small amplifier — phantom powered — which has a pot on it which also allows us to drive the bass in at line level, bypassing the mike transformer completely, making the entire mike input transformerless. That tends to help clean it up a bit.

The Fat Box has bottom on it. It's the idea that it doesn't do *anything*. That's the trick. It does less than any of them. We use the Comtec direct boxes — Deane Jensen transformers - they're real good, too. They get a really good sound, but the Fat Box seems to be . . . It seems to, you know, work the best.

Limiting

In the past it's been sort of a rule-of-thumb always to throw a limiter on the bass. The Inovonics limiters are excellent on the bass — it's a broadcast-type limiter — and a lot of people are using it. That and I use a filter, like the Audio Design and Recording Selectable Bandpass Processor — it's a tuneable bandpass filter. Or a tuneable filter, is what really it is. Meaning you can *tune* the frequencies you're filtering. I use the Urei filters — the 550's — a lot, 85-, 90-cycle roll-off. That gives the bass a very bright, clear sound. I've always thought that you can always make it fatter when it's bright, but you can't make it thinner when it's fat. And it also seems to react on the tape better, too — it doesn't tend to bubble the tape and if the machine isn't quite biased right on, you don't get rocks and whatnot. So a very clean, clear bass sound is achieved that way — filtering with a little touch of compression. But lately less compression and backing the fader off and letting the headroom in the board handle it a little more. It seems that the instrument . . . This *really* depends on the player, too. That movement around the neck — especially into very low frequencies and into upper frequencies, in example going up to . . . D above middle D on bass, which is at the 7th fret — moving that far up, not limiting tends sometimes, depending on the player, to give more uniformity of sound. The bigger the wave, the more the limiter's going to act, of course. The lower the note, the larger the bass wave, the more the limiter is going to react. You know, it really depends on the player. Lately if you have a really good player, you can sometimes get away with not limiting at all, other times you can't.

See, you get the bass dropping out in both cases. If you're limiting a great deal, the limiter's attack and recovery time may not be set just right, so if he hits a very low note, the limiter's going to be working in a very extreme sort of way and if you then go to a very high note, the limiter may not recover in time, depending on the passage, depending . . . See, it depends on the music, you know — it *really* depends on the music. It depends on the notes that you're playing and the time-rhythm that you're playing in. You might have to limit more for a player who's playing fast than if he's playing a very slow ballad, where the limiter has a lot of time to attack and recover. If the guy is playing very fast, sort of jazz things, the limiter may never ever really get a chance to recover.

Equalization

I usually do add top. If the part seems to need better uniformity, dynamically throughout the arrangement, sometimes EQ is applied in higher registers.

I can't say that I always do it a certain way. You know, it's like I search the equalizer for the notes that I'm looking to attenuate, and when I find it I attenuate it. I used to have a set thing, where I always would go like right to 1,400 on my console, and boost it 3 dB, and roll 120 or 400 off so many dB, you know. I would do the exact same thing every time. I've tried to stop doing that. I'm not always going to the same EQ that I know works. Those are tried-and-true methods that took a while to develop, but like I'm not trying to do that anymore — I'm trying to . . . listen to the whole arrangement and hear what it needs. I used to start with each instrument individually without ever hearing the song. Now I'd never do that — now I push all the faders up with no EQ in, to start at a level — at a fairly low level — and listen to the entire arrangement to see what's going on. As they run it down two or three times, I start making my equalization adjustments. So I'm looking at the *total*, rather than

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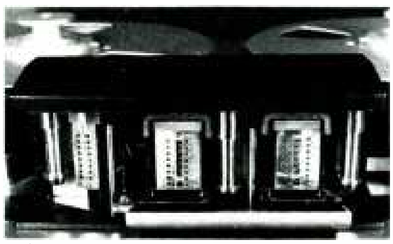

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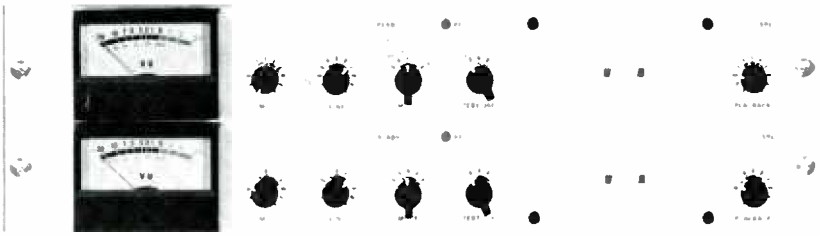

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I'm not into that trebly bass sound at all. I know what that is — like Stanley Clarke and people like that — that's very popular, but to me the bass is always down in the bottom where it belongs. It should be on the bottom down there where it was meant to be. It's not really a lead instrument — it's too clunky an instrument. It's like you don't have a lot of French horn leads, in most classical music. Or any contrabass solos, you know? It belongs down in the bottom. I saw Return to Forever live once, and to me all of that lead bass, all of that busy bass playing was really nothing. I just didn't enjoy it at all.

Pick sounds can be way up. I had just finished mixing an album, for 20th Century, and the bass was very poorly recorded. I didn't do it — someone else did — and I had a great deal of difficulty with the bass because there was so much clicking. It was up around 7k, so I was like using notch filters and things to try to get it out. I think part of the thing of recording a good bass sound is making sure your machine — the bottom end of your machine — is aligned right and your bias is correct, because *bass* tends to bring out incorrect biasing in a tape machine more than anything else. You'll get bubbles, rocks, kkkkkkkkkkkk — kind of a whooshing sound. It's quite common. We don't have that here at Davlen.

Bias is a whole trip. There has to be a certain amount of bias voltage applied to the record head so that the carrier — the bias carrier, which is an ultra-high frequency — is applied in a uniform manner, in which to align the oxide particles to a specific way so that they can then have a *signal* placed on them. And if they're not exactly aligned right — the bias carrier doesn't go on the tape right — the bass, being as broad and low-frequency an instrument as it is, tends to throw off this . . . this forming of iron oxide particles which form the sound, and a lot of extraneous noise is introduced at that point. High-frequency extraneous noises, like bubbles and clicking and then non-uniformity and non-linearity . . . This sounds like a dissertation on bias, but that's sort of what it is. I think having the machine aligned properly is very important, so that the bass sound comes back off the tape like it is on the buss. It's really important.

The bad resonances are usually very low. A dead E and A string. Open E and open A are often too low, and don't match up with the rest of the thing. Doesn't match up with the D and G strings at all. That's where you run into problems. Also rattles — to try to get rattles and pick sounds out . . .You could do that with equalization or slapping the player's fingers until he behaves.

Track Allocation

There was a time when I was taking amped and a direct track extensively, but I don't anymore. 'Cause I would find later on I'd either combine the two, or find I'd be using the direct one.

If I do take two tracks of direct bass, what I usually do is run one through a flanger, and *synthesize* the sound that you would normally get with all the guitar toys. I'd rather do *that*, than use like an MXE or a . . .What is it? Mutron, or any of those things. I'd rather do it in the control room — get a phased bass sound from the control room using a flanger, as opposed to doing it with those things. I like to do *that* better — it's cleaner. So I use two tracks for that — one is the flanged sound, and I can integrate just the amount of flanging I want, *or* if a part in the song changes, you can like then suddenly change the quality of the bass, by adding this new sound. It has its advantages.

One Particular Player (Chuck Rainey)

With Chuck I use a limiter — usually I use an Inovonics or an 1176. The Inovonics if I'm going for a very tight, short sound — rather compressed, rather pick-y type of sound. Chuck is one of those players that if the band is 14 takes, he got it exactly the . . . What he did on the first take and on the 14th take are exactly the same. He's so professional and so uniform — you know, he really is right in there. Now with him I tend to roll off the bottom and compress his bass. That's been for stuff that's like out of his field, like not jazz things — things that were more "popular" — the things that he is not normally hired for. He seems to do really well on those. I love his playing on things that are out of his bag so to speak.

He has a couple basses — he has two Precisions I've seen, and he has a Yamaha bass, which sounds very awful to me. We used it once here and I had him go back to the old Precision he normally uses. Sometimes I mike his amp, it depends. Usually my setups have been such that a little bit of bass leakage into the rest of everything has been kind of nice. 'Cause since we don't use baffles or anything anymore — you know, it's been one of those things."

Ron Malo

continued from page 60 . . .

He has spent the 70's in Los Angeles, working mainly out of Devonshire Sound in North Hollywood, and with artists like Paul Anka, Kenny Rankin, Mac Davis, the Osmond Brothers, Andy Williams, Dizzy Gillespie, Weather Report, and Cannonball Adderly.

Approach to Bass

What I like to hear in the bass is due to my earlier training, in making AM 45's or AM 78's or whatever — an AM record — is that extreme bottom on the bass does me absolutely no good. It creates problems in my mastering — I can't get the level on the record that I want . . . I'm thinking of the end-product when I'm in the studio, or even *before* I get in the studio, knowing that what I want to get on is the loudest sound that I can — the *fullest* sound — without creating a problem down the line. So I want to hear good definition on the notes, even dynamics, a nice round sound without being excessively . . . heavy. Not too much bottom — extreme bottom. When you're boosting a lot of bottom in the studio, it doesn't get *out* of the studio. It means the mastering engineer's gonna have to roll the bottom off the record, and if he has to roll it off just to get the bass corrected, that means he's rolling it off everything else. Because he's dealing with the total record, not just the one instrument.

The amount of energy that every piece of equipment has to consume to get extreme bottom on, so that the ear thinks there's a lot of it, is *tremendous!* That's a tremendous amount of energy. When you look at the difference on the Fletcher-Munson curve between say 50 cycles and 1,000 cycles, for the ear to think that it's the same volume, you're talking . . . It's 50, 60 dB or something like that, depending upon the listening level, of course.

Then there's the other point that's important about recording the bass: what monitor level are you going to record it at? The ear's frequency response changes drastically, just between 80 and 110 decibels or something like that, the bass response of the ear changes, not to mention

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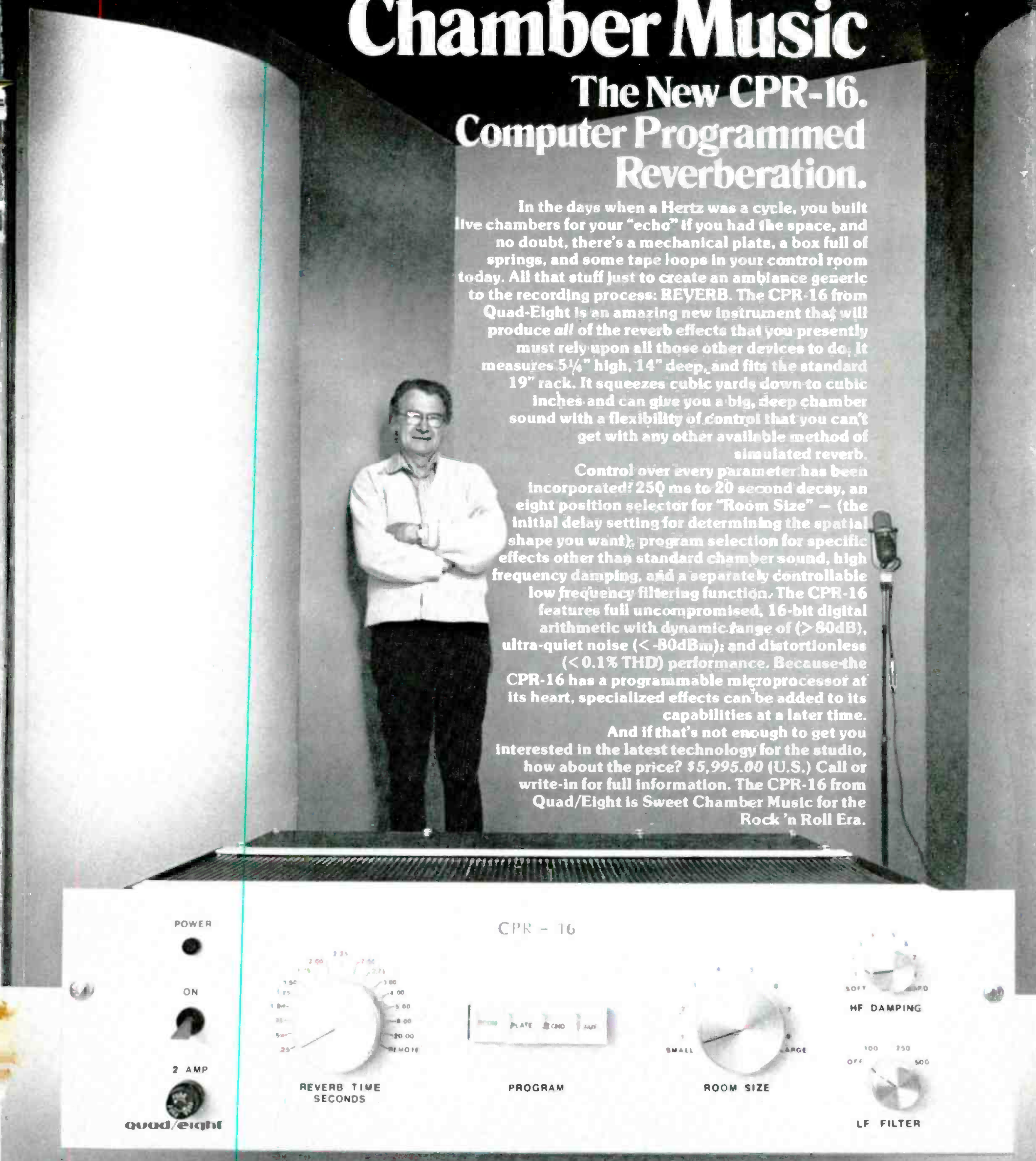

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the top end, too. Also, the other phenomenon that occurs is how're you gonna tell if it's in tune? The ear's pitch sense is extremely sensitive to level at the lower frequencies. In other words, the human ear does not sense pitch on a linear basis, or with level. It changes. Or rather, its *reference* of pitch, at different frequencies, is not a linear thing. If you change the speed of a tape machine, all frequencies move up by the same amount. If you double the speed, it's exactly one octave higher. The strange thing that happens with *volume* is that, as you increase the volume or change the volume level, some frequencies change pitch more than others. So if you have it extremely loud, it may be in tune extremely loud, and as you turn the volume down, it's out of tune.

Favorite Bass Sounds

When I think of bass, and thinking of *sounds* of bass, we're thinking of something other than the rhythm instrument in the bottom of the band, so we're talking about an acoustic bass, or someone like Jaco Pastorius, or a *solo* bass player, who's creating a musical solo – playing harmonics and doing things like that. So now we're talking about a different aspect of it. I hear some of the bass that's on the German records – the Fender bass that's on some of the Deutsch Gramophone records . . . And I'm not speaking specifically of rock & roll records, I'm speaking of . . . some of those old Caterina Valente records and stuff like that, where they managed to get a bass that has definition, sounds big, but doesn't really have any bottom, but it sounds like it has a lot of bottom. It sounds loud, without being loud. It's really a pleasant sound – clean . . . How they do it or what they did I don't know. Did I tell you what happened in London? I worked in George Martin's studio – in AIR – and every time I go to England, do you want to know what they want to know first? How did we get the "American Sound"?! They are *copying* the American sound, and that whatever we hear coming out of England is their best copy of what we do! So I told *them* that, "Do you realize that there's a movement in the United States trying to copy your copy of us?" And they broke up. It was a total breakup, and I talked to the guys that work at AIR, and at Advision Studios, and at Audio International, and some session guys that were there. And they said, "How do you get that drum sound, and how do you . . .?" And I'm not talking about just the Chess stuff. The grass is always greener on the other side of the ocean, so to speak.

The Player

I'd say that the *biggest* variable I've observed is in the musician, not necessarily in the bass or how you record it. How they play it, the technique that they use. I mean, I'm talking about the finished product – the sound of a bass. Okay. Every producer or artist has an idea of how *he* wants it to sound – *not* necessarily the way the bass player wants it to sound. Okay, what *is* a good bass sound? You got a producer . . . he says, "I want this", so you try to give him what *he* wants, you have a musician out there that may want to hear the bass he's playing completely different than the way you want to record it. So you have little fights occasionally as to how you're going to pick him up and how you're gonna record him, etc. And *that* is an agreement that has to go on between the producer, the mixer, the bass player . . .

With the caliber of musicians that I'm working with today – as opposed to 15, 20 years ago – I'm working with the top men in the field.The very best bass players available in Los Angeles, if you will. Of the dozen best, I'm working with Wilton Felder, Jerry Scheff, Lee Sklar . . . Chuck Rainey . . . I mean, when you take Wilton Felder and Joe Osborne – these people – you're talking about the epitome of bass players as far as current records are concerned. So, these people all have styles, and they're hired for their *style*, not their sound. Now a lot of times people confuse what is the sound against the style. I mean, that *is* the man's sound if he plays on a hit record and they say, "That's a great bass sound", then that becomes the style of it. It's how he picks it, whether he uses a pick, whether he uses his fingers, whether he plays down by the pickup or up by the frets . . . All of that affects the sound coming out of the instrument. Now, the engineer cannot control the way the man plays. He can't . . . How can he control the *artistic* playing? If you're having a problem getting a bass sound that you want, over a period of years you start to figure out, "Well, okay, you can do this and do that and do that . . .". How far can you bend the particular musician? Now a person that's just doing semi-amateur, run-of-the-mill recording, out there in the world, he doesn't have Wilton Felder or Jerry Scheff as bass players or one of these people. Now, first the person he's recording probably doesn't have a lot of studio experience, so you've got a problem like to start with.

Most of the gigging bass players – the working bass players – that play in groups or whatever – I find their whole sound, if you go to the stage, is all bottom, and no "point" on the bass, no definition of the notes. No percussive quality. They're playing loud, and they're only trying to "move air" with the bass. So they're not too concerned with . . . They've let their articulation go – they're not fretting as tightly as they could – so if you turn all the highs off of the bass and boost the bottom, all you hear are the notes and if the string buzzes on a fret, you don't hear it. So, you've got a bass player that may not be playing . . . *correctly*, and you bring him into the studio and he's bom-bom-bom-bom . . . It's not giving you the definition. Say he's playing all the right notes and he's playing the right figures and everything, so you say, "Well, okay, we want to brighten it up", and the guy says, "Well, no, man, that's not my sound". Then if you *do* get him to brighten up, all of a sudden you're hearing "bssst, bssst, bssst, bssst . . .". Because he's not *fingering* properly. So you have a problem. No, there's no way to *solve* that problem.

The Equipment

I *like* the Hofner myself. They made two Hofners – they made a long neck and a short neck. The reason I *liked* the bass is because it had very good brilliance. I mean, because of the dual pickup arrangement or whatever. However, it was built, you could get a beautiful, bright definition on the top end – almost guitar-like sound at the top end. Knowing the Hofner basses that I had recorded, it was a beautiful bass to record – and you still got the extreme smooth bottom. In other words, it was a hi-fi bass, if you will. I mean, it had . . . top, and bottom, and mid-punch, and with the adjustments – the tone adjustments – that were on it, you could get a nice sound. It was a very "musical" bass – maybe that would be a definition for it. A very musical-sounding instrument, as opposed to a bom-bom-bom-bom Fender bass.

The thing that they're working with now is the fretless bass. To get the musical sound. Some of the bass players . . . Jaco Pastorius, who I just recorded with Weather Report, uses a fretless bass, for some things. You don't have frets in your way, and it's a very bright-sounding bass.

I've never recorded a Rickenbacker bass that I'm aware of, so I don't know if I could comment on a Rickenbacker.

The Fender bass – all of the Fender basses – characteristically have a very tight sound, as opposed to . . . a "singing", "musical" sound. Out here some of the guys have Gibsons for a particular sound, but it was always either the Fender or the Hofner.

But talking about the different basses, and how you treat them . . . *I* don't treat them different. In other words, if we wanted a Hofner sound, we would use a Hofner. If we wanted a Rickenbacker sound, we would use a Rickenbacker. And that's up to the bass player. Wilton Felder I think carries two basses with him, and if one bass doesn't give him the sound that he wants, he'll use the other bass. And he'll say, "Ah, this one doesn't have quite the sound that I want for this record".

As far as those Roto-Sound strings go, again, that is the musician's prerogative. Because of the musicians that I'm working with, I don't have to go out and adjust their drums, I don't have to go out and adjust the bass player's bass . . . I *used* to have to do that, certainly, but currently I may go out and say, "Hey, that's a great bass sound. What're you doing?", and he'll tell me, but I don't have to go out and tell *him*, "Hey, that's the wrong . . . Let's try these strings or something like that".

I just finished mixing an album for American Flyer, that they had done in Toronto, I believe. And they had that very bright, pointy, *clangy* bass sound of you will — ba-*tong* . . . you know. It's a very metallic sound.

Miking

I stopped miking the amp about 15 years ago. None of the Chess records had miked bass, and none of the Rolling Stones that I did were either. The guitar amps were direct and the bass amp was direct, so any ambience that was in the room on any of the Rolling Stones records that you've heard that I've done was strictly being picked up by the piano or the drum mikes. I'm sorry — the drum *mike*. The bass drum mike or the one overhead microphone.

Occasionally you do mike the bass. Again, with the players that you have, they're so good that the direct is all you need. And I'm not doing groups. I'm not doing that many group recordings, of rock groups, per se. I'm doing Top 40-type single material with Paul Anka . . . If I *do* rock & roll . . . I did the Maxine Nightengale release that's out now. I'm doing straight-ahead music, as opposed to artistic compositions by a group, by a self-contained rock group. Then when you do those things sometimes you do go into different sounds — putting a room mike up, and doing things of that sort. So we're talking about maybe a different class of recording.

I would use any mike I got left over. I mean, that's a decent microphone. There are certain mikes that you *wouldn't* use, but . . . I'd sometimes use a Shure series microphone. On Jaco's bass amp, let's see, we used a Shure for some of the pickups — 545, the very cheap one. It's basically the same microphone as the SM57. It's the same capsule element and the same design. It's a *grading* system, so it's the same microphone. It's the same basic characteristic. I'm sure that the top-of-the-line ones are a little better quality than the bottom-of-the-line ones, but for the uses of the microphone and the normal limitations of that particular capsule, they're all in the same ballpark.

Direct

Basically I record all my basses direct. I don't generally mike a bass amp. I can't remember when the last time I miked a bass amp was. Oh, I miked Jack's bass amp, very slightly. And it's miked right on with the direct. And it's hardly there at all.

I don't do a thing significantly different with any bass player that comes in here unless it's a *problem* bass. The *formula* that I've developed, or whatever, to record a bass . . . We have these directs that they use here at this studio, basically just bridging transformers. I think it's 60,000 ohms to a 200 ohm primary or secondary or something, and that just feeds the mike input of the console, and bridges across his bass output — the bass instrument output. That feeds into the console, I feed it through an LA-3A Teletronix limiter. And if I do any equalization at all to the bass, it will be at 1,800 cycles, and peaking, and it may be anywhere from nothing to 10 dB, depending upon the bass player. Very seldom do I have to do anything to the bottom end of the bass — boost, cut, roll.

So we record it one way — direct — possible equalization is at 1,800 cycles. On this console that happens to be an equalization point. I found that it happens to work very nicely, with giving a fairly sharp sound to an electric bass, and it just happens to feel right. And it works with, again, *most* of the bass players.

The direct boxes that we're using, and the only ones that I've ever used are basically a UTC transformer that bridges the output of the bass, and feeds microphone level into the console, so it's just a transformer. No resistors or anything. Now we *did* have, like I say, at Chess basically the same transformer arrangement except with clip leads, to clip across the speaker terminals or a jack-to-plug into the external speaker jack, to get some of the amplifier sound into the bass or into the guitars. Kind of going backwards to the old Chess sound.

Limiting

The bass I *do* limit. I use the LA-3A by preference, because . . . maybe it's not too fast. The characteristic of the limiter seems to work very nice on vocals and bass. And it's . . . well the *result* is pleasant.

Equalization

Now the difference in sound with whatever the setup is is up to the bass player. Some bass players are brighter . . . We have a thing with Jerry Scheff — excellent bass player. Played on Presley things, and he's played on a lot of great records . . . Jerry would hear his direct bass in the earphones and it would sound too bright to him, so at the beginning of the session I'd get the bass sound, right?, then he'd put the cue phones on, and as we're going, *he's* turning the highs on his bass and *I'm* equalizing up, and we'd end up at the end of the session that I'd have 14 dB of equalization in, putting top end back on his bass. He turned the actual tone control on the Fender, just turning it down as I went up. We didn't communicate . . . It bothered him to have to hear it that way, and we've finally come to an agreement — we had a little discussion and I said, "All right, I'll meet you halfway", and so in extreme cases now for Jerry, as I put *my* equalization in that I want on the tape, and then on the monitor I de-equalize it with the reciprocal equalization, just for him. He doesn't care what you put on the tape, as long as that's the sound *you* want, but for him to be able to *play*, he has to hear it a certain way, and I guess that's true of any musician. You base what you do with your fingers or mouth or hands or feet or whatever on what you hear coming *to* you, whether it's an acoustical sound or in a room or whether it's what's coming through the earphones on your head. And the current way of recording out here is everyone uses headphones and you have a separate headphone cue mix that's totally different from what you're listening to in the control room. And specifically tailored for the purpose of musicians. This studio has four cue systems, so it's feasibly possible to have four totally separate and different cue mixes in the studio at any given time. The singer could have one, the bass player could have one . . . Of course, somebody's gonna have to share if you've got more than four people in the studio.

Absence equalization is something that a lot of people forget about. Everybody thinks boost, and sometimes taking *out* will produce a more pleasant effect than a lot of boost. I do as little equalization as necessary, to get the sound that I get — you either love it or you hate it. I prefer not to equalize at all if it is possible. I do what I have to do to make it right, and sometimes I'll take the . . . What you have a tendency to do is you take a course of action that you think is right, and you will keep continuing in that course of action of equalizing, boosting it, and rolling and everything else, and you may lead yourself down a road that's totally the wrong direction. And so the best thing then is to take it all out — get the equalizer out of the circuit, flip the in/out switch, pull it out. Because your ear becomes accustomed to these changes, and the lack of it sometimes is . . . Well, you end up with the totally wrong thing. I've seen people in here with the boost and cut controls operating at the same time at the same frequency. That's a cute trick. On the Lang equalizer. I actually came in and I saw someone boosting and cutting at 50 cycles! On the same equalizer! It must have been purposeful — it had to be purposeful. I don't think they understand the function of the equalizer. On the Lang, there is a separate boost and a separate cut control. Actually it's been done because they are separate switches and you can select different frequencies so you could boost at 200 cycles and cut at 25 cycles, if you wanted. But if you cross *over* the opposite way, and you cut above your boost, you have the interaction going on. It's not a parametric, where you can sit there and get very narrow frequency bands. Very strange phenomena occur and basically what you're getting probably is just distortion. Because of the two circuits interacting with each other. But again, if it sounds right . . . It's good if you know *why* something is happening. You should know why that something is working, so that you can take a logical course of action. If you know that there's a particular problem, you should know how the equalizer reacts or will control it. So it's good to have some background. I mean, if you turn someone that knows nothing about it loose with an equalizer, he doesn't know . . . You say, "That's for the highs, that's for the lows", and that's pretty broad. There's a broad spectrum in there.

I'm still working with this 1,800 cycle frequency point, for some strange reason. I've got half-octave steps here on these particular equalizers, and I just swept down through it and everything else. The particular sound that I like and the producers that I work with like seems to be at 1,800 cycles. And that's either more at that frequency and then if there's a noise problem or something, we roll off at 10,000 to get rid of anything that's occurring out there. You might boost something or cut something in the *lower* areas, depending upon the particular character. And this is done *after* the limiting on this particular setup here. In other words, the equalization is done after it's been compressed a little bit. And I only use enough compression so that the notes come out consistent so that, if the bass player hits a note a little lighter than another note, you still get the constant beat, but not so much that you've taken his *dynamics* away from him and he can't play *louder* in one passage and softer in another passage. Again, with the type of people and the type of sessions that I'm working with, I have to produce a record in a given period of time. I don't have time to spend an *hour* on a bass sound and two hours on a drum sound, and 15 minutes on this and that. In other words, they want to be able to start recording within that three-hour session, because I'm locked into that three-hour time schedule as opposed to a 12-hour or 24-hour situation with a group. So I have to come up with a setup and a group of conditions that will produce a good record, 99% of the time. Immediately, and be ready to move on the options. So a lot of the questions that you're asking are pertaining to group recording. And I'm talking about the everday studio-type recording where we get sidemen to come in and play. And it's a little different condition.

I *do* record group musicians — the bass player that we just had . . .What was the bass player the other day? Oh, he came in with . . . He has an Orban Parametric equalizer, he has an Eventide Omnipressor, and an Eventide Instant Flanger, and a Crown D60 amplifier. And that is his package, and *he* feeds you the output of *his* package. Not the direct off his instrument, and so I told him that he was a little noisy and he says, "What frequency? At what frequency?"! I said, "Could I get a little brighter sound?". He's a good bass player, and it was just a matter of us . . . We'd never worked together — he was a substitute because Wilton Felder was supposed to be on the date, and the answering service apparently screwed up the scheduling, and here we got a studio full of musicians and no bass player. And we were able to get this fellow to come in, and he's great. He's playing on a lot of big records . . . I'm trying to think of his name. Anthony Jackson, I believe it is.

One Particular Player (Jaco Pastorius)

Jaco is a . . . a very *musical* bass player — He's into the music that he can make with a bass as opposed to just a rhythm instrument. He's an *excellent* musician. Now Jaco creates his sound by hearing what's coming out of his amplifier, but what we're picking up on the direct pick-up from the bass — directly — seems to be a better sound, but if he doesn't have the amplifier out there to work against, he has problems.

Actually it was a very enjoyable situation, recording Jaco. He was a new bass player with Weather Report, so it was a different . . . I think he played a couple of tunes on the previous record that I did, and on the last album — this "Heavy Weather" — he was the exclusive bass player with the group, because they were in a transistion period. Alphonse Johnson was the bass player with Weather Report previously.

When I record him, I mix the direct with the microphone, till I get whatever sounds right. Mostly direct, and a little bit of his amplifier. 'Cause his amplifier doesn't have the definition that the direct has, and he's very happy with the bass sound. When he comes in *here*, he's happy. Now most of the time when a bass player comes in or any musician comes in, he can't tell *me* how to get my sound. All right? I say, "Let me do it my way, you come in and you listen, if it's not right then we'll make whatever adjustments that are necessary to do it". In other words, "Play a little bit of bass, or play your drums, or whatever, then you come in the control room and you listen, and if it's not the sound that you're happy with, then we'll adjust to make it correct. But give me *my* shot first, because I *am* the mixer". I said, "Rather than make me change what I know works most of the time . . . ". I don't use *exactly* the same amount of limiting and *exactly* the same amount of equalization *every* time — there are fine variables within this, but that's where I *start*. I know on the LA-3A, for instance, that I'll set the threshold at a certain point and I set the output at a certain point, and *that*, coupled with the gain structure of this particular console, and those pickups out there, will be my starting point. Then I either add more mike gain, change the threshold, change the equalization or whatever to get the particular sound for *that* particular bass player on *that* particular tune. Now sometimes it's changed within a tune. If it's a tune and there are changes that occur within the tune — changes in sound that have to occur within the tune, then you make those changes. Again, we're making records for ears not for graphs and charts and things like that. It's what the end-product sounds like.

I try to make *hit* records as opposed to hi-fi records, and I've always contended we're in the business to make money and not to make records for our own personal enjoyment. I mean, what my personal tastes are and what I do may not be the same. I have to make a commercial phonograph record, and I think that's the job of *all* recording engineers — to make a commercial phonograph record. A saleable record.

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A better way to adjust Azimuth

by
NEIL HOPPER

Virtually all instructions supplied with professional tape recorders and alignment tapes direct that azimuth be adjusted for maximum output of a high-frequency tone, regardless of the number of tracks. However, most engineers realize that when two or more tracks are available, relative phase of the tracks is much more sensitive to azimuth error than is output level. Relative phase of two tracks can be observed on a dual trace oscilloscope or on any oscilloscope set up for a Lissajous (x-y) display. Because most recording studios consider oscilloscopes to be service-bench rather than control room equipment, many engineers use the console to sum two or more tracks and adjust azimuth for maximum sum, which obviously occurs when the tracks are exactly in phase. The peak obtained with this method is much sharper than the peak observed on individual tracks, *but we can do better.*

Table 1 shows that the peak occuring when two sine waves are in phase is quite broad when compared with the null which occurs when they are 180 degrees out of phase. A phase shift of 10 degrees will only cause a 0.03 dB drop in the sum (difficult if not impossible to see), but when observing the null, the same phase shift will produce a VU meter reading of -15 (compared to no signal, or minus infinity VU), which is easy to see. If that isn't easy enough to be seen, or if even better resolution is wanted, the gain can be increased. (You can't increase the gain with the sum method; in fact, you have to *decrease* the gain by at least 3 dB to get an on-scale reading.)

If your console doesn't have phase reversing switches built into the input modules, it will be necessary to construct a phase-reversing patch cord (make sure it's *clearly* labeled as such). A phase-reversing patch cord can be used to adjust azimuth without going through the console, provided your machine has balanced output and the meters on the machine bridge the output lines. Simply connect one output to the other with the phase-reversing patch cord, and the VU meters on the machine will indicate a null when both tracks are in phase. Or, if you can find a convenient location, you can mount a momentary double-pole single-throw switch on your machine and connect it as shown in figure 1. In either case make sure the outputs are balanced and *stay* balanced (the inputs to your console or monitor system might be unbalanced), because if the low side is common, connecting them together out of phase will connect both high sides to common (in other words, a short circuit). This will *look* like a null no matter what the two channels are doing with respect to each other.

If your machine has unbalanced outputs, all is not lost. All you need is an extra meter, which should be connected between the two high sides as shown in figure 2. When the outputs of the two channels are at the same level and in phase, there is no voltage difference between the terminals of the meter, so the meter will indicate a null. If there is a phase difference between the two channels, the meter will read as shown in the difference column of table 1. This setup can also be used to monitor relative phase of the two channels of a stereo mix. The more out-of-phase information present in the mix, the higher the meter reading. Notice that a phase difference of 60 degrees will cause a reading of 0 VU, and anything over 90 degrees will pin the meter.

This system is actually *too* sensitive for some situations. The measurement system must be matched to the size of the quantity being measured. For example, you can use a magnifying glass to watch an ant moving around, but a high power microscope would be useless in the same situation because the ant's motion would be larger than the microscope's field of vision. A similar problem arises when adjusting azimuth on a multitrack machine. Because of gap scatter and different amounts of phase shift among different tracks, phase relationships between two adjacent tracks on a multitrack machine is not necessarily the best way to measure azimuth. It's better to compare two tracks farther apart from each other, closer to the edges of the tape. The trouble with this is that an azimuth change of only a small fraction of a degree will cause a phase difference of several cycles between the two tracks, so there will be several nulls in the general vicinity of correct azimuth. Even at the proper null, slight variations in azimuth caused by less-than-perfect tape tracking may cause wildly fluctuating meter readings, a situation like the ant under the microscope. So the less sensitive summing method is more useful in this case. Even so, we still have the problem of several cycles of phase difference between the tracks for relatively small azimuth error.

TABLE 1

	SUM		DIFFERENCE	
	$2 \cos \phi/2$	dB	$2 \sin \phi/2$	dB
0	2.000	+6.02	0.000	$-\infty$
1°	2.000	+6.02	0.017	-35.16
2°	2.000	+6.02	0.035	-29.14
4°	1.999	+6.02	0.070	-23.12
6°	1.997	+6.01	0.105	-19.60
8°	1.995	+6.00	0.140	-17.11
10°	1.992	+5.99	0.174	-15.17
20°	1.970	+5.89	0.347	- 9.19
30°	1.932	+5.72	0.518	- 5.72
40°	1.879	+5.48	0.684	- 3.30
50°	1.813	+5.17	0.845	- 1.46
60°	1.732	+4.77	1.000	0
70°	1.638	+4.29	1.147	+ 1.19
80°	1.532	+3.71	1.286	+ 2.18
90°	1.414	+3.01	1.414	+ 3.01

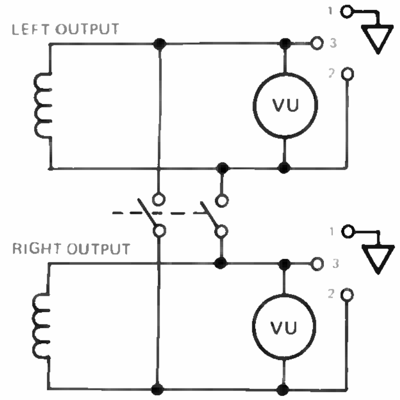

Figure 1

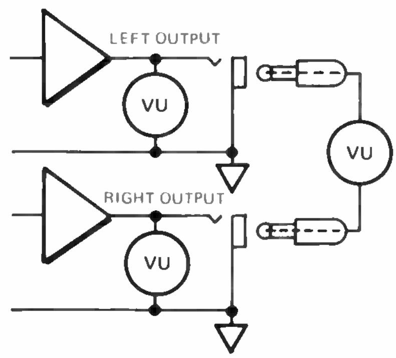

Figure 2

An easy way around this is to add a few tracks from in between the outer tracks; all the tracks will be in phase only when the azimuth is correct. There are a couple of ways to sum more than two tracks. Besides the obvious way of summing them through the console, you can patch them all to the same mult (or you can build an *octopus* patch cord) and then feed the mult to an auxiliary meter. (If the meters on your machine bridge the output lines, you won't need the auxiliary meter.) If you decide to use the console for summing, keep in mind that every time you double the number of tracks, the level in your summing amplifier increases by 6 dB, so decrease the levels on the *input* faders accordingly.

If you are adjusting reproduce azimuth to tones on someone's master instead of a full-track reproduce alignment tape, watch out for the possibility of the tones having been printed 4, 8, or 16 tracks at a time instead of simultaneously (not everybody has 24 busses on their console). In this case, you can't compare phase between two tracks near the edge of the tape. The best you can do is to compare phase between two simultaneously recorded tracks as far apart as you can find them. Again, throw in a few tracks from in between to make sure you've found the right peak.

This method is so sentitive that it's best to start your azimuth adjustment with a mid-frequency tone rather than a high-frequency tone. There are a couple of reasons for this. First, it's just about impossible to adjust azimuth to the wrong null at a mid-range frequency, although a 360 degree phase shift between tracks at 15 kHz occurs not far from zero degrees phase shift. Second, different amounts of phase shift may occur on different channels at different frequencies. Certain amounts of phase shift are expected on all channels because of the equalization circuitry; but because of track-to-track variations in the head, the exact amount of phase shift at any given frequency will vary among the tracks. This equalization-related phase shift is present at high frequencies with any equalization characteristic, but with NAB equalization it is also present at low frequencies. Azimuth should not be adjusted at low frequencies because the magnitude of phase difference due to equalization difference is greater than that due to azimuth error.

So the initial azimuth adjustment should be made at a frequency located on the flat portion of the equalization curve. Equalization should then be adjusted for flat response (or at least identical response between the two channels), and then the phase relationship between channels should be checked at several higher frequencies. Little or no azimuth adjustment should be required at this point. If the phase relationship between channels varies considerably with frequency, azimuth should be adjusted so that phase difference is minimized over as broad a bandwidth as possible.

After reproduce azimuth has been adjusted as carefully as possible, record azimuth should be adjusted to match reproduce azimuth (while monitoring reproduce phase). Bias should always be adjusted before record azimuth, because the signal is not being recorded on the tape at the gap itself, but at some point after the gap, the exact point varying with the strength of the bias field. Once bias has been adjusted, record azimuth is adjusted the same as reproduce azimuth — adjust azimuth at a mid-range frequency, adjust equalization, then check phase relationship at higher frequencies.

Of course, this is not really a different way of adjusting azimuth, it's a different way of measuring azimuth error. Everything else remains the same. However, because this method is more sensitive than the method you may have been using, it may bring to light problems in your equipment of which you were unaware but which have been there for some time. An unstable reading will result if your machine has high peak-to-peak azimuth variations. Even though the reproduce head is secure in its position once it has been adjusted, if the path of the tape varies as it passes over the head, the effective azimuth angle between tape and head varies. This can be caused by either your transport or the tape itself. If your alignment tape has seen lots of use, even very slight wear on the edges of the tape will cause instability in tape tracking, which will in turn cause variations in azimuth. Sometimes even brand-new tape is slit out of tolerance (yes, even from the most reputable manufactuers — nobody's perfect). If the tape is slit too narrow, it has lots of room to skew, and if it's too wide, it can bind in the guides or even be squeezed out of the guides (which can cause worse skewing than too-narrow tape). Tapes from different manufacturers are often slit to different widths. If possible, it's a good idea to standardize on one type of tape and have your reproduce alignment tapes manufactured on the same type of tape.

Transport problems causing unstable tape tracking are more common and usually more severe than tape problems. Worn bearings, sticking brakes, bent reel flanges, and other defective components will obviously cause fluctuations in tape tension. Tension must be correctly adjusted when these problems have been eliminated, of course. If any of the heads or guiding components are even slightly displaced from vertical, the tape tension will cause an upward or downward force on the tape. The capstan must also be vertical and the pinch roller must apply equal pressure across the width of the tape. On a closed-loop transport the vertical *position* of the pinch rollers is critical. A very slight misalignment in the height of either pinch roller can cause drastic shifts in the tape path. Worn guides and/or heads will sometimes cause a *bistable* tape path — sometimes the tape will go through in one path, sometimes another. It may even *snap* from one path to the other while the tape is in motion. Usually the only cure for this problem is replacing the guilty guides and replacing or relapping the heads.

If your equipment is in good condition, this method is better than the summing method because it's more accurate, and easier than the oscilloscope method because it doesn't require as much equipment. It is even possible to adjust azimuth to program material when no tones are supplied, provided that there is a sufficient amount of high frequency information common to both channels (Lead vocal usually works fine.) Simply monitor the difference signal and adjust azimuth to the point where the center channel disappears.

Correct azimuth adjustment is more important than many people realize, especially when you consider that most stereo program material is heard most often in mono (even on FM, most people are listening on mono sets). This method should make it easier for studios (and radio stations, too, if we're lucky) to keep their azimuth consistent, with the end result being bettter quality audio at the listener's end.

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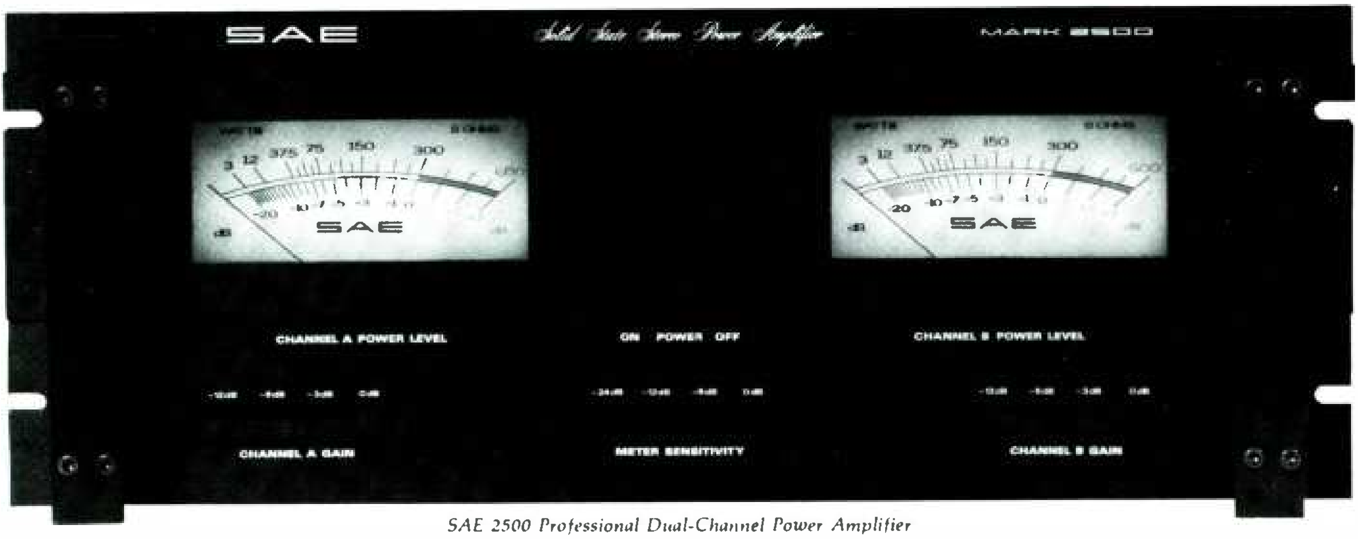

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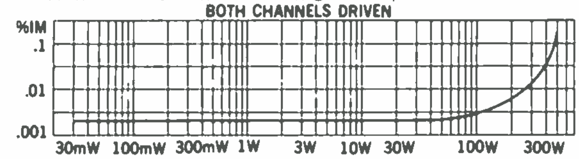

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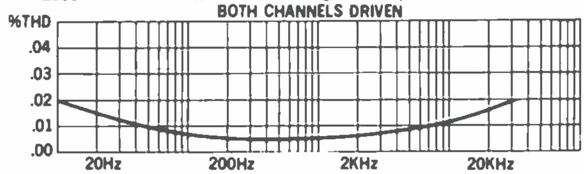

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POWER... how much is enough?

by chris foreman

Just about everybody from Altec to Yamaha now builds a "super power" amplifier (or maybe two or three models). In fact, the audio consumer is currently faced with Alvin Toffler's *(Future Shock)* "overchoice" dilemma: there are so many different amplifiers, with such widely varying specifications, features and prices, that making a wise purchasing decision becomes a real challenge.

This article presents some guidelines for amplifier selection, and discusses the meaning and usefulness of some amplifier features and specifications. In addition, it examines the foundations for the initial decision to buy a "super power" amplifier: How much power *is* enough?

An Unofficial History of the High Power Audio Amplifier

We might arbitrarily define a "high power" amplifier as one able to deliver more than 100 watts continuously to a loudspeaker load — on a per channel basis for a multi-channel amplifier. By this definition, the high power amplifier in tube-type designs, got its start before World War II. Many of the later tube-type high power amplifiers, such as those produced by Altec, McIntosh and others in the 1950's and early 1960's, are still performing well in recording studios, Elk's Lodges and other installations.

When transistor amplifiers first gained acceptance in the 1960's, they were usually limited to lower-power designs, and were often failure-prone. The transistors themselves were part of the reason for early failures; most designs used germanium power transistors which, in general, have limited thermal capabilities when compared to modern silicon devices. The primary reason for many failures, however, was that the amplifiers were not designed to adequately protect themselves against excessive input signals or overheating or against the common overload condition that results from simply hanging too many speakers on the output. Most tube-type amplifier designs, however, were inherently resistant to these abuses; I can remember connecting three and four 8-ohm speaker cabinets to the "8-ohm" output on my tube-type Fender guitar amplifier!

Armed with their new protection circuit designs and higher power silicon transistors, the manufacturers began to develop higher power transistor amplifiers to meet the demands of an increasingly watt-hungry audio buyer. Thus, the buyers and the builders reinforced each other, and soon the first "super power" amplifiers appeared. Again, arbitrarily, a "super power" amplifier could be defined as capable of delivering 200 watts (or more) per channel, continuously, to a loudspeaker load.

At least one manufacturer set out to educate the buying public to the advantages of a super power amplifier for even small home stereo systems. Their reasoning went something like this:

Reasons For Using a Super Power Amplifier

Many speaker systems, especially home stereo speaker systems, achieve their smooth, natural sound and compact size by sacrificing efficiency. In trying to reproduce realistic listening levels from a pair of these speakers a small amplifier, of say 50 watts or less per channel, will be driven into extreme clipping, producing up to 40% or more distortion on musical peaks. In other words, the 0.1% or 0.01% distortion rating of a low power amplifier is meaningless when the amplifier is driven into "clipping". By using a much larger amplifier, of 200 to 500 watts per channel or more, the system has enough peak power capacity to avoid most clipping and lower the distortion levels significantly, without raising the average listening level.

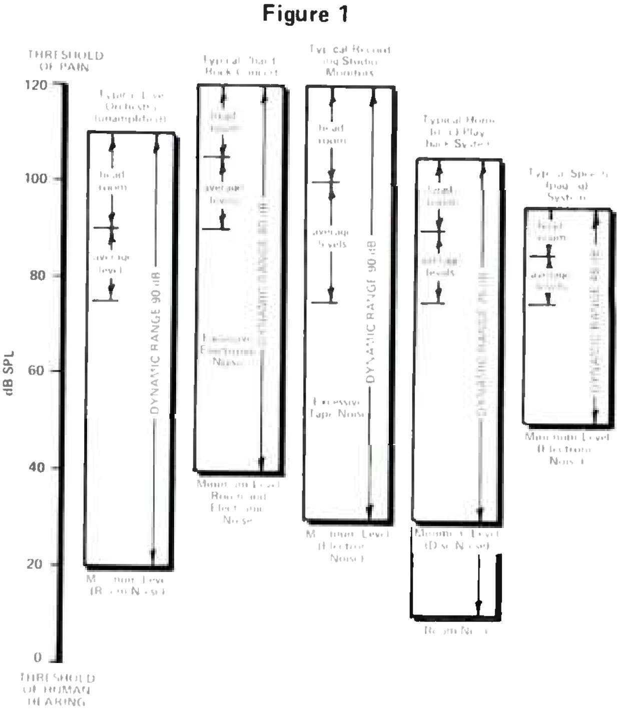
An Examination of the Above Concepts, and General Criteria for Selection of an Amplifier

For the owner of a home stereo system using low-efficiency "bookshelf" style speakers who demands realistic concert-hall listening levels, these ideas are valid and accurate (provided the speakers can withstand the high power peaks). For the more general situation, it is necessary to understand more of the background behind the ideas. In particular, it is useful, at this point, to discuss the relationships between amplifier electrical power output, speaker system acoustic output (SPL) and the effect a room may have on that acoustic output. These relationships affect the decision of how much power is enough, and should be considered for each application. Then, secure in the knowledge that a potential buyer will carefully calculate the total power needed for a given application, the discussion turns to an examination of other aspects of amplifier choice.

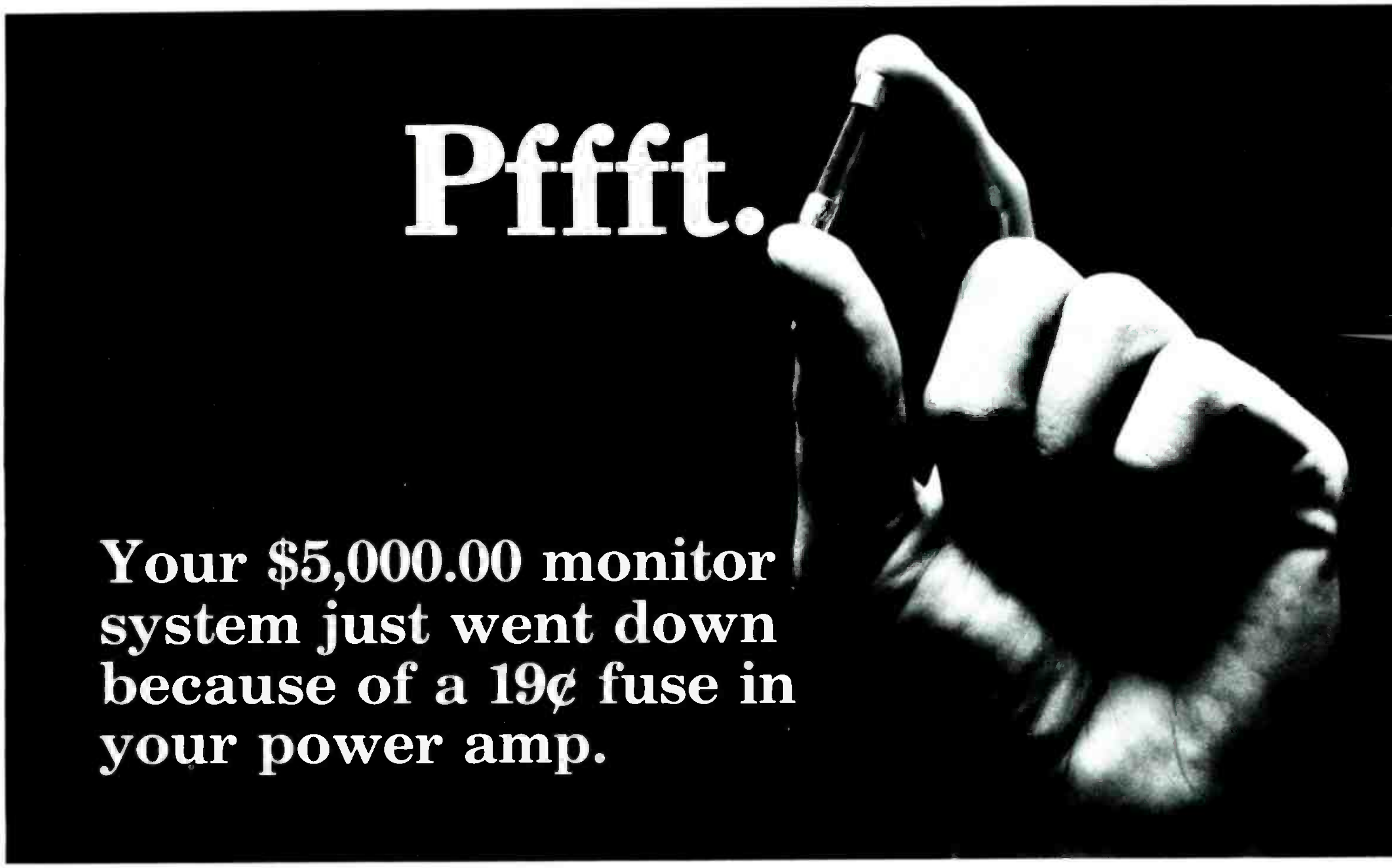
Commonly Encountered SPL Levels

One of the first steps in designing any sound system, is to determine what range of SPL levels it will have to produce. The chart in Figure 1 shows some typical levels for various types of systems. It also shows possible ambient noise level, nominal operating level and dynamic range for each of these systems.

Figure 1



COMMONLY ENCOUNTERED SPL AND DYNAMIC RANGE



Dynamic Range

Dynamic range can be defined as the difference, in dB, between the loudest and the softest levels in a performance. However, the softest levels of a performance must be audible above any ambient room noises or sound system electronic noises. Thus, another definition of dynamic range is the difference in dB between the loudest level of a performance and the ambient noise.

The dynamic range of a live performance may be as great as 100 dB or more. For a concert of classical music, the 100 dB dynamic range may represent SPL levels ranging from a low of 20 dB SPL to a high of 110 or even 120 dB SPL.* This 100 dB dynamic range implies a power variation of *ten billion to one* (see Figure 2). In other words, for a live concert sound system with 10,000 watts of peak power capability, the minimum levels would be about one-millionth of a watt — quite a variation! The 60 dB dynamic range of a good classical record (phonograph disc) represents a power variation of "only" one million to one. Thus, for a home stereo system with a peak power output of 100 watts, the minimum power levels would be about one ten-thousandth of a watt — still an appreciable variation.

The average or "nominal" level of a performance may be anywhere from 10 to 30 dB below the maximum. In a sound system, the difference between the maximum output and the average level is called "headroom". If a sound system can produce a maximum of 100 dB SPL at a given distance from the speaker system, and is required to produce 90 dB SPL average level, then its headroom is 10 dB. If the musical performance being reproduced by this system has peaks of 120 dB SPL, with a 90 dB average level, this same sound system would "clip" the highest 20 dB of the performance, causing considerable, audible distortion.

Once the system's maximum levels and "headroom" requirements are known, the next step is to calculate the electrical power required to produce these levels from a given speaker system. Loudspeaker sensitivity ratings are the key to computing the required electrical power once the desired SPL levels are established.

Using Loudspeaker Sensitivity Ratings

The sensitivity rating of a loudspeaker relates its sound pressure level at a given distance (usually four feet, one meter or 30 feet) to its electrical input power (usually one watt or one milliwatt). While the sensitivity rating is similar to an efficiency rating, it is not the same. An *efficiency* rating relates acoustic power output to electrical *power* input, and includes sound eminating in all directions from the loudspeaker. In contrast, the *sensitivity* rating relates the sound *pressure* level (not acoustic power) to the electrical *power* input, and includes only on-axis measurements.

In a free-field condition (an anechoic chamber or, to a good approximation, outdoors) the sound pressure level at any distance from the loudspeaker can be calculated by the following formula: (SPL values are expressed in dB, distance values in feet or meters)

Formula I

$$SPL_{D_1} = SPL_{D_2} - 20 \log \frac{D2}{D1}$$

Where D_1 is the distance used in the sensitivity rating, SPL_{D_1} is the SPL given in the sensitivity rating, SPL_{D_2} is the unknown SPL, and D_2 is the distance at which the unknown SPL is measured.

** If the room noise is appreciable, it may limit the lower SPL figure to 30 or even 40 dB SPL (and thus limit the dynamic range as well).*

This formula leads to the simple rule, known to most sound engineers, that sound pressure level drops 6 dB each time the distance from the source is doubled (when $D_2 = 2D_1$). For example, if a loudspeaker produces 80 dB SPL at a distance of 4 feet, then the SPL will be 74 dB at 8 feet, 68 dB at 16 feet, and so on.

Indoors, the calculations are complicated by reflections from walls, ceilings, floors and furniture. In order to accurately calculate the attenuation of sound at a given distance from the speaker system indoors, two new factors are needed:

1. The "Q" of the speaker system, a measure of its directivity.*

2. The room constant, "R", for the indoor area under consideration ("R" describes the room's sound absorption).*

Given these additional parameters, Q and R, calculate the room attenuation by following the following steps: First, convert distances D_1 and D_2 to dB loss values.

Formula 2

$$D(\text{dB loss}) = -10 \log \left(\frac{Q}{4\pi r^2} + \frac{4}{R} \right)$$

Where r is the distance from the source, and $\pi = 3.14$.

Second, find the total attenuation of the sound from distance D_1 to distance D_2 by subtracting the two loss values just calculated:

Formula 3

$$\text{Attenuation from } D_1 \text{ to } D_2 = D_2(\text{dB loss}) - D_1(\text{dB loss})$$

Subtract this value from the SPL value at D_1 to find the actual SPL level at D_2.

This whole process can be simplified by making some assumptions. First, assume that sound levels indoors attenuate at 6 dB per doubling of distance, just like the outdoor case, up to the so called "critical distance". (At the critical distance, the levels of direct sound from the speaker and reflected sound from walls, etc. — reverberant sound — are equal). Second, assume that after reaching the critical distance, the sound does not attenuate appreciably with increasing distance.

These two assumptions are usable for rough approximations, provided the critical distance is known. The critical distance depends on both the room constant ("R") and the "Q" of the speaker system. To get an idea of the critical distance in a particular room with a given speaker system, play a constant volume source through the speaker system (pink noise will work well). Then measure the SPL close to the speaker system, but farther away than the speaker system's largest dimension. Next measure the SPL far away from the speaker system, in the reverberant field. (You know when you are in the reverberant field because the sound level does not vary appreciably when you walk closer or farther away from the speaker system.) Then use the following formula to approximate the critical distance:

Formula 4

$$D_c \cong D_1 \times 10^{[(\text{db close} - \text{dB far})/20]}$$

Where D_c is the critical distance, and D_1 is the distance where the first (close) SPL measurement was made.

To use these approximations for finding the attenuation at any distance up to the critical distance, use Formula 1. To find the attenuation at or beyond the critical distance, substitute the critical distance D_c for D_2 in Formula 1. (Past D_c, the SPL will not attenuate appreciably.)

Now, to demonstrate the usefulness of all these formulas and concepts, here's a sample problem: Find the amplifier power required to produce 90 dB SPL at a distance of 100 feet assuming the following parameters:

Speaker system sensitivity = 101 dB SPL at 4 feet from 1 watt of pink noise, band limited from 500 to 3,000 Hz.

Speaker system pink noise power capacity = 50 watts over its operational bandwidth.

Speaker system "Q" = 7 for the bandwidth from 500 to 3,000 Hz.

Room Constant "R" = 10,000.

Using formulas 2 and 3:

$$D(\text{dB loss at } 4') = -10 \log\left(\frac{7}{4\pi x16} + \frac{4}{10{,}000}\right) = 14.5 \text{ dB}$$

$$D(\text{dB loss at } 100') = -10 \log\left(\frac{7}{4\pi x10{,}000} + \frac{4}{10{,}000}\right) = 33.4 \text{ dB}$$

Thus, the attenuation at 100 feet = 33.4 dB - 14.5 dB or 18.9 dB. This means that if 1 watt is fed to the speaker system, producing 101 dB SPL at 4 feet, the level at 100 feet will be 101 dB - 18.9 dB = 82.1 dB, or approximately 82 dB. This is 8 dB less than needed, and to produce the required 90 dB SPL at 100 feet, the 8 dB loss must be recovered by adding 8 dB to the speaker's 4 foot output level, raising it to 109 dB SPL.

One way to recover the 8 dB is to increase the power fed to the speaker system. To calculate the total amount of power necessary to add this extra 8 dB of SPL, use this next formula:

Formula 5

$$\text{Pwr req} = \text{Pwr (in sens rat)} \times 10^{(\text{needed dB}/10)}$$

Thus, our power needed = 1 watt x $10^{(8/10)}$ = 6.3 watts.

Since this is well within the speaker system's power capacity, the goal has been accomplished. However, there is no

Figure 2

Level Change in dB	Equivalent Power Change
1 dB	1.3 times
2 dB	1.6 times
3 dB	2.0 times
6 dB	4.0 times
10 dB	10 times (10^1)
15 dB	32 times
20 dB	100 times (10^2)
30 dB	1,000 times (10^3)
40 dB	10,000 times (10^4)
50 dB	100,000 times (10^5)
60 dB	1,000,000 times (10^6)
70 dB	10,000,000 times (10^7)
80 dB	100,000,000 times (10^8)
90 dB	1,000,000,000 times (10^9)
100 dB	10,000,000,000 times (10^{10})
110 dB	100,000,000,000 times (10^{11})
120 dB	1,000,000,000,000 times (10^{12})

allowance for headroom in this calculation. Remembering the dynamic range discussion, a musical program may have peak levels that are considerably higher than its average levels. If the 90 dB SPL required in the above example is adequate for peak levels, then the power calculated is adequate; if the 90 dB SPL is to be the *average* level, then the system needs additional power to allow for these peak musical levels. For a live reinforcement system, 10 dB of headroom is usually considered adequate. Adding 10 dB to the above power figure multiplies it by a factor of 10 (see Figure 2). Thus, instead of 6.3 watts, the system needs 63 watts of amplifier power to be able to produce the peak levels of this program. If 20 dB headroom were required, for reaching 110 dB SPL peak levels, the power required would be 630 watts! It becomes apparent that it is important to carefully set headroom and average levels.

It is also useful to examine what would happen if a speaker system with a different sensitivity were used. The chart in Figure 3 shows the power and SPL figures for the same sample problem if the speaker system sensitivity is varied. The extreme importance of the sensitivity rating is apparent from this chart. Even a small 3 dB reduction in sensitivity rating means that an amplifier of twice the size in watts will be needed to give the same SPL rating at 100 feet.

Figure 3
Results of Example for Varying Speaker Sensitivity

Speaker System Sensitivity (1 watt 4' rating)	Power to Achieve 90 dB SPL at 100'	Power to Achieve 90 dB SPL at 100' with 10 dB Headroom
101 dB SPL	6.3 watts	63 watts
95 dB SPL	25.2 watts	252 watts
89 dB SPL	100 watts	1,000 watts
83 dB SPL	400 watts	4,000 watts
77 dB SPL	1,600 watts	16,000 watts

Multiple Speaker Installations

Doubling the number of speakers, each receiving the same amount of power, increases the SPL level by approximately 3 dB. Thus, two speakers, each receiving 50 watts of amplifier power produce approximately the same SPL as one (same type) speaker receiving 100 watts of amplifier power. The "approximately" results from the fact that stacking one speaker on top of another will normally result in an increase in the effective "Q" for the combination. Thus, when compared to a single speaker receiving 100 watts, two speakers (same type as single speaker), each receiving 50 watts of amplifier power, may produce a total SPL at a given distance that is greater than the 3 dB gain that would be expected. The exact amount of "Q" increase for a given speaker system configuration is hard to calculate, however, and experimentation may be the best way to measure the "Q" increase in any given situation.

To review: the amount of amplifier power needed for a given sound system depends on the maximum SPL needed at the listener's position, on the amount of headroom required to avoid peak clipping, on the environment (room) in which the system will be located and, in particular, on the speaker system ratings.

After going through the calculations and finding the amplifier power required for a given system, it might be tempting to "double" that power in order to double the available sound level, and increase the headroom to a point "far beyond" that which is needed. After all, the "super power" amplifiers are available at a relatively low cost per watt. However, doubling the power will only add 3 dB to the total SPL, and in increase of 3 dB SPL level is not detected by the human ear as a doubling of apparent loudness. This physchoacoustical effect is stated by a physiological principal called the Weber-Fechner Law.

The Weber-Fechner Law

In its general form, the Weber-Fechner law applies to all our senses: The amount of additional stimulus needed to produce a perceptible change is dependent on the amount of stimulus already present.

In mathematical terms, the Weber-Fechner law suggests that the human ear responds to changes in sound level in a logarithmic manner. More simply, this means that *for a sound to seem twice as loud,* it requires approximately *ten times as much amplifier power.*

Biamplification

The "brute-force" method of increasing headroom is to simply add more and more power. Unlike the "brute-force"

* *For a detailed discussion of "Q", "R" and the entire subject of sound system design, see "Sound System Design and Engineering" by Don and Carolyn Davis, published by Howard W. Sams Company.*

method, biamping (or triamping, etc.) a system can significantly increase the amount of headroom with little or no increase in the total amplifier power capacity.

Program material (music or speech) is made up of many different frequencies and their harmonics. Most music, especially popular music, is bass heavy; that is, the low frequency material contains much more energy than the high frequency material. When both high and low frequency material, such as a flute and a bass guitar, are present in a program, the high energy bass frequencies can "use up" most of the power in a power amplifier leaving none for the high frequencies. The result can be severe clipping (distortion) of the high frequency material. By biamplifying the system with an electronic crossover, the high frequency material can be routed to its own power amplifier, which can be considerably smaller than the bass power amplifier and still avoid the clipping problem. This results in an effective increase in headroom that is *greater* than would be obtained by simply using a larger, single amplifier rated at the same power as the combined bass and treble amps in the biamped system.

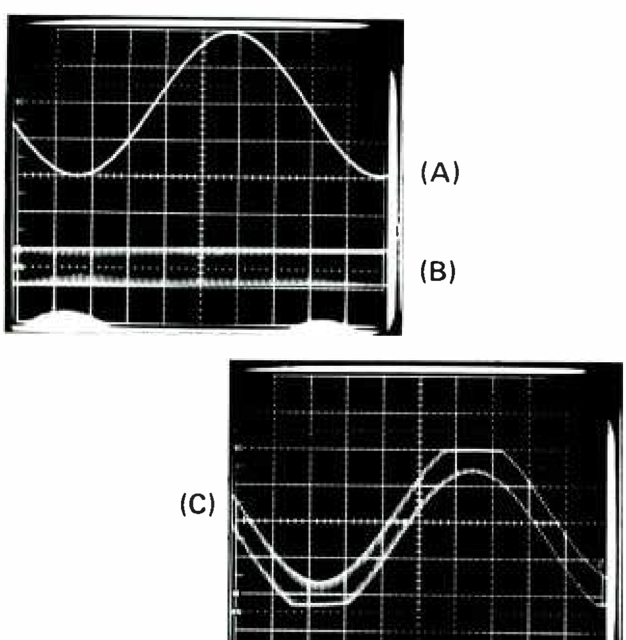
Figure 4A shows a low frequency waveform from a power amplifier output. The peak-to-peak voltage of the waveform is 127 volts, corresponding to 45 volts RMS. If this voltage were applied to an 8-ohm speaker load, the power level would be 250 watts.

Figure 4B shows a high frequency waveform from a power amplifier output. The peak-to-peak voltage, RMS voltage, and power into an 8-ohm speaker load are less than shown in Figure 4A and correspond to a 16 watt output into an 8-ohm load. The levels of these high and low frequency waveforms are fairly typical of musical content.

Figure 4C shows the effect of adding the signals of Figure 4A and Figure 4B, corresponding to a low frequency note and a high frequency note being played at the same time. Note that the total peak-to-peak voltage (which would be 56 volts if it were not clipped) is greater than the peak-to-peak voltage of either signal by itself. For an amplifier to produce this voltage into an 8-ohm load, it must be rated at 392 watts (power is proportional to voltage squared). Since the amplifier used for these examples is only capable of 250 watts, this waveform is clipped, especially the high-frequency component.

If the same two waveforms in Figure 4A and Figure 4B were reproduced by two separate amplifiers, the total amplifier power capacity needed would only be 266 watts (the sum of the two powers), not 392 watts. This power could be provided by one large and one smaller amplifier. Thus, using one large and one smaller power amplifier to produce these two waveforms reduces the total needed amplifier power capacity. Or, by using two larger amplifiers, there is a substantial increase in headroom.

FIGURE 4



Note that the actual power delivered to the speakers remains the same in the biamplified case as it would have been in the non-biamplified case. The advantages of biamplification are realized in increased headroom, and therefore in a reduced need for amplifier power *capacity*.

Features and Specifications

Once the "super power" decision has been made, the ultimate selection of one brand of amplifier over another must be made on the basis of cost, reputation, availability, and on the relative merits of each amplifier's features and specifications.

There are a wide variety of different features on the various brands of power amplifiers. Many add significantly to the price of the amplifier, yet can add significantly to its usability as well. Decisions need to be made on the merits of any given feature versus its cost. One way to determine the approximate cost of a feature would be to compare the amplifier with one not having the feature.

Balanced Inputs are useful for professional installations because a balanced line tends to reject outside hum and noise pickup better than an unbalanced line. However, most balanced inputs utilize transformers, and to use an *audio input* transformer on a power amplifier, with its large *AC power* transformer, means using a considerable amount of magnetic shielding to prevent the input transformer from picking up hum from the power transformer. It may be less expensive and more effective to use an amplifier with unbalanced inputs, and to build an input transformer box that can be located some distrance from the amplifier. By remotely locating the transformers, less expensive input transformers (having less magnetic shielding) may actually yield lower hum levels. Some manufacturers are offering XLR input connectors that lead to an unbalanced input circuit as an alternative. The XLR connector offers professsional quality and avoids accidental disconnection, but it costs far less than the combination of XLR connector and input transformer. Other manufacturers offer XLR input connectors and a socket for a high quality input transformer that can be added at extra cost.

VU Meters are useful when speaker power levels need to be constantly monitored, or when the two channels of a stereo amplifier must be accurately balanced. They also perform the simple function of indicating whether or not the amplifier is receiving a signal and delivering an output. Some manufacturers are offering "peak-reading" meters (PPM's), which allow an accurate reading of the amplifier's maximum power output and help the operator avoid overpowering a speaker system or clipping the waveform. A variation of the peak-reading meter is a strip of LED's which light successively to show increasing peak power outputs. These will not show variations as precisely as a meter (precision is limited by the number of LED's) but the LED's are probably less expensive. Other manufacturers are offering separate VU meter panels which can be mounted in a rack along with the amplifier. This gives the buyer the choice of buying or not buying the option.

Monophonic Operation is offered on some stereo amplifiers. This may be accomplished by a simple switch, by a circuit modification or by using a special type of input splitter transformer. In the "mono" mode, the two channels operate together in a "push-pull" configuration into a single load (speaker system). Since the effective output voltage is doubled, the potential power output, compared to a single channel, is quadrupled (power is proportional to voltage squared). The main

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limitation is the amplifier's power supply capacity. Another advantage of this mode of operation is that the output signal is balanced. Thus, in many cases, the amplifier can drive a balanced "constant voltage" speaker system. Depending on the size of the amplifier, this may be a 70-volt line, a 25-volt line or some odd rating.

Input Attenuators (volume controls) may be useful in some systems, and superfluous in others. In some systems, such as a commercial sound system where the installer doesn't want the operator to have access to "too many knobs", front-panel input attenuators may be more of a hindrance than a help. Some manufacturers offer "straight-through" amplifiers with no controls for these applications. Others offer simple linear potentiometers, and still others offer sophisticated dB-calibrated, stepped input attenuators. This last type of control can be very useful for a portable system that must be set up the same way night after night in different rooms and for different shows, or for a disco or studio monitoring application where the system may need to be "faded" and then brought back up accurately to the previous level.

Cooling Fans are built-in to some amplifiers, and other amplifiers have mounting provisions for them. This option is good for an amplifier that will be operated at high continuous power levels in a confined space, or in a high ambient temperature environment (such as a constricted portable rack, or in an outdoor "blockhouse"). Even if the amplifier can operate in the high temperature environment without the cooling fans, its lifetime may be increased by adding them. Some amplifiers that advertise "massive heat sinks" may have them physically located in a position that makes it very hard to direct airflow onto them from an external cooling fan.

Protection Circuits All modern amplifiers offer some degree of electronic protection. In some cases, however, the protection circuits protect the amplifier adequately, but may offer little or no protection to a loudspeaker load. In other cases, the protection circuits themselves may actually harm a loudspeaker in attempting to protect the amplifier! This last type of circuit reacts to an overload or to a highly reactive load by delivering high power spikes (transients) to the output stages which may sound like static or a faulty input cable. Other protection circuits offer a high degree of protection to both the amplifier and the loudspeaker load. Some amplifiers have delayed turn-on and fast turn-off circuits in the form of output relays or special semiconductor circuits. These help reduce the possibility of loudspeaker damage from turn-on/turn-off transients from the amplifier or any other electronic equipment connected to the amplifier. These same circuits may also disconnect the loudspeaker in the event of a large amount of DC voltage or excessive audio signals appearing at the output. DC voltage can be disastrous to a speaker voice-coil. Other amplifiers offer AC coupling at the input or a choice of AC or DC coupling at the input to allow protection against DC offset from a preamplifier, mixer or other device. Thermal protection circuits, which may be as simple as a thermostat that shuts down the AC power, should be smooth-acting and, when possible, should also be self-resetting. To prevent thermal shutdown during a performance, one manufacturer offers a thermal protection circuit that, rather than completely shutting down the amplifier, automatically limits the output power to 40% of normal.

Other Features Various warning lights, AC accessory outlets, multiple output connectors, plug-in preamplifiers, and so on, are useful in some situations, and are offered on some units. Other important "features" are the manufacturer's warranty, the overall "serviceability" of the amplifier, and its general physical ruggedness. Does the manufacturer provide easy access to parts and service information? Is there a warranty service center near enough to do you any good? Is the amplifier physically and electronically designed for easy servicing? Is it rugged enough to withstand the rigors of a concert tour? Watch out for power transformers that are poorly supported, and could physically pull off the chassis under extreme "g-forces". Similar considerations apply to internal components. Regarding serviceability, a plug-in circuit card would be infinitely more desirable than a circuit card held down by umpteen wires and located between a "massive heat sink" and an even more massive power transformer. Warranty and serviceability are intangibles until something goes wrong.

UL Approval can be useful for some installations, and may be required to meet bid qualifications in some cases. An amplifier that meets the UL specifications has passed a *very* rigorous set of tests with regard to its electrical and fire safety. Other governmental agencies, such as the City of Los Angeles, also offer safety approval services, and their stickers may be a requirement on other installations. These certifications say nothing about audio performance, however.

Specifications

Modern amplifiers, in general, have such high standards for their performance specifications that comparing them on that basis is a hit and miss proposition, at best. Yet without the aid of a set of expensive test equipment, most buyers depend on the manufacturer's specifications (or those provided by an independent test lab) as the only measure of relative performance available.

The question, then, is how good is good enough? Is an 0.1% T.H.D. rating good enough, or does the application call for a 0.005% rating? What does a 45 volt per microsecond slew rate *sound like?* And why is that better than 25 volts per microsecond? The questions are endless, yet valid, and the answers would be very useful to a potential buyer. The following ideas are attempts at some non-technical explanations of a few specifications. Where applicable, guidelines are presented for determining just what is good and what is bad. These are by no means "rules", however, since the importance of any specification depends heavily on the application and on the "ear" of the listener.

Types of Power Output Ratings

Peak power refers to the maximum undistorted power output of an amplifier. Most amplifiers cannot sustain their peak power ratings for long periods of time due to thermal and/or power supply limitations. Because there are many different methods of rating an amplifier's peak power, and because peak power is generally only a few dB higher than average power capacity, few amplifiers are rated by this method.

"RMS" power is actually a misnomer for *average power.* Average power is usually measured with a sine wave input signal, and is equal to the square of the amplifier's maximum RMS output voltage divided by its load impedance. Because RMS voltage is used in the formula, the resulting power rating is commonly called "RMS power". To be more accurate (in terminology), many amplifiers are now rated in watts of "continuous average sine wave power", which is measured in the same manner.

Professional power amplifiers, not sold for home hi-fi use, are not required to meet the power rating standard set by the FTC (Federal Trade Commission), a standard meant for consumer (hi-fi) power amplifiers. However, most professional power amplifiers have been designed for operation under severe operating conditions and would easily pass the FTC ratings. In addition, a UL rated amplifier has passed thermal tests that are probably considerably more severe than the FTC tests. Many amplifiers are UL-certified for 8-ohm operation, but are unable to pass the 4-ohm UL tests due to their severity. However, these amplifiers may work well into 4-ohm loads under normal operating conditions.

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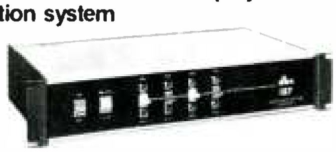

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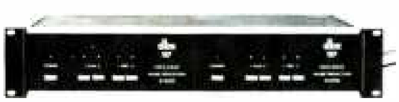

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Power Output at Clipping

This rating commonly indicates the maximum peak output capabilities of an amplifier at 1 kHz with an 8-ohm load. The amplifier may or may not be able to produce this amount of power at frequencies near the ends of the audio spectrum, far removed from 1 kHz.

Power Output versus Load

Within their power supply limits, most power amplifiers act like perfect voltage sources, that is, their power output rises linearly with decreasing load impedance. When the load impedance drops below a given value, the amplifier's protection circuits may begin to limit the power. This specification is often given in graph form.

Distortion

There are many different forms of distortion, and comprehensive distortion ratings offer a means to compare the performance of different amplifiers. At the same time, distortion ratings are among the most confusing when it comes to making an actual buying decision. Perhaps the confusion is even greater because there seem to be more factors affecting the sound than those types of distortion commonly specified.

Harmonic distortion, is characterized by the appearance at the amplifier output of harmonics of the input waveform which were not present in the original input waveform. *Total Harmonic Distortion*, or T.H.D. is the sum total of all of these unwanted harmonics expressed as a percentage of the total signal.

Harmonic distortion, in an amplifier, can be created in any of several ways. The T.H.D. rating of a power amplifier refers to creation of unwanted harmonics by the amplifier during "linear" operation (normal input and output levels, impedances, etc.). Harmonic distortion is also created by "clipping", a form of "non-linear" operation, which occurs when the signal level at an amplifier's input is high enough to drive the amplifier beyond its rated maximum output. The amplifier, in attempting to reproduce this signal, reaches its maximum output voltage swing before it reproduces the top of the signal waveforms. Since the output voltage cannot rise any farther, the tops of the waveform are "squared off", or clipped. Clipping distortion adds odd upper harmonics (3rd, 5th, etc.) to the original signal.

Another form of harmonic distortion that occurs in some power amplifiers is called *crossover (or notch) distortion.** Crossover distortion can be caused by improper bias in the output transistors of an amplifier. The *amount* of crossover distortion products remains the same whether the signal is large or small, so the *percentage* of distortion goes *down* as the signal level goes *up*. Thus, an amplifier with crossover distortion may sound relatively distortion free at high output levels, yet sound "fuzzy" at low levels. Some amplifiers have internal adjustments which enable a service technician to control the amount of output transistor bias, and therefore control the distortion. Others have automatic biasing circuitry which needs no adjustment and minimizes crossover distortion under normal operating conditions.

One trick in interpreting harmonic distortion ratings is deciding which of the above types of distortion the rating refers to. For example, crossover distortion can be considerably more irritating than simple production of harmonics of the input waveform. And, in general, higher harmonics (above the 7th or thereabouts) are also more irritating than lower harmonics, and odd-order harmonics (3rd, 5th, etc.) are more irritating than even-order harmonics (2nd, 4th, etc.). This is because the lower harmonics, especially the even-order 2nd harmonics are "musical" in nature, and simulate naturally occuring harmonics from a musical source. The difference in the type and level of these different harmonics in the distortion products of different amplifiers is one reason why two amplifiers with the same distortion ratings may sound very different indeed.

Intermodulation distortion, or I.M. is characterized by the appearance in the output waveform of frequencies that are equal to the sums and differences of integral multiples of two or more of the frequencies present in the input signal. The difference between intermodulation distortion and harmonic distortion, is that two or more different frequencies must be present to produce intermodulation distortion (only one frequency is needed for harmonic distortion to appear), and that intermodulation distortion products are not always harmonically related to the original frequencies.

Intermodulation distortion may be more irritating than harmonic distortion since it produces distortion products that are not harmonically related to the input signal. Some manufacturers are now giving low-power I.M. ratings as well as high power ratings. This practice should probably be encouraged and extended to harmonic distortion, as well, since much of the output of a sound system may take place at very low amplifier power levels, as shown by Figure 1.

Another interesting question about distortion is just how much distortion can actually be heard? Realizing that some types of distortion are more "hearable" than others, it still is questionable whether an 0.01% distortion rating is actually audibly better than an 0.1% distortion rating. Harry F. Olson in *Acoustical Engineering* (D. Van Nostrand and Co., Inc.,) states the results of tests which showed that distortions as low as 0.5% were audible to critical listeners when background noise levels were low and the sound system bandwidth was wide (allowing high frequency distortion products to be heard). Whether or not lower distortion figures than indicated by this study are audible in certain situations is open to question.

Frequency Response

The *frequency response* of a power amplifier describes the variation in its output signal level as the frequency is changed while the input signal level is held constant. A "flat" frequency response curve is an indication of an amplifier's overall quality and its ability to respond to upper and lower harmonics of audio frequency signals.

Because extreme stability is necessary for some types of commercial sound applications, notably 70-volt lines, some manufacturers of commercial sound amplifiers restrict frequency response or allow relatively high distortion in return for increased amplifier stability. However, many of the newer "super power" amplifiers can, to a considerable degree, be said to embody the "best of both worlds" because they have good frequency response, low distortion and high stability under difficult loads.

Offset Voltage

This specification indicates the amount of DC voltage naturally present at the output of the amplifier. A high DC voltage could damage the loudspeaker load. Offset voltage is a rare specification, but could be valuable information to a prospective buyer.

Unit Step Function Response

A unit step function is like the leading edge of a square wave which goes up, but never comes down. The response to this input indicates the output of the amplifier for a DC input signal which might come from a faulty, direct-coupled preamplifier or mixer. The response to a unit step function will be another unit step function for a DC coupled amplifier, but will look something like a single pulse of a sagging saw-tooth wave

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**"Crossover", in this case, refers to the transistion between the positive half and the negative half of the output voltage waveform in a "push pull" class B or AB power amplifier; it has nothing to do with the crossover used to divide frequencies in a speaker system.*

The Peavey CS Series

Last year when Peavey introduced the CS-800 Stereo Power Amp, professional sound men and engineers acclaimed it as the most versatile high performance power amp available for under $1,500.00.

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Peavey Electronics, Corp. / Meridian, Mississippi 39301

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for an AC coupled amplifier. Again, this is a rare specification, but is of considerable interest in protecting loudspeakers.

Power Bandwidth

The *power bandwidth* of an amplifier is a measure of its ability to produce high power output over a wide frequency range. The limits of the power bandwidth are those points where the amplifier can only produce 1/2 the power that it can produce at 1,000 Hz. While the frequency response is measured at relatively low power output (1 watt), the power bandwidth is measured at the amplifier's full power output (before clipping).

Phase Response

The *phase response* of an amplifier is a measure of the amount of time delay it adds to different frequencies. An amplifier with perfect phase response would introduce equal time delay at all frequencies reproduced.

An amplifier with poor phase response would change the shape of a waveform that was made up of a fundamental frequency and several harmonics by delaying each harmonic differently. It is difficult to say just how much phase shift is tolerable. One audio engineer says that I can demonstrate that phase shift can cause very audible differences in the sound, yet he claims he can demonstrate just as well that phase shift causes absolutely no audible difference!

Channel Separation

This specification indicates the output from one channel when a signal is fed to the other channel.

Hum and Noise

Hum or noise from a power amplifier disrupts a program, and is irritating to a listener. Both poor separation and high hum and noise levels could be considered forms of distortion.

Rise Time

Rise time is a measurement of the amount of time an amplifier requires to respond to a square wave input signal. The rise time of an amplifier is an indication of its high frequency response. A fast rise time corresponds to a wide frequency response. A common way to measure rise time is to use a 1,000 Hz square wave output signal of one volt peak-to-peak amplitude. The rise time is the time the amplifier requires to change from 10% (0.1 volt) to 90% (0.9 volt) of its output. To improve measurement accuracy, the first and last 10% are normally not included in the test (any slight non-linearities that occur in the test signal or the amplifier could lead to measurement error).

Slew Rate

Slew rate is a measure of the ability of the amplifier to follow a fast-rising waveform at higher frequencies and higher power outputs than are used to measure the rise time.

It might seem reasonable to assume that the fastest slew rate for an audio waveform occurs at 20 kHz. However, this is not the case. When one frequency is superimposed upon another, the combined waveform has a slew rate that is greater than the slew rate of either signal by itself. The actual value of the slew rate of one of these waveforms (or any waveform) depends not only on the frequency, but on the amplitude of the waveform as well. Thus, the criteria for a good slew rate specification, which indicates that an amplifier can reproduce these combination waveforms, varies with the maximum power output capability of the amplifier. The higher the power, the higher the required slew rate.

The question, again, is what does a good or bad slew rate do to the *sound* of an amplifier, and again, this does not have an easy answer. Changes in sound from a poor slew rate specification (a slew-limited amplifier) would probably come at high power outputs with complex program material and would be most noticable at high frequencies.

Input Sensitivity and Gain

An amplifier's *input* sensitivity indicates the input drive voltage needed to produce its rated output.

Gain (from the IEEE definition) is the ratio of an amplifier's output *power* to its input *power*, usually rated in dB. Many manufacturers specify the *voltage* gain of their amplifiers converted to a dB value (dB = 20 log voltage gain). The difference is important, and the method of gain calculation should be given along with the specification.

Output Impedance

The *output impedance* of an amplifier indicates its ability to act like a perfect voltage source. An amplifier with a very low output impedance is a good approximation of a perfect voltage source and will deliver increasing power levels into lower impedance loads in a linear fashion (according to Ohm's Law).

Damping Factor

Damping factor is a term that is derived by dividing the load impedance (speaker or other load) by the amplifier's output impedance. Thus, a high damping factor indicates a low output impedance at a specified load.

The cone/voice-coil assembly of a loudspeaker gains inertia during its back and forth movements. This inertia can cause it to "overshoot", that is, to continue movement in one direction, even when the amplifier is trying to pull it back in the other direction. An amplifier with a low output impedance can "damp" (reduce) unwanted loudspeaker motions, as explained below.

During the "overshoot" movement, the voice coil of the loudspeaker interacts with the loudspeaker's magnetic assembly to produce a voltage called "back E.M.F." (electromotive force). This action is similar to the operation of a dynamic microphone. If the amplifier's output impedance is low, this "back E.M.F." voltage is shunted through the amplifier's output circuits to ground, and back to the voice coil. Since the path from the voice coil, through the amplifier's output circuits, and back to the voice coil is a complete circuit, a current flows in the voice coil. This current, causes the voice coil to act like an electro-magnet; the electro-magnet (voice coil) interacts with the magnetic assembly of the loudspeaker, and the unwanted overshoot is reduced (a magnetic braking action).

If the amplifier's output impedance is low (considerably less than the impedance of the loudspeaker voice coil), this damping action is limited only by the resistance of the voice coil combined with the resistance of the speaker lead wires. While the value of a high damping factor in reducing cone overshoot is disputed, a high damping factor is evidence of good overall engineering design.

In addition, in many cases, the damping contributed by a properly designed speaker enclosure is far more effective in controlling unwanted cone movement (overshoot) than the damping ability of the power amplifier.

Types of Amplifiers

There are a number of new technologies being applied to consumer power amplifiers which, as they mature, will certainly be applied to professional designs as well. At least one type, the Class D or switching amplifier, is already available from one manufacturer in a professional model.

Classes

The "class" of an amplifier describes the operation of its output stages:

Class A amplifiers are designed such that current in the output

Nashville's Hottest Group

Claude Hill, the rotund lead vocalist for Audio Consultants was recently asked why he felt his group has had such phenomenal success. To quote Claude: "In addition to giving our public all the standards we have come to know so well, my group has continually worked out new material. You know, fresh approaches to the same old tunes. I'm no prima donna either. I couldn't do it alone. I count on everybody in Audio Consultants to hold their own. It's real team work."

"Regardless of the gig, after we perform there's a real sense of joy—a job well done."

Well said, Claude. Like anything worth waiting for, we'll just have to see if this group can hold on to their position at the top. They know sound like no other group in the business and if what they've already achieved is any indication of the future, then keep an eye (and an ear) on Claude & his Audio Consultants.

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Don Woerner is the man in Dallas.

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stages flows all the time. A class A amplifier stage operates essentially over a linear portion of its characteristic. Most preamplifier, and predriver stages are class A, and some newer consumer amplifiers offer class A operation for the output stages.

Class B amplifiers are designed such that current in the output stages flows only part of the time (usually about half the period of a sinusoidal input signal). Because an audio amplifier obviously requires amplification of an entire input waveform, class B audio amplifiers have two output stages, operating in "push-pull". One stage amplifies the "top" of the waveform, the other amplifies the "bottom" of the waveform.

Class AB amplifiers are designed so that current flows in the output stages for slightly more than one-half of the time for a sinusoidal input signal. In a push-pull configuration, a class AB amplifier avoids much of the crossover notch distortion that occurs in a push-pull class B design during the transistion from operation of the "top" half of the push-pull circuit to the "bottom" half.

Class C amplifiers also operate only over a portion of the waveform, usually less than half of a sinusoidal input. Thus, even in a push-pull configuration, class C amplifiers do not have any direct application to audio. However, one manufacturer now offers an amplifier deemed to be "*Class G*", and another offers an amplifier designed for "current dumping". Both of these amplifiers operate using output stages that are similar to class C designs.

Class D amplifiers are relatively new, although the concept is not new. The audio signal to a class D amplifier controls a pulse-width-modulation circuit which, in turn, controls a very high frequency switching power supply. The switching power supply directly feeds the speaker load through an audio low-pass filter that removes the "carrier" frequency (the frequency at which the power supply operates).

A well designed push-pull amplifier inherently cancels even-order distortion harmonics. A class B amplifier is as high as 78% efficient. Thus, most professional high power amplifiers are push-pull class B (or class AB) designs. The preamp and driver stages are usually class A because a class A amplifier, operating entirely in its linear range, and not in a push-pull configuration, does not have to be compensated for crossover distortion. The reasons given for the use of class A output stages in some new consumer amplifiers is that the class B and AB designs require non-linear circuits to compensate them for crossover distortion, and that these circuits may add more distortion than the push-pull configuration cancels. In addition, it is claimed that the compensation circuits in a class B or AB design, which may involve large amounts of negative feedback, can cause extra distortion in preamp stages at high frequencies (where the feedback loop cannot respond). This may be similar to the so-called "TIM" distortion. In any event, a class A amplifier will never be more than 25% efficient (with an output transformer, a class A amplifier can achieve 50% theoretical efficiency) and thus they are not practical for professional amplifier outputs. With the top 25% efficiency, a class A amplifier would produce 3 watts of heat for every 1 watt of output power! Even with an expensive output transformer, the efficiency would still be lower than a class B or AB design.

Some hope for improved efficiency along with controlled low distortion lies in the class D switching amplifier. Its high efficiency results from the fact that there is never any current flow in the output stages in the absence of signal. Its low distortion could be compared to a computer "error rate" which might be reduced merely by increasing the number of "bits per second". Most of the common distortion-causing circuits and components in a conventional analog amplifier are simply not there in the class D amplifier. The major drawback, at this point, is cost, and that should be solved in time through higher production rates and better technology. In addition, due to the extremely high frequency, high voltage pulses generated in a class D amplifier, there is some chance for a kind of distortion that would be similar to slew limiting.

Transistors vs Tubes vs VFET's vs IC's

Except for a few low-power paging quality amplifiers, today's professional power amplifiers use bipolar (conventional) transistors. A few use integrated circuit front-ends, and there are any number of older tube-type amplifiers still around. Consumer amplifiers, on the other hand, are offered with tubes, bipolar transistors, vertical field-effect transistors, integrated circuits or just about any combination. Again, some of these technologies may be showing up in professional amplifiers as soon as their cost drops a little.

Our faithful companion, the electron tube, or "valve" as my British friends would say, has just about had it. Tube-type amplifiers had some definite advantages, however. Their distortion characteristics were "different", and for this reason, they could sound "nicer" on certain program material. They were usually easy to repair, because the most common malfunction was a bad tube. In general, most tube-type designs were well enough established that they had had all the "bugs" worked out of them, and most servicemen were familiar with them.

Unfortunately, a super-power tube-type amplifier tends to be rather large and produces a great deal of excess heat. Therefore, their uses are pretty well limited to permanent installations. A carefully designed transistor amplifier, on the other hand, has an indefinite lifetime which, considering the high cost of labor these days, is a real advantage to the non-technical buyer. Finally, the quality and availability of original and replacement tubes is diminishing rapidly, partly because of the advent of all solid state television. So goodbye to our old friends. However, at least one new technology, the vertical field effect transistor or "VFET", claims to be the replacement technology for tubes. Claims for the VFET include little or no saturation at high voltage output, no thermal runaway, low output impedance, no carrier storage and the possibility of designs using very small amounts of negative feedback (which, all of a sudden, seems to be an enemy, rather than a friend). Assuming all these claims are true (and truely advantageous) the VFET will probably appear in a professional amplifier, again as soon as the cost comes down.

Bipolar transistors, of course, are far from dead in conventional designs, and may be the best choice for the newer switching designs as well. Integrated circuits are becoming more popular for preamplifier stages, but are not used for output stages except in some low power consumer devices such as car radios. Even the higher-power hybrid IC's are not yet cost-competitive with discrete designs, partly because of the difficulty of adequate heat dissiaption from an IC package.

How Much Power is Too Much?

Regardless of what class or type amplifier, there is an upper limit to the desirable power output for a given application, as well as a lower limit. Decisions need to be made on the basis of maximum SPL allowable (thinking about legal aspects, noise pollution and hearing conservation), speaker power capacity and the available budget for amplifiers.

There is enough data available to persuade almost anyone that sustained high SPL levels can cause permanent hearing loss. Unfortunately, rock music fans are still subjected to incredible SPL levels at some concerts, and often go away shaking what's left of their heads in disappointment. That's something for promoters, musicians and sound system owner/operators to think about. If these fans are staying away from concerts they might enjoy simply because of the SPL level, then that is hurting the *pocketbooks* of those promoters, musicians and owner/operators, as well as the ears of the fans. Enough.

The only comment that needs to be made about the budget is that it is always a good idea to double check all calculations used to determine needed amplifier power, and the data used in those calculations. Make sure that amplifier ratings are adequate, yet not excessive for the application. A "small" 3 dB error would lead to the purchase of twice as many amplifiers as needed (or half as many!).

Speaker system limits may be the most difficult to determine, and to plan for. Speaker system ratings are varied and it's hard to interpret them for one manufacturer, much less to compare the ratings between two different manufacturers. However, there are a few general ideas concerning speaker system power capacity that may aid a decision about how much power is too much.

Loudspeakers, whether cone-type or compression drivers, have two predominant failure modes. Thermal failure happens when too much *average* power reaches the speaker voice coil, causing it to burn up. The other type of failure, mechanical failure, happens when too much *peak* power, usually at a low frequency, reaches the speaker, causing overexcursion. Mechanical failure may be evidenced by torn speaker cones or suspensions, or torn or shattered compression driver diaphragms, or voice coils that have come loose from their mountings.

For a given SPL, a low-frequency signal requires greater voice-coil excursion than a high-frequency signal. This means that for a high-frequency driver, frequencies below its rated cutoff may cause overexcursion. Thus, for a high-power system, higher crossover frequencies are in order. As a rule-of-thumb, the power rating of a driver must be divided by four each time the crossover frequency is lowered by an octave. In other words, for a driver rated at 20 watts of pink noise from 2,000 Hz to 20 kHz, using a 1,000 Hz crossover frequency would mean that a maximum of only 5 watts could be safely delivered to the driver. This rule does not necesarily work in reverse. In other words, the example driver will not necessarily handle 80 watts of pink noise above 4,000 Hz. Woofers have similar low-frequency limits, although for excursion these limits are largely determined by the type of cabinet design. A properly designed woofer cabinet can do a lot to save a woofer from mechanical failure.

In some cases, a higher power amplifier may actually cause less abuse to a speaker than a low power amplifier. If the low power amplifier is forced into clipping on high power musical peaks, the tops of the clipped output waveforms are similar to a square-wave signal. The speaker cone (or drive diaphragm) is forced to move in one direction, then *stay there* for as long as that portion of the waveform is clipped, then move back in the other direction and *stay there* as long as that portion of the waveform is clipped. While the speaker cone or driver diaphragm is hanging out there in one place, it cannot produce any sound, yet it is still receiving energy from the amplifier. This energy becomes heat, and may result in thermal failure. Since a higher power amplifier would be less likely to clip the waveform, the possibility of thermal failure is reduced. On the other hand, the high power amplifier may be able to push the speaker cone or driver diaphragm beyond its excursion limits, so it's a good idea to consider both possibilities.

End

In general, the overchoice available in super-power audio amplifiers can be a good thing for an informed consumer, and with competition and technological changes being what they are, there is probably no end to the future possibilities. Have a good time.

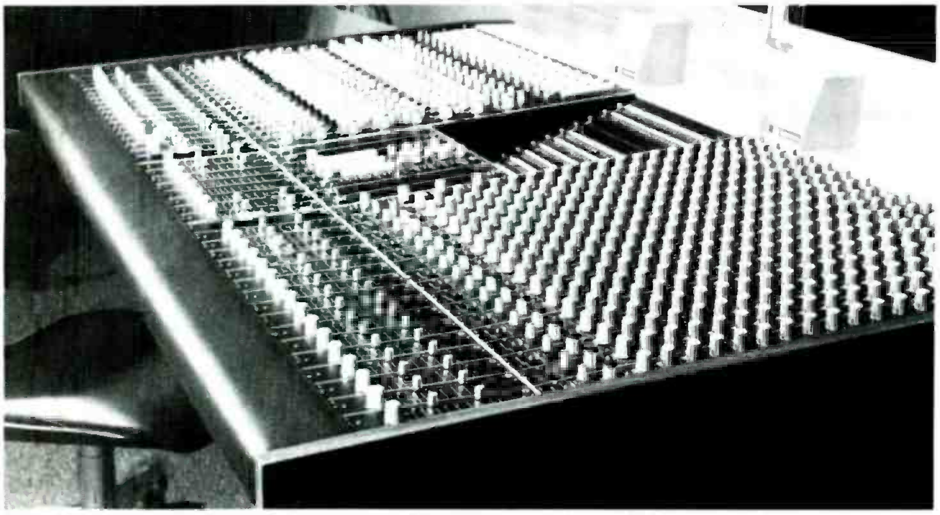
Gratis

Special thanks to Deane Jensen, of Hollywood, and Don Davis, of Synergetic Audio Concepts, for their help in editing.

About the Author

Chris Foreman is an audio consultant who works closely with Gary Davis and Associates, of Topanga, California, a technical writing firm.

New Products



AUDITRONICS 600 SERIES CONSOLE

Auditronics, Inc., has announced its 600 Series, a new line of audio control console specifically engineered for use in the arts. Filling the range of needs from theatrical sound effects to sound reinforcement, A.V. and television production, the 600 is said to offer the user a tool designed with special requirements of "in performance" situations in mind.

The completely modular consoles offer two mike and line inputs per each input channel. Extensive facilities are provided for multiple external effects devices, multi-track recording simultaneous with effects or reinforcement use, and complete interfacing of the console's monitor and talkback circuits with the house headset intercom system. Separate output channels for foldback, paging and reinforcement are standard, with an additional 16 x 16 channel preset matrix for effects outputs. All faders are DC controls, allowing easy interface of console with synthesizers, computers and a wide range of control and effects-generating devices.

Offered in a standard version of 16 input by 16 output channels, the console may be expanded to 32 x 32 or ordered in configurations down to 8 x 8 with provisions for easy future expansion.

Designed by theatre professionals and backed by Auditronics' many years of proven performance in the recording and broadcast fields, the 600 offers the first full-featured alternative for the needs of the live performing arts.

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MEMPHIS, TN 38104
PHONE: (901) 276-6338

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RUGGED NEW ELECTRET CONDENSER MICROPHONE ANNOUNCED BY ELECTRO-VOICE

The new Model 1776 microphone being introduced by Electro-Voice, Inc., Buchanan, Michigan, is a ruggedly built cardioid condenser with all the fine performance characteristics of a professional condenser. It is eminently suited for live-performance sound reinforcement and superior quality recording.

The 1776 is designed for both hand-held and stand use. Built with a strong die-cast and machined case and tough metal screen protecting the head, the microphone will withstand the most severe handling and use — so is ideal for on-the-road applications. Since the 1776 electret has a permanently charged element, the microphone needs no bulky and troublesome electronics for powering, making it a highly reliable and easy-to-use instrument.

The electret condenser generating system is mechanically uncomplicated adding to its reliability and high quality performance. Frequency response of the 1776 is 60 to 18,000 Hz; its transient response, or reaction to sudden tone burst such as a drum beat, is excellent; it has exceptionally high output resulting in more sound output from the amplifier and also allowing the 1776 to be used in-

to almost any tape recorder input. The microphone is a Single-D cardioid, according to the company, with good off-axis reduction of sound pickup. Using the microphone close up is said to emphasize bass tones, a sound desired by many vocalists. The 1776 has excellent gain-before-feedback characteristics making it superior for difficult sound reinforcement jobs.

The Electro-Voice Model 1776 condenser microphone has on/off switch, is finished in non-reflecting gray, is furnished with a 15-foot professional cable and stand clamp. User net price is $99.00. It can be purchased with 25-foot professional cable with 3-pin connectors on both ends as Model 1776P at $105.00. (Suggested prices slightly higher in Western states.) The microphones were engineered and are being manufactured totally in E-V United States facilities.

ELECTRO-VOICE, INC.
600 CECIL STREET
BUCHANAN, MICHIGAN 49107
PHONE: (616) 695-6831

for additional information circle number 66

ORBAN/PARASOUND NEW MODEL 622 PARAMETRIC EQUALIZER

Encompassing all of the features of the field-proven Model 621 series, the new 622 offers in/out switches on each of its four bands, standard balanced input with output transformer option, extensive RF shielding, and 115/230 volt, 50-60 Hz AC power supply included in the package.

A new proprietary parametric band-pass filter is virtually immune to the effects of control wear, and complements their unique "constant Q" design by permitting -40 dB notches to be consistently obtained. THD has been reduced to less than 0.025%, 20-20,000 Hz @ + 18 dBm.

All other features of the 621 series are retained: four cascaded sections, each with non-interacting, continuously variable center frequency, bandwidth, and amount of boost or cut. Each section tunes over a 25:1 frequency range, with broadly overlapping coverage for maximum flexibility. An overload light is provided which monitors all potential overload points in the circuit, and overloads can be easily corrected with the integral gain control.

The 622 will be offered in single and dual-channel configurations on a 3½" x 19" rack mount.

For more information contact:

PARASOUND, INC.
680 BEACH STREET
SAN FRANCISCO, CA 94109
PHONE: (415) 673-4544

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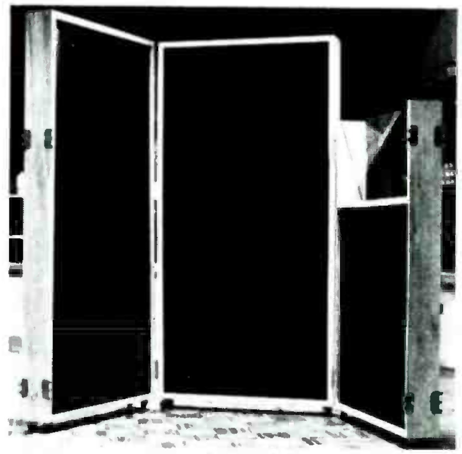
PORTABLE SOUND BAFFLES FROM SUGARLOAF VIEW

Sugarloaf View, Inc., introduces a series of interlocking studio "Gobos". Separation is better than .85 NRC. Two



widths are offered – 3'-0", and 3'-6" in all three models. Smaller models are designed for drum platforms and seated instrument positions – sloping plastic top portions provide vision and variable sound reflection. All models are on high quality brass casters and utilize an interlocking hinging system for system connection and support without "legs". Durable washable fabrics are available in many colors and fireproofed if required.

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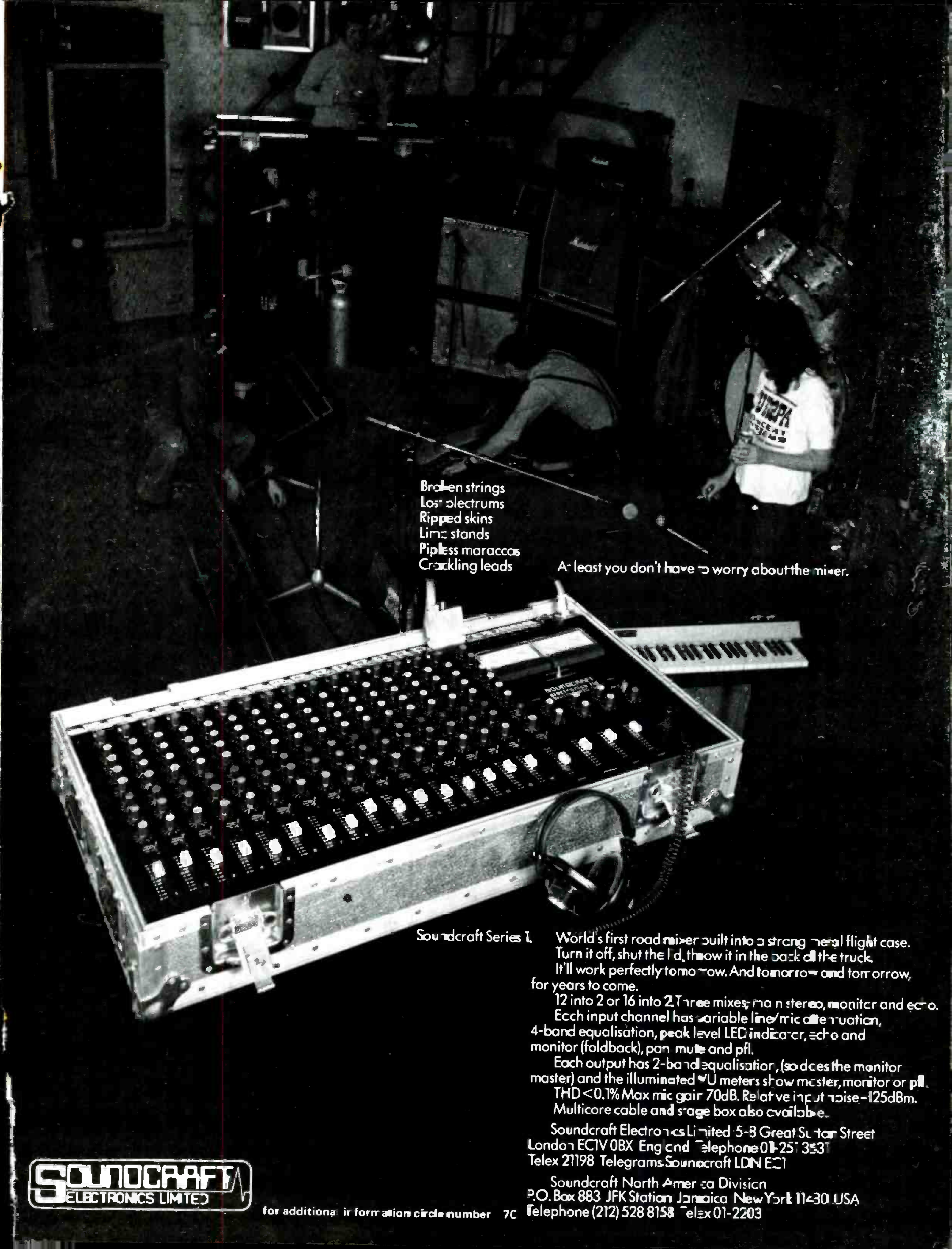
SYNARE PERCUSSION SYNTHESIZER

The Synare Percussion Synthesizer, first in a family of revolutionary electronic music instruments from Star Instruments, is the first fully integrated percus-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At least you don't have to worry about the mixer.
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World's first road mixer built into a strong metal flight case.
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It'll work perfectly tomorrow. And tomorrow and tomorrow, for years to come.
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sion synthesizer. Synare is a performance oriented instrument that provides the drummer with unlimited sound possibilities.

The four percussion pads can be tuned like a timpani or Roto-Tom to any desired pitch or timbre range. Striking a pad will produce the sound that has been set by the front panel controls. The pads are "zone sensitive" allowing the performer to dynamically shape and control the sound as he plays. The synthesizer has four sound sources (VCA, white noise, pink noise, ring modulator), full mixing, a wide range voltage controlled filter, and dual voltage controlled amplifiers. Synare mounts on a standard tom-tom stand and may be positioned to be compatible with the arrangement of the drum set.

Suggested retail is $795.00.

STAR INSTRUMENTS, INC.
DEPARTMENT R
BOX 71
STAFFORD SPRINGS, CT 06076
PHONE: (203) 684-4421

for additional information circle number 71

ASHLY CONSOLES FEATURE VERSATILE EQUALIZATION

Ashly Audio has introduced a line of high quality, portable mixing consoles designed for sound reinforcement or recording.

The S.E. series combines all signal processing components for a complete sound system in a rugged, portable modular package. The mechanical design allows the use of the lightest possible materials without sacrificing durability or strength. A 32-in, 8-out console weighs only 130 lbs.

The signal path of the console is designed to be as clean and quiet as theoretical limits will allow. State-of-the-art integrated circuit amplifiers in plug-in sockets are used where possible without compromising performance requirements. Hybrid or discrete circuits are used in noise critical areas.

The S.E. input is designed for extreme dynamic range (126 dB) and includes a shielded input transformer, low noise mike preamp, and LED peak overload indicator.

Input equalization is provided by a 3 band semi-parametric equalizer. Said to be much more flexible than fixed or selectable frequency types, this equalizer allows continuous adjustment of both amount and frequency of equalization. Range is ±15 dB, equalization curves are broad, constant "Q", precise reciprocals to avoid "peaky" tone control sound.

The specially designed summing amplifiers used in the S.E. series have a noise level within a few dB of theoretical limits, making 100 dB signal to noise ratios a practical reality in actual performance.

A peak limiter maintains close control of sub-master levels, has adjustments for attack, release, and ratio, and may be bypassed if not needed.

Tri-colored LED arrays are used to meter the console outputs. These meters can read true peak, or average level.

Four band parametric equalization is used for sub-masters and outputs. Each band provides ±15 dB amplitude, 50:1 fequency range, and bandwidth adjustment from a broad 3-1/3 octaves to 1/20 octave for true narrow band filtering.

Of special interest is an all solid state preview-solo system with independent level control for monitoring individual inputs in true stereo position.

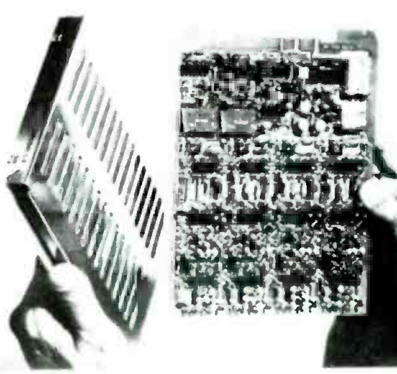
The S.E. series is available in studio and sound reinforcement formats and as a multiple-mix stage monitor system.

ASHLY AUDIO, INC.
1099 JAY STREET
ROCHESTER, NY 14611
PHONE: (716) 328-9560

for additional information circle number 73

TELEFUNKEN NOISE REDUCTION SYSTEM READY FOR DELIVERY

After three years of intensive development efforts, Telefunken has started deliveries of its unique Telcom c4 Noise Reduction System.



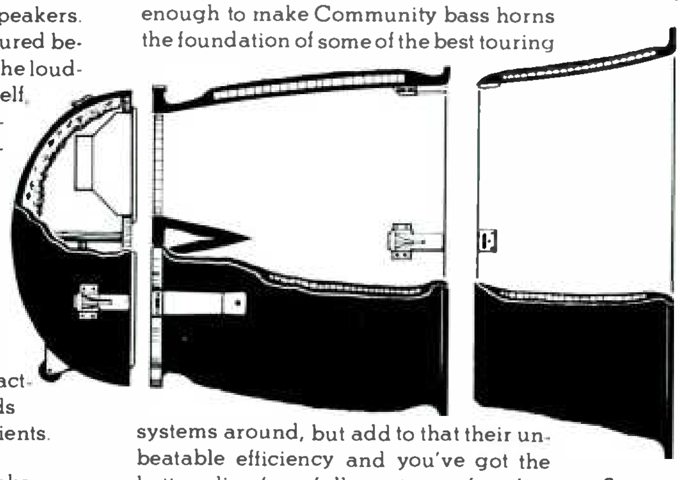
for additional information circle number 72

LEVIATHAN BASS HORN

This is the legendary Leviathan, our fiberglass bass horn for two 15" loudspeakers. It comes in three sections as pictured below: the back pod which houses the loudspeakers, the 48 Hz flare horn itself, and the optional extension for increased frequency range, projection and efficiency.

Not shown are our other bass horns: the FRC/B, designed to provide true horn performance in the smallest possible package, and the aptly named BLT, or Bass Long Throw, which does exactly that over several hundred yards with the closest attention to transients.

Like everything else that we make, our Levi, FRC/B and BLT are rock solid, portable, and built to last. That's reason enough to make Community bass horns the foundation of some of the best touring



systems around, but add to that their unbeatable efficiency and you've got the bottom line for a full spectrum of professional applications.

What does efficiency mean? Because of our design criteria any Community bass horn's output is typically 4-6 dB above its wooden competitor's. To you, the professional sound person, this means that you need fewer bass horns to fulfill your requirements and, consequently, less drivers and electronics to power them. In addition, our bass horns weigh thirty to forty percent less than the old wooden horns meaning an additional savings in reduced installation and freight charges.

Need a couple of bass horns? See your Community dealer. You might only need one.

SPECIFICATIONS	EXTENDED LEVIATHAN	BLT	FRC/B
Flare Rate	48 Hz	52 Hz	66 Hz
Operating Range	from 50 Hz	from 60 Hz	from 75 Hz
Driver	Two 15"	One 15"	One 15"
Size (HEIGHT/WIDTH/DEPTH)	43¼"/69¼"/64"	44"/44"/56"	30½"/40"/44"
Weight (less drivers)	175 LB	90 LB	65 LB

Community

COMMUNITY LIGHT & SOUND, INCORPORATED □ 5701 GRAYS AVENUE, PHILA, PA 19143 □ (215) 727-0900

for additional information circle number 74

Telcom is a compressor/expander (compander) which applies brand new and patented (No. 3,969,680) methods to the production of a stabilized gain controlled amplifier system. It divides the audio signal into four separate pass bands and applies the proper attack/release behavior to each of these bands separately, a process which has evolved over the last twelve years as, it is said in company literature, to be the best approach to compander design. Telcom then produces a constant compression factor over the entire dynamic range which assures that the expander will always track the compressor properly, even if the expander is not closely aligned to the level of the compressor. This makes it possible for the first time to use a four-band system in long lines transmission, sattelite communications and other areas where such alignment cannot be guaranteed. The 30 dB of dynamic range improvement is an enormous step forward in making the tape medium meet the capability of the disc record.

In its initial release, the Telcom c4 is being sold as a retro-fitting card for units manufactured by Dolby Laboratories such as the 360, 361, and the multi-track units of the MH series. Re-alignment of the system is not necessary when switching over from the cat.22 card to the Telcom c4D.

The Telcom c4D card sells for $700 and is available through Gotham's franchised professional audio dealers.

GOTHAM AUDIO CORPORATION
741 WASHINGTON STREET
NEW YORK, NY 10014
PHONE: (212) 741-7411

for additional information circle number 76

WESTLAKE AUDIO HEADPHONE MULT BOX, MODEL 1200

The HPM 1200 provides source selection (off-stereo, cue 1, cue 2) and level control for four seprate headphones.

This compact box measures 4.5 x 2.5 x 7.5 inches (11.5 x 6.4 x 19.1 centimeters) and weighs less than 2 pounds (4.4 kilograms). For user convenience, both phone and XLR connectors are supplied.

The HPM 1200 utilizes total printed circuit construction and is complete with an output multing connector for easy system expansion and two watt current limiting resistors for protection of voltage sources.

The box can sit on its rubber feet on any flat surface, snap on to mike or music stands with optional clamps provided, or be wall mounted using convenient screw slots on the rear of the box.

Priced at $189.00, the unit is available for immediate delivery.

WESTLAKE AUDIO
6311 WILSHIRE BOULEVARD
LOS ANGELES, CA 90048
PHONE: (213) 655-0303

for additional information circle number 77

MXR: ANALOG DELAY

MXR Innovations, Inc., recently introduced their new Analog Delay system. Designed to electronically perform the functions normally associated with tape or disc echo units, the MXR Analog Delay provides the user with variable delay times from 33 to 500 milliseconds, with a dynamic range of 80 dB.

A regeneration control provides multiple echoes with varying decay times from a single repeat to the point of feedback. A mix control allows any combination of dry and delayed signals. Special low-noise circuitry ensures quiet operation in the most critical of applications.

for additional information circle number 75

IF YOU'RE LOST

Everytime you search for program material on your tape machine? Let the El-Tech Take Finder find it for you!

'The Take Finder'

Shown Actual Size

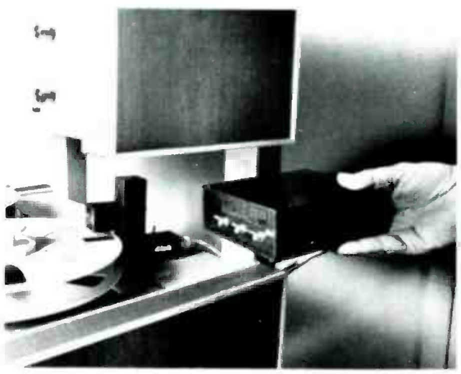
Locating program material on tape machines has been a problem since the early wire recorders. To overcome this problem on expensive multi-track machines, manufacturers have recently provided a remote digital readout which indicates exact tape location. Unfortunately these readouts have not been available for most machines since they were designed specifically for these recorders. Now, the El-Tech Take Finder gives the owner of any tape machine a simple inexpensive tape location digital readout.

The El-Tech Take Finder indicates tape location on a large 5 digit L.E.D. display, which can be located up to 25 ft. from the tape machine. A small cable connects the display unit to a sensor which optically senses reel rotation. The sensor picks up reel rotation without any mechanical inter-connection by illuminating the edge of the reel and sensing the amount of reflected light. By placing small pieces of black tape on the reel edge light reflection will be interrupted as the marker passes under the sensor. The sensor is easily adjustable for any tape width which means you can use it on your 2 track, 16 track, or any type machine.

The display unit contains a memory for holding selected numbers and also gives a relay contact output when the memory and display equal. By connecting the relay output to the stop circuit of a machine automatic return to a memory number can be obtained.

Many hours spent searching tape can be saved with the Take Finder. The Take Finder allows you to find any position on tape without the hassels normally encountered.

Only **$349.95** from the following distributors:



patent pending

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615-546-3467

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el·tech

The MXR Analog Delay is intended as a low cost, maintenance free alternative to existing tape and disc echo systems.

With a suggested retail price of $299.95, the MXR Analog Delay is designed for a wide variety of applications, including electronic musical instruments, P.A., and sound reinforcement.

MXR INNOVATIONS, INC.
P. O. BOX 722
ROCHESTER, NY 14603
PHONE: (716) 442-5320

for additional information circle number 79

SPHERE ANNOUNCES NEW MASTERING CONSOLE

This mastering console is designed to meet today's requirements of a highly sophisticated system with the flexibility to make corrections in the sound quality and to improve quality of masters made under less than ideal conditions. The console employs A and B input groups to allow preset equalization or filtering to be changed between bands when cutting. Each group is switchable or will follow lathe spiral control. Graphic equalizers are featured. Push-button controlled inserts are provided for ancillary equipment. Lamps in illuminated push buttons verify relay operations, assuring function has taken place.

SPHERE ELECTRONICS
20201A PRAIRIE STREET
CHATSWORTH, CA 91311
PHONE: (213) 349-4747

for additional information circle number 80

SOUND WORKSHOP UNVEILS NEW DISCO/BROADCAST MIXER

Ultra-high performance and flexibility in a low-cost phono mixer/per-amp is offered in the Sound Workshop 421 Broadcast/Disco Mixer. Designed for broadcast production, disco, and home use, the Sound Workshop 421 provides 2 stereo phono inputs (magnetic; RIAA), 2 stereo high level inputs, and 1 microphone input (low or high Z.) Any of the stereo inputs can be assigned to the "active summing" cue buss whether or not that input is "on the air" or not.

VCA controlled "talkover" can drop the music level as much as 20 dB when the microphone is punched in. A 3 position EQ switch on the mike input provides for flat response or ±8 dB @ 100 Hz for maximum voice intelligibility. A sharp low-cut filter (switchable) eliminates power absorbing rumble from the

for additional information circle number 81

turntables without adversely affecting the program material.

Sound Workshop's Tri-Lite LED Readout gives accurate indication of both average and peak output levels (nominal level is internally adjustable from -10 dBm to +4 dBm.) The monitor section can select either the cue or program buss, and drives the internal 3 watt headphone amp (plug-in card) or an external monitor amplifier. The program output utilizes a highly linear booster amplifier which is stable into any value of capacitance load and will provide drive levels of up to +20 dBm into 600 ohms or greater, and +26 dBm into 300 ohms. Link-jack patch points provide access for system EQ or other effects (reverb, delay, etc.) while maintaining line drive capabilities.

Low noise op-amps are used throughout, all IC's are mounted in plug-in sockets, and maximum THD is .1%.

The Sound Workshop 421 Broadcast/Disco Mixer carries a 2 year parts and labor warranty and sells for $500.00.

SOUND WORKSHOP
1040 NORTHERN BOULEVARD
ROSLYN, NY 11576
PHONE: (516) 621-6710

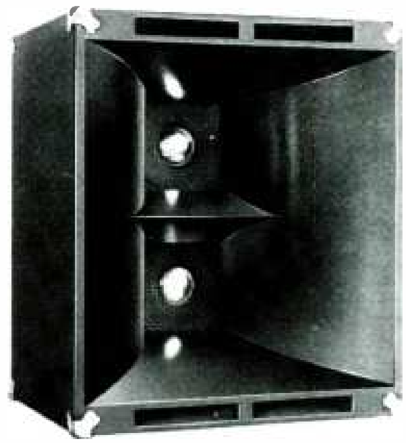
for additional information circle number 82

NEW FORSYTHE AUDIO LOW FREQUENCY SOUND REINFORCEMENT HORN

The new SR-215 bass reproducer is designed for applications requiring response to 40 Hz at extremely high sound levels. The horn is built with sufficient rigidity to handle the power of any commercially available bass drivers, and its cross-grain laminated 11-ply hardwood construction makes it less prone to panel resonances than other low frequency horns of this type.

The horn flare is maintained at a true exponential rate to deliver a smoother frequency response and better efficiency than the radius or quasi-exponential flares usually used. Unlike folded horn designs, the SR-215 maintains proper loading to above 1 kHz to permit crossover to the mid-range driver anywhere in the favorable 500 to 800 Hz region. The two-driver SR-215 further insures smooth operation by use of a center phasing plug to minimize interference between the two woofers.

Maximum sound output is 136 dB SPL on axis at four feet, when equiped with Gauss 5840 drivers. Rated frequency response in this configuration is 55 to 1,200 Hz ±3 dB. The unit measures 42"H x 36"W x 28"D, and weighs 149 lbs. less drivers. The SR-215 is equipped with "roadie" type corners to withstand the rigors of road use, and its black Durane polyurethane finish provides excellent



for additional information circle number 83

moisture resistance and scuff protection.

Standard convenience features include steel mesh screen to prevent foreign objects from contacting woofer cones, woofer spacer plates to prevent cone slap at extremely high levels, and tee-nut fasteners to facilitate quick woofer change. Roadie wheels and handles are available as an option.

FORSYTHE AUDIO SYSTEMS
75 NORTH BEACON STREET
WATERTOWN, MASS. 07172

for additional information circle number 84

AUDIO KINETICS (U.K.) TO MARKET AK 4000 CONSOLE IN U.S. THROUGH EVERYTHING AUDIO

The AK 4000 console, equipped with 40 inputs, 40 Perfect Q parametric equalizers, 32 group outputs with digital readout, 6 cue systems incorporating Supercue, 8 echo returns, 44 limiter/compressor/de-essers, 44 noise gate expanders, 672 position patch bay, 32 track remote control, 36 meters, a digital timer/clock, and Autofade, is all contained within a 7' 10" width.

Included within the group/monitor area of the module are full multi-track controls, the track number corresponding to that group and monitor. This integration allows the console logic to control the master sync, replay and record conditions of the multi-track; enabling simplified and more carefree operation by the mixer. The ultimate advantage, however, is Supercue — a logic controlled dropping-in system which always feeds the musicians cans with personal foldback, regardless of the drop-in routine selected by the engineer. Also, for the first time, the engineer can pre-sync before drop-in. It is said to be the fastest, most foolproof, one button drop-in system yet devised.

For the first time in a production console, AK have introduced a VCA controlled Compressor/Limiter/De-esser plus a noise gate/expander in every channel. The side chains of each device can be routed to the adjacent module allowing stereo quad compression and/or expansion as a routine mixing aid.

A new Perfect Q parametric equalizer is introduced which maintains the harmonic bandwidth correctly when using frequency shift.

EVERYTHING AUDIO
7037 LAUREL CANYON BOULEVARD
NORTH HOLLYWOOD, CA 91605
PHONE: (213) 982-6200

for additional information circle number 86

CONCEPT 1 SERIES CONSOLE ANNOUNCED BY AUDIO CONCEPTS/ DAVE KELSEY SOUND

Available in both 16 x 8 and 24 x 16 configurations, the consoles will feature full modular construction with push button sub group selection, independent stereo monitor/mix down busses with pan and positional soloing on all cues, echo sends and returns and sub group outputs. Monitor and cue functions are through the input modules (16 and 24 track mix down) and each input position has two patch points plus a direct out. A full balanced and normaled patch bay is included and Jensen transformers are used throughout.

Concept 1 is tentatively priced at $9,600.00 and $17,000.00.

AUDIO CONCEPTS, INC./
DAVE KELSEY SOUND
7138 SANTA MONICA BLVD.
HOLLYWOOD, CA 90046
PHONE: (213) 851-7172

for additional information circle number 87

for additional information circle number 85

PEAVEY CS-400 STEREO POWER AMPLIFIER

The Peavey Electronics Corporation model CS-400 Stereo power amplifier is designed for commercal sound reinforcement applications as well as studio and home applications. Each channel of the CS-400 is capable of 200 watts (rms) output into 4 ohms. The CS-400 boasts a very wide frequency response from 20 Hz

to 60 kHz (+0, -1 dB). Total harmonic distortion is less than .05%, intermodulation distortion is less than 0.1%. Other features include LED overload indicators on each channel; steel reinforced, zinc die-cast front panel; forced air cooling; and a 19" rack mountable chassis.

PEAVEY ELECTRONICS CORP.
P. O. BOX 2898
MERIDIAN, MISS. 39301
PHONE: (601) 483-5365

for additional information circle number 88

Addendum from pages 23 & 40:* **CTS, *London and* **Musicland**, *Munich, were used for string overdubs during production of "BREEZIN'".*

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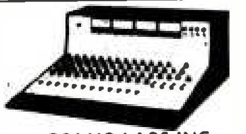

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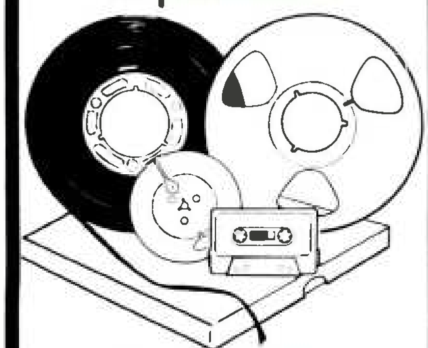

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EMPLOYMENT

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Manufacturer of high quality stereo equipment is seeking a young, aggressive individual to plan, implement and manage its sales program.

Applicants must have proven record of Sales and Management abilities in the consumer electronics field.

Submit resumes in confidence to:
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EMPLOYMENT

PRODUCTION ENGINEER

Opportunity for experienced production engineer in top recording and production facility in mid-Atlantic market. Multi-track music experience necessary. Send resume and production sampler to:
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ALL AROUND AUDIO MAN

Production run & systems checkout, trouble shooting. Challenging job for self-starter, working with a leading console manufacturer. Southern California.
SEND RESUME OR CALL BOB BUSHNELL
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PRODUCT PLANNING SPECIALIST

Ideal opportunity for enthusiastic individual seeking a responsible marketing position in new product development with TAPCO—one of the nation's leading professional audio equipment manufacturers. Responsibilities include origination of new product ideas, product idea evaluation, presentation of new concepts and programs, and a close working relationship with engineering and marketing groups throughout the design and market introduction stages of new product development.

Some experience with state-of-the-art electronics circuitry is required, EE preferred. A thorough knowledge of products, procedures and techniques used in professional sound work is essential, as is a fundamental understanding of pro-audio sales and marketing practices.

Working with a forceful team of new product planners, the qualified individual will play an integral role in the future of this dynamic and fast-growing company. TAPCO offers competitive salaries, superb employee benefits and, of course, all of the environmental benefits of the great Pacific Northwest.

All qualified persons are invited to apply for this excellent opportunity. Send all pertinent data, including salary history to Doug Schauer, Product Planning Manager.

TAPCO
TECHNICAL AUDIO PRODUCTS CORPORATION

3810 - 148th Avenue, N.E.
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(206) 883-3510

Sales Manager Professional Audio Products

Fast growing, established manufacturer of digital audio products needs proven sales manager with good track record in audio industry to manage pro audio dealer and commercial sound contractor sales.

Hands-on experience in recording studio operations or commercial sound systems desirable. This new position reports to President.

Please send resume or call.

lexicon

60 Turner Street, Waltham, Massachusetts 02154/617-891-6790

for additional information circle number 96

. . . continued from page 19

AUTOMATION "DITHER"

allows the use of a *Rewrite* mode, providing *(if desired)* a smooth transistion to and from *write* mode.

It is MCI's belief that the user of an Automated console does not want to and should not be required to "know what's going on in *(the Automation's)* little binary brain." The mix engineer wants to sit down at a mixing console with minimum training, and do an automated mix with greater ease and less effort than would be required if the console had not been automated. The mix engineer and producer *want* to be freed of the operational limitations to devote their efforts to creativity.

With the MCI Auto Nulling logic the "little binary brain" worries about level shifts, getting back to the read value, and all the other bothersome chores of automation.

The Harrison/Allison package offers no method of observing the actual operation of the automation. This problem has been completely eliminated with the MCI automation system. MCI provides a Plasma Display meter which can be switched to a VCA DC status mode. This metering mode shows the operator the DC voltage returning from the automation to each fader. The operator can see at a glance the relative position of each fader in the mix. The ability to see as well as hear the levels is a valuable aid when doing any automation manipulation.

In conclusion, we would like to comment, with pride, on our record in one area mentioned by Mr. Buff – reliability. MCI is the *only* company which has demonstrated a *working* automation system on the floor of an AES show for everyone to see and abuse. The JH-528 console we shipped to the Paris show was installed within three hours and worked perfectly throughout the show. It was then shipped to Conney Studios in Cologne, Germany, and installed by an MCI dealer. He completed his work within one day. The console *with automation* has been working ever since. By the time of the AES show in Los Angeles we will have over half a dozen automation systems installed and working.

LEXICAN REDUCES DELTA-T PRICES UP TO 23 PER CENT

Lexicon, Inc., has announced a major price reduction on its Delta-T 102 line of digital audio delay systems.

The precentage price reduction will be dependent on system configurations. For example, a typical studio stereo system with two independent input channels, 240 ms of delay, three outputs, and VCO special effects module will carry a net professional price of $4,700 versus a previous price of $5,894. A large mono 102B configuration with 320 ms of delay and five outputs is reduced to $4,900 from $6,380, a 23% reduction.

According to Ron Noonan, President of Lexicon, the company elected to pass on to its customers production cost savings in its 102 series line due to a higher volume of sales. The Delta-T 102 series is designed for studio use and large sound reinforcement systems.

Lexicon has recently introduced a low cost Model 92 digital delay system at a net professional price of $1,560 for small installations such as churches, school auditoriums, and discotheques, as well as studios.

WILLIAM G. DILLEY TO RECEIVE DISTINGUISHED ENGINEERING AWARD

William G. Dilley, founder of Spectra Sonics of Ogden, Utah, has been named to receive the Distinguished Engineering Alumnus Award of the University of Colorado College of Engineering and Applied Science. The award is the highest honor that the College can bestow upon an alumnus.

Mr. Dilley played a creative role in the American space vehicle development, then founded his company for the manu-

facture of advanced electronic equipment. Spectra Sonics has become a leading designer and manufacturer of sound reinforcement equipment, recording studio equipment, and electronic apparatus for testing and research.

Born in Sterling, Colorado, Dilley graduated from North High School in Denver. His engineering career at Colorado University was interrupted for three years while he served as a fighter pilot in World War II. He was decorated several times for valor in combat.

He returned to CU to earn the bachelor of science degree in architectural engineering in 1951. As a professional engineer he made major contributions to the missile program, some of them still classified. Others include a system for the Thor vehicle that reduced countdown time by 50 per cent.

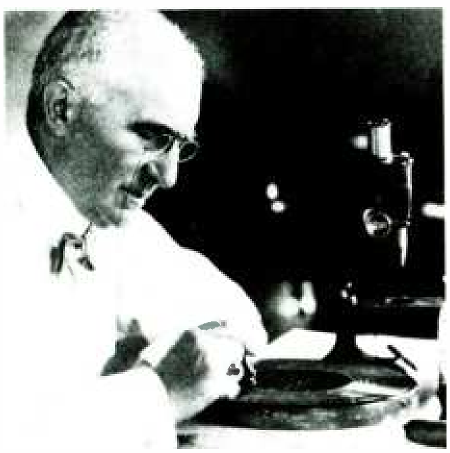
Dilley was responsible for determining what data were required from airborne launches, for designing the instrumentation with which these data were obtained, and for processing of the information received.

He designed, manufactured, and supplied international timing signal converters for all missile launch complexes at Vandenburg Air Force Base. He has published more than 250 engineering papers and technical articles and holds 14 U.S. and foreign patents.

As a civilian pilot Dilley has competed in U.S. and international air races and has set aircraft speed records. He is a member of national associations for broadcasting, television, motion pictures, audio engineering and aviation.

He and his wife Jean live in Ogden. The have two children.

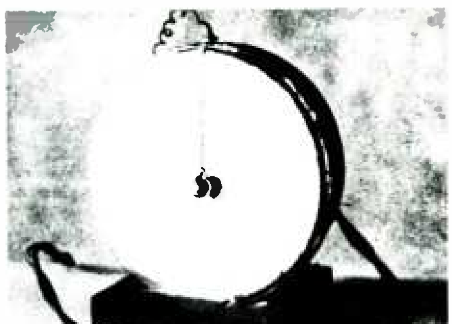
APRIL 1977 MARKS 100th ANNIVERSARY OF MICROPHONE



1977 marks the 100th anniversary of the invention of the microphone by Emile Berliner, who at age 25 and a penniless immigrant youth from Germany, who went on to give the world another of its greatest benefits in the form of the disc record and player, the method of mass-producing discs from a single master, and the famous "His Master's Voice" trade mark.

Emile Berliner's microphone made practical telephony possible, and its acquisition by the then-fledgling Bell System saved the firm from destruction by the then-powerful Western Union and paved the way for Bell's becoming the world's largest corporation. The loose-contact principle introduced by Emile Berliner and still in use throughout the communications world today, was deemed to have passed the limits of scientific credibility at that time.

Upper photo shows the original microphone of March 4, 1877. In the lower photo is the telephone transmitter, forerunner of all talking pieces used in the world's telephones today, included in the Berliner Caveat of April 14th. With mouthpiece added it was acquired by The Bell System.

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Affordable Automation
From HARRISON
2824
Automation ready means different things to different people. At Harrison Systems, it means that our consoles can be connected to an existing and available automation programmer with a simple multi-pin connector. Every Harrison console ever produced has this capability. Some are now being used for automated mixing.
Some manufacturers of recording consoles *claim* to be automation ready. Interface with existing programmers, however, requires additional circuitry not included in the console, or the programmer.
Harrison Systems not only has all circuitry for automation interface included in all its consoles, but beginning January 1977, all consoles shipped will have been interfaced and tested with an automation programmer during final test. This insures plug-in compatability should you automate in the future.
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part of the NO COMPROMISE philosophy at
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Station Master.

You can virtually drive coast-to-coast without leaving the sound of a radio station using a Shure microphone. In fact, you'll encounter almost as many different models of Shure microphones as you will states.

Case in point: the Shure SM7. It features a wide-range, ultra-smooth frequency response with show 'n' tell switches that allow the user to select any of four microphone response curves: (1) flat response; (2) presence boost; (3) bass rolloff; and (4) presence boost with bass rolloff.

The SM7 also uses an innovative "air suspension" integral shock mount for super-isolation against mechanical and shock noise.

Ask your Shure dealer for a demonstration of the show 'n' tell SM7. It's one Shure "show" worth telling everybody about.

Shure Brothers Inc.
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TECHNICORNER
The Shure SM7 is a unidirectionc ic microphone with a 40 to 16,0 frequency response. Noise rec systems cut mechanical noise m. "pop," wind, and electromo ry. "Add-on" filter devices are filter The SM7's integral foam win h reduces even difficult close ohms sounds. Impedance is rated to for microphone inputs rate dB = 300 ohms. Output level: circuit 1 milliwatt per 10 microbar voltage: −79 dB (0 dB = microbar).

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