JUNE 1975
VOLUME 6 / NUMBER 3

Recording engineer producer

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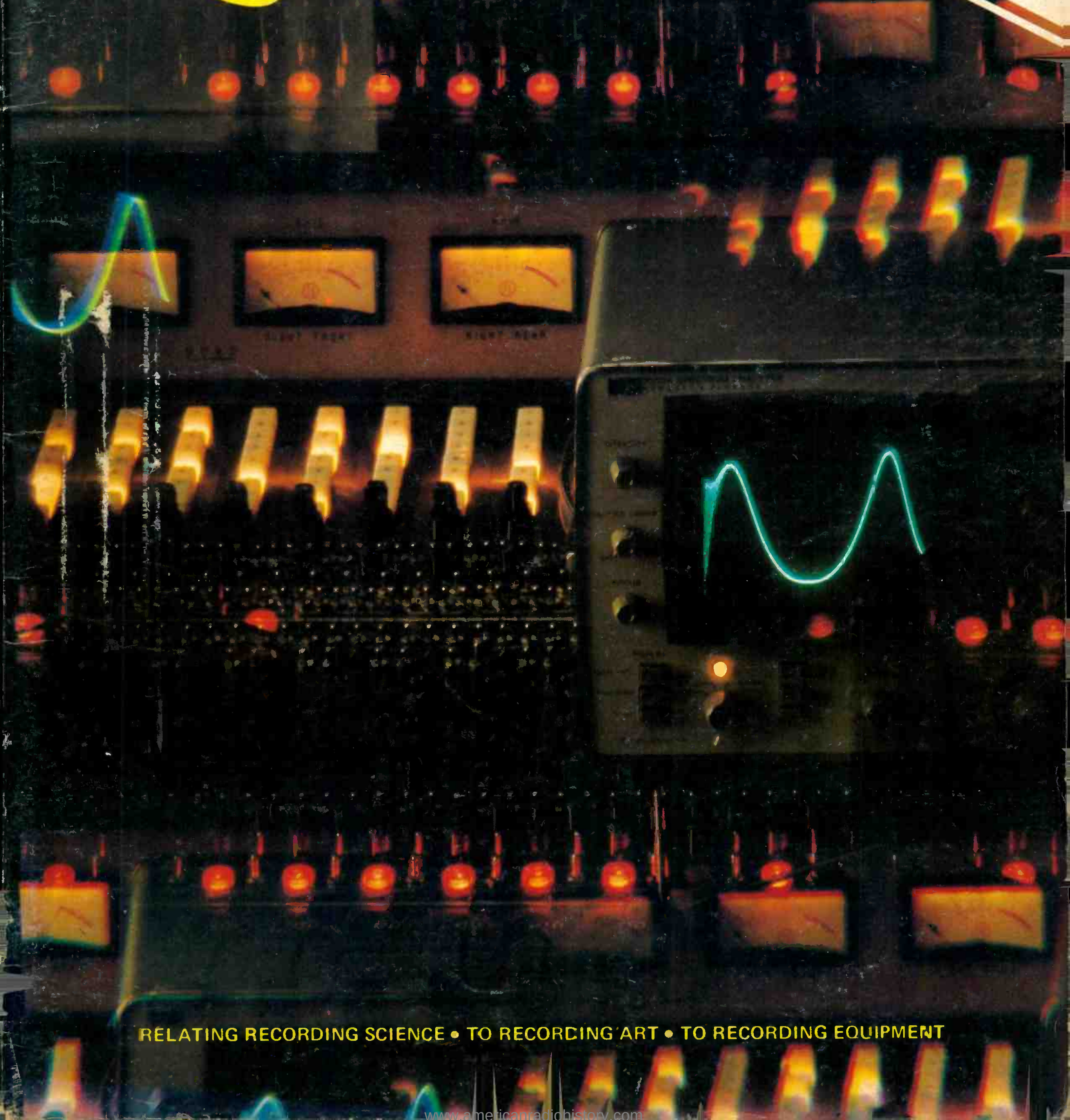

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RECORDING engineer/producer

– the magazine to exclusively serve the recording studio market . . . all those whose work involves the recording of commercially marketable sound.

– the magazine produced to relate . . . RECORDING ART to RECORDING SCIENCE . . . to RECORDING EQUIPMENT.

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'RECORDING engineer/producer' is published bi-monthly (six times a year) by RECORDING & BROADCASTING PUBLICATIONS, 1850 No. Whitley Avenue, Suite 220, Hollywood, CA 90028, and is sent to qualified recipients in the United States. Subscriptions for other than qualified individuals or companies may be purchased at $6.50 per year (6 issues). All foreign subscriptions: Surface Mail – $7.50, Air Mail – $13.00.

Controlled Circulation postage paid at Los Angeles, California.

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THE COVER: By using a multi-image prism along with a special diffusion filter, the illuminated highlights of the H.P. Spectrum Analyzer and the Spectra Sonics Console form the montage background for the amplitude response vs. frequency trace of the equalizer being analyzed. Peter Butt's article beginning on page 42 explores the uses of this new generation of calibrating equipment.

Letters & Late News

FROM: Charles P. Repka
Vanguard Recording Society
New York, NY

I'm a bit upset over your article on alignment tapes in the February 1975 issue of RE/P. I'm upset because the article perpetuates the myth that the Dolby Noise Reduction process works only at a flux level *of 185nWb/m or 320 nWb/m. Nonsense! A Dolby box uses volts, not magnetic flux and will work properly (to its own parameters) at any flux level as long as the tape recorder has been properly aligned.*

For example: Lets say you have a tape machine that has been aligned to the 185 nWb/m and has a 0 VU output of 1 volt at that flux level. If you now wish to recalibrate to a 250 nWb/m flux level, the recorder's electronics will be adjusted to produce a 0 VU output of 1 Volt at the 250 nWb/m when following the standard alignment procedure. A Dolby box connected to this tape recorder will not be able to "tell the difference"between the two different flux levels since all it will ever see is a 1 volt input.

The article did not really get into the problem of the lack of an official magnetic reference standard. All we have are some proposed NAB standards, a semi official Ampex standard and the DIN standard. The recording industry is in need of a definitive standard for magnetic recording and it is up to the people working in the industry (through its professional organizations, such as the AES) to set these standards.

Perhaps RE/P could have some future articles about standards in the recording industry plus comments from other readers.

FROM: Steve Katz
Los Angeles, CA
(reply to Charles P. Repka)

You are correct in stating that Dolby Level, as seen by the Dolby unit, is a voltage. This voltage, however, is derived from a magnetic flux. Therefore it is essential that Dolby tone be recorded at the head of each track of tape so that playback level can be correctly calibrated.

For more information Dolby System alignment for different flux levels, Dolby Laboratories offices in New York or London should be contacted.

The difficulty in setting level standards today is due to the constant state of flux of tape technology, no pun intended.

FROM: William Wittman
Variety Sound
New York City

The Venereal Studio

Well, now that I've caught your interest allow me to explain that what I'm talking about is Venereal terms or Terms of Venery (or the hunt) such as: a school of Fish, a Pride of Lions, an Exaltation of Larks, a Slate of Candidates, an Unkindness of Ravens, a Sentence of Judges, and a plenitude of others.

And as it would show ludicrous ignorance to say "There goes an herd of fish" rather than the correct school of fish, so it must be with a Siege of Herons . . .

And why can't we studio creatures get in on the fun? We can. So get with it out there! Following are my suggestions. Prizes for the best suggestions will be your own satisfaction in adding to the English language.

In the Studio
(a/n/)
Scale of Musicians
Score of arrangers
Inaccuracy of copyists
Roll of Drummers
Pound of pianists
String of violinists
Breath of flutists
Deafening of Guitarists
Lateness of Rock Groups
Overdub of Vocalists

In the Control Room *(a/n/)*
Splice of editors
Ohm of technicians
Breath of compressors
Space of echo
Flick of meters
Gate of Keypie
Percentage of managers
Vague of producers
Fade of engineers
Sad but often true
Hiss of Tape / Hum of Cable
Falling of Relays and a
Disappearance of Headphones

Readers who wish further enlightenment on the subject are recomended to the very thorough and entertaining book "An Exaltation of Larks" by James Lipton.

BROCK . . . HAMILTON TO HEAD NEW AUDITRONICS DISTRICT OFFICES

Auditronics Systems Division, from headquarters in Memphis, has announced the appointments of: BILL BROCK to head a new regional sales and service office in Nashville, and BILL HAMILTON to head the new regional sales and service office in Philadelphia.

The new offices were created to work with established, as well as newly created recording studios, sound reinforcement applications and broadcast facilities, providing assistance with equipment, system design, acoustic design and financing.

Both Hamilton and Brock bring a reserve of engineering depth and wide ranging pro-audio experience to their respective offices.

Bill Hamilton can be reached at (215) 328-9889 in the Philadelphia area.

Bill Brock can be reached at (615) 794-7529 in the Nashville area.

NEW CBS TECHNOLOGY CENTER IN STAMFORD, CONNECTICUT ASSUMES CORPORATE R&D ACTIVITIES OF CBS LABORATORIES AND NEW APPOINTMENTS ANNOUNCED BY CBS

With the completion of the transfer of the Professional Products Department of CBS Laboratories to Thomson-CSF. S.A. of France, in a transaction announced on April 2 and concluded May 1, CBS President Arthur R. Taylor announced that former CBS Laboratories research activities in broadcasting and audio recording have been transferred to a new CBS Technology Center reporting to Harry E. Smith, Vice President, Technology.

Benjamin B. Bauer, Vice President of the predecessor CBS Laboratories, will head the CBS Technology Center as Vice President and General Manager.

The CBS Technology Center on High

Continued

Ridge Road in Stamford, Connecticut, will be organized with four scientific departments along the following lines: Advanced Television Technology, headed by J. Kenneth Moore as Director; High Density Recording Technology, with Robert A. Castrignano as Director; Audio Systems Technology, Emil L. Torick, Director; and Sound Reproduction Technology under Louis A. Abbagnaro as Director.

Advanced technology activities will be totally funded by CBS and its operating divisions, and will support long-range corporate and divisional research and development goals. The new concept has been described by Mr. Taylor as "a continuing commitment to broadcasting, audio recording and other CBS-related technologies."

Benjamin B. Bauer, Vice President and General Manager of the CBS Technology Center, was formerly CBS Laboratories' Vice President, Acoustics and Magnetics. Since he joined CBS in 1957 he has guided major developments in a broad range of the communications sciences. He is a 1932 graduate of Pratt Institute with a 1937 E.E. degree from the University of Cincinnati. His post-graduate studies were in physics, mathematics and acoustics at Chicago and Northwestern Universities. Prior to joining CBS, he was associated with Shure Bros., Inc. from 1937 as development engineer, Director of Engineering and Vice President. He is the author of numerous papers, the holder of more than 50 patents and a member of the National Academy of Engineering of the United States of America. He has lectured widely and served as a Visiting Professor of Engineering Acoustics at Pennsylvania State University. He is a Fellow of the Institute of Electrical and Electronics Engineers and of the Acoustical Society of America. He is a Fellow of the Audio Engineering Society, of which he is also a past President and a recipient of its Gold Medal Award.

J. Kenneth Moore, Director of Advanced Television Technology at the CBS Technology Center, joined CBS Laboratories in 1958. As General Manager of the Laboratories' Electronic Systems Department he was actively engaged in work on systems for image data handling and phototransmission and such technologies as laser scanning, electron beam recording and ultrasonics, as well as the Vidifont graphic arts quality titling system for broadcast television. He earned his B.S. and M.A. degrees in physics at Williams College, where as a teaching assistant he had a graduate fellowship from the Sprague Electric Company.

Robert A. Castrignano, Director of High Density Recording Technology at the CBS Technology Center, has been with CBS since 1938 and served as General Manager of the Television Technology Department of CBS Laboratories. He earned his B.S.E.E. degree at City College of New York in 1949 and his M.E.E. at Polytechnic Institute of Brooklyn in 1956. He has directed activities associated with black and white and color television systems and specialized in the development of television signal enhancing techniques, high resolution scanning, rapid film processing and pulse circuit design. He is a Fellow of the Society of Motion Picture and Television Engineers.

Emil L. Torick, Director of Audio Systems Technology at the CBS Technology Center, was Manager of Electronic Systems Research at CBS Laboratories. He joined CBS in 1958. A 1953 graduate of Duquesne University with a baccalaureate in music, he earned his B.S. in physics at the University of Pittsburgh in 1958 and an M.B.A. at the University of Connecticut in 1970. He is a Fellow of the Audio Engineering Society and the holder of patents in signal control circuitry, noise control and broadcast devices. He was an instructor in physics at the University of Pittsburgh and a violinist with the Pittsburgh Symphony and other orchestras.

Louis A. Abbagnaro, Director of Sound Reproduction Technology at the CBS Technology Center, joined CBS in 1964 upon graduation from Yale with a B.S.E.E. degree. He received his M.S.M.E. degree at the University of Bridgeport in 1969. He served as Branch Manager of Acoustics Research at CBS Laboratories and has been extensively involved in the design of advanced acoustic transducers, transducer arrays and signal processing techniques, as well as in underwater communications systems, noise measurement and control, high performance bone conduction microphones and advanced noise-canceling microphones.

BOLLINGER NEW DIRECTOR OF MARKETING AT CAPITOL MAGNETIC PRODUCTS

William C. Bollinger has been named Director of Marketing for Capitol Magnetic Products, recording tape division of

Capitol Records, Inc.

Bollinger will be responsible for all marketing activities of Capitol's consumer, professional and audio-visual product lines.

Before joining Capitol Magnetic Products, Bollinger held various marketing positions with Memorex Corporation. He was brought to Memorex in 1970 as a part of the original marketing team charged with the responsibility to introduce Memorex's first consumer product line.

Capitol Magnetic Products is headquartered at 1750 North Vine Street, Los Angeles, Calif. 90028.

SHURE THEME FOR 50TH ANNIVERSARY CELEBRATION: "THE 51ST YEAR OF OUR COMMITMENT TO EXCELLENCE"

A half-century of growth and success as a distinguished member of the sound industry is being celebrated this year by Shure Brothers Inc., Evanston, Illinois.

The company was started as the Shure Radio Company on April 25, 1925, by S.N. Shure, to this day president and chief operating officer of Shure Brothers Inc.

From its beginning as a distributor of radio parts, Shure has grown into one of the world's largest manufacturers of microphones, high fidelity phono cartridges, and sound reinforcement components. Shure products are presently sold in over 100 countries throughout the world.

The company moved into its present headquarters at 222 Hartrey Avenue, Evanston, IL, in 1956. Since 1966, Shure has also maintained a manufacturing facility in Phoenix, Arizona. An additional manufacturing facility and a distribution center begins operation in 1975 in Arlington Heights, IL.

As a theme for its 50th Anniversary celebration, Shure has selected "The 51st Year of Our Commitment to Excellence" to exemplify the company's continuing dedication to providing products and services of the highest quality and reliability.

ARTHUR A. SCHUBERT NEW WARD-BECK DIRECTOR OF ENGINEERING

Ward-Beck Systems, Limited, manufacturers of professional audio control equipment for the television, radio, and motion picture industries, is pleased to announce the appointment of Arthur A. Schubert, Jr., to the position of Director of Engineering. Mr. Schubert comes to Ward-Beck with twelve years professional experience in the broadcast and audio engineering industries. Mr. Schubert was a member of the CBS Television Engineering Department for nine years, and most recently served as Chief Development Engineer for Neve Electronic Laboratories Ltd., in England.

In his new position with Ward-Beck, Mr. Schubert will be responsible for engineering management in the design and production of Ward-Beck audio consoles and related products. Mr. Schubert is a graduate of Pennsylvania State University in Electrical Engineering, and is a member of the Audio Engineering Society and the Society of Motion Picture and Television Engineers.

CONSULTING FIRM FORMED BY LEON WORTMAN

L.A.W. ASSOCIATES has been formed by Leon A. Wortman to provide specialized marketing and sales consultant services to audio and closed circuit television firms. Wortman was formerly National Marketing and Sales Manager for Ampex Corporation's Professional Audio and Industrial Video product lines. He is author of the CCTV HANDBOOK and numerous articles on audio and business management. Wortman is also Vice President, Western Region and Member of the Board of Governors of the Audio Engineering Society.

Current accounts with L.A.W. Associates include Bechtel Incorporated (CCTV) of San Francisco, California; Burwen Laboratories (noise filters) of Burlington, Massachusetts; The Center for Professional Advancement (marketing seminars) of Somerville, New Jersey; and Training Services, Incorporated (sales training programs) of Rutherford, New Jersey.

L.A.W. Associates is located at 743 Holly Oak Drive, Palo Alto, California; telephone (415) 494-2613/2614.

MAKER OF THE MICROPHONE AWARD TO NIPPON VICTOR

Toshiya Inoue, a Victor Co. of Japan director, and manager of the firm's audio research center, flew in from Tokyo to present an AES Los Angeles Convention paper on the firm's CD–4 stereo disc system and to also receive the Maker Of The Microphone Award for an outstanding contribution to the world of sound, from Oliver Berliner, grandson of Emile Berliner who invented the microphone, disc record and record player, and in whose honor the annual award is made.

"... my beautiful record sounds terrible over the radio ... not at all like I mixed it ... WHY?"

WHAT RECORDING ENGINEERS and PRODUCERS SHOULD KNOW ABOUT

BROADCASTING

(and maybe were afraid to ask about!)

An R-e/p report based on material from a conference of engineers with experience in both Recording and Broadcasting.

One of the recording engineer's or producer's greater professional disappointments is hearing his latest release played over the radio; either AM or FM. The frequency response and equalization so carefully tailored and selected through hours of mixing somehow gets all jumbled up and re-arranged coming out of the radio speaker. The dynamic range of the broadcast version doesn't even begin to match the test pressing and is nowhere near the stereo mix.

"What's going on here?" is a question often occurring to those instrumental in recording and production. To get to the root of that question requires a bit of discussion. It's not one of those simple short answers that straightaway illumines the consciousness of the questioner. It's a long way from mix-down to kitchen or car radio.

Radio broadcasting is a fairly complicated process involving several general considerations, technical, fiscal, and psychological. To grasp the implications of these factors as they manifest themselves in the way a record sounds as it emanates from the listener's receiver, it might do to take a good look at the obstacle course to be overcome on the way to the radio station antenna. Then, too, we must consider the impact of the spectrum of receiving equipment available to the cross-section on the listening audience.

THE BROADCAST AUDIO CHANNEL

The *package* that a record broadcast over the airwaves has to fit within is dependent on a *weakest link* sort of situation. The poorest segment of the program channel determines the maximum amount of spectral information getting through to the listener. For AM radio, the audio channel bandwidth can be reliably assumed to cover the 50 Hz to 5kHz range. Some stations operating on *clear-channel* frequencies and/or in areas where adjacent channels aren't too crowded, can extend the upper frequency range somewhat. Since even the more expensive AM receivers are down 6 dB at anywhere from 5 to 8 kHz, it's not clear how many AM stations take the trouble to take their audio signal out that far, even if adjacent channel interference considerations would permit it.

The low-frequency roll-off of an AM audio channel will most generally be sharply rolled off below 100 Hz. This is done because the signal information below about 100 Hz tends to be lost in the receiving system and also takes up a disproportionate amount of modulation percentage for the difference it makes at the listener's radio.

The FM radio channel bandwidth looks a little better. It runs from about 50 Hz to 15 kHz. It is difficult to do more than speculate on what proportion of the FM stations' audio signal chains actually can reliably pass that bandwidth within a 2dB tolerance. There has to be a rather sharp upper frequency bandwidth limitation in the case of the stereo station, which most FM stations now are. This is because the stereo information is multiplexed to provide two audio channels on a single RF carrier channel frequency.

THE STEREO FM SIGNAL

This is done by direct summing of the left and right channels and transmitting the sum in the 50 to 15 kHz low-frequency, or *baseband*, portion of the FM modulation channel. The two stereo channels are also combined as a difference signal, in a left-minus-right manner, and double-side-band, suppressed-carrier, amplitude modulated at a 38 kHz carrier frequency. The suppressed 38 kHz carrier is divided by 2 and transmitted as a 19 kHz CW signal, sandwiched in between the lower base-band signal and the suppressed-carrier difference signal centered at 38 kHz. This whole composite of signals is then used to deviate the frequency of the transmitter.

This whole process is then un-done at the stereo receiver where the 38 kHz difference signals are demodulated with the help of a 38 kHz carrier generated in synchronism with the 19 kHz *pilot tone* sent along for that purpose at the transmitter. The sums and differences are then re-combined in such a way as to separate the base-band monophonic L + R composite back into two-channel stereo again.

SIGNAL CONDITIONING

The net dynamic range of the broadcast audio channel is determined by a number of factors, some of which are outside the control of the broadcaster. First, there is the maximum signal level that his transmitter will accept, consistent with the physical and legal limitations imposed upon it. At the listener end, there is the ambient noise level in which the radio receiver is placed. The residual noise within the receiver itself, and inherent in the broadcast station program chain, are generally less significant in any environment other than the truly high-quality home listening situation. Presently, residual noise floors are running as low as -70 dB below 100% modulation for very high quality FM stereo receivers and as low as -50 dB for very good AM receivers.

The broadcaster tries to condition his audio signal for the very worst case as far as noise floor is concerned. This is because the interference from acoustic sources at the point of reception may be very high indeed. Consider the case of the noisy household or automobile at high speed. The useful dynamic range for the majority of radio listeners, AM or FM, is probably less than 20 dB.

The broadcaster tries to counteract this high residual noise level by restricting the dynamic range of his signal by using combinations of compression and limiting.

The louder he sounds, the greater is the liklihood that the broadcaster can capture a large share of the listening audience. This means ratings and ratings translate as dollars. This is true of both AM and FM program chains. In the case of the AM station, there is a slightly different approach to the limiting technique. The AM signal differs from FM, among other ways, in that it is sensitive to the polarity of the modulating signal swing. There is no theoretical limit to the upper level of the modulation. There is a limit to magnitude of the negative peaks.

Legally, the AM station is bound to keep its modulation peaks below 125% on positive peaks, and below 100% on nega-

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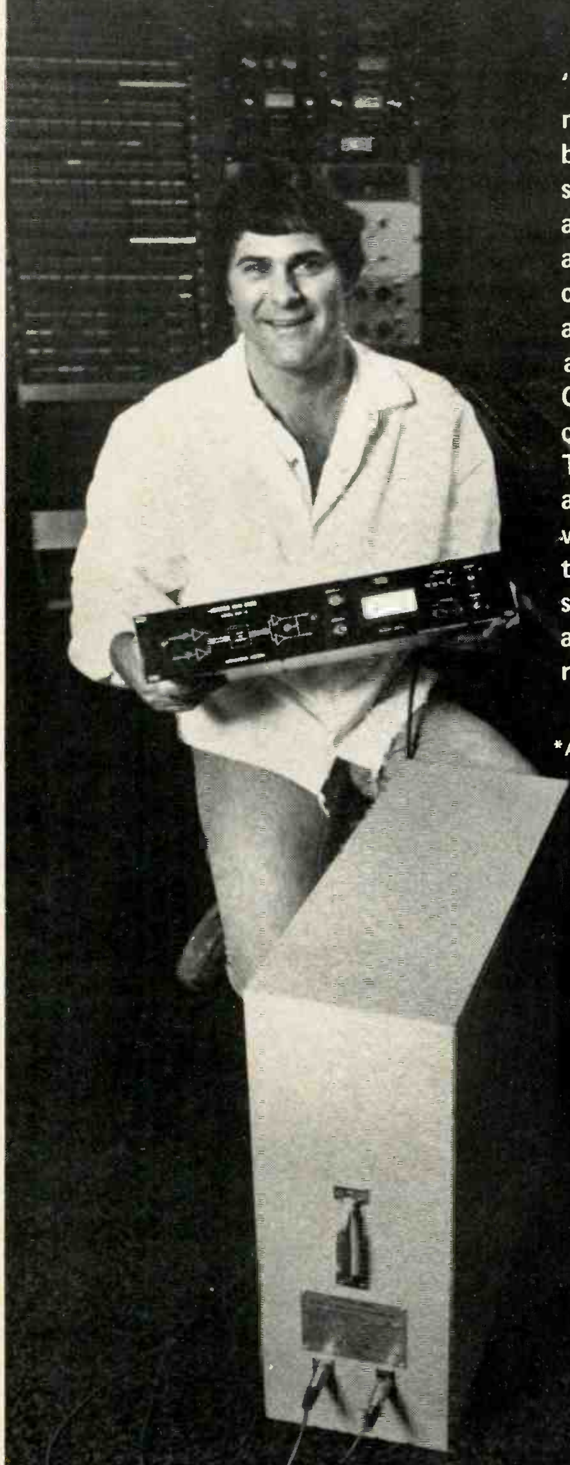
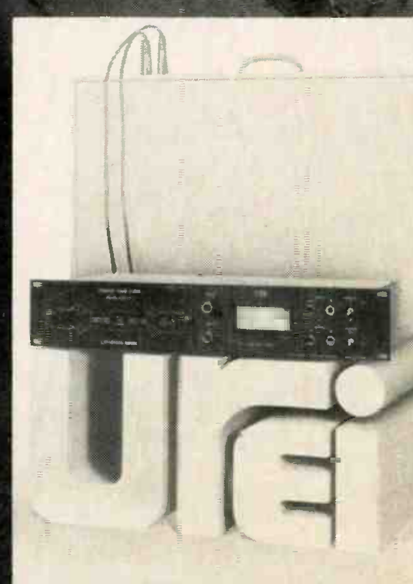
Don Sciarrotta, Rufus engineer, "tells you something good" about the Cooper Time Cube.



"I think the Cooper delay is something good. It has more of a room sound and it makes the room sound bigger. The electronic delays have an 'electronic' sound. The Cooper's versatile, too. I can use it with a tape machine by itself, or I can put one before it and after it, or I can patch it through my echo chamber. When we recorded the 'Rags to Rufus' album*, we used just the Cooper with the voice bag and the voice at the same time. The double delayed Cooper was used on the voice for the verses and choruses, the bag was Coopered only on the choruses. The result was a great doubled voice sound. The album was gold and also had two gold singles. One was 'Tell Me Something Good'. That's why I like the Cooper Time Cube. With it we got just the sound we wanted. I've had my Cooper for two and a half years without any trouble at all . . . I've replaced one light bulb."

*ABC Records, Bob Monaco Producer



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tive peaks. This is accomplished by using a type of limiter that has a different compression ceiling for signal peaks of different polarities. These are known as asymmetrical limiters. Every dB counts in radio.

The FM limiting processes are not without their adaptive modifications, either. The FM transmitter features a 75 micro-second pre-emphasis characteristic that dates from the time FM was written into the Communications Act of 1934. This was originally done to minimize the subjective effect of the white noise characteristic of the FM RF channel. Those were the days of 5 kHz microphone response, so typical modulation of that era didn't extend up into the 15 kHz portion of the audio spectrum. Since there is a lot of signal information in that region with more modern program material, the 75 micro-second boost poses a significant problem to the broadcaster. His modulation level is frequency dependent and his limiting systems have a 75 micro-second pre-emphasis in their control sections to compensate for the transmitter HF boost.

In the case of both AM and FM limiters, hard diode peak limiting may be employed to absolutely prevent audio peaks above predetermined levels. It all sounds a little vulgar to the studio recordist, but that's the way things are in radio land.

Some FM stations have a background music service riding along on their modulation signal package. This sort of thing is known as *Sub-Carrier Authorization* or SCA. Each SCA service channel takes up about 10% of the modulation capability of the FM channel, leaving less room for the music. The station generally tries to make up for the reduced stereo music modulation capability by using even more compression.

Many stations have an equalization network that is placed in the program signal chain to try to compensate for relative insensitivity of the human ear below about 1 kHz and above about 5 kHz. Everything getting out on the air gets shoved through that equalizer.

REMOTE TRANSMITTERS

Broadcast transmitters are often located at a place geographically separated from the studios where the program signal originates. There are two general ways of getting the audio from the studio location to the transmitter: telephone land lines and microwave radio links.

Telephone land lines aren't too bad from the standpoint of residual noise and frequency response. They are usually guaranteed to be flat within 2 dB from 50 Hz to 15 kHz. Noise is generally down 60 or more decibels from a +8 dBm reference level.

The microwave approach, called STL for Studio-Transmitter Link, will usually be capable of somewhat better performance than the land line. Since the STL is a line-of-sight signal situation, the likelihood of serious interference from outside sources is minimized. STL links are sometimes subject to fading. They also beat the phone line in stereo applications where the phase shift between program channels must be held to low tolerances.

One of the difficulties with land lines is that they have to be heavily equalized and precisely impedance-matched to achieve a very flat frequency response. The equalization may be quite radical in the case of longer distances and the phase response may be difficult to predict. This is not very good for a stereo signal, as most producers and recordists already know. It is doubly bad in the case of stereo broadcasting because the baseband signal is the sum of the left and right channels, as discussed earlier. Dis-similarities in phase response between channels can cause the mono mix, as heard on mono FM receivers, to be quite disagreeable to the listener. FM broadcasters are very aware of this problem and are prepared to go to extensive lengths to preserve the phase integrity of their mono signal. This is because 70 to 80% of the FM receivers in the hands of listeners are relatively inexpensive and are monophonic only. Stereo is appreciated most in the home living room-type of environment.

PROGRAMMING FORMATS

Music gets on the air by way of a number of different formats. The important ones, in order of relative importance, are: continuous-loop tape cartridge, the disc recording, open-reel ¼-inch tape, and Philips-type cassette.

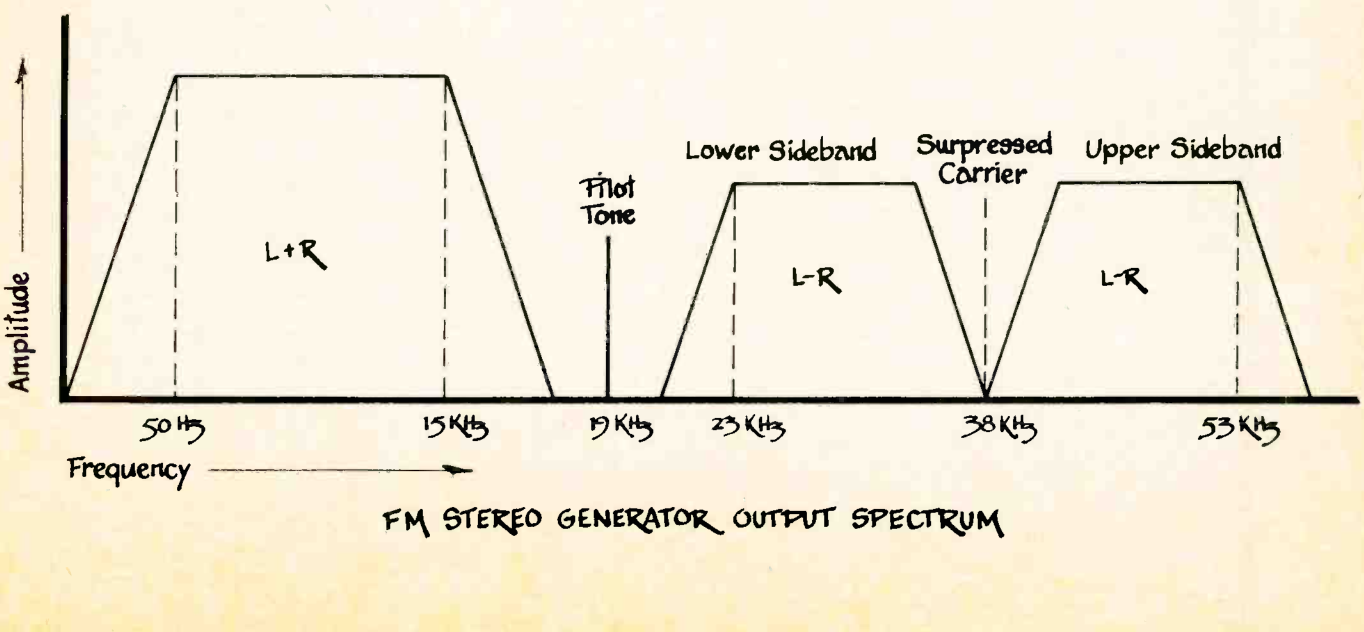
CONTINUOUS LOOP CARTRIDGES

The reasons for using these formats rather than the disc itself are several. A cartridge is fairly immune from handling damage. It automatically returns to its cueing position after a playing cycle. It is easy to locate, being stored in an orderly rack. Transferring the disc recording to the cartridge format also allows the station to control the equalization and the maximum and minimum levels of the music. Any questionable lyrics can be excised and the maximum playing time of the record can be adapted to the stations' requirements.

As far as equalization goes, this is usually done to conform to what the program director feels is the right kind of sound for his station format. At times, echo is added. Peak limiting is sometimes done at the time of transfer, adding still more limiting to the on-air program chain. Records are sometimes speeded up by as much as 3% to meet timing requirements and to increase tempo.

DISCS

Discs are used directly by stations hav-



FM STEREO GENERATOR OUTPUT SPECTRUM

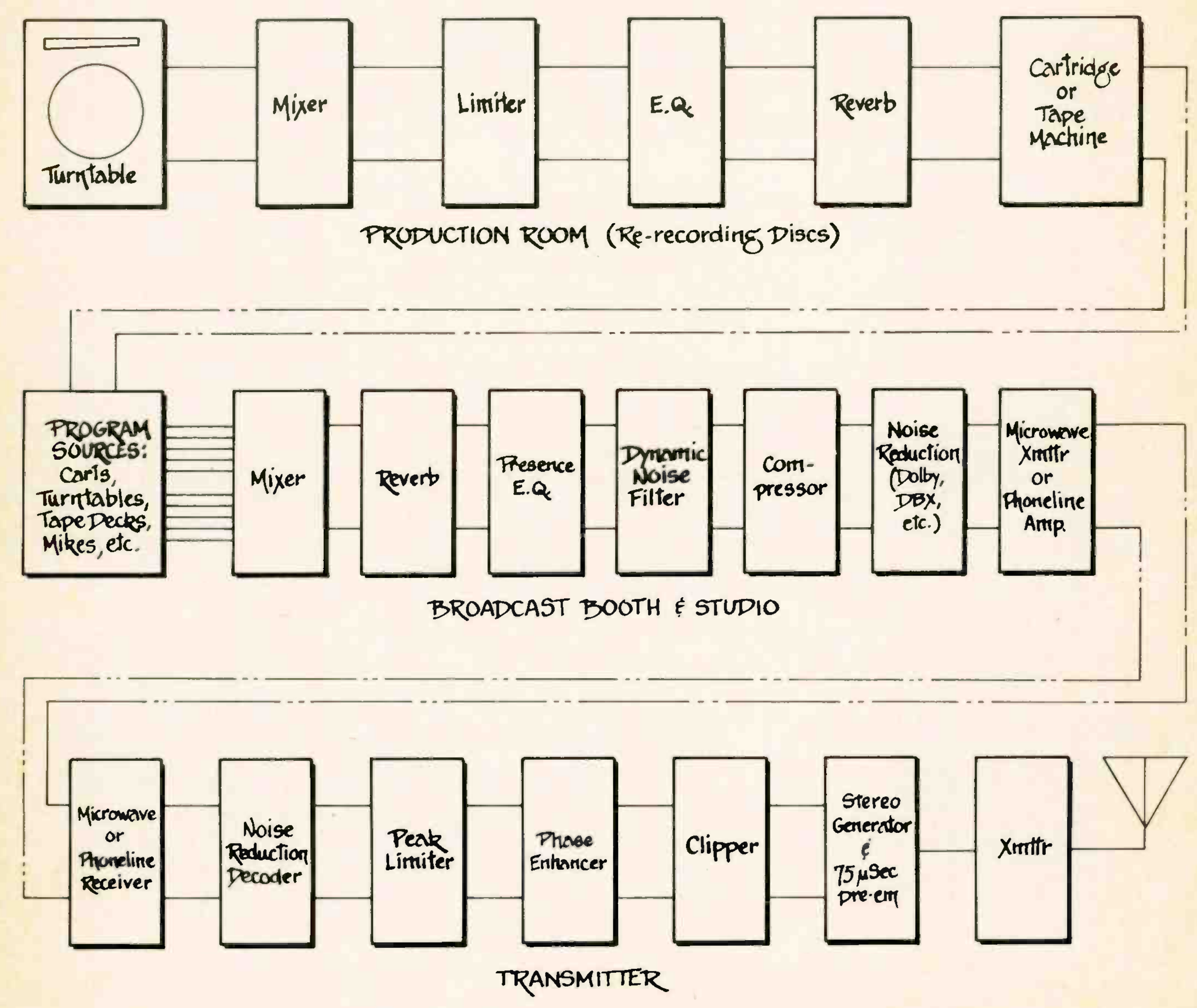
ing either very large play lists or essentially free-form programming. In these cases it becomes very expensive to commit their entire music programming to cartridge. In the course of normal use, records and pick-up cartridges take a severe beating in the form of hurried back-cueing, careless handling and storage. Broadcast pick-up cartridges are very often used at their maximum tracking pressure to prevent skipping and to permit the record to be slipped on a felt-covered turntable.

The RIAA playback equalization curve in the turntable preamps may be taken for granted and thus there may be no periodic check of the conformance of the preamp characteristic to any kind of tolerance of acceptability. That certainly will affect the on-air sound of the record.

OPEN-REEL TAPE

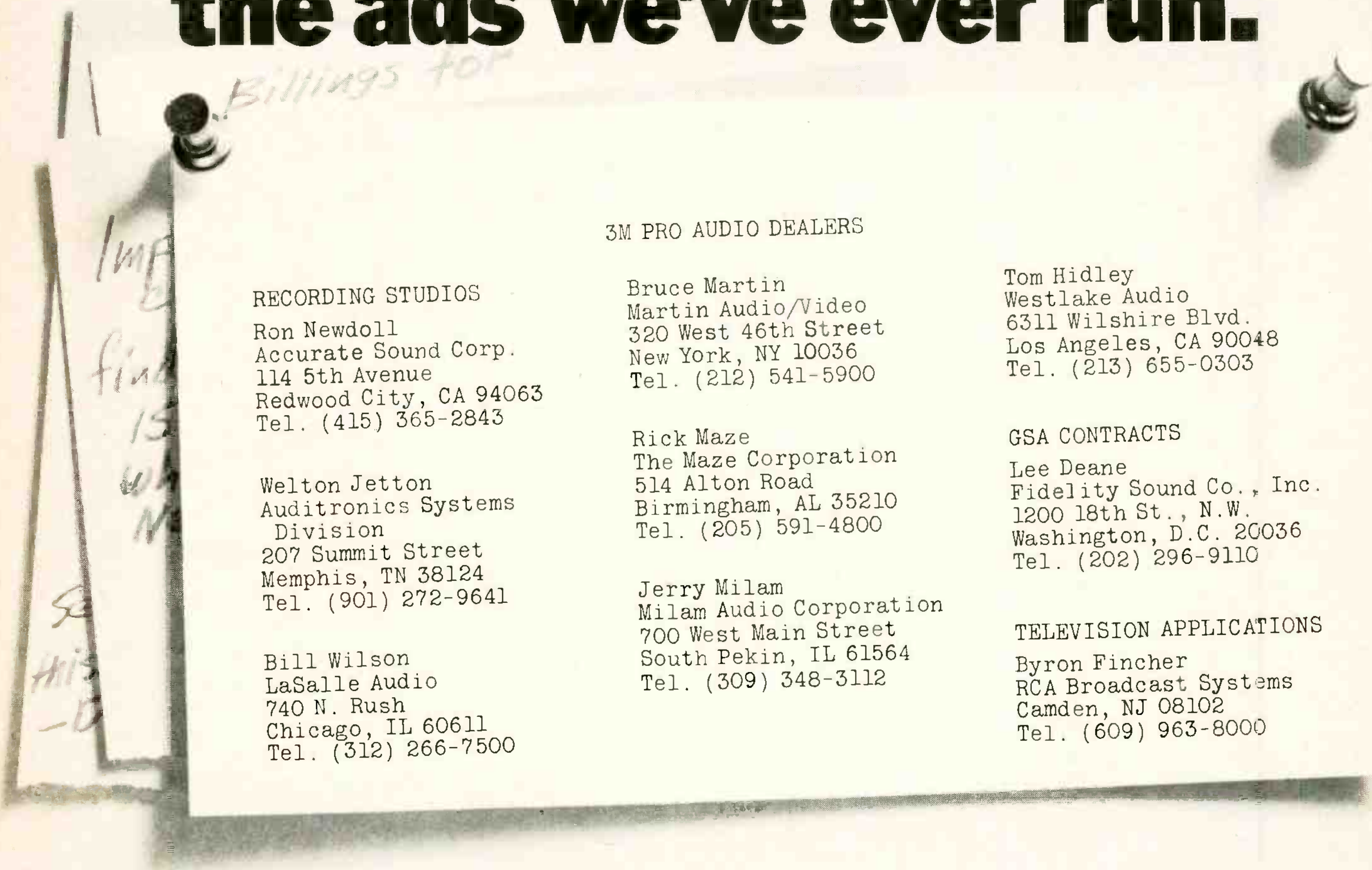
Open-reel ¼-inch tape is used in cases where the station programming originates from an automated system. The open-reel tapes are played back in some pre-determined sequence with commercials, time-checks, weather reports and station ID's being inserted from cartridges at the proper time. The open-reel tapes are generally recorded at 7.5 i.p.s. and have the capability of yielding quite good fidelity when properly aligned.

The majority of broadcast automation syndication services supply their program material at 7.5 ips in either a full-track mono or NAB 2-track format. A few employ Dolby "A" or DBX noise reduction. In this case, the syndicator exercises a great deal of control over the sound of the music that they supply. The automation equipment requires the inclusion of certain control tones that are used to indicate the end of a given program segment: a commercial, music sequence, station ID, etc. This takes the form of a 25 Hz tone recorded directly on the program track(s). The program material is pre-filtered through a 50 Hz, 18 dB per octave, high-pass filter to prevent low-frequency information in the music from accidently tripping the 25 Hz cue-tone



TYPICAL PROCESSING ELEMENTS IN AN FM BROADCAST CHAIN WHICH CAN INFLUENCE THE SOUND OF A RECORD

A call to one of these guys will tell you more about 3M pro audio gear than all the ads we've ever run.



If you know the recording industry, you know at least one of the men on the sheet above. And you know you can count on getting straight answers from them.

So if you want to know where 3M is really at in this business, these guys can tell you.

The fact is, 3M audio equipment has worldwide acceptance, and we back it up with a superb organization, our Business Products Service, with offices in New York City, Memphis, Washington, D.C., Detroit, Chicago, San Francisco, Los Angeles, Atlanta, Pittsburgh, Providence and Grand Prairie, Texas.

And we have a special flying squad of troubleshooters that we can dispatch from our St. Paul headquarters on a moment's notice for big problems. So when it comes to service, 3M leads the industry. Something you might want to keep in mind when some other companies have been cutting back on their service capabilities.

And we think it's great when the guys in the sales organizations around the country praise 3M equipment.

Because we think word-of-mouth advertising is the best kind there is.

Our ad agency hates it, though.

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Circle No. 111

for the channel gains must be assumed when the composite encode and decode circuits are designed. FM stereo separation should be about 40 dB between stereo channels, dropping to about 25 dB at 15 kHz. It is certainly less than that for stations using the composite method for their stereo tape program sources.

CASSETTES

There have been experiments in the use of the Philips-standard cassette as a program source. There has been no widespread use of cassettes as of this writing. American Forces Radio and Television Service successfully concluded a pilot test of the idea for mono programs within the last year. The program information was recorded on one track with noise-reduction processing, while control tones were recorded on the other. Stereo format cassettes were used, although the programs were entirely mono.

Regardless of the tape format used by the stations, the music program material tends to go more generations from the copy initially received by the station than it would for direct disc air play. This is because much of the processing of the recorded music is done at an intermediate stage in tape form. If noise and distortion were prominent on the record, no improvement will be realized through several more generations.

detection circuits in the automation control system. This obviously has an effect on the low-end response of the music as heard over the air. This will be true whether the programming is done by a syndication service or by the station itself.

Another factor in the use of syndicated programming material is that such music tapes are usually generated on high-speed duplicators. Any neglect of very high quality standards on the part of the duplicator of those tape copies will result in poor mono compatibility in the case of FM stereo. Changes in record/playback azimuth of as little as two minutes of arc will cause a phase-shift of about ninety electrical degrees and result in a 3 dB cancellation at 10 kHz. Cartridges are also very susceptible to this sort of degradation due to the slight mechanical variations in tape tension and guidance within the cartridge from one unit to another.

To try to minimize the phase-cancellation problem, a few stations are using a composite combination of the stereo program in the same way as is done in the transmitter stereo generator. The L+R and L-R signals are derived and recorded onto the tape medium. Upon playback, they are added and subtracted again to regain the two stereo channels. The major disadvantage of this approach is that separation is impaired by deviations of the channel gains, since very precise values

NOISE-REDUCTION

Some stations will attempt some sort of noise reduction measures at the program formulation level. They may use techniques as simple bandwidth restriction and Kepexing or other, more sophisticated devices such as the Burwen dynamic noise filter.

THE STUDIO PROGRAM CHAIN

Regarding the on-air program chain, the first thing the music must contend with is the broadcast console. These mixers typically are of considerably lower quality than their recording studio counterparts. Noise floors are generally not much lower than -60 dB with only 10 or 12 dB of head room.

The board output is then passed through a compressive gain-riding device. Typical of these is the CBS Audimax. Other manufacturers supply counterparts that achieve the same purpose. These compressors are used to maintain the highest average signal level for transmission to the transmitter. This helps maintain a high signal-to-noise ratio, although these devices tend to raise the level of electrical noise and any disc surface noise that may become prominent during whatever low-level passages that may have escaped the notice of whatever pre-air production the station may employ.

There are stations that use a reverberation device directly in the program line to give all of their program material a sound that they feel distinguishes that station from others in the same listening area. This sounds like a rather odd practice, but it is done.

PHASE ENHANCEMENT

Another little goodie finding increased usage in the FM broadcasting signal chain is some sort of phase enhancement device. What these devices do is constantly monitor the phase error of frequencies above some cut-off frequency and shift the time delay of one channel or the other to minimize the phase error. The idea of this is to try to cancel out all the phase irregularities occurring in the signal chain. This sort of device may be installed either at the studio or transmitter end of the audio chain.

Other processing devices may be used to attempt to dynamically alter the spectral distribution of all or part of the signal to meet some general subjective criteria established by the station programming staff.

As far as STL or transmitter telephone lines are concerned, there is an increasing number of stations employing either Dolby "A" or DBX noise-reduction systems in those signal channels. These measures are fine if they've taken care of business before the sound gets to the processors.

THE TRANSMITTER

At the transmitter end of the link, we arrive at a final stage of the audio signal limiting operations. This device is generally a fast-acting peak limiter of the type described earlier in our discussion of transmitter modulation characteristics. The peak clipping action may seem like an extreme measure to those familiar with recording studio signal processing. It is reserved as a last-ditch effort to restrict modulation input to the transmitter. That may make it an easier pill to swallow. The odd-harmonic distortion generated by this clipping tends to be reduced by subsequent low-pass filtering so it doesn't sound quite as bad in practice as might be expected.

The limiters and compressors used in the broadcast audio signal chain are not optimized for specific types of program material. They are set to whatever attack and release times and threshold levels as are deemed appropriate by the stations' technical staff and left at those settings. "Pumping" and "breathing" are often audible due to incorrect limiter settings or to inability of a given set of limiting panometers to adequately handle all program material.

DISTORTION

As long as we're discussing distortion, it might be well to mention that most AM receivers will yield about 3 to 5% harmonic distortion from a 100% modulated AM signal while high-quality FM receivers can maintain a distortion contribution below 0.5% in stereo mode and about half that in mono mode.

The reason for the higher distortion in stereo reception goes back to the demodulation of the double-sideband, suppressed-carrier difference composite discussed earlier. The carrier must be regenerated locally within the receiver using the transmitted 19 kHz pilot tone as a phase and frequency reference. If there is phase error between the 19 kHz pilot tone and the 38 kHz suppressed-carrier, a form of distortion will occur that closely resembles the cross-over distortion encountered in class AB complementry-symmetry solid-state power amplifiers. The demodulator itself is also subject to the traditional non-linearities and the resulting distortion products.

NEW DEVELOPMENTS

Recently, a few FM stations have been using the Dolby "B" noise-reduction process for the purpose of improving their audio signal transmission characteristics. The Dolby "B" system is used in conjunction with a 25 rather than the standard 75 microsecond FM pre-emphasis. The purpose of the change in pre-emphasis in conjunction with the noise reduction is to yield a more subjectively compatible signal for those listeners who do not have a Dolby "B" decoder in their receiver. The Dolby "B" processor and the 25 microsecond pre-emphasis change are mandatorily used together by FCC regulation.

The major benefit in using the Dolby process is that greater dynamic range in the modulation signal can be tolerated without the effective coverage area loss that would normally accompany such an increase in dynamic range. Because of the reduced pre-emphasis used with Dolby noise-reduction, the station can gain a few extra dBs of modulation level at high frequencies without the risk of overmodulation. The combination of the two permit the station to reduce its compression somewhat without a loss in effective coverage area. At this writing about a hundred FM stations are using the Dolby "B" system.

Since broadcasting is a numbers game, increased program quality with no loss in listening area will likely seem attractive to many broadcasters. With any kind of luck, the quality fever may spread to the benefit of the listener, the broadcasters and the recording industry. With the advent of some very sophisticated home receivers there is some incentive for the radio stations, FM stereo especially, to strive for higher sound quality. This will have to suffice for a summary of whatever light may be at the end of the proverbial tunnel.

REMEDIES

With that summary of the audio obstacles standing in the path of your records' journey to the waiting public's ears, most producers may be inclined to just hang it all up as a bad job and forget about degradations inherent in radio air play. That decision may be a bit hasty. There are some remedial measures and precautions that can be undertaken in the studio to minimize the aesthetic risks of running the radio gauntlet.

MONITORING

When mixing a record specifically for air play, it would be useful for the producer to check his mixes in mono level and stereo on speakers representative of those found in fairly low-cost radio receivers. The normal control room monitors should be used as well, in both mono and stereo, to see that there are no big surprises in the way of signal information outside the pass-band of the El Cheapos lying in wait for the hapless disc mastering engineer.

An oscilloscope or other phase indicator should be used to monitor the phase of the mix as well as to ensure that all of the tape equipment used to generate the stereo or mono mix-down will maintain the phase integrity of the record.

Any corrective balance and equalization changes to the mix are a lot easier to handle at the multi-track mix-down stage than they are after the record has been pressed. If there is less reason for the program director to attempt to modify the sound of your record, there's that much more chance that he'll leave it alone. The mix-down is the place for removal of any lyrics that may prove sensitive to the program directors' ears. In the light of recent FCC decisions regarding questionable lyrics, the deletions would be better left in the hands of the producer than forced upon the program director.

METERING

It would be well to use peak-responding level meters to get some idea of how the radio station modulation monitors would see the music. Radio station modulation monitors are peak-responding devices having threshold lamps that light when thresholds are instantaneously exceeded.

In the case of a mix intended for FM airplay, it would be well to have a special meter constructed incorporating a 75 microsecond pre-emphasis. The mixer could then get some idea of how his mix would look to an FM peak limiter. This whole process of monitoring and metering is an attempt to achieve a compromise final mix that is most likely to sound fairly good over the sound systems on which it will generally be listened to.

A side benefit of the 75 microsecond pre-emphasized metering is that the RIAA disc recording characteristic also incorporates the 75 microsecond time constant. Therefore, the pre-emphasized meter should yield a fair indication of what the cutting head will have to contend with during the disc transfers.

Peak responding metering can be achieved in several ways. Burwen Labs offers an active module that converts an ASA standard V-U meter to a peak reading device switchable to its original characteristics. The Neumann disc mastering console has a couple of light-beam peak meters in it. Copies of those meters may be available through your friendly, neighborhood Neumann dealer. Quad-eight offers a peak-responding V-U meter having LED indication in a couple of packages. Your phase scope can also be used as a peak meter by recalibrating the graticule in decibel units.

ANTICIPATING THE PROGRAM DIRECTOR

The final equalization of the mix should be checked at both low and high levels to get an idea of how it would sound to a listener under the same circumstances.

Knowing of the "loudness race" in both the AM and FM bands, the producer can attempt to choose a final equalization that will bear up well under the presence-boosting techniques typically used by the station programming people.

SPECTRUM ANALYSIS

A real-time spectrum analyzer such as the White or Amber would be a genuine aid toward tailoring the spectral content of the final mix. If the upper spectrum, say above 8 kHz is rather hot, you may as well save yourself the cutting problems. If the record is for AM, it won't get out of the antenna. If it's played on FM the infamous peak limiter and clipper will wipe it out before it hits the stereo generator.

If the production budget won't stand the strain of special mixes for DJ promotional pressings, you'll just have to decide where your best course of action lies. It's a cinch to say, though, that a record with a 60 dB dynamic range is not likely to emerge from a radio receiver in anything like it was at the final mix.

MONO FROM STEREO

Many times the same mix is used for both AM and FM promotion. In this case a stereo record will be sent to both stereo and mono stations. Many AM stations will treat a stereo record as if it was mono in a couple of ways. First, there is the quick and dirty approach of strapping a stereo pick-up cartridge so that the right and left outputs combine in series to yield a straight L+R combination of the record. Some of them may simply drop a channel or may re-mix the stereo signals for the mono mix that they feel fits in with the way they want to sound to their listeners.

The practice of strapping the stereo cartridge for mono has been used for years and still persists widely. This is why it is vital to the success of promotional records that they be checked for mono compatibility at every stage of the production process. You will recall the composite form of the baseband audio signal as it's fed to the FM transmitter modulator from the stereo generator. Same story there, too. If it's not mono compatible, it ain't gonna fly.

The truly dedicated producer may go to the extent of mixing his product in the front seat of a '55 Chevy. That is a somewhat extreme approach to the problem of verifying mix compatibility in the context in which it will be heard. Some studio facility constructors may be moved to introduce a monitor having the acoustics of a GMC Dieselcab for C&W records, a surfer van for FM stereo rock records, and a statistically selected kitchen for mixes going to schmalz stations. There must be equipment marketing possibilities there somewhere.

FIGHTING THE PHASE ENHANCER

The phase enhancers coming into wider use by FM broadcasters are worth consideration from the recording end of the business, too. Use of very short time delays between stereo channels will drive one of those things crazy. The phase enhancer looks at the higher level signal components in establishing its phase correction function. A one millisecond delay will cause the phase enhancer to alter the phase delay of the stereo program channels to cause interference nulls at 1 kHz intervals throughout the mono combination spectrum. The thing to watch is use of short delays between stereo channels. If the net delay between the channels is the same, you're still OK. Extreme separation between stereo channels for any length of time over a couple of seconds can also cause the phase enhancer to go spastic. If the program content between left and right sides doesn't have some sort of commonality, it won't know what to correct for and wreak havoc with the mono composite signal.

The delays originating from multiple miking of single sound sources at the time of recording should also be watched carefully at the time of recording. The acoustic delay has the same effect as the artificial kind. It is important to realize that the time delay problem will be hard to see on a scope for a broad band signal. It will show up for a single frequency but not for a complex waveform.

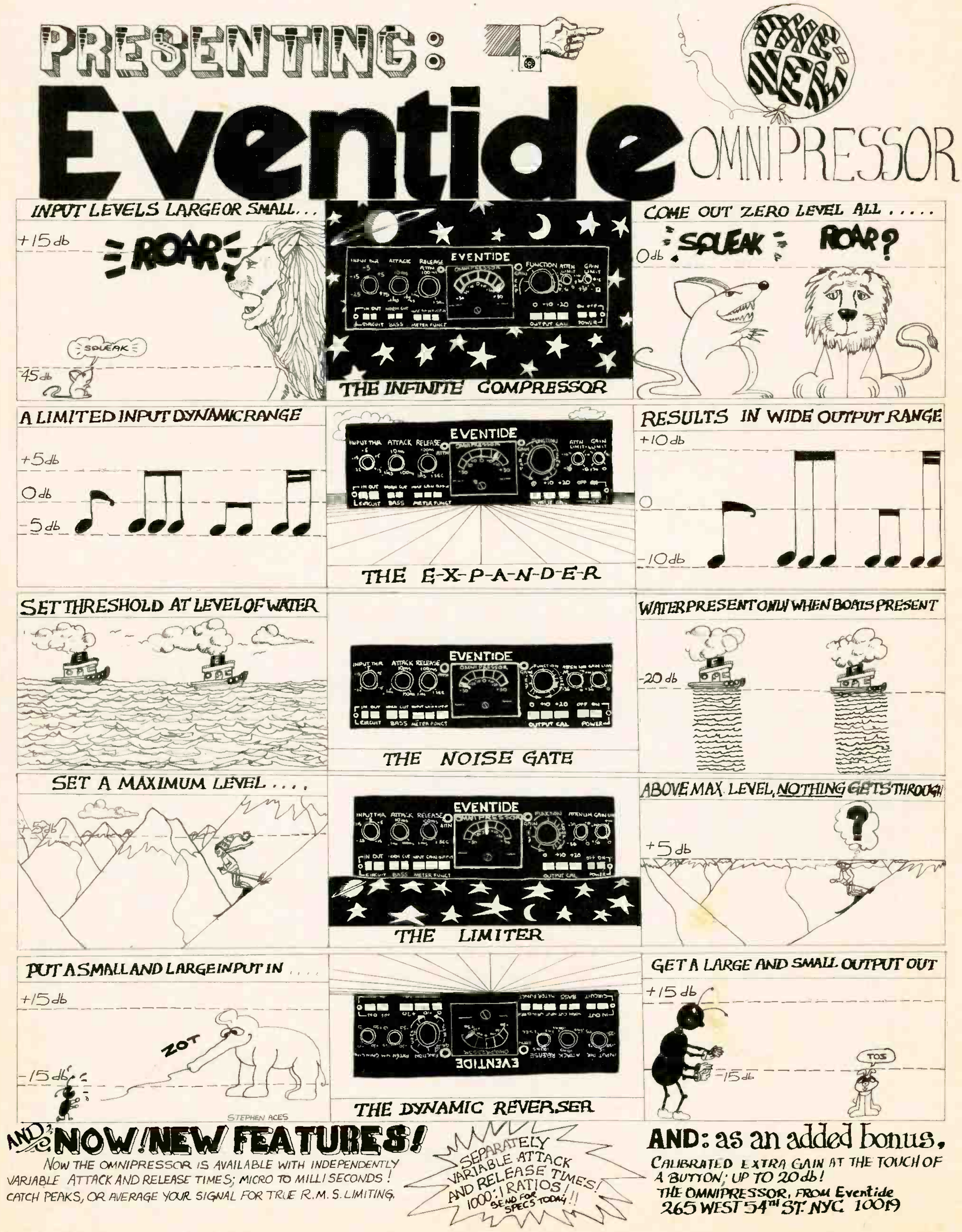
PARTING THOUGHTS

Generally speaking, it is in the best interest of the record producer and the broadcast stations receiving his record that the recording be of the highest possible quality. The route to this end ought to be well known by the entire recording industry at this late date.

Careful and thorough equipment maintenance, all along the way, matters very much. Use of noise reduction will yield a quieter recording that will not worsen the noise problems of the stations themselves. Care to minimize distortion will also result in a cleaner on-air sound. Both of these effects are cumulative in nature and the radio station obstacle course won't make them any better.

In general, the producer should try to have his product in a form that the radio stations can use with a minimum of modification. Any editing should be done before the stations receive the record than after. He should attempt to anticipate and compensate for as many of the dandy goodies in the broadcasters' bag of tricks as he can.

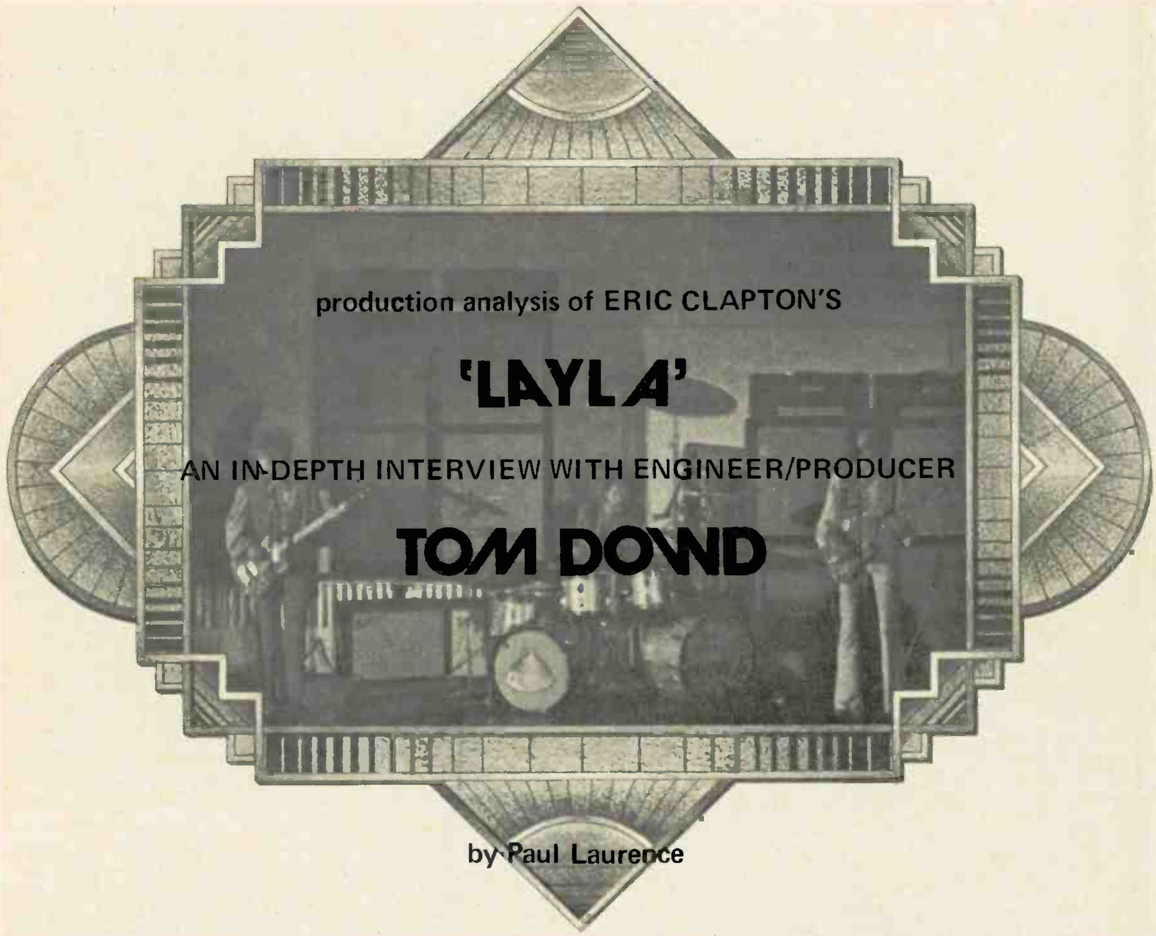
The record is caught in a numbers game just as the radio stations are. Realization of this fact is the foundation of a mixing philosophy that will be more likely to get air play for your record.



Circle No. 116

Tom Dowd has participated in as much recording history as, maybe, anyone around today. During his 28 years in the business, he has recorded and/or produced, among others, John Coltrane, Charlie Parker, Herbie Mann, Ray Charles, The Coasters, The Drifters, Aretha Franklin, King Curtis, Otis Redding, Dusty Springfield, The Young Rascals, The Allman Brothers, Stephen Stills, and Joe Walsh.

He is probably most noted, though, for his longstanding association with guitarist Eric Clapton. Originally Cream's engineer, and in recent years, Clapton's engineer/producer, he is the closest professional associate of the man who, by either musical or sociological standards, must be considered the most influencial musician of our time. Clearly, Eric Clapton, more than anyone else, has defined the style and set the standard for the era's most prominent instrument – electric guitar.



production analysis of ERIC CLAPTON'S

'LAYLA'

AN IN-DEPTH INTERVIEW WITH ENGINEER/PRODUCER

TOM DOWD

by Paul Laurence

"Layla," the only studio effort from Clapton's short-lived Derek and the Dominos, has met with unqualified success both critically – very rare for a double album – and commercially. Stylistically, it was among the first of the "Southern Boogie/Funk" records. Additionally, it contains a few slow blues, some country & western, and fleeting elements of jazz, folk, and Polynesian. For this project, Dominos Bobby Whitlock (keyboards), Carl Radle (bass), and Jim Gordon (drums) were joined on guitar by the late Duane Allman, then all but unknown to most of the American audience. Almost certainly, "Layla" is the definitive recording of those precious few occasions where two legendary rock soloists have collaborated in the studio.

Technically, it is a pretty basic album. There was a definite emphasis on performance, with many live vocals, often no overdubbing, and little in the mix to render the tracks otherwise. With regard to effects, there is almost no dynamic panning, little echo, and only a moderate amount of equalization and limiting. All the tracks are placed relative to the five-point stereophonic spectrum.

The album's vocals (usually only two per song) are pretty natural-sounding, maybe even a bit thin at times. Clapton's is usually the louder, drier, and hence more up-front, Whitlock's being just the opposite. Interestingly, they were occasionally sent to placement points 2 (Whitlock) and 4 (Clapton), roughly approximating their positions on stage.

The sometimes-stereo acoustic guitars and keyboards are likewise pretty unmodified, occasionally sounding a bit thin as well.

The electric guitars often have an attenuated low end, with occasional limiting. As Clapton played his then-favorite Stratocaster at some comparatively low volumes, they often have a somewhat tinny, "attack"-y sound.

The laterally-placed bass guitar is rich in the lower registers, and occasionally limited.

The drums are pretty "airy" due to liberal distant miking, and are given a full stereo spread. Level-wise, they are moderate, with but a moderate amount of kick drum.

PAUL LAURENCE: Tom, were there any people who influenced you as an engineer

Continued

" . . . I knew that I could make a career out of this business . . . I had the good fortune to run into two young gentlemen by the names of Herb Abramson and Ahmet Ertegun . . . that was the beginning of Atlantic Records . . . "

and producer, or were you too early in the game to be "influenced"?

TOM DOWD: Well, in those days, there *were* no recording engineers per se. "Recording equipment" was usually hand-me-down radio equipment and recording engineers were, for the most part, radio engineers who were working extra time or relegated to doing recording instead of radio broadcasts. There were no "recording engineers" because there was no recording equipment!

PAUL LAURENCE: Did you get into recording by being "relegated" in this way?

TOM DOWD: Actually, no. I'll tell you how I got into recording. It was 1947, and I'd returned from three years in the service and had gone back to school for a year, and decided I deserved a holiday. Looking through the "New York Times" – being a native New Yorker and reading it faithfully every Sunday – I saw an ad for a recording studio that needed somebody for a summer job. Though I was a physics major, I had always enjoyed music – having been in various orchestras and bands through school as a musician – and felt that this would be a great deal of fun. I went to work for that studio, and within a short period of time realized that the recording technology as a whole that existed in those days was easily within the grasp of any training I ever had with my engineering and my physics. I knew that I could make a career out of this business and have a thoroughly pleasant time the rest of my life with it.

During the first two or three months I worked there, I had the good fortune to run into two young gentlemen by the names of Herb Abramson and Ahmet Ertegun who were starting a record company. That was the beginning of Atlantic Records.

PAUL LAURENCE: Were they specifically aiming to form an R&B label?

TOM DOWD: No. The first records on Atlantic were jazz and gospel. Boyd Raeburn, Tiny Grimes, the Harlemaires, and the Gospel Harmoneers out of South Carolina – those native traditional groups were the kinds of things that Atlantic was interested in. What we now call rhythm & blues was then called "race music." Ahmet and Herb were primarily interested in jazz and gospel music, or at least that was what their initial endeavor was. Like me, they were professionally trained but fancied music as a career. Herb Abrahmson was a dentist – I believe he had just graduated from Georgetown in Washington – and Ahmet was a Thomas Aquinas scholar and graduated from St. John's University in Maryland. I might add that they were both very knowledgeable about music. It was more than just being "learned" – they had the facility for determining good from bad, and pure as opposed to derivative.

PL: I had always put you in the category of "engineer/producer." Do you consider yourself that?

TD: No, not any longer. It's not really fair for me to accept the title "Engineer" any more. The time that I devote to producing has taken away those hours that you must spend to be an engineer today in order to update yourself on all the new techniques and equipment.

" . . . you try to ascertain how real the project is – how much you can satisfy the artist and still satisfy the public's image of the artist . . . "

PL: Do you feel out of touch as far as your engineering goes?

TD: Ah, I'd say that I'm "in touch" but not in the mainstream. Let's first define the term "engineer." To me, "engineer" implies that a person is learned in the current state of the art. He should know how to best utilize the equipment from the studio floor, through the console, to the recorder, back to the mixdown, and onto disc, so that from beginning to end, he knows the abilities and limitations of every piece of equipment he has. He's got to be able to picture that record, and determine how best to get there with all that is available to him *That's* an engineer. There are too many managers, hangers-on, and people who might even be musicians or singers in the group who have an ability for hearing and arranging sound the way they want to hear it, but have no knowledge of the equipment. They are often the ones who are the most sorely disappointed with the product, because they don't realize that there are things that you can entertain yourself with doing and still not be able to get onto the record exactly the way you want.

PL: What is the extent of your engineering today?

TD: I re-mix most of the albums I get

involved with. On some I'll institute the initial recording because I might think that there are some ways of using the studio that differ from the way the house staff does it, and so I might get involved there too.

PL: In your experiences as a producer, do you have any "guiding principles" or basic jumping-off points as to what a producer should be or do?

TD: Well, it depends on the project and the artist as to exactly what hat you'll be wearing. If I'm working with a group, I should be familiar with their styles, what their limitations are, and what they're extra good at. After we've determined the material and the goal for the project, I work with the individual members to get the ultimate contribution out of each one of them, and give confidence or advice where it's needed. I try to have a one-to-one relationship with all the members of a group.

When I'm dealing with a single artist, I try to find out what image he has of himself or would like to have, what records he likes, and generally try to gain his trust. Then, in a bit of soul-searching, you try to ascertain how real the project is – how much you can satisfy the artist and still satisfy the public's image of the artist. Sometimes an artist will get carried away, and might spend too much time doing a song, half an album, or a whole album that is really only rewarding to him, which will hurt because the public won't accept it. With respect to this, I often influence material, choice of keys, musicians, etc.

PL: I'd really like you to elaborate on something you said earlier. Let's talk about the "limitations" and the "what they're extra good at" of some of your artists. How about Aretha Franklin?

TD: Aretha is one of those most unusual artists. Aretha does not need a producer – she needs a confidante, that's all. She just needs somebody there when she's singing with whom she can share what she's trying to do. Sometimes, when she hears back a performance that has completely captivated you – you'll say "This is the best singing I've ever heard you do!" – she'll listen to it a few more times and say "I can do one better." She means it – it's not an ego trip and it's not theatrics. She actually *knows* that there's something in there that she can do better. Aretha has an incredible facility for judging her own performance and knowing how much room there is for improvement.

PL: Would you say she has any "limitations"?

TD: No. She can do absolutely anything she wants to do.

PL: By contrast, how about the Allman Brothers? What are they good at, and where might they need guidance?

TD: I have not really done anything with the Allmans since Duane's passing. The last project I did with them was, I think, "Eat a Peach" or the Fillmore resumes. The Allman Brothers needed what I guess you'd call a disciplinarian more than anything else. First off, it's unusual for a band to have two drummers. It's also unusual for a band to have two lead guitar players as good as Duane and Dicky Betts. Much of what I did was simply ironing out the polyrhythmic confusion that often existed as a result of those two guitars and two drums. Now you can't just go in there and say to them "You play this and you play that" – you have to put it diplomatically. It would be more like "Why don't each of you take turns on that lick, and then that will make room . . . ", etc.

PL: Were they generally amenable to your suggestions?

TD: All the time. To this day they are. I've been doing some recording in Macon, and I see Dicky once in a while, I see Gregg, I see Butch, and we'll talk and they'll say "Will you listen to some sides we've done?" They always realized that when I would say something. it was never taken as "criticism" as much as "I like what you're doing but it's not happening as well as it could be."

" . . . there are some times when you sit in the studio . . . and hour seems like a day . . . 'cause nothing is going right . . . "

PL: What do you consider their strong points — where they need you the least?
TD: Oh, I could never tell them about their solos — they knew.

PL: This might be kind of an unfair question, but can you name the five projects that you most enjoyed being a part of?
TD: Hmmmm. Well, the first one is definitely off-the wall. It was a recording I did in September of 1952 with Wilbur De Paris, who had a Dixieland band in New York. His brother was Sidney De Paris, the clarinet player was Omer Simeon, and I can't remember the drummer or the bass player. Anyway, there was a chap in the metropolitan area at that time who was advocating stereo recording — in those days *binaural* recording. His name was Emery Cook and he was a wild-haired genius engineer and a recording enthusiast who fancied doing raucous things. He proposed to us recording this band in binaural, and as it was jazz, Atlantic was quite interested. I had spent some time with Emery and was quite captivated with the sound and clarity he could get, and I told them "You're gonna like the music, I'm gonna like the recording, let's do it!" In September 1952, we made our first stereo recording. To me, it was a real milestone, especially because stereo didn't really happen to the American public till eight or nine years later.

PL: What was the tune?
TD: It was a Dixieland *LP*, actually. It was like a live concert — we hired a hall, put the band on stage, and put two microphones up. If they took four minutes for each number and they did six numbers, it took half an hour to do and boom we were done. They were very professional — good performances, good solos, and for those days sensational sound.

PL: Was it actually released as a stereo record?
TD: It was released as a binaural record initially. It involved two cuts on the same side of the record that you played with two pickups simultaneously — one was the left channel and one was the right channel. Musically, it was acknowledged as a fine album, but there were not too many people who wanted to spend the money to buy the equipment to play it the way it was best reproduced, so it was put out monaurally too. It's still in the catalogue, if I'm not mistaken, and reissued periodically in those "Best of" series.

I guess the next little project would be my becoming familiar with Les Paul and real "multitrack" recording. After I went up to his place in New Jersey and saw *his* equipment, I went into Atlantic and said "Hey, people are arguing about 2-track

" . . . so I went . . . in 1957 . . . and I ordered an 8-track recorder . . . I was the laughing stock of the industry . . . New York thought I was crazy . . . "

recording. Forget it. There's a recorder available now where you record on wide tape — eight tracks — which is a much better way of storing the information." And they went with me — they said "If you believe that it's going to help us make better records, get it." So I went, in 1957, and ordered an 8-track recorder. I was the laughing stock of the industry, New York thought I was crazy, everybody was bananas. From the time that machine arrived until about 1962, I saw every other record company and every other studio in the country go through the painful process of going from 2-track to 3-track to 4-track. Every year, they'd amortize the equipment or write it off, and go up another track. Ultimately, they all went to 8-track anyway. We just took a shortcut. There is a pile of records that we made the first year on that machine — the Coasters, Lavern Baker, Wilson Pickett, Bobby Darin — any one of which could have paid for it with one week's sales! I'd say that step — getting that 8-track recorder — was a milestone for me. I look back on myself and I say "Boo" to the world. I was five years ahead of them.

Another one of my all-time favorite projects would have to be the Otis album that we did in Memphis — Otis Redding's first album. That's the one that has "Satisfaction" on it, "Respect," "Down in the Valley." Instead of doing nothing but slow songs like "Pain in my Heart," they did more rhythmic, up-tempo stuff. You know, there are some times when you sit in the studio where an hour seems like a day, 'cause nothing is going right, and there are other times when you sit there and say "My God, I've done an hour and a half's worth of music and it's only 4:00!" This was one of *those* albums — we did it all in one day, I think.

There are still more. The "Layla" album, of course. Can I overlook a John Coltrane experience? I can't ignore the first Ray Charles album — that's certainly a milestone. The Herbie Mann "Memphis Underground" album? I can't say no to two or three Aretha albums either. And this is not to forget one I'm doing now with Rod Stewart and some of those same Memphis musicians. After about 25 minutes of recording, Duck Dunn and Al Jackson came over to me and said "It's like the feeling we had with Otis – listen to that boy sing! Where you been hidin' him?" It was all quite reciprocal, because later Rod came walking out of the studio and said "Mai gawd, wha' a band!"

PL: Do you have any favorite hardware, like mikes or limiters?

TD: No. I'm a firm believer in using whatever equipment will do the job. There are some microphones that are designed for a very specific application that I wouldn't say that you shouldn't use in an unusual fashion. Just because it's designed for PA, that doesn't mean you shouldn't use it in the studio, or if it's designed to be nine inches from the vocalist, you shouldn't have him "hug" it. I'm appalled by some of the things that are put on the market – some by little fly-by-night firms, and others by large reputable companies – that violate the obvious chronology of progress. You know, things that revert back to an old method, and they say "But this is the better way." I'm not talking about home equipment – I'm talking about consoles that can cost up to $150,000! I'm against a manufacturer mass-producing an item, saying "This will solve all your problems" where it's not an improvement, it's just a bad permutation of something that's already failed once and they're trying to foist it on you another way.

PL: Okay, but let's set up a completely hypothetical situation. Suppose that you're the only engineer in a totally unfamiliar studio where they have every single type of hardware ever made. What would you start out with – are you partial to Neumann or Kepex, for example?

TD: Well, first off, the musicians should position themselves where they're comfortable. If I cannot take advantage of what the studio recommends as the best placement in the room for those people, then the house choice of microphones might have to be altered too. Where I might have had physical separation, now I might have to use a high front-to-back rejection mike because the bass and guitar amps are six inches apart. If the studio is accustomed to having the drums in one corner, the piano over here, and the guitar and bass in traps, and they're using omnidirectional microphones, that's all very well and good *if* the musicians are comfortable. However, if they're not comfortable that way and end up standing on 14 square feet of a 20x40 room, I can't use omnidirectionals. Once the musicians are physically comfortable, then I can try to give them the sound they want.

Certainly, I'll use directional microphones where directional microphones are necessary, but I'm not too concerned with whether they're Electro-Voice or AKG. An instrument like a guitar, I would, for the most part, record as a mono track. In a case like this, you normally go for a very tight focal field on that source of sound, as opposed to something like drums, where the man is flailing about over a large surface area. With drums, I want to capture the motion and the depth, so I'd want a big spread, meaning distant miking. You don't want them very tight, where you have to *manufacture* the sound he's creating – you want to be able to capture his technique and dynamics just as he did it.

PL: So you normally don't limit as a rule.

TD: I try not to. Often it depends on the complexity of what you're trying to record, but I believe that you can usually get away without limiting anything on an initial recording.

PL: What were the circumstances surrounding the making of "Layla"?

TD: Well, Eric had this new group, and they felt that they'd better find out what they're all about and do an album. I'd always had pleasant dealings with Eric and Ginger and Jack, and with the Stigwood Organization, and when Eric wanted to record, I was asked. At that time, the best place to do it was in Miami because that was where I was working. If I was in New York, it would have been done in New York.

". . . with drums, I want to capture the motion and the depth . . . so I'd want a big spread . . . meaning distant miking . . . you want to capture his technique and dynamics . . . just as he did it."

PL: How long did it take to record?
TD: About two-and-a-half weeks.

PL: Had they rehearsed beforehand?
TD: They had a concept for each of the songs, but as I say, they had played them and listened to themselves in rehearsal halls, but they had not ever heard themselves back in a mirror image.

So they came into the studio and started horsing around. While they were doing that, we were setting up and working on the sound. Then we did a couple of passes so everyone could hear what they sounded like on tape, so we could make adjustments. Jim would say something like "Gee, I wish I had more bass drum," or Carl would say "I don't like this amp" and that would be changed.

PL: So each musician exerted a strong influence over how his instrument should sound.
TD: Oh yeah. You have to look at a group with that talent level and remember that each one is a soloist. You can't say to one "Well, I'm making a vocal record, to hell with you." Clapton is very very strong but extremely quiet. He will sometimes say something like "I don't like the way the bass sounds," but that doesn't mean that the bass player can't say "Well I *do* like the way it sounds." Or Eric might say "That's lovely," and if the bass player wants it changed, Eric will say "Let's hear it that way then." He's the leader, but there's wisdom and judgment – he would never say "This is what I want and I don't care about you."

So they were playing around, jamming, and whatnot while we were still getting things straightened out. This is what's been coming out in the last few years – alternate takes from "Layla," which were really just rehearsals.

Once we got everyone sounding the way they wanted, then we could sit down with them and find out various things about the song, so we could start trafficking tracks. How many voices will it ultimately have? How many guitar parts? Is it going to be piano *and* organ? You've got to plan ahead.

PL: I noticed you got a fairly "windy" drum sound. How did you mike the drums?
TD: Jim Gordon is a very tasteful, very strong drummer. Because he has such incredible facility, you have to be careful that you don't over-mike him – you could miss some of his dynamics because you have too many mikes fishing around. If he had five tom-toms up, and you miked every doggone one, if he hit a cymbal that was anywhere near those tom mikes, it would be leaking in too many directions. Speaking of cymbals, he had a little dinky cymbal that once in a while we would put up. It was just a toy, really – he could have never gotten that sound with any of the other cymbals. Every once in a while, he'd just reach over and smack it. No matter where we put it, he'd manage to hit it hard enough so that it always came through!

That sound you're referring to was a result of distant miking. We used the overhead mikes – 67s or 87s – like "spotlights." You have to adjust them initially for height and angle so that their fields don't overlap and create a "hot spot." Once you've got the right focal plane in the down line, then you work on the vertical axis, so that you can catch a better proportion between a tom-tom sitting down on the floor and a cymbal way up on top of a stand.

PL: How about the organ?
TD: For the organ, we used an omni – on the top and an RCA 77 ribbon mike on the bottom. We usually took two tracks – the high end on one and the low end on the other. The high mike is in back, and the one for the low is down by the lower baffle and around the side of the cabinet where you're protected from the rumble of the motor. If you look at a Hammond cabinet from the back, you have a shelf, then your rotating horn device, and then you have this big dumb dodo of a woofer that rotates and makes all that horrible rumble. For the top vent, you can place a microphone at about a 20 or 30 degree angle to avoid the wind and draft deflection that the rotor causes. Down here, to get rid of the "woof-woof" of that thing rotating, you would go to the side of the cabinet so that the baffle affords you some screening.

PL: How did Duane Allman come to be involved in "Layla"?
TD: Well, Duane and I were into a recording project – maybe it was "Idlewild South" – around when I got the call from the New York office saying that Eric was wanting to record Derek and the Dominos. The next time I saw Duane, I said "You've got to meet Eric Clapton," and he said "Oh, I'd be embarrassed to, he's such a fine guitar player." Soon after that, we had to part company as he was going on tour or something. Now when Eric came down with Bobby and Carl and Jim, I said "I was just working with a fine guitar player named Duane Allman last week, and I'd love for the two of you to meet." Eric looked at me and said

"God, I love the way that guy plays, but I'd be too nervous to be in the same studio with him." They both stood in awe of each other, both two very soft-spoken gentle human beings. As fate might have it, after we'd started doing Derek and the Dominos, the Allman Brothers were doing a concert in the Miami area. One night in the studio, I said to Eric "Would you like to go down and see Duane Allman and the band play tonight?" and he said "I'd love to, but don't let him know I'm there. I'd be embarrassed if he asked me on stage." So I called up Duane and said that I might be able to bring Eric by, and he said "Don't tell me if he's there 'cause I'll freeze. I can't play in front of *him.*" To make a long story short, they finally visited with each other at the concert that night, and later on, the Allman Brothers came up to the studio. Eric and Duane went off in the corner and spent like seven hours talking to each other and trading licks. The first time that Duane had available, he came back and played on the "Layla" album. The two of them just fell in love with each other.

PL: Did you feel early on that this was going to be a landmark album?

TD: When we were making it, I felt that it was a mighty good album. I knew that the music was good, the songs were good, and the performances were outstanding on the part of every musician. When I finished mixing it down, I walked out of that room and said – and several people have teased me about this – "That's the best damn album I've made in 10 years!"

(at this point, the "blindfold test" was initiated and "Layla" was put on)

I LOOKED AWAY

TD: Hah, there's that Delaney Bramlett influence! The sequence of the songs on this album in the order in which we recorded them. Actually, I should qualify that a bit – some of the tunes were done before Duane arrived, and he was later added to them.

PL: It's strange to me that Clapton's made so little effort to preserve the Cream licks.

TD: Yeah. He's not ashamed of that stuff – he's proud of it. It's just that he doesn't believe he has to wear that coat the rest of his life. Eric doesn't walk into the studio and say "I've got to make a record as good as . . . ," he says "This is what I want to do now."

PL: Were any of these tunes actually written in the studio?

TD: Bobby Whitlock indicates that a few of them did come out of things that transpired in the studio. We've talked about this a couple of times, and he said that there are one or two songs that they only had the faintest clue to when they walked in. I don't know which songs he's referring to, but I would guess some of the later ones.

BELL BOTTOM BLUES

PL: Was this a case of ultimately using both lead guitar tracks, or was it originally conceived of that way?

TD: Well, when you're playing one part, you'll often hear another in your head. Then you'll decide to re-do it *that* way, still keeping the first one. When you play them back together, you realize that they're complementary and should be together.

The vocal harmonies here are excellent.

PL: Were all the vocal parts worked out beforehand, or were some generated in this same way – by hearing what they already had on tape?

TD: Both things happened. They'd usually do two live, and there was always the possibility that Eric would go one over Bobby or Bobby would go one over himself.

Then they suspend the ending, which is just as bizarre as everything else they did on this song.

KEEP ON GROWING

TD: This is the tune where Jim re-did the drum track.

PL: Do you have any of the original drum

track in there?
TD: Yeah, I'm sure there are some elements of the original track. I remember that when we put the tabla on, he said "I want to re-do the drums."

PL: Isn't that considered very difficult?
TD: Well, I wouldn't trust it to but one or two drummers that I've ever worked with, and he's one of them. I might go with Al Jackson the same way. The problem is overdubbing drums isn't really meter, but *feel.* A drummer might have the best time in the world, but he still has to be responsive to the music as a whole and the other musicians. If one of the guys staggers or lays back a bit, he has to make an instantaneous decision: does he lay back with them, or does he do something to complement their laying back? That's the spontaneity of playing drums – how perceptive are you and how quickly can you respond? The reason Jim wanted to re-do the drum track was because the lyrics didn't turn out to be where he thought they'd be. When he finally heard the lyrics, he realized that there were some places that called for him to break out and "punctuate."

Oh, that ending. They still do that and it aggravates me. They sometimes get cute and put these little post-mortems on.

NOBODY KNOWS YOU WHEN YOU'RE DOWN AND OUT
TD: This is one of those songs where the performance and the sincerity of the endeavor really come through. It happened one night when we were talking about old jazz blues, and somebody said "You know, nobody ever plays 'Nobody Knows' . . ." – one of those things.

PL: I really like Clapton's dynamics here – the way he brings out that arpeggio as a fill, almost.
TD: Exactly! If I had a limiter on it, those dynamics would all have been erased. This kind of thing is always a challenge, when you're not sticking a limiter on to make sure that it doesn't overload or whatever. You're sitting there having a great deal of anxiety, hoping that you can anticipate what they're going to do. Instead of "painting" the picture with equalizers, add more echo, do this, do that, you're taking a "snapshot" – trying to capture it just the way they did it.

I can promise you, when they do things like that, the earphones are off. I'm not against earphones – don't misunderstand that – but often when you're using phones, you don't relate to the other musicians as well.

PL: Was Eric playing as softly as it sometimes sounds, or did he have an amp that just wouldn't break up?
TD: I must say that there was a dramatic change from my last contact with Eric in Cream to Eric in Derek and the Dominos. With Cream, it was always three or four Marshalls, wide open, feedback, earth-shattering levels, and so forth. Of that group, Ginger was the softest member. Between Jack and Eric with their Marshalls, I couldn't hear Ginger when I walked into the studio, and Ginger Baker is a loud drummer! He always used to protest and say "They're making too much noise!" 'cause he couldn't even hear himself!

When we go in for "Layla," Eric shows up with a Champ and an old Gibson – one of those straw-colored things midway in size between a Champ and a Princeton. Duane came in with something out of that old school too – God knows what it was, those oldy goldy amplifiers. They played so softly that if you weren't close-miked on the amplifiers, the fret noises would have been too loud, which is to say nothing of the other instruments leaking in. Really, if you opened up one of the studio doors, the rush of air would pin your meter!

I AM YOURS
TD: I forgot all about this one! I didn't see it at the time, but it's a very strong cousin to "I Looked Away."

PL: This has a definite Caribbean or Polynesian flavor to me.
TD: To me, it sounds like Mediterranean. If it sounds like Trinidad or Jamaica *and* it sounds like Mediterranean, then it must be African, because that's where they both feed from.

ANYDAY
PL: Why haven't there been more albums featuring two "super guitarists"?
TD: Well, it's nice to do for people who are guitar buffs, and it's good for the guitar players, but then all of a sudden you're in a very delicate position. You get into that "jazz" category, where you're making records that you know are musically this and that and you're doing something to preserve a tradition that you believe in. You're trying to educate the public. Unfortunately, when you're talking about mass education, you may not be talking about mass tastes.

KEY TO THE HIGHWAY
TD: This was influenced by a record that was being done that they heard in the hallway. I had worked with Sam the Sham – Sam Samudio – and Ronnie Hawkins, and we'd done songs like "Key to the Highway" and other traditional, spiritual-type things. That night, someone was making a tape copy or something in another room, and when we broke, Duane heard it and said "Hey, that's a great old hymn, it goes like . . . " and Eric said "I remember . . . !" Eric is very deep in American blues – he knows it extremely well, better than a lot of American musicians.

PL: He often treads a very fine line between a lead and a rhythm part.
TD: Well, he enjoys playing rhythm more than anything in the world. Most people don't know what a really excellent rhythm player he is. He'd have a delightful time if nobody ever asked him to play a solo or go "Boo." Really, he'd be tickled silly.

PL: I've always said that you can tell this is a live vocal because of that interplay between the voice and guitar. Also, sometimes a word or two will be off-mike, like he's moving around.
TD: I think this *is* live – most or all of the blues numbers were. Eric, at that time, was quite insecure about his singing. He didn't feel he was an adequate vocalist, and he really didn't want to sing.

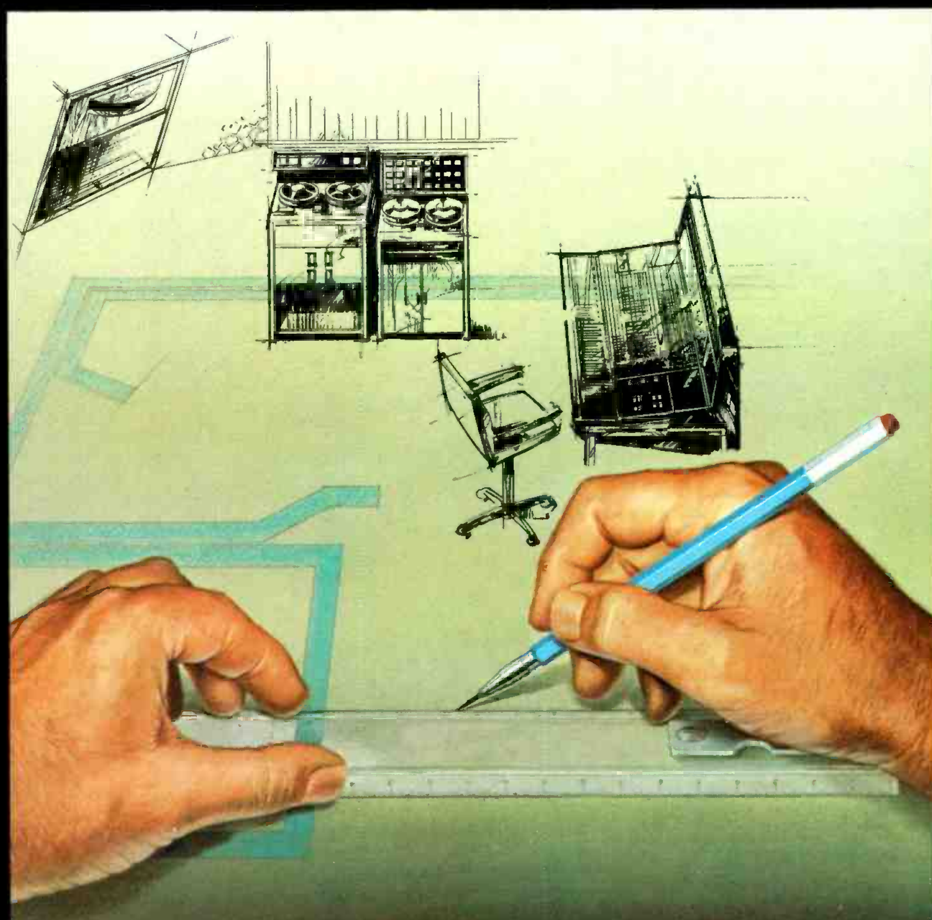
TELL THE TRUTH
TD: Tell the Truth! It's interesting that this version came out as slow as it did, because if you were to hear it now – like how he did it on the "Rainbow" album, or how it's been done by a few other groups – they all play it much faster.

Duane was really an incredibly sensitive musician. You know, he'd be playing a part and all of a sudden think "What I'm doing is not that significant, so I can just as well take it the hell out and not bother anybody with it," and he would make room for the people that are playing to play more. Too often, someone will be playing and they'll figure "The song is this long, so in here I'll just play rhythm." When they think they're contributing, whatever they're doing might be tipping some intricate rhythm pattern that exists between two of the other musicians. Sometimes, when you stop doing your part, the best thing you can do is not do anything – stay away.

WHY DOES LOVE GOT TO BE SO SAD?
TD: I met a disc jockey at Rod Stewart's house last night, and we were just talking about this song. He swears that a recut of this song could be a hit. I never saw the man before in my life! He came up to me and said "You're Tom Dowd – you made the "Layla" album. There's a song in there . . ." and he started talking songs. He actually proposed that Rod record it. We were talking about doing it samba-like or reggae. Rhythmically, you can give this type of song a pattern like "Grapevine" or "Shame Shame Shame" – it's very easily done. You could put it into a push-one playing double time in the rhythm configuration, even though the chord changes are only playing two

Continued on page 35

Acoustical Guarantee



Performance Specifications

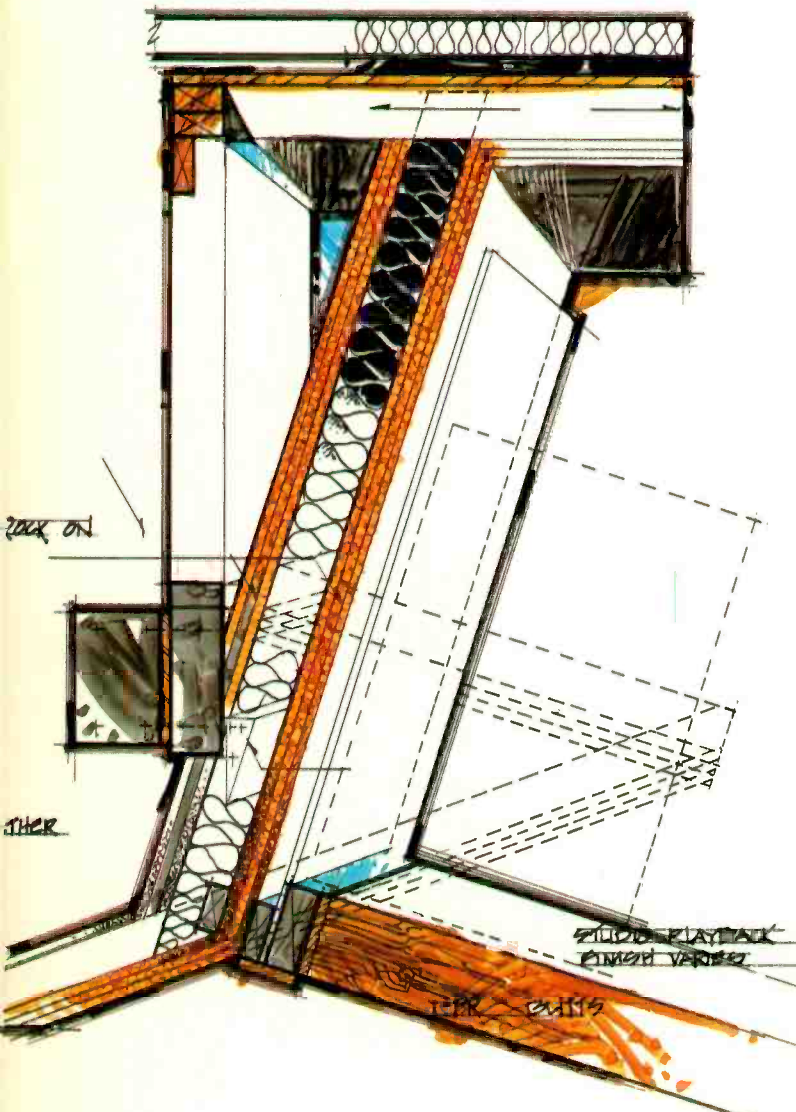
from acoustic design
to down beat...

Westlake Audio

"The control room sounds good here but not over there. Stand up and you've lost your mix. Lean back in your chair and all the bass is gone. The monitor has to be loud to hear it. Turn your head and big changes occur. The stereo image moves."

"The drum leakage in this studio is terrible. The strings sound great but the bass is loose and muddy. This room is so dead the sound isn't happening and the musicians can't get into it."

These are subjective observations which producers and engineers have made and lived with for years in many studios. We at Westlake are prepared to talk to you about a guarantee *against* those things happening in your studio.



Guarantee of Acousti by Westlake Audio*

WE WILL GUARANTEE YOU A CONTROL ROOM WHICH WILL:

Allow you to stand . . . sit . . . lean forward or back . . . move left or right and subjectively not change your mix.

Let you accurately pinpoint any musical instrument within a 360° quad listening environment.

Permit monitoring loud or soft while retaining a tight and musical sound.

Keep your stereo "locked center" on all instruments panned to the middle.

Response:	±3 dB upon speaker installation, 31 Hz-16 KHz measured with B & K ⅓ octave pink noise source. *Between speakers, ±1 dB.*
Dispersion:	±2 dB @ 10 KHz across a minimum 10 foot horizontal plane at the console (from left of the engineer to the right of the producer or vice versa) from any one of the four monitors, measured with pink noise source.
	±2 dB @ 10 KHz across a minimum 10 foot horizontal plane front to back in the mixing area from any one of the four monitors, measured with pink noise source.
	±2 db @ 10 kHz from 6" above console vertically to 6" down from ceiling.
Power:	116 dB SPL minimum, linear scale, with broadband pink noise source from one monitor measured at the mixer's ear. The control room potential with four monitors is a minimum of 128 dB SPL.
Source Location:	Within 2 dB of *total sum* from any two sources in the 360° quad circle environment.

*Available on all new projects from Jan. 1975 on.

cal Performance Specifications

WE WILL GUARANTEE YOU A STUDIO WHICH WILL:

Have a tight rhythm sound under all recording conditions yet allow the producer and engineer the option of changing the midrange character anywhere from "dead" to "very live" in less than sixty seconds. — *Any location in the room.* —

Provide drum cages which are live inside, something that the drummer can get into, allowing you to get a bright drum sound from an open drum cage.

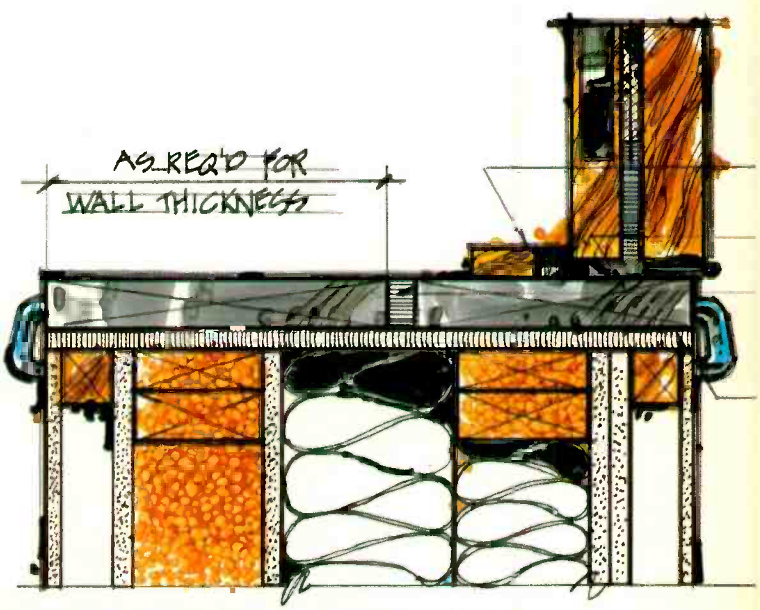
Let you obtain a natural piano sound with excellent isolation from loud electronic instruments. — *With the piano in the room, lid open and not caged in.* —

Provide an echo chamber with low end "mud" removed by trapping in the chamber, resulting in a chamber that "sings."

Room Character:	The characteristic "room sound" which results from recording in a three dimensional area is eliminated by the utilization of an active ceiling. From 40 Hz up, this produces an infinite third dimension such as would be present in an amphitheater.
Separation:	Active traps are built into the studio walls which allows "in-studio" vocals, eliminating the need for the usual vocal booth. 30 dB of isolation can be provided between the band and a vocalist only 10 feet away, resulting in 30 dB of isolation @ 40 Hz or tuned frequencies.
Traps:	Drum cages, bass traps and broad band attenuators will provide in excess of 24 dB isolation @ 40 Hz. The piano can be recorded in the studio while still providing over 20 dB broadband rejection of unwanted sound to the piano mikes *with lid open!*

THE CONTROL ROOM AND THE STUDIO ARE YOUR TOOLS AND SHOULD WORK FOR YOU... NOT AGAINST YOU.

THAT'S WHAT AN ACOUSTICAL GUARANTEE IS ALL ABOUT!



DOOR JAMB DETAIL

Kent R. Duncan, President, Kendun Recorders, Burbank, California: *"The new room has been in operation for six months now and our success is as much a tribute to Westlake Audio and Tom Hidley as it is to our long hours and attention to detail (and possibly some good engineering). Our Westlake room made us a 2 studio operation but instead of just doubling our gross, we went from $12,000 a month to $60,000 a month. The incredibly accurate planning of our Westlake turnkey installation resulted in completion exactly on time, response precisely as promised, all equipment functioning within one day of installation, and all within budget! In the past six months we have mastered such acts as Stevie Wonder, Bob Dylan, America, Buddy Miles, Fleetwood Mac, Rick Nelson, Tower of Power, Livingston Taylor, Isley Bros., Rod McKuen, Nitty Gritty Dirt Band, Emitt Rhodes, Richard Greene, El Chicano, Nana Mouskouri, Cleo Laine, Bola Sete, San Sebastian Strings, Jo Stafford, Maxayn, Pharoah Sanders, Archie Shepp, Ballin' Jack, Vickie Lawrence, Maureen McCormick & Chris Knight, Don McLean, Vikki Carr, Bill Medley and even Rodney Allen Rippy. Over half these acts were recorded on Westlake monitors in various studios around the country, attesting to the fact that truly, you are the professional."*

Christopher Stone, President, Record Plant Recording Studios, Los Angeles: *"As you know, we have used Westlake Audio and yourself since the inception of the company for all of our studio design, construction, electrical interface and implementation. During the past four years you have designed and implemented eight studios for us in New York City, Los Angeles and Sausalito. Obviously we are known as a Westlake-designed operation. We have built our total reputation around your studio design and have always been happy with our decision to utilize you on an exclusive basis for all our acoustical requirements and equipment consultation. The success of your design speaks for itself in the form of our success as an independent studio operation."*

John Sandlin, Vice President A & R, Capricorn Records, Macon, Georgia: *"Words alone cannot express my appreciation for the friendly and courteous atmosphere I enjoyed while at Westlake mixing Bonnie's (Bonnie Bramlett) album.*

It was really a pleasure to work with such extremely competent and dedicated people. Thank you for giving me an opportunity to experience the automated mixing facilities and to work around the type of people I love and can relate to.

Take care of Baker, he's incredible."

John Boylan, John Boylan, Inc., Hollywood, California: *"First of all, this is my third project in a row to be mixed on your monitors and once again it looks like we have a winner — a record that sounds as good at home as it did in the control room. From a producer's nontechnical viewpoint, this ability to trust a studio monitor and come out with even results is extremely satisfying. Secondly, the Westlake Monitor never seems to vary in any substantial way from studio to studio, in the control rooms that you've designed. So I have no worries about consistency in today's widely dispersed recording scene."*

Complete, unedited photocopies of these and many other testimonial letters are available on request from Westlake Audio. Phone or write direct to Tom Hidley, President.

WE PUT OUR MONEY WHERE OUR MOUTH IS!

Below are excerpts from a typical acoustical system acceptance from a client authorizing the release of the final portion of the construction monies from a trust account.

SYSTEM PERFORMANCE ACCEPTANCE

In accordance with the terms set forth in that certain agreement contained within Westlake Audio's invoice number 3930 dated March 1, 1974 mutually accepted by Westlake Audio, Inc. and Sounds Interchange, the undersigned hereby:

1. Acknowledges receipt of and accepts a final sound measurement report from Westlake Audio, Inc.
2. Agrees that Westlake Audio has, as relates to the design and construction of the Sounds Interchange studio facility, Toronto, Canada, it met or exceeded all performance specifications as set forth in the Westlake Audio brochure entitled Acoustical Design The Key To The Success Of Your Studio as amended and signed by T. L. Hidley on February 8, 1974.
3. Acknowledges that all work has been completed in a satisfactory manner and that all materials have been delivered.
4. Acknowledges the fact that Westlake Audio, Inc. has complied with and fulfilled all the terms set forth in a certain Letter of Credit drawn in favor of Westlake Audio, Inc. and hereby instructs the advising bank — Bank of America, Westlake Boulevard, Westlake Village, California, U.S.A. to honor and pay at sight said Letter of Credit on or after December 6, 1974.

SOUNDS INTERCHANGE LTD.

By ____________

Dated Dec 6/74

THAT'S WHAT AN ACCOUSTICAL GUARANTEE IS ALL ABOUT!

Continued from page 30

bars of cadence inside of one measure of change.

One thing interesting about this song is the flow of words. They deliver them in a unique fashion, so that the words are the percussion — "Why-Does-Love-Got-To-Be-So-Sad?" — like a cowbell part or a tamborine part. They're actually using words to give the illusion of percussion.

HAVE YOU EVER LOVED A WOMAN?

PL: When Clapton's playing the solo live, would he junk an otherwise-good take if the solo didn't meet his standards, or does he go for the best overall "feel"?

TD: At the outset, if there was a solo that was shaky on Eric's part, but the track felt good, he'd say "Let's try some more but save that one." Two or three days later, the jury would come in and we'd sit down and try to determine whether or not the track could be saved. "Is the solo really that bad, or should we try to do it again?" This kind of exchange existed among all the people in the band — if Whitlock'd say "I can do a better organ part," Eric might say "I liked my solo but I'll try with you," and they'd go in and do it again.

PL: Did you ordinarily try to baffle them in such a way that you could re-do the solo?

TD: No. As a result, there were some situations where we *did* have leakage to the point where we had to scrap takes that we might have saved otherwise. We didn't go for that "studio-sterile" miking where you could isolate everything because that wasn't the sound we wanted.

PL: Is he really meticulous about his solo as an overdub? Will he play it ten times, for example?

TD: Oh yeah, like in that Aretha Franklin tradition. When he hears that solo back and thinks he can do it better, he'll do it. He knows.

PL: Does he look for a particular kind of development, or will he try a solo a number of different ways?

TD: When Eric is playing traditional music — blues or any kind of "historical" composition — it is the spontaneity of the performance that completely determines if it's used or not. If he believes that his rendition of that song, at that given moment, is what he felt and what he way trying to do, that's the way it stands. He would not then go back and alter the solo to make it something he wants it to be now. When Eric writes a song, he knows what kind of solo he wants, he knows what he wants every corner of that song to sound like. When he's playing an old blues, he might play something right now that he's in love with, but a month from now he'd say "I can't imagine what I was thinking when I did that." Still, he wouldn't go back and re-do it, because that would be incongruous with what was happening at that time.

PL: On this album, did he do any punching-in — re-doing part of a solo?

TD: No, Eric is not inclined to be fragmentized with his solos. He might on section parts, but not on solos. There might be an awkward gliss that he'll want punched *out*, but to punch in two bars or something isn't his nature.

LITTLE WING

PL: How did this one come to be recorded?

TD: Well, Eric always had a great deal of respect for Jimi Hendrix, and they wanted to try and make a record of that song. Obviously, the recordings he'd made were unique and classics, but they figured that they were up to being able to make *one* as good. They spent hours preparing this.

Jimi was a dynamic human being who made some great contributions. I just don't think we ever got to the meat of Jimi Hendrix — we just got all the sparks. We never really got into what he was all about.

PL: You never recorded Jimi Hendrix, did you?

TD: Yeah, as a matter of fact, I did, before he became the Jimi Hendrix of reputation. *That* Jimi Hendrix played guitar in a band for King Curtis along with Cornell Dupree — Jimi was the second guitar player. He also played with the Isley Brothers and Little Richard. I knew him then as a guitar player who had some most unusual ideas and fantastic ability. It was just a matter of when it was going to happen. He was going to do something, that was for sure!

PL: This is the only place on the album that really says "studio effect" — that delay off the lead guitar.

TD: Yes, that should be a 7½ tape delay. It was done in the mix.

PL: How much did Clapton participate in the mixing? Would he move the faders?

TD: No, he'd just sit there and listen, and nod approval or protest.

PL: Does he ever ask for a definite sound? Would he say "I want a delay off my

THANKS TO BEN JORDAN & DAVE TRUSMAN.

guitar" or "Compress the middle eight"?
TD: No, he wouldn't use terms like that, but he'd say "I want it to sound more "hall-like" or "I want it to repeat." Then we'd put on some slap-back and adjust the speed on it to where it fits with the feeling that he has.

IT'S TOO LATE
TD: This was done by Chuck Willis originally. Back about 1956, Chuck Willis was a very popular singer. At that time, the music was called Stroll, and he was "The King of the Stroll." Last year when we did "461 Ocean Boulevard," Eric and Carl asked me if I could get them any Chuck Willis records, and I went back into the Atlantic archives and had tape copies made of everything we ever recorded on Chuck Willis and gave them a set.

I think that's Eric on a Telecaster. You know, he doesn't really play hard – I mean physically. He doesn't attack an instrument – he's delicate. He's got a very light touch and he uses very light gauge strings. One day, we were kidding with someone and the guy picked up Eric's guitar and played a few chords, and all of a sudden, two strings came undone. He looked at him and said "My God, why do you use such light strings?" and Eric said "They're not light to me."

LAYLA
PL: You rolled off a lot of bottom on these guitars, didn't you?
TD: Yeah, you had to. Because of the number of guitars, the complexity of what they're playing, and with the exception of one, that they're all voiced the same. You get a confusion, especially with echo chambers, down in the low strings, so it might be that you'll want to roll off the lows in that portion that is going to the echo chamber, so that the chamber doesn't overconfuse.

PL: Was this tune originally intended to be the album's central track?
TD: Yes. It was based on a personal experience Eric was having around that time. This and "Bell Bottom Blues" had some significant meaning for him.

PL: Was the ending written separately?
TD. Yes, it was. From the time we started recording "Layla" until the time we got that whole first part done, Jim Gordon and Bobby Whitlock had been talking about a part that could possibly be added onto the end of it. It was to be a concerto-type theme. Jim wrote the part, but we could never put it in with the guitars and the organ. It just never fit the track, and we abandoned it. Finally, when the first part was all done and we were listening to it, Jim said "Let me go in and play the piano part on it right now," so we played the tape and set the mood up for him, and when the track came to an end, we punched in on an adjacent machine and Jim continued right in the tradition that he'd been saying it should be done all along. Then we backed up the tape and added the other parts. We could never do it on the fly – it just never materialized, no matter how many times we tried it, until the song was done.

PL: I'd always thought that last little sound was a whistle with a tape delay, but a friend told me it's Duane Allman doing something strange on his guitar.
TD: He's right. Eric showed Duane how to get the harmonics way up at the top of the neck. Duane did it, but not with a steel bottleneck. He was using one of those little coffee creamer jars. Remember you used to get cream in a jar in restaurants, and it would be in a small bottle? We used to go into coffee shops and truck stops and snitch a couple.

THORN TREE IN THE GARDEN
TD: This is interesting. It was done directly to quarter-inch tape. Bobby, Eric, and Duane are playing guitars and Carl is playing bass – all seated around two microphones. It was a real *stereo* recording. The microphones were omnidirectional, and if I recall, were about nine inches apart. I just had them sit around them, did a little adjustment, and it was made in like two or three passes. Live vocal, everything. That was the ideal opportunity to make, in the true sense of the word, a *stereo* recording. If you were to listen to this on earphones, it would be grossly different from things where you'd have five or six mikes going, cross-fading,and mixing and all.

PL: What was your first contact with Cream?
TD: Very simple. At the time I was doing a lot of work for Stax, Ahmet was going to Europe and got interested in a lot of English acts. One of the acts that he was deeply interested in – and you could understand why, he being into jazz and so forth – was Cream. To him they were representative of a new breed, a new form of jazz. It was akin to blues but not blues, akin to jazz but not jazz, and he was tremendously impressed by them. Of course, he was also aware of the groups that each one of them had come from, and here was an opportunity to get all of them into one group. He made a deal with RSO – Robert Stigwood – for the rights to the group in the United States.

Anyway, they were over here on tour as part of a package, and Ahmet called me up one day and said "There's this fine English group that's currently on tour, and they're going to finish the tour in the next few days and we should make an album with them before they go back." It was like that – you had to grab them before their work permits expired and they had to leave the country! For the first album, the equipment arrived on a Thursday, they walked in on a Friday, played the numbers they had been playing on tour, and on Sunday the limosines came right to the studio and took them to the airport. That was it. That was how I met them.

PL: Let's talk about "Disraeli Gears." I've always called it "the album that introduced 'lead guitar' to the American audience."

TD: Yeah, I know what you mean. It was threatening to happen for a long time – it was just a matter of being at the right place at the right time. You're always close, but some times you get luckier than other times, that's all. It was not an "endeavor" – it was just that that was what we had to go with, we believed in it, and we did it.

PL: I would imagine that you are significantly responsible for Jack Bruce's recorded bass sound. What did you do to get it?

TD: Well, there was a big problem that I had recording them, especially on the first album. All of their equipment was 50 cycles/220 volts, which is the European standard. When they had done their concert tours, they had transformers stepping up and stepping down, so that they never had to alter their Marshalls. They had original English Marshalls and that's the way they travelled. When they got to my studio, we had none of this step-up and step-down equipment. This was at Atlantic Records in New York – 11 West 60th Street. Being that it was a weekend, we had limited access to transformers and things that could correct some of the problems and so we had some very unusual power factors working. They were playing under hardship and I was recording under hardship because none of the equipment was really running the way it was designed to run. That's the truth! Their amplifiers were tube, and running in this unorthodox fashion, they were heating up continually. They were physically heating up and we'd have to stop every once in a while to let them cool down. It was not just a matter of air circulation, either – we were absolutely *abusing* some of the electronics, and as a result, it altered the sounds. It altered the sounds considerably.

PL: So was the bass taken only as an amplified channel?

TD: No. I had mike and direct running simultaneously on Eric and Jack in the studio.

PL: Were the miked and direct channels combined at the time of recording, or did you take each one down on a separate track?

TD: I took them down on two tracks. Sometimes, though, when we had to back up for a lot of overdubs, the luxury of two bass tracks was gone and we had to start combining them.

PL: When you were recording "Sunshine of Your Love," did you suspect that it would be such a classic – the first real "hard rock" single?

TD: Yeah, I believed in the song intensely. If I remember, Ginger was playing almost Indian-type drums, emphasizing the backbeat. Everything else had a complete *downbeat* influence. I know that it took about an hour for the dust to settle after I suggested we get downbeat-happy instead of backbeat-happy. They played it for the tape once or twice, and when I played it back to them, they realized that it accented just what they wanted and then away we went.

PL: I read somewhere where Clapton said that on "Sunshine" and "Strange Brew" there were no effects on his guitar – just

his Les Paul through a Marshall with everything at 10. Is this true?
TD: Yep. You couldn't stay in the room when he was playing. It was a gold-and-brown Les Paul. I remember the guitar.

PL: Were you in on the editing to the single version of "Sunshine of Your Love"?
TD: Oh yeah.

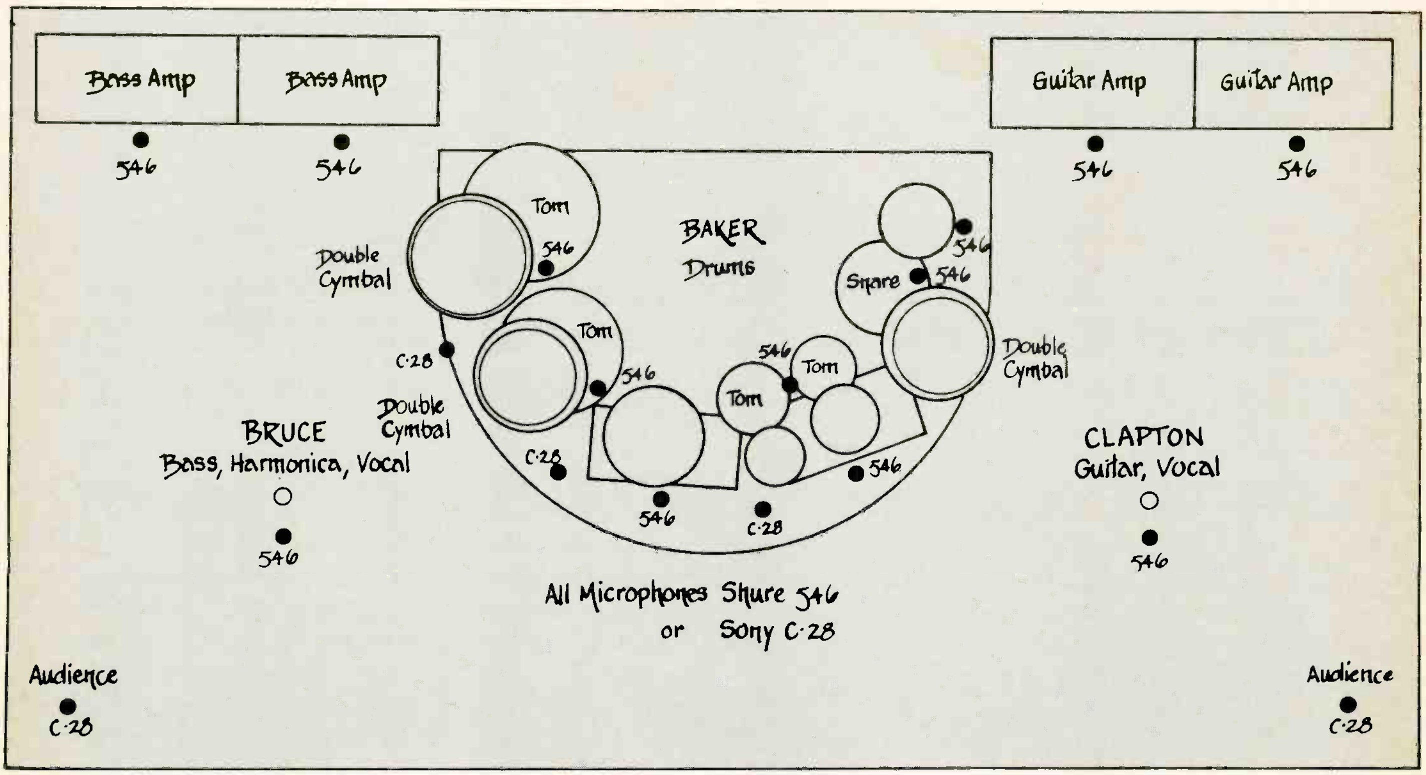
PL: You slowed it down a percent or so, didn't you? It's noticeably slower.
TD: I could imagine that it might be slower, but it was not intentional. I did edit it, yes. You know, I enjoy making records the way the artist likes to do it, but when I put on that "record company" hat and when you want to get it on the air . . .

PL: So you took out that riff one out of every two times.
TD: Exactly. I become a monster, I become a policeman! I'll sit down and notate the arrangement and say "This goes, that goes, you said that lyric once already, forget it, zap, zap, zap."

PL: How about the "crowd noises" on "Take It Back"? Was that an overdub, or were you actually miking the control room while the basic track was being recorded?
TD: Well, initially when we recorded Cream, it was just the three of them and their roadies and myself. Then came Felix, and by the time we got into the "Wheels of Fire" album, when Cream showed up it would be with Janis Joplin and everyone else! Every once in a while, you'd be aware that those people had an influence — they were really contributing to what was going on. For example, when you'd play something back, they'd jump up and say "Hey, that's great!" or "Why don't you try this?!" — you know, you get that enthusiasm. For "Take It Back," we thought "Maybe we could use that."

PL: How did you actually record them?
TD: We sent them into the studio and put the track on loudspeakers. I said "Hey, you like it so much, let me play it back to you and you can make some noise." I guess there were about 15 or 20 people out there.

PL: How about "Mother's Lament"? How did you come to record that?
TD: With Cream, there were times when the tension used to be very high, when they were very serious about what they were doing. Ginger might be saying "You're playing so well and the part I'm playing is terrible, and I'd like to do it again except that I wouldn't want to do it again if you didn't think that you could play as well . . . " — that sort of thing. Now the day *this* happened was just the opposite — we were all very jovial. We were sitting around listening to some quarter-inch roughs we had done, when somehow or other we got talking about English music halls. We were talking about some of the things you used to see



Stage set-up for live recordings at the Fillmore West and Winterland of Cream during their Summer 1968 U.S. tour (excerpts from which appear on disc in "Wheels of Fire" and "Live Cream Volume II").

Said Bill Halverson: "This was the second remote recording I had ever done. It was very unsophisticated back then — "The Stone Age of Live Recording" — and no one had really done any remotes, at least not of rock & roll. At that time, I didn't like rock & roll, but when I heard these guys stretch out like some new sort of jazz group, I was very impressed.

The original Fillmore was a nice hall to record in. Even though it wasn't that big, it had a very live quality. That and the short delay time made it like recording in a huge living room. At Winterland, we used pretty much the same set-up.

For the Cream performances, we used a little 12-position rotary console that had a left, a middle, and a right. It had a ±3 and 6 at 100 and 7500 — that was all the EQ that was on there. I only had room for the vocals and drums through the board. Everything else I ran through Ampex mixers, padded way, way down.

For the miking, I used what were then Shure 546s on almost everything, which is absurd. We used them because that's what Heider's happened to have at the time — that and three Electro-Voices, which I couldn't stand. I also used two Sony C-28s — an old tube model — for the audience. They were positioned right at the edges of the stage, usually pointing at the center of the back of the hall. We used three more C-28s as overheads for Ginger.

This was the first time that I had ever recorded Marshalls. The key to recording them is that you put the microphone right in the middle of the four speakers — anywhere else and it'll blow the poor microphone's brains out. You keep it in really tight — about three inches away. This way, it doesn't have any sound coming directly at it and it's surrounded by a "wall" of sound which blocks out all the leakage from the drums. That was really important because there was often only a foot between the amps and the drums."

on the English stage – the comedians and the pub scenes and so forth – and Ginger just sat back and recited this little ditty. I completely busted up – I said "We have to put that in the album." I thought it would be great with all those other heavy things going by. They said "Great idea," ran out into the studio, and made a pass at it, and boom. That was "Mother's Lament."

PL: I wanted to ask you about the live Cream dates you recorded.

TD: Well, the first live dates I did with them were for the "Wheels of Fire" album. They wanted to make it a two-pocket album, with part of it live and part of it studio. We wanted to be able to get enough things in the live recordings that weren't on any albums and still have one or two live versions of songs that *had* appeared before. If I recall, we did three nights of recording for "Wheels of Fire" – two nights at the Fillmore and one at Winterland, or maybe it was the other way around.

PL: Were those gigs supposed to be the "high points" of the tour?

TD: They might have been, I don't know. I think that really it was another case where they had to leave the country soon.

PL: What track tape did you use?

TD: 8-track.

PL: How were the tracks allocated?

TD: Well, for the live stuff we could really cheat. We had two tracks for the audience, a vocal for Jack and a vocal for Eric, and then a bass track and a guitar track. What we would sometimes do is drop the guitars onto the audience mikes, so we got them from the amps *and* distant. We get into cutie little stunts like that. The drums were recorded on two tracks. It might have been three – we might have folded the two vocals together one night and not on the next, I don't know. For the most part, though, I think Ginger was recorded on two tracks.

PL: What procedure would you go through in recording a live performance? Suppose I call you up and say "Tom I want you to come down to the Fillmore and record a Cream gig." What do you do from then on?

TD: First I go about finding out what the best remote truck is. Then you find out about the crew. Your crew is really important – they're a living, breathing part of the whole operation. They shouldn't be out there running microphones all over the place, but should be operating with a definite plan. For Cream, I used Wally Heider Recording and Bill Halverson, both of whom I used to use quite a lot when I was recording in California.

Knowing what Cream was like in the studio, I could imagine how much more bizarre they might get on stage, and so I figured I'd better give Bill some preparation. I gave him diagrams and sent him notes, telling him how I got the way it sounds on disc. I'd note to him the microphones I used and the distances – really, ". . . this kind of mike that far away . . ." and so on. By that time, I had evolved a classic sound pattern – whatever that means – for Cream. It was just the way I saw them, having Ginger spread with multi-miking, and then Jack and Eric going mad on the sides. We got a very heavy, ominous-sounding record going. I wasn't saying to Bill that he should use these same mikes, I was just telling him what I'd done to get that sound. The intent was to get the group like they sound on disc and live at the same time.

Bill did a great deal of homework for these dates – he probably spent a week or two taking the things apart. He's a very conscientious chap. I sent him some 8-track out-takes so that he could listen to the individual tracks, then I sent him some rough mixes so he could see it at that stage, then the records, and so on. We spoke on the phone – everything! By the time I finally walked into the Fillmore for the first recording, it was all set up. You know, there might have been some miking that was different from the way *I'd* done it, but when the group was playing and I went back into the truck, Bill had it covered. It was clean, it was virile.

The next major thing I do for a live recording is check out the hall. One of the biggest problems is always the acoustics of the place where you'll be recording. You get into what are now theaters – motion picture houses – that are carpeted and with velour upholstery on the chairs and so forth, and these rooms are heavy and dead to play in even when they're empty. When you take a sound check in a room like that, all you can do is make sure that the microphones are working. You can't really evaluate the sound until you get all the people in there.

PL: So you have a limited amount of time in which to get the miking straightened out.

TD: Right, sometimes you have to make a very quick guess. If it's an upholstered room, it's going to call for a more spatial-type recording. This is as opposed to a live, reverberant room where you'll want very tight, close-up recording, because the ambience in the room is going to be coming down all the up-front mikes anyway. With a dead room, you can back off and let everything breathe more, because there's so little ambience. If you didn't, it would sound like it was done in a studio instead of a concert hall.

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SPECTRUM ANALYSIS APPLIED TO AUDIO SYSTEMS DIAGNOSTICS

By Peter Butt

The complexity of modern audio mixing systems need not be dwelt upon at length. Casual inspection of recording console descriptive literature will serve to describe systems having thirty-two inputs and as many as twenty-four program outputs, not counting echoes, cue circuits, and monitors. Program equalizers in common use feature as many as fifteen discrete peaking and notching characteristics, and ten or more shelving curves, all of these in two or three decibel increments. Depending on the specific characteristics and features of a given program equalizer, 500 or more specific measurements would be required to verify the equalization frequencies and incremental steps of each particular frequency dependent function.

Even more complicated are the parametric equalizers introduced within the past couple of years. These devices permit independent variation of their characteristic center frequency, filter shape factor, and relative boost or cut, independently of one another.

Those are just the equalizers. What about the response of high-pass input filters, microphone preamplifiers, summing amplifiers, and line drivers? Extending this progression still further, what about distortion characteristics, residual noise, gain variations and input and output drive levels? These are all tests that serve to characterize significant aspects of audio component performance that the entire system quality rests upon.

How many of the above tests are periodically performed as a means toward the detection of system performance degradation and verification of original specifications? Some of the above? Any of the above? I would hazard a suspicion that "None of the above," would outnumber the other two by an unsurprising ratio.

The focus of this little rhetorical sermon is that it simply takes too much time to perform a complete set of electrical measurements on any sort of system module for the task to be done in any but the most dire of circumstances.

Costs of down time and quality test equipment being what they are, many studios have simply eliminated adequate measurement facilities because the majority of clients aren't as interested in voltmeters and distortion analyzers as they are in direct production facilities.

At issue here are not the catastrophic failures of the fire-and-smoke or "it-don't-work-no-more" variety. Those kinds of failures rarely fail to draw swift and sure response from the maintenance staff.

Time does get to most of us, one way or another, and recording consoles are no exception to that rule. Over a period of months or years switches corrode, resistors and capacitors drift in value, solder joints crystallize, semiconductor junctions change. The slow rate of these effects make them difficult to track subjectively. The crushing burden of making and recording test results by traditional means just about eliminates them as a realistic method of detecting console performance degradation.

Since most of us know all that already, there need be only minimal further delay in delivery of the gospel to the millions. There is a way of rapidly performing and recording all of the tests mentioned. The speed with which they can be performed is such that the condition of all active elements comprising a typical twenty-four by sixteen console could be determined in a period of eight hours or less.

There are a couple of types of analysis systems that have been available for some time and are in prevalent use for audio system analysis. They are the fixed-tuned parallel filter arrangement, most widely known as a "real-time" analyzer, and the older technique involving a sweep generator driven in synchronism with a chart recorder or an oscilloscope display.

Although widely known, a brief discussion of both of these approaches toward reducing the time element of audio system analysis may be worth while. There are many applications where limited capability is entirely adequate. Directing our attention to their limitations will, I think, serve as a reference for comparison with later developments.

Figure 1 illustrates an elementary approach to dynamic display of a frequency response characteristic. I would expect that just about every TV service shop in the country uses this technique to verify television receiver RF and IF amplifier bandpass characteristics. Most of us are familiar with the use of the late General Radio Company 1350 test system consisting of a GR 1304 sweep generator synchronized to a GR 1521 chart recorder that yields frequency response curves in a fairly reliable manner.

Continued ——

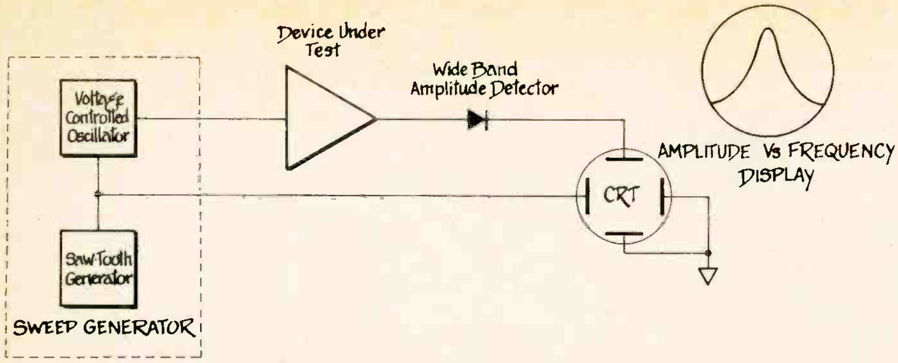

Figure 1 ELEMENTARY SWEPT-FREQUENCY ANALYSIS SYSTEM

General Radio has discontinued the 1304 sweeper, and thus the 1350 system. They've replaced it with the 1523 chart recorder which, when used with the 1523-P2 sweep generator module, will indicate true rms response from 1 Hz to 500 kHz. All this at a higher price, of course. Anyone who does not already own the less sophisticated 1350 system will have to settle for its new, improved successor, or rummage around the used test equipment market.

The Amber 4550 spectrum display system uses a columnar display composed of light emitting diodes. It covers the same approximate spectrum range, but does it in ten full-octave bands. The LED display has a resolution of one or two decibel increments. The predominant application of the Amber 4550 seems to be as an indicator of music spectral content during mixing and mastering operations.

The White Instrument Model 140 Analyzer displays 1/3 octave data also using a light emitting diode matrix. The total spectrum SPL is displayed on a separate column. The display range is either ±20 dB or ± 10 dB of center reference. The White 140 contains its own pink noise source as does the DuKane analyzer.

Bruel & Kjaer still offers their 1022 beat frequency oscillator in combination with their 2305 strip-chart recorder for these kinds of measurements. It should be mentioned that none of these instruments constitute the last word in portability and those vulnerable to hernia should regard them with this disadvantage in mind.

The persistent factor to be recognized in the application of this type of response recording system is the time required to generate a trace of useful quality for the purposes at hand. Although the strip-chart method accumulates data in a more detailed manner and in less time than would a tabulation of output level versus frequency, compiled manually, the enormity of the mixing consoles in common use make anything resembling thorough system performance verification just about impossible in any reasonable period of time.

Practical experience involving the GR 1350 system and a set of four Olive 2100 equalizers indicates that verification of only the maximum peaking, notching and shelving characteristics took between 45 minutes and one hour for each of the four units. Truly monastic dedication would be required to complete only these rough checks for a set of sixteen or more similar equalizers. It should be emphasized that these were tests of frequency response only. No consideration was taken of the time necessary to verify insertion gain or loss, distortion, residual noise, or maximum output level.

A significant contribution toward the reduction of frequency response analysis time was the introduction of devices utilizing parallel, fixed-tuned filters whose center frequencies and bandwidths were determined to cover the spectrum of interest geometrically. This, of course, is the most widely known and used spectral analysis tool covering the audio range. Examples of this type of analyzer are the Hewlett-Packard 8050, marketed through Altec-Lansing, the General Radio 1921 analysis system, and the Bruel & Kjaer 3347 real-time 1/3 octave analyzer.

Although the HP 8050 prices out around three to four thousand dollars, the GR and B&K instruments are truly the "big guns" of real-time instruments, featuring a variety of data outputs and displays as well as computer interface potential. For applications involving sound and vibration analysis, acoustic system performance, and impulse or transient analysis, these heavy instrumentation packages are pretty much a necessity. Basic systems range from about twelve thousand dollars, upwards, depending on configuration.

DuKane, Amber, and Communications Company have introduced real time analysis devices recently. The DuKane and Communications Company devices are used with an X-Y oscilloscope and cover the range of approximately 31 Hz to 16 kHz in 1/3 octave bands.

Figure 2 shows a generalized signal path of the real-time type of device. The term "real-time" is descriptive of the fact that all portions of the spectrum of interest are continuously monitored at all times. Since the read-out device most generally used is a cathode ray tube, the entire series of filter detectors can easily be scanned at a frequency comparable to the highest frequency in the spectrum of interest. For all practical purposes, this amounts to simultaneous display of each detector output.

Common application of the real-time analyzer involves use of a random noise generator of one kind or another. The noise is filtered through a 3 dB/octave, low-pass filter and then fed to the system to be tested. The output of the system under test is then fed to the real-time analyzer and its display interpreted. Any change in the system characteristics is indicated by a change of height of the histogram display pattern. The kinds of systems being analyzed using this method are generally acoustic. As far as the technique impacts the professional audio industry, it is often used to equalize and verify response characteristics of control room monitoring environments.

There are a couple of disadvantages to the multiple filter approach to system analysis. One is that the limited resolution of the fixed filters tends to hide response

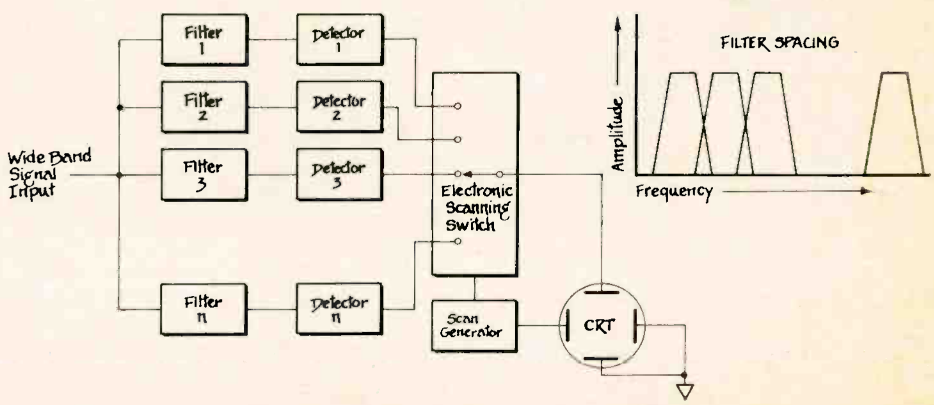

Figure 2 REAL-TIME MULTICHANNEL SPECTRUM ANALYZER

Circle No. 127

nulls that occur within a filter band and the displayed output tends to favor the maximum level observed. For example, a continuous-wave tone would appear as if it occupied an entire filter band instead of its theoretical 0 Hz bandwidth. In the case of a parametric or other device having a sharp notch in its frequency response, the dip would not be precisely shown with respect to its depth, slope, or center-frequency characteristics.

Another limitation is due to the reliance of the real-time analyzer on a random signal source. Since the signal to be measured is statistical in nature, the measurement will be statistical also. The observed statistical fluctuations in any given spectrum band will be greater for narrower bandwidths and will require a longer sample averaging time to arrive at an average value within a given accuracy limit. For a very low error limit, the sampling time could be fairly long.[1] For a 1/3 octave filter centered at 31.25 Hz, an averaging time of about 8.6 seconds would be required for a 99% probability that fluctuations in the observed reading would not exceed ±1 dB of the true average. For a 1/3 octave filter centered at 12.5 kHz, the averaging time need only be about 20 milliseconds for the same accuracy limits. The effective bandwidths for the two cases are about 7.24Hz and 2895 Hz, respectively.

The real-time analysis approach seems to be most effective in verification of flat response rather than system responses that deviate from flat in some very precise way. An example of such a non-flat system would be the RIAA record or reproduce characteristic.[2] In the absence of a known complimentary network to yield net flat response, exact determination of turnover frequencies and response slopes may be somewhat difficult. Resort to discrete frequency plots may be required for determination of precise turnover frequencies and relative boost.

For all the benefits of the real-time analysis technique, we are again reduced to the oscillator and voltmeter approach when very precise response determinations are to be made. For all our lengthy discussion to this point, we have not seriously impacted the time factor in audio systems measurements. Take the case of the dauntless reader who has clung to every word in this discourse; he now has eyestrain and boredom to add to the engineer's traditional occupational hazards of insomnia and chronic hearing loss.

Those with their fingers on the fiscal pulse of our industry cannot help but be impressed with price statistics of many of the instruments previously mentioned. "If progress is our most important product," the critical reader may ask, "why can't we sell it like General Electric?"

A question cogently posed, to be sure. I am here to proclaim that our salvation lies within the ever-advancing fronteirs of human knowledge and understanding.

Over the last three or four years a number of manufacturers have introduced rather sophisticated devices whose potential impact on the professional audio industry has yet to be widely considered. I am speaking of the low-frequency spectrum analysis equipment now available from at least two manufacturers in a variety of configurations, capabilities and prices.

The fundamentals underlying the application of these various devices were explored and perfected for use in the radio frequency region of the spectrum many years ago. Radar and microwave system rf amplifiers, mixers and intermediate frequency amplifiers have been evaluated and aligned using swept-frequency techniques for almost as long as the higher frequency portion of the spectrum has been utilized.

With improvements in filter design, circuit stability, oscilloscope storage tube performance and integrated circuits, it has become possible to generate a staggering amount of system performance data in a relatively short measurement time.

The general means toward this end is shown schematically in Figure 3. The incoming signal enters the first mixer through the low-pass input filter at the left of the drawing. The other input to this heterodyne mixer is the output of a voltage tuned oscillator whose frequency varies upward from the center frequency of the intermediate amplifier. The range of this upward frequency sweep is adjusted to correspond to the portion of the input signal spectrum to be inspected. When the frequency difference between the input signal frequency and the voltage-tuned oscillator frequency is equal to the IF amplifier system center frequency, the signal is detected and a vertical deflection appears on the CRT screen. The system functions as if it were a wide-range, tunable band-pass filter, controlled by the sweep generator output voltage.

The advantages of this approach to display of spectral content are not to be dismissed lightly.[3] The sensitivity of the system is very high due to the gain of the IF amplifier. One of the important aspects of spectrum analysis is the ability to discriminate between spectral components narrowly separated in frequency. This ability, the resolution of the system, is determined by the IF band-pass filter. Changing this bandwidth changes the analyzer resolution. The narrower the IF bandwidth, of course, the longer is the time required for the filter to reach a stable signal output condition after application of a stimulus. The resolution of the analyzer, then, places a restriction upon the speed with which an input can be scanned without sacrifice of indication accuracy due to filter response time.

Because of this response time limitation, the rate at which a given frequency range can be swept is limited and the entire spectrum of interest cannot be observed simultaneously. The application of the heterodyne analysis technique is therefore limited in regard to time-varying phenomena. For the purposes at hand, this is not a serious limitation.

The length of analysis time required for treatment of the entire frequency range of interest can be compensated for with relative ease by one of two methods. The first is the use of a storage CRT as a display mechanism. In this way a non-recurrent sweep as slow as may be necessary can be recorded and observed indefinitely without deterioration. This is the approach taken by Tektronix for their 5L4N spectrum analyzer.

The second approach involves digitization of the vertical signal and recurrent circulation of the digital data series in a shift register. As each digital word is recirculated, it is converted to its analog value and displayed on the face of a normal persistence CRT. Hewlett-Packard applies this method to their 3580A spectrum analyzer.

Application of the superheterodyne analyzer to frequency response measurement requires addition of some extra hardware to provide a sinusoidal signal that will follow the center frequency of the IF bandpass filter as it is tuned across the spectrum. This facility is shown within the dotted line at the lower left side of the block diagram in Figure 3.

The output of the voltage-tuned oscillator is mixed with a fixed-frequency oscillator whose frequency is adjusted to match the center of the IF amplifier bandpass very closely, typically within one or two hertz. The output of the mixer is fed through a low-pass filter and the result is a signal whose frequency corresponds closely to the center frequency of the IF band-pass at that instant of time. This circuit is called a Tracking Generator or Beat Frequency Oscillator.

The tracking generator output is applied to the input of the circuit to be tested while the output of that circuit is fed into the analyzer input port. The result is a single-line trace in the scope face giving a graphic representation of the network amplitude versus frequency response. Damned clever, those occidentals.

The proof of this theoretical pudding you will doubtless agree, dear reader, lies in the empirical tasting. To that end, tests were performed on audio consoles of recent design in daily use in the Los Angeles area using Hewlett-Packard and Tektronix equipment supplied by those manufacturers. The results should prove to be interesting.

The trial of the HP 3580A was performed using a custom console employing a popular line of brand "x" components. The HP 197 oscilloscope camera was used to record the CRT traces. In all cases, the logarithmic sweep function was used for

Circle No. 128

QUALITY: **SPECTRA SONICS** audio control consoles show the care and attention to detail that are the mark of the skilled American craftsman. The internal wiring, module construction, console housing and the control display reflect the precision and distinctive craftsmanship that is characteristic of SPECTRA SONICS.

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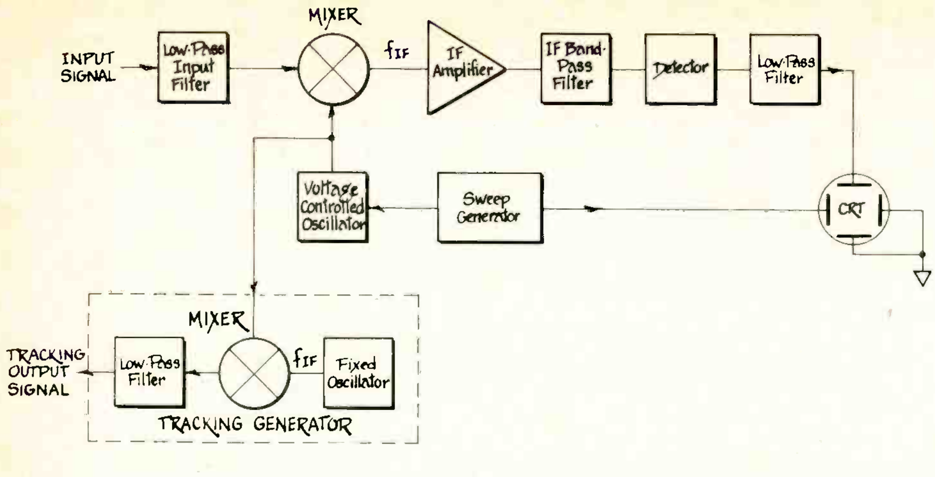

Figure 3 SUPERHETERODYNE SWEPT-FREQUENCY SPECTRUM ANALYZER

generation of the response curves while linear sweeps were used for distortion measurements. All of the photographs made with the HP 3580A show no CRT graticule. This makes them a little difficult to interpret, however some reference lines were generated by varying the tracking generator output attenuation by ± 10 dB and ± 12 dB from the reference level with the equalizer in by-pass. The HP 197 has an option that permits photography of the CRT graticule, however a camera with that option could not be obtained within a time frame compatible with test and publication schedules.

With that apology, then, let us turn our attention to Figure 4. Shown are the 12 dB shelving characteristics of two different equalizers of the same model. The upper set of shelves are almost perfectly symmetrical about the reference line. Its companion does not quite match it in this respect. Note the 50 Hz cut curve as compared with the 50 Hz boost. Note also, the low frequency set of curves, shown as maxima and minima, exceed the ± 12 dB nominal values by almost 2 dB as judged by the ± 12 dB reference lines, in both cases. The high frequency shelves exceed nominal maximum boost and cut by about 1 dB.

The 50 Hz – 15 kHz "filter" function is also shown with the shelving family. It appears to be identical for both equalizers. Note the upward curve of the center reference line. It rises about 1/2 dB at about 15 kHz to about 2 dB at about 40 kHz. This appears to be a common characteristic of the brand "x" equalizers tested. The HP 3580A tracking generator output checked out flat across the band within .2 dB of reference at 5 kHz. All amplifiers in the console were tested with a 604 Ohm load termination. The cause of this rise was not determined nor was it established that this rise was at all troublesome.

Figure 5 shows the same two equalizers in the same order as Figure 4. Comparison of the maximum and minimum extremes shows that all agree closely with the ± 12 dB nominal.

On the upper set of curves, locate the fifth notch from the left. This is the 400 Hz curve that is duplicated in the LF and MF sections of the equalizer. This curve appears slightly heavier than the others because the LF and MF 400 Hz center frequencies are not exactly identical. They do overlay each other very closely in the upper family. Comparing this in the lower set of curves, we see that the LF and MF 400 Hz notches do not match as precisely as in the upper set.

These are variations noted in two out of thirty-two equalizers. It would have been interesting to have checked more units, had time permitted, to see how great the range of deviations from nominal performance might be. After some practice, it was possible to generate a full set of extreme shelving, peaking and notching curves in about 11 minutes, taking time to apply the Polaroid preservative solution to each of the two photos and to make identifying notes regarding each. Both shelving and notching could have been recorded on a single photo had the HP 197 camera been adjusted to sufficiently offset the two tracks. Doing

Figures 4 through 9 were generated on the HP 3580A and depict the performance of Brand 'X' console modules. While the CRT graticule markings are not seen in these photos, an accessory is available which illuminates the CRT background, making the graticule visible.

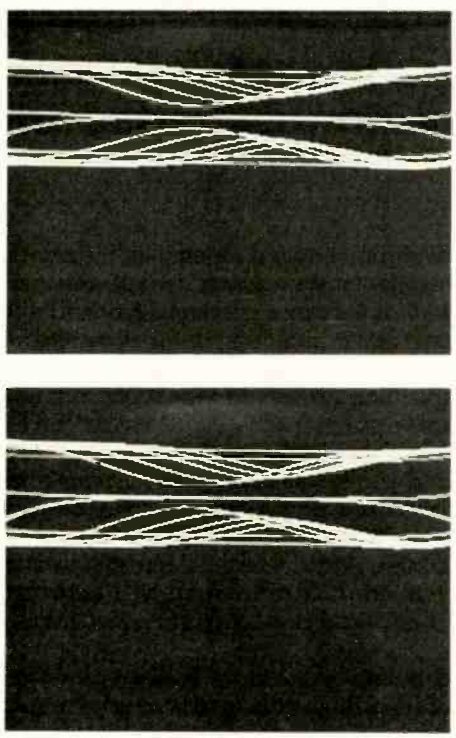

Figure 4: Shelving equalizer family of curves for two separate equalizers. 10 dB/vertical cm with horizontal reference lines at ±10 dB and ±12 dB, log frequency sweep, 20 Hz to 43 kHz.

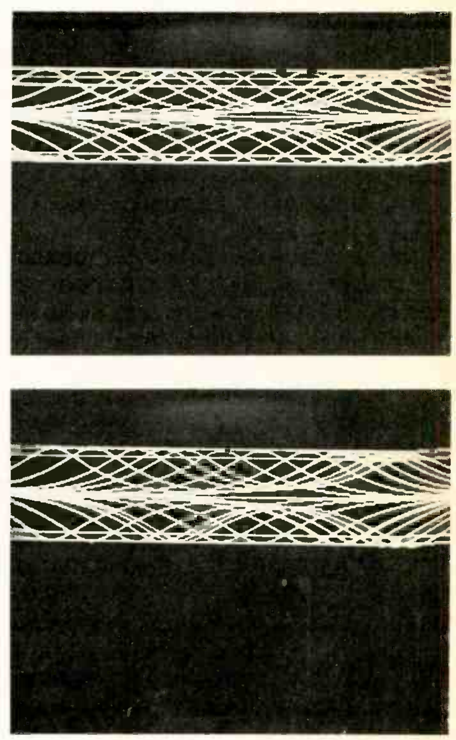

Figure 5: Peaking and notching curves for two separate equalizers. 10 dB/vertical cm with horizontal reference lines at ±10 and ±12 dB, log sweep, 20 Hz to 43 kHz.

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The entire set of thirty-two equalizers could have been checked in about six hours, excluding interruptions of biological or other necessity, and the test results inspected at leisure.

Figure 6 shows the distortion spectrum of a 500 Hz signal fed to the input channel mike input with the equalizer in the signal path and its frequency controls set for zero. The channel output level is +2 dBV, the second and third harmonics are -68 and -61 dBV, respectively. These figures calculate out to about 0.09% total harmonic distortion.

Cross-talk measurements have always been a hassle using conventional methods. Even with a real, live wave analyzer, getting a good idea of how much unwanted signal is leaking into where at what frequency takes the patience of a monk. Plotting the curves from manually recorded data is always time consuming. Figure 7 shows an adjacent channel cross-talk sweep from 0 Hz to 50 kHz. One input channel was driven with a -50 dBV sweep from the 3580A tracking generator while an adjacent channel output was observed on the display screen.

Although the numbers aren't apparent due to our lack of graticule, the leakage lies below -70 dBV up to about 10 kHz, rising to about -50 dBv at around 45 kHz. The driven channel gain was adjusted to yield +4 dBm at 1 kHz. The lower trace shows the program channel noise, with the equalizer in the circuit set for flat, over the 50 kHz band. The estimated average, unweighted noise over the bottom 20 kHz band is approximately -77 dBV. A 2 kHz per division horizontal scale factor would have yielded somewhat more accurate data. The cross-talk curve took about 20 seconds to trace while the noise curve took about 100 seconds.

The HP 3580A has three vertical scale deflection factors: linear, 10 dB/division, and 1 dB/division. The 1 dB/division scale isn't terribly useful. A 2 dB/division factor would be a lot easier to use.

Figure 8 shows the incremental peaking response of a brand "x" equalizer at 100 Hz and 15 kHz. The +2, +4, +6, and +9 increments show up nicely and exhibit remarkable precision, but the +12dB setting is off the screen. Marker traces are shown at reference level and at 2 dB increments. The traces of Figure 9 show the HP 3580A's response to decibel level changes generated by a precision attenuator in 1 dB and 10 dB increments. The effective display dynamic range is about 80 dB in the 10 dB/division mode.

The Tektronix 5L4N was used in conjunction with a popular production model brand "Y" console. The 5L4N is an oscilloscope plug-in module that works with one of the Tektronix 5000 series oscilloscope mainframes. The 5103N/D11 storage oscilloscope mainframe is recommended for use with the 5L4N and that was the equipment configuration used to generate the following photographs. An unantici-

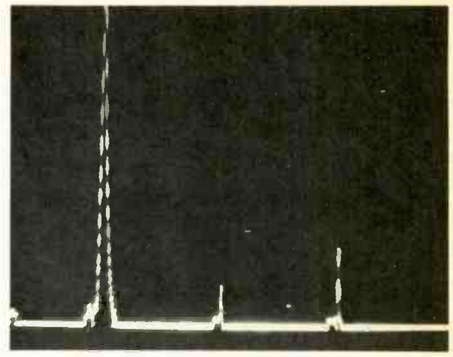

Figure 6: Program channel distortion. +2 dBV line output, -20 dBm microphone preamp input, 500 Hz fundemental. 10 dB/ vertical cm, 200 Hz/horizontal cm (linear sweep), 3 Hz resolution bandwidth.

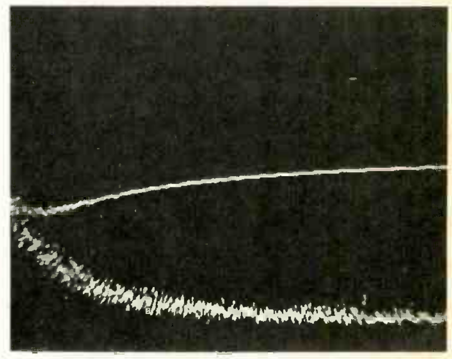

Figure 7: Adjacent channel cross talk (upper trace), and channel residual noise (lower trace). 10 dB/vertical cm, 5 kHz/horizontal cm (linear sweep). See text for details.

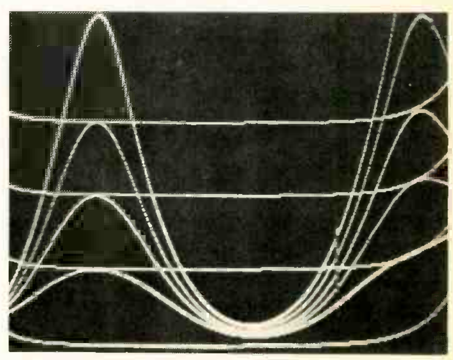

Figure 8: Equalizer peaking curve boost increments, at 100 Hz and 15 Hz center frequencies. Expanded scale, 1 dB/vertical cm with calibration lines at 2 dB intervals, log sweep, 20 Hz to 43 kHz.

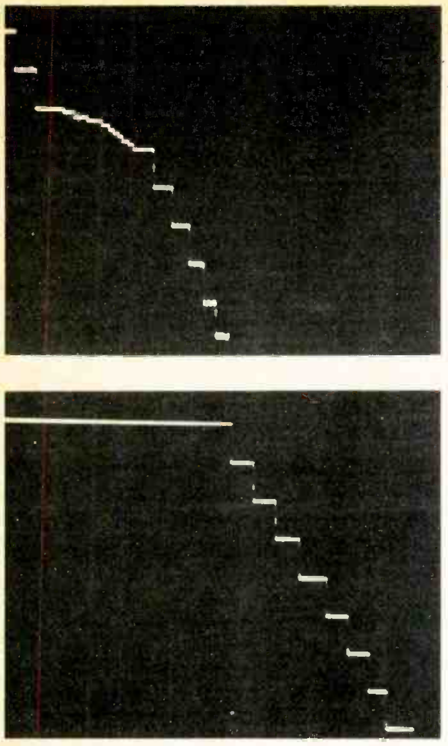

Figure 9: Vertical log scale calibration lines. Upper trace 10 dB and 1 dB steps shown at 10 dB/vertical cm. Lower trace, 1 dB steps shown at 1 dB/vertical cm. Top line is 0 dBV for both traces.

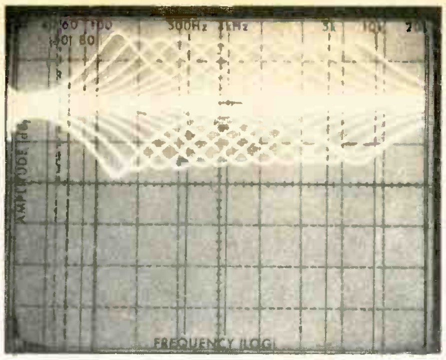

Figure 10: Peaking and notching family of curves. 10 dB/vertical cm, log frequency sweep, 20 Hz to 20 kHz.

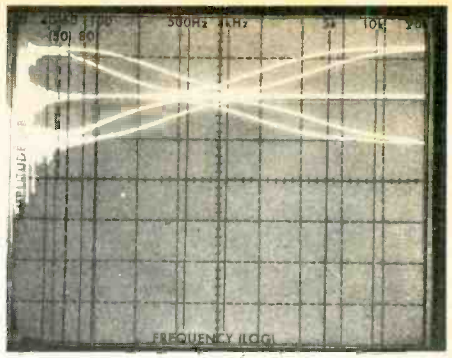

Figure 11: Maximum shelving contours. 10 dB/vertical cm, log frequency sweep, 20 Hz to 20 kHz.

pated side benefit of the storage scope is the background illumination from the CRT phosphor in storage mode. It backlights the CRT graticule markings and makes the traces easy to read against the graticule calibrations.

As with the HP 3580A, all response curves were generated in the log sweep mode while noise and distortion are shown in a linear frequency sweep mode.

Figure 10 shows the maximum peaking and notching characteristics of the brand "Y" equalizer. All of the traces conform closely to the ± 14 dB nominal maximum boost or cut with the exception of the 150 Hz curve which appears to be more like ± 17 dB.

The shelving contours are shown in Figure 11. The nominal turnover frequencies are 50 and 100 Hz and 7.5 and 10 kHz. There appears to be a slightly higher boost than cut in the HF characteristics.

A more detailed look at the 100 Hz and 7.5 kHz shelves is shown at 2 dB/division in Figure 12. The suspicions spawned by Figure 11 are fairly well confirmed. Note that the maximum 7.5 kHz shelf reaches its plateau by 10 kHz in the upper family while the lowest plateau hasn't been reached by 20 kHz in the lower set of curves. The difference is less than 1 dB at 10 kHz so this would probably not constitute a life or death situation. This set of photographs does serve to demonstrate the ability of the method to clearly indicate rather small deviations from the nominal or ideal.

Each of the curves generated by the Tektronix 5L4N in the logarithmic sweep mode took about 10 seconds. All of the trace families shown in Figures 10 through 12 took less than 5 minutes of test time to generate.

Here, again, we have the capability of generating accurate data characterizing system performance in a very short period of time. The data appears in a highly readable, familiar form that can be preserved for indefinite future reference. Amplitude accuracies of ± 1/4 dB over the frequency band of interest and frequency determinations having less than ± 5% uncertainties ought to be good enough for anybody's rock & roll.

A check of the brand "Y" equalizer incremental peaking and notching accuracy was run at the 3.5 kHz frequency setting. The results are shown in Figure 13. There is as much as 1 dB error in this particular equalizer's boost or cut setting. All increments are nominally 2 dB. The error is as great as 1 dB in the ± 6 dB and ± 14 dB settings and as little as about 1/4 dB elsewhere. How this kind of error affects the spectrum of the program material depends on how heavily the complementary properties of the curves is relied upon for additive or subtractive equalization through successive generations. The deviations from nominal in one equalizer lead one to suspect that there may be similar variations in others. The ear of the beholder figures in to some extent also. Similar evaluation of another equalizer was not attempted due to neglect on my part, so we'll never know who closely two different equalizers compare.

Figure 14 shows the program channel noise as measured with the line input open. The equalizer was switched out of the channel for this sweep. The single spike at the center graticule line is feedthrough from the board test oscillator at 10 kHz. The oscillator was used to verify the frequency calibration of the 5L4N just prior to the taking of the photograph. The very top graticule line corresponds to -50 dBV. It is evident that the majority of the noise energy lies below -110 dBV, for the line input condition.

Referring back to the brand "x" equalizer noise spectrum of Figure 7, note the pink nature of the noise in that photo. A significant difference that should be taken into consideration when comparing Figures 7 and 14 is the horizontal scale factors and the resolution bandwidths of the analyzer. Figure 7 was made with a 30 Hz resolution bandwidth while Figure 14 was made with a 10 Hz resolution bandwidth. The wider bandwidth will account for about 4.8 dB observed increase in noise level over what would have been indicated had the 10 Hz bandwidth been used in both cases.[4] The inclusion of the mike preamp as a noise source in Figure 7 makes it difficult to compare noise performance between the two program channels.

The brand "Y" program channel distortion was measured for a +2 dBV line input at 500 Hz having a residual second harmonic distortion of 0.036%. Figure 15 shows the channel line output at +2 dBV. The calculated THD is about 0.046%. The photograph clearly indicates the orders of distortion present as well as their magnitudes.

Availability of a pair of high spectral purity signal sources and a resistive summing network would have permitted records of the intermodulation characteristics of the consoles.[5] This could have been done with respect to each isolatable active console element if desired, the only limiting factor being the availability of patch bay access.

Intermodulation distortion is still not as widely used as an index of audio system performance as it might be. The use of the heterodyne spectrum analyzer makes it possible to conduct several different IM tests on the same test unit by

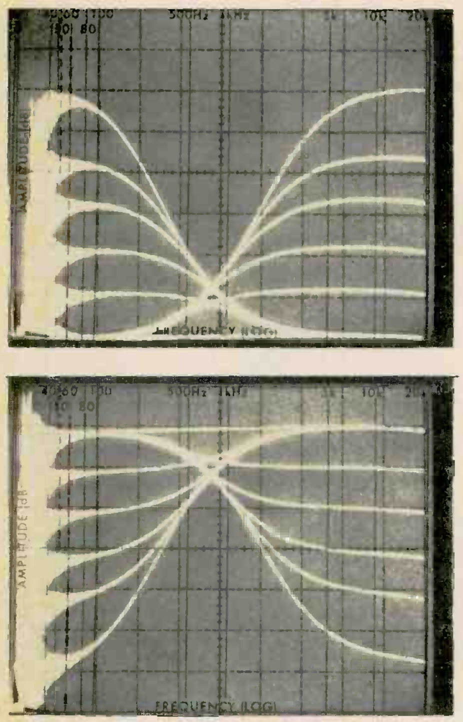

Figure 12: Shelving equalizer characteristics. Expanded scale, 2 dB/vertical cm, log sweep, 20 Hz to 20 kHz.

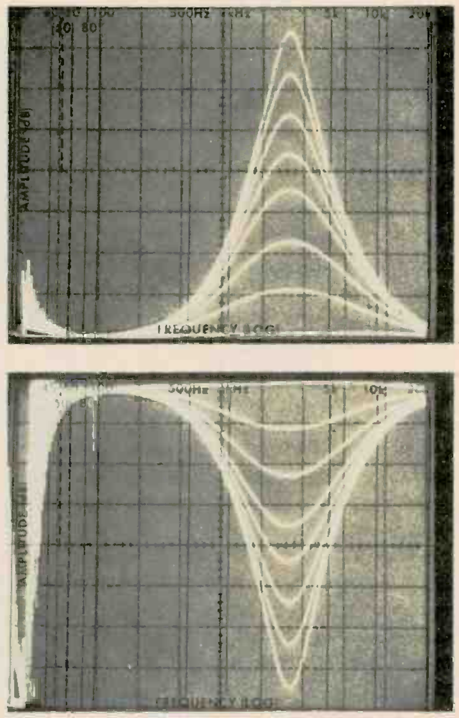

Figure 13: Incremental peaking and notching curves, 3.5kHz center frequency. 2 dB/vertical cm, log sweep, 20 Hz to 20 kHz.

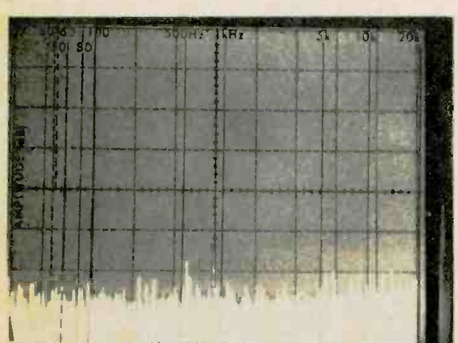

Figure 14: Program channel noise. 10 dB/vertical cm, 2kHz/horizontal cm (linear sweep), 10 Hz resolution bandwidth, -50 dBV top line reference.

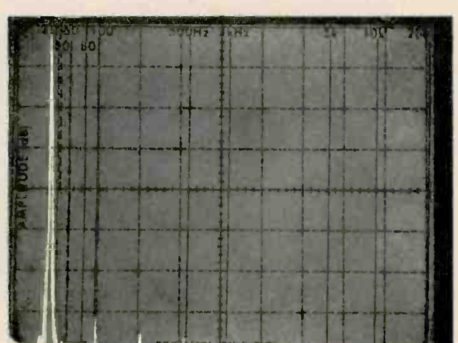

Figure 15: Program channel distortion. 10dB/vertical cm, 500 Hz/horizontal cm (linear sweep)

simply varying the signal input frequencies and relative levels. Use of the conventional IM distortion meters limits the test performed to that for which the individual IM analyzer was designed.

There are other tests that could also be performed using the spectrum analyzer and appropriate signal sources. The recently infamous Transient Intermodulation Distortion can be investigated quickly over a range of input levels. The tracing simulation devices used for reduction of geometrically induced distortion in lacquer master discs can be checked by cutting a 1 kHz CW signal at several groove velocities and checking the harmonic content of the reproduced signal.

My brief association with the art of master disc recording has led me to believe that identical response and gain characteristics are very desireable, not only between program channels, but for preview channels as well. Deviations from the nominal noted in Figures 4 and 5 could be detected by inspection of photographs generated in less than an hour. Program channel RIAA equalization characteristics could be similarly verified against each other as well as against the published standard. The response and distortion characteristics of limiters versus compression could also be recorded and determined graphically. Extended white or pink noise analysis would yield system frequency response.

Spectral analysis of system square wave response was attempted with the Tek 5L4N, but I'm not sure the results indicate anything of real significance. The time-domain observations of input and output waveforms indicated nothing undesireable or unexpected. Even harmonics were slightly reduced in relative level at the output against the input spectra, but that's no big surprise for an AC coupled system.

The professional recording industry has impressed me as being somewhat light as far as preventive maintenance is concerned. The economic necessity of keeping things running and the conflict between full studio bookings and the time required for system performance verification is not hard to understand. I like to eat regularly too.

Test equipment is a different breed of cat from the electronics that comprise the signal-handling systems in the recording industry. Test equipment requires periodic recalibration if confidence in the accuracy of test results is to be retained. Five thousand dollars spent on a new tape machine is a far more direct income producer than a like amount spent on test equipment. Deterioration of audio system performance over extended intervals is difficult to detect subjectively and thus merits no particularly high position on studio priorities. I can't recall a console being replaced for reasons other than configuration obsolescence.

The extent and complexity of modern recording systems is, to me, staggering. I suspect that there may be some who will regard mere existence of the diagnostic methods described above with some suspicion. The whole idea of performance verification will no doubt be greeted, by some, with the same warmth reserved for a mirror salesman at an Ugly convention. After all, the ostrich approach to problem solution has sustained mankind for millenia. Sleeping dogs left lying never caused rabies.

For those of us who realize that the present state of the art is only a prologue, I sincerely hope that I have contributed to a rise in the general level of awareness. As Dr. Kinsey once said, my goal is to shed some light on those dark areas of life, little discussed in polite circles. Or was it Masters and Johnson?

Although I don't care to assume the role of Jiminy Cricket to the recording industry, my basic sense of insecurity causes me to sustain the belief that things could be better than they are at present. If we take the time to establish any point of departure in the present, the direction of future improvements will become more definite.

I am indebted to several individuals whose interest, courtesy and tolerance contributed to the enhancement of my consciousness in the compilation of this moral epic. They are:

Bill Brennan of B&K Instruments, Inc.
Anaheim, California
Dave Breville of Hewlett-Packard Co.
North Hollywood, California
Jerry Ferree of ABC-Dunhill Records
Los Angeles, California
Norbert Gonsman of General Radio Co.
Irvine, California
Robert Loft of Tektronix, Inc.
Van Nuys, California
Dave Luttropp of Hewlett-Packard Co.
Loveland, Colorado
Ron Malo of Devonshire Sound
North Hollywood, California

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JBL ANNOUNCES MODEL 6233, A 600-WATT RMS POWER AMPLIFIER

The 6233 is JBL's first in an extensive new series of professional power amplifiers and related electronic products scheduled for release this year.

The 6233 is said to be unique among high power amplifiers; it weighs 35 pounds, is virtually distortionless and delivers its full rated output continuously in ambient temperatures as high as 120 F. Plug-in crossover modules can be added, making the 6233 a self-contained biamp system. It is designed for use in recording studios, wide range sound reinforcement systems and similar applications requiring stable, distortion-free amplification at sustained high power levels.

The 6233 is approximately one-third the size and weight of conventional amplifiers having comparable performance. Such lightness was made possible by incorporating a 2000-Watt solid state inverter power supply instead of the massive transformers commonly used. Inverter power supplies have been used extensively in space flight vehicles but the 6233 is the first audio amplifier in its power class to feature such a device.

Each 6233 is extensively pretested, then individually certified to meet or exceed its published specifications. To achieve certification, each amplifier is operated non-stop for 24 hours under conditions simulating extremely severe field use. Without being allowed to cool down, it must then produce a minimum of 300 Watts RMS per channel into 4-ohms and 200 Watts RMS per channel into 8-ohms with both channels driven. Under these same conditions, THD and IM distortion cannot exceed .1% from 20-20,000 Hz at rated power or less.

Advanced design is apparent throughout the 6233; output stages are full-complementary and twelve 150-Watt output transistors per channel are featured. This provides outstanding sound quality and greatly increased reliability since each transistor never operates at more than a fraction of its output potential — even when the amplifier itself is driven to full rated power.

The amplifier is also stable under all load conditions and fully protected from short circuits, mismatched loads, installation errors and excessive temperature rise. A thermally activated two-speed fan assures safe operation in ambient temperatures as high as 50 C (120 F).

Because VU meters cannot reveal amplifier clipping or power reserve accurately, particularly when power line voltages are low, the 6233 features five front panel lights per channel for monitoring power output. Four lights become illuminated in sequence as higher power levels are reached and the fifth glows red at the threshold of clipping. A special circuit assures accuracy even when power line voltages are low.

JAMES B. LANSING SOUND, INC., 3249 CASITAS AVE., LOS ANGELES, CA 90039

Circle No. 135

WHITE INSTRUMENTS SERIES 4000 ACTIVE EQUALIZERS

The new Series 4000 Active Equalizers are based on a combination of LC tuned circuits and the latest integrated circuit operational amplifiers to assure high linearity and stability. A unique circuit utilizing all negative feedback provides equal Q in both cut and boost conditions. There are 27 channels on ISO 1/3 octave centers from 40 Hz to 16 kHz. 10 dB of boost or cut is provided on continuously variable controls. In addition, there is a low end roll-off control which can be varied from 20Hz to 160Hz with a 12dB/octave slope.

The Series 4000 units have a 20 kilohm input impedance and a recommended operating level of 0 dBm. Maximum output before clipping is +18 dBm. Noise and hum are better than -92 dBm due to low noise circuitry and magnetic shielding. Distortion is less than 0.2% for levels to +18 dBm.

Dual independent outputs are provided. Each output will drive a 600 ohm or greater load. An accessory socket is provided prior to the buffered outputs for insertion of a low level crossover network for bi-amped systems. Thus, equalization and electronic crossover is accomplished in one unit. An equalizer by-pass switch is also provided on the rear panel.

Sealed Mil-spec type rotary potentiometers are used throughout for long life and low noise operation. Dials are calibrated for ease in logging or repeating settings. The unit is contained in a 3½" x 17" black anodized aluminum case and is approximately 8½" deep. Weight is approximately 11 pounds.

Model 4001 is for sound reinforcement. It comes complete with rack mounting end pieces and security panel. A transformer input with a floating primary is provided. Connections are made on a barrier type terminal strip.

Model 4002, the music reproduction model, has end pieces with rubber feet for table use, phono jack type connectors, and a level control on the input.

Both models are priced at $635.

WHITE INSTRUMENTS, INC., P.O. BOX 698, AUSTIN, TEXAS 78767.

Circle No. 136

VEGA INTRODUCES DIVERSITY WIRELESS MIKE SYSTEM

This new system utilizes the VEGA PRO Series transmitters and receivers in a diversity reception mode that virtually eliminates all fades and dead spots.

Feeds and dead spots are caused by interference between direct and reflected radiation that cancel, resulting in loss of signal. The problem is most prevalent in "studio" operations, but also occurs outdoors.

In the diversity mode, two VEGA PRO receivers, placed three feet or more apart, both receive the transmissions. Because the two receivers are more than one-half wavelength apart, both will not have signal cancellations at the same instant. Both receivers feed a Model 62 Diversity Combiner that selects the receiver with the best signal strength within microseconds. The switching is immediate and noiseless. The resultant audio, the best of both receivers, is noise free and drop-out free.

The VEGA Diversity System is composed of Models 54 or 55 Transmitter, two Model 58 Receivers and Model 62 Diversity Combiner. For technical application notes on this new concept, call or write Kenneth L. McKenzie, Sales Manager, Wireless Products.

VEGA, DIVISION OF CETEC CORP., 9900 BALDWIN PLACE, EL MONTE, CA 91731.

Circle No. 139

FET POWER AMP FROM YAMAHA

Utilizing a new power FET developed in Japan, Yamaha has introduced the model B-1 FET stereo power amplifier capable of delivering 150 watts/channel into either 8 or 4 ohms, with both channels driven, and a full power bandwidth of 20 – 20kHz.

According to Yamaha, the power FET offers several advantages over bipolar transistors in the output stage. 1) Because of its excellent linearity, the high order odd harmonics (those most irritating to the ear) are virtually absent. The even order harmonics are cancelled through the use of a carefully designed push-pull circuit. The increased linearity also permits a design with less negative feedback,

which results in increased stability. 2) FETs do not exhibit the carrier-storage effect associated with bipolar transistors. This improves the transient response of the amplifier as well as eliminating the cause of crossover distortion. 3) The FET exhibits an inverse relationship between temperature and current flow, and as a result is thermally self-protecting, making it immune to thermal runaway.

The theoretical and technical advantages of the FET combine to produce audibly better sound reproduction, according to Yamaha, producing a natural and smooth quality, free from the harshness sometimes attributed to bipolar transistor amplifiers.

Frequency response is 5 – 100kHz, +0, -1dB, with both total harmonic and IM distortion at .1% at rated power.

YAMAHA INTERNATIONAL CORP., MUSICAL INSTRUMENT DIVISION, 6600 ORANGETHORPE AVE., BUENA PARK, CA 90620.

Circle No. 140

YAMAHA MODEL B-1 FET POWER AMPLIFIER (FRONT VIEW) WITH COMPANION CONTROL UNIT MODEL UC-1.

BACK VIEW OF MODEL B-1 FET POWER AMPLIFIER.

POLYPHONIC GUITAR SYNTHESIZER INTRODUCED BY 360 SYSTEMS

The 360 Systems Polyphonic Controller makes it possible to use an electric guitar to play six music synthesizers. Each string of the instrument produces outputs from the Polyphonic Controller for pitch, envelope, and trigger. All outputs are electrically compatible with control voltage formats used in most synthesizers. System response is extremely fast, and reproduces all pitch and loudness variations played on the guitar strings. Complete six channel synthesizer packages are also in production for both studio and live performance work. Exceptional musical stability and ruggedized construction are featured in the Polyphonic Controller and all Polyphonic Synthesizer packages. The guitar synthesizer system produces up to 18 notes from a single chord, with independent control of the voicing on each of six channels. Exclusive from:

360 SYSTEMS, 2825 HYANS STREET, LOS ANGELES, CA 90026.

Circle No. 141

VARI-DEPTH COMPUTER FOR DISC MASTERING

Capps & Co., Inc. has developed a new Vari-Depth Computer for disc mastering. This depth system has been designed to operate in conjunction with the Capps Vari-Pitch Computer.

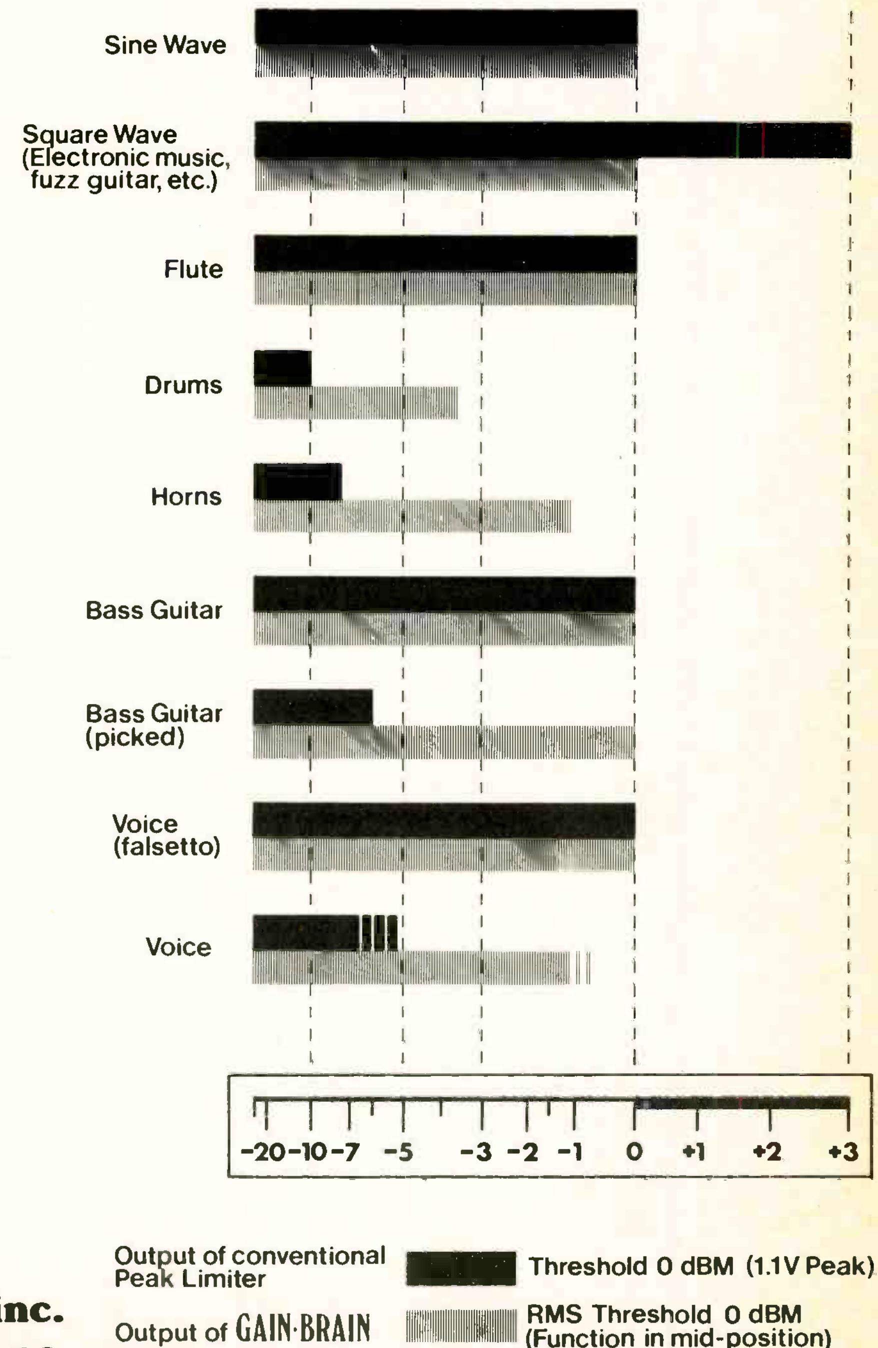
This completely computerized disc mastering system is available at a moderate cost affordable for even the smallest disc mastering studio. The complete system can be installed on any Scully or Neumann lathe in a matter of hours.

The new vari-depth unit utilizes state-of-the-art low power CMOS logic elements to digitize the incoming audio and transfers this information to the master disc for amazingly high cutting efficiency. The Capps vari-depth unit is compatible with all Westrex, Neumann and Haeco solid state systems.

When using the new vari-depth computer, the operator may program the system to deepen on lateral as well as on vertical information. This unique feature, when used with the Capps vari-pitch system, permits high level cutting at nominal pitches above 500 G.P.I. The

GAIN·BRAIN™ limiting or conventional Peak limiting?

A COMPARISON OF OUTPUT LEVELS FOR DIFFERENT PROGRAM MATERIALS



If you use a conventional limiter . . . this is what you get. Sure the peaks are level, but what about the sound? If you really want the horns 7db under the flutes or the bass 10db louder than the drums, go ahead and use your limiter; it doesn't recognize the apparent loudness of the program. It's activated only by the peak value of the input waveform and you know how little that resembles the actual audible power content.

GAIN·BRAIN™ knows what you want to hear. It's activated by both the peak and the RMS content of the input waveform. The result is a really accurate control of the output level, for *all* the instruments, plus an absolute control of peaks. Depending upon your needs, the GAIN BRAIN may be adjusted to act only as a peak limiter, or only as an RMS limiter or anywhere in between.

Light emitting diodes sequentially indicate the gain reduction and the mode of limiting.

allison research inc.
nashville, tennessee
(615) 385-1760

Circle No. 142

vari-depth system can be programmed for automatic deepening during lead-in, banding, expand and finish operations if desired.

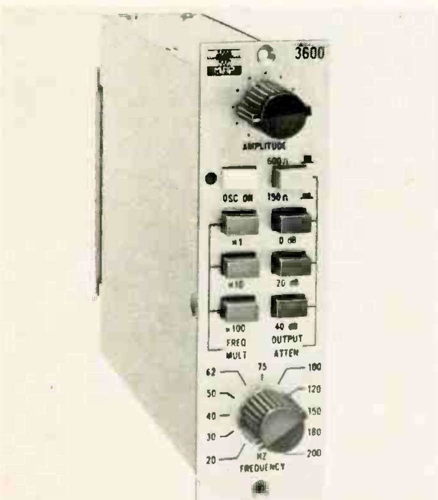
CAPPS & CO., INC., 20 ADDISON PL., VALLEY STREAM, L.I., N.Y.

Circle No. 143

PRECISION OSCILLATOR

A new Precision Oscillator designed for performance testing of high quality audio equipment which accept input levels from -80dBm to +18dBm is available from Modular Audio Products, a unit of Modular Devices, Inc., Bohemia, New York.

Known as Model 3600, the new unit, with low distortion (typically .02%) tests microphone preamplifiers, line amplifiers, power amplifiers and complete audio systems. The unit is equipped with its own output transformer and attenuator for normal testing, and an unbalanced low impedance output for use in slate/tone systems.



Other features include — stable output level of ± 0.25dB, 33 calibrated frequencies from 20Hz to 20kHz, selectable output impedance of either 600 or 150 ohms, transformer coupled output and calibrated output attenuator.

The Modular Audio Precision Oscillator measures only 1-15/32"W x 4-5/8"H x 5-3/4"L and weighs 2 lbs. Delivery is from stock to 30 days.

MODULAR AUDIO PRODUCTS, 1385 LAKELAND AVENUE, BOHEMIA, NEW YORK 11716.

Circle No. 144

6 IN 2 OUT RECORDING/REINFORCEMENT MIXER

The TAPCO 6200 is a rugged, high performance rack mountable 6 by 2 mixer designed for the professional musician and recording engineer.

A remarkable -132 dBm equivalent input noise is claimed (with a 150 ohm source impedance) for each of the balanced inputs, with a frequency response of 10 Hz to 40 kHz ± 1 dB, and harmonic distortion at .08%. The input preamp

contains a single transistor operated at low gain and carefully matched to a custom built input transformer. This combination, according to the designer, pushes the signal-to-noise ratio right up against the theoretical limit.

The input circuit is next to impossible to overload, handling mike levels up to 500 mV RMS. Automatic padding level controls expand the system's headroom, permitting it to handle extremely hot input signals without clipping, and allowing the mixer to optimize signal-to-noise regardless of input level.

Each channel has individual gain and panning controls, ± 18 dB bass and treble equalizers, and auxiliary high level inputs for tape mixdowns. Output capability is +16 dBm into 600 ohms, or 10 V RMS into 2 k ohms and higher.

The stereo output section includes left and right controls for volume, +9 dB @ 20 kHz mike equalization and -6 dB @ 100 Hz rumble filter (6 dB/octave). A 600 ohm stereo headphone output is provided. Each output channel also has a continuously adjustable LED level indicator.

For studio use the 6200 can be stacked with additional units to create 12, 18, and 24 channel recording capability. Like all TAPCO, the 6200 is compact, sturdy and reliable and has been designed for heavy duty use with maximum professional performance.

Professional net price, $389.

TAPCO, 405 HOWELL WAY, EDMONDS, WA 98020.

Circle No. 145

SHURE ANNOUNCES SMALL, LIGHTWEIGHT, PROFESSIONAL MICROPHONE

For broadcast applications where an inconspicuous microphone with professional quality performance is needed, Shure Brothers Inc., Evanston, Illinois, is now offering the Model SM62 unidirectional dynamic microphone.

This new Shure microphone combines flat, uncolored response, uniform cardioid pickup pattern, excellent performance characteristics and minimized feedback in a uniquely small unit. The SM62 measures only 124 mm (4 29/32") long.

Other features include an extremely effective "pop" filter to suppress wind noise and explosive breath sounds, and internal shock isolation to keep noise caused by unstable stands, stages, and podiums from reaching the microphone.

In professional recording applications, the SM62 is especially well-suited for picking up brass, drums, strings (guitar), and vocals. Its compact styling, balanced feel, and wind filter make the SM62 ideal for hand-held stage and remote interview applications. When used on podiums or desk stands for newscasts, panel discussions and speeches, the SM62 "disappears" into the surroundings so that users speak to the audience, not to the microphone.

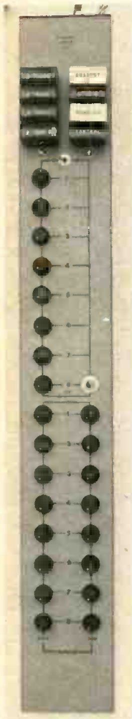
Professional user net price of the Shure Model SM62 Unidirectional Dynamic Microphone is $84.00.

SHURE BROTHERS INC., 222 HARTREY AVENUE, EVANSTON, IL 60204.

Circle No. 146

A MONITOR/CUE MODULE FOR THE TASCAM MODEL 10 CONSOLE DEVELOPED BY ACCURATE SOUND COMPANY

This eight channel monitor/cue module is available exclusively from Accurate Sound Company. All components and controls are mounted on a standard Tascam 3" panel and no modification of the mother board is required. This module accommodates 8 channels of monitor and cue. The cue section, with individual gain control for each channel can facilitate one 8 channel machine or one 4 track machine with selection by a toggle switch. The monitor section features individual gain and panning controls for each channel.



A headphone jack is provided on the module to sample the mix sent to the studio headphones.

There is also switching to punch up any one of the 4 multi channel tape monitor speakers in stereo or quad. Module is complete with all necessary cables and connectors.

ACCURATE SOUND COMPANY, 114 5th AVENUE, REDWOOD CITY, CA 94063.

Circle No. 147

HELPINSTILL PIANO SENSOR

The Helpinstill Piano Sensor uses three electromagnetic pickups of a new flexible design to convert any upright piano into an electric instrument.

These sensors slip into position behind the strings near the top of the piano, and stick to the frame magnetically. Unlike contact mikes or transducers, they sense only the string movement (like the pickups on an electric guitar) and are not dependent on vibrations of the sounding board. This results in a clean, no-feedback signal from the piano with all the attack and presence necessary to compete on an equal footing with guitars and other electric instruments.

The complete kit consists of three sensors in lengths of 14", 16", and 18". These lengths have been selected to enable a perfect fit on nearly every style verticle piano whether it be a spinet, console, studio, or upright grand. The passive mixer box features three balance controls to regulate the three sensors, volume, and tone controls; outputs on the box allow connection to low-impedance microphone inputs or high-impedance guitar amp inputs. The resulting sound is the same quality as the original Helpinstill Piano Pickup for grands, which has become the choice of virtually every concert artist in the world. Price $195.00.

HELPINSTILL DESIGNS, 6124 JESSAMINE, HOUSTON, TX 77036.

Circle No. 148

ROBINS INTRODUCES UNIQUE, MULTI-TIP HEAD DEMAGNETIZER

The Consumer Products Division of Robins Industries Corp., Commack, N.Y. 11725, has introduced an unusual, universal head demagnetizer with interchangeable tips.

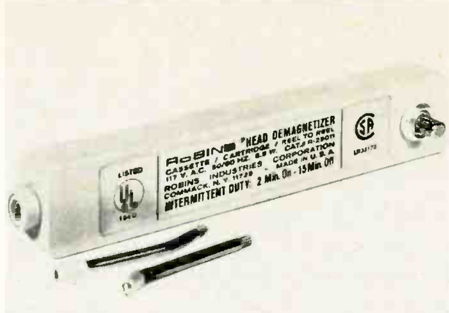
The new Robins head demagnetizer, comes with one tip in place and two other tips of different shapes to fit all types of equipment. Removal and replacement of tips, as needed, is done by simply screwing them into place.

Thus the Model 25011 represents a solid, single investment in an easy-to-use

Circle No. 149

LC #73-87056
ISBN #0-914130-00-5

accessory essential for optimum recording and playback. A momentary control switch built into the flame-retarded, impact-resistant plastic case adds to convenience.



Model 25011 is UL approved. ACSA-approved unit is available for Canada.

Recommended usage is after 20 hours of play-record time, although it can be applied to advantage after every operating period. The demagnetizing process takes only a few seconds.

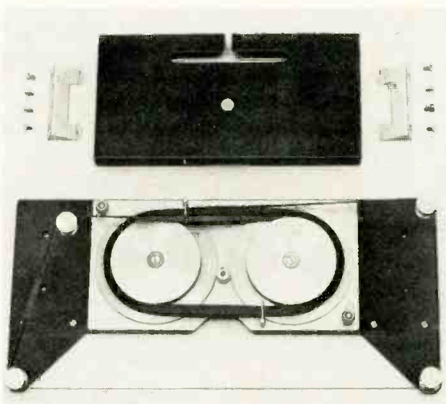
Model 25011 lists at $15.00.

ROBINS INDUSTRIES CORP., 75 AUSTIN BLVD., COMMACK, L.I., NY 11725.

Circle No. 150

DELAY/ECHO UNIT FOR REVOX A77

Revox Corporation is now offering a delay/echo unit that incorporates an endless loop cassette that can be used with any Revox A77 tape deck by removing the deck plate and reel turntables and attaching a cassette mounting plate. The changeover takes less than 10 minutes.



Applications include: tape echo, automatic message repeating; continuous short-term program monitoring; time delay of programmable machinery. For the tape echo application, the variable speed control unit may be used to extend the range of available time delay.

Price: including preloaded tape: $187.

REVOX CORPORATION, 155 MICHAEL DR., SYOSSET, NY 11791.

Circle No. 151

NEW STEREO FRAP FOR PIANO

Using two channels of the same preamp electronics as other studio FRAPs, the new model FS-200 features outputs for stereo and/or mixed mono and built in variable low frequency roll-off filters for each channel.

Since many customers prefer using two FRAPs to pick up piano, the FRAP people developed this versatile stereo unit. It has also proved effective for picking up guitar, harp, and instruments with spread out or uneven sound sources. The two transducers can be used on two different instruments and fed to either the stereo outputs or mixed through the mono output.

This unit uses the same high quality transducers that FRAP is known for. They are quickly attached and removed with FRAP's inert adhesive wax. The specifications (available from FRAP) are the same as for the other studio FRAPs. The built in variable low frequency filter rolls off at 6 dB per octave from approximately 30 to 300 Hz. This filter is also available for use with the F-100, F-200, and F-250 FRAPs.

FRAP systems are available at most places where professional music and audio equipment are sold or they may be ordered direct. The stereo/Piano FRAP retails at $650, and the Variable Low Frequency Roll-Off retails at $25.

FRAP, BOX 40097, SAN FRANCISCO, CA 94140.

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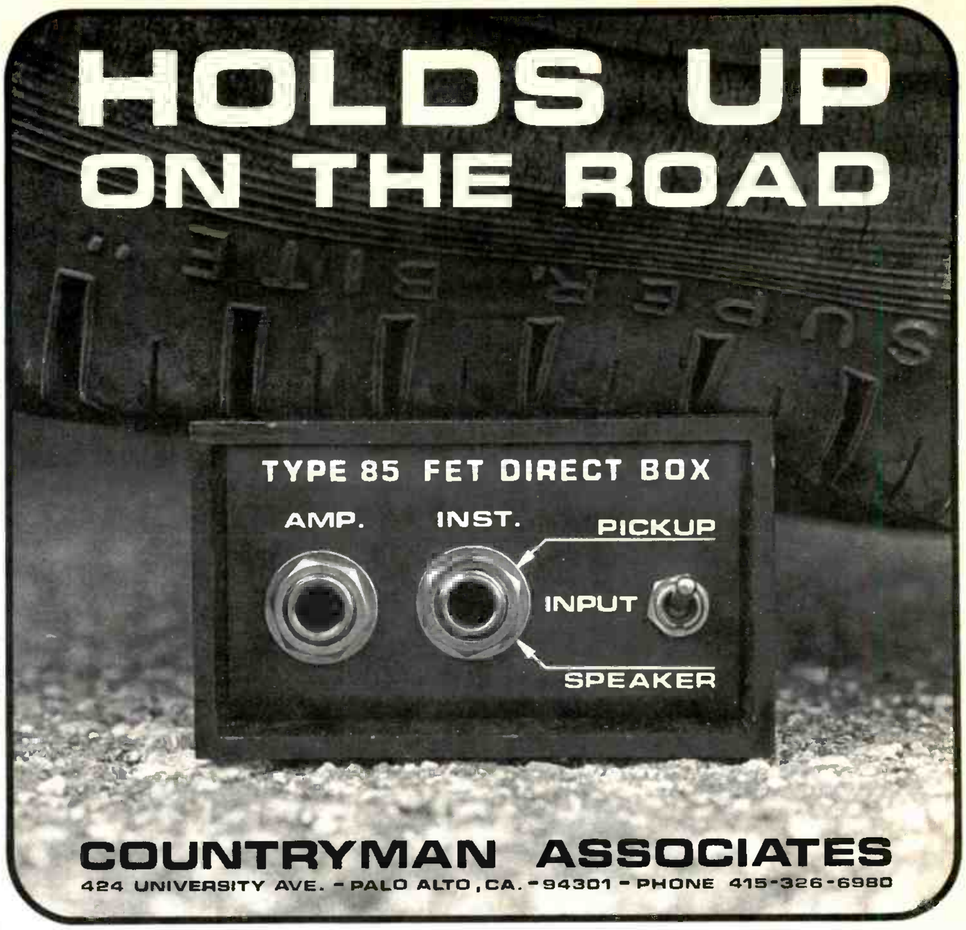

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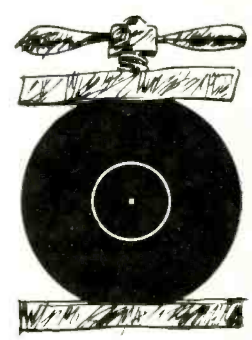

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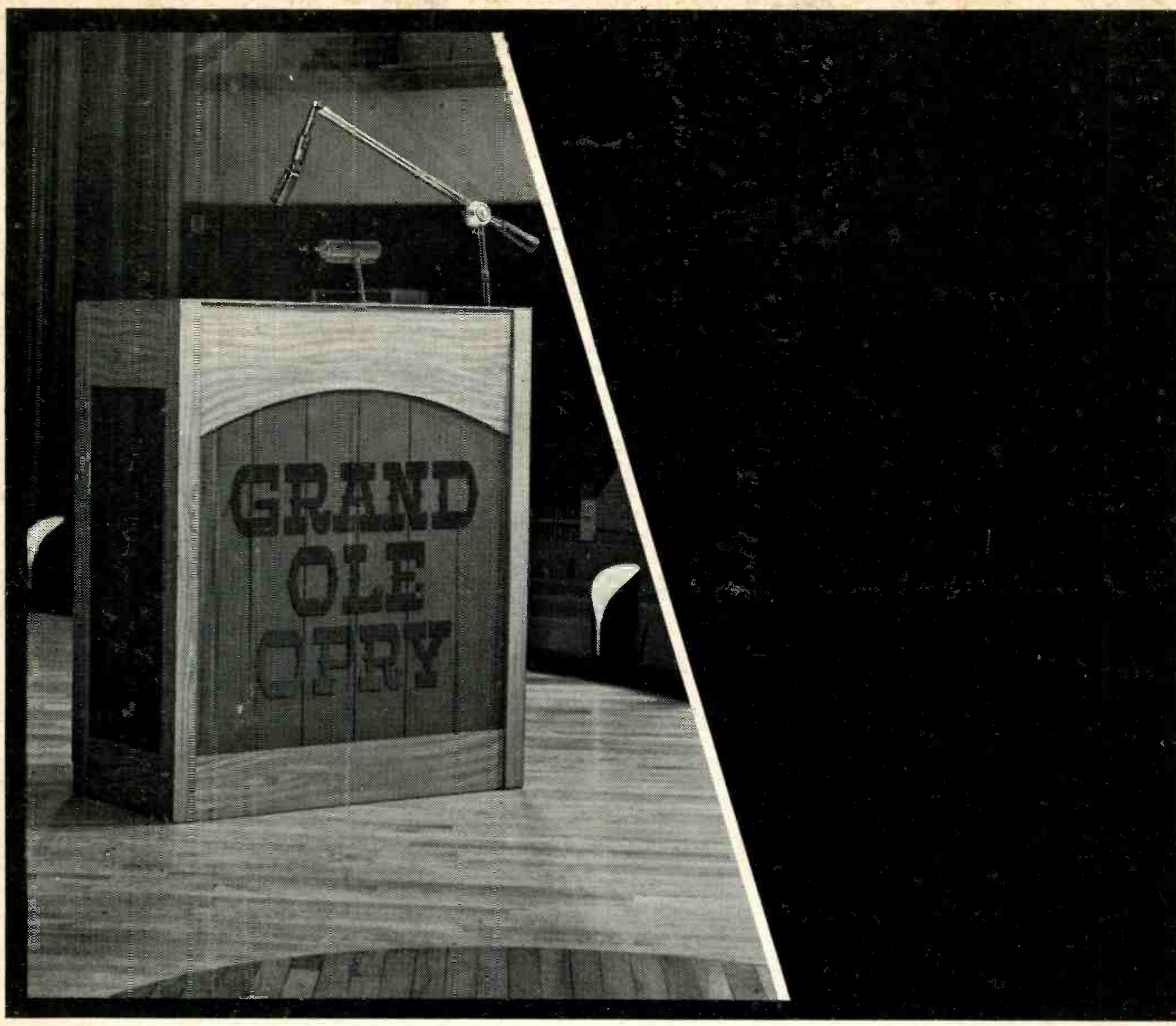
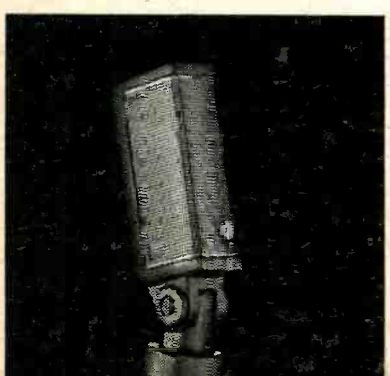


Grand New Opry.



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