

The LEDETM Concept for the Control of Acoustic and Psychoacoustic Parameters in Recording Control Rooms*

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An approach to the standardized control room is found in the "live end-dead end" (LEDETM) approach. Desirable features include control of the initial time delay, psychoacoustic removal of the directional clues belonging to the control room, and control of the early reflected sound field's density, spacing in time, and acoustic level. This results in an exceptionally neutral acoustic environment and allows development of a sound field at the mixer's ears which correlates remarkably with the sound field appearing at the microphones in the studio, thereby allowing precision judgments to be made at the mixing console.

0 INTRODUCTION

The concept of the LEDETM control room (Fig. 1) evolved from observations of the interaction of very early reflections with the direct sound emitted by a monitor loudspeaker in a contemporary control room. These observations were made at the mixer's head position behind the mixing console by means of time-delay spectrometry (TDS).¹

TDS is a unique and valuable measurement technique which allows us to study the reflection-by-reflection construction of what is popularly called "room modes." In addition, TDS enables us to measure the direct sound field L_D as well as the total sound field L_T , thus providing accurate values for the calculation of the reverberant sound field level L_R . In cases where the ambient noise level value L_N is within 20 dB of any of the levels mentioned above, its effect should be compensated for in the usual manner (see Table 1).

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¹ TDS was developed and patented by Richard C. Heyser in 1968 [1]–[3].

1 KEY OBSERVATION

The key observation which stimulated our interest was a simple one, easily recognized as fundamental and yet relatively overlooked in the literature, and where noted, such as in [4], it was not applied to loudspeaker mounting considerations.

What has been overlooked is that a reflection combining with the direct sound from a loudspeaker, both reflection and direct sound having nearly equal levels, causes "comb filters," and that the shorter the delay experienced by the reflection, the broader are the individual response anomalies which constitute the observed comb filter. Figs. 2 and 3 illustrate the bandwidths and spacing of the response anomalies generated by two sound sources which are separated by various delays.

2 RESULT OF OBSERVATIONS

It was realized that broad-band response anomalies would be more audible than narrow-band response anomalies. The early work by one of the authors (Don Davis) in the development of sound system-room equalization made the critical bandwidth theory and application very



Fig. 1. The first LEDE™ control room.

Table 1. Sound fields present in control rooms.

$$L_T = 10 \log (10^{(L_D/10)} + 10^{(L_R/10)} + 10^{(L_N/10)})$$

$$L_R = 10 \log (10^{(L_T/10)} - 10^{(L_D/10)} - 10^{(L_N/10)})$$

where

L_T = total sound level, in dB

L_D = direct sound level, in dB

L_R = reverberant sound level, in dB

L_N = ambient noise level, in dB

All levels in dB re 20 μ Pa.

The addition of two signals having approximately equal amplitudes and phases (they may be at different frequencies, however) constitutes coherent addition of sound pressures. Such addition can only occur when precisely equidistant from two sources in an essentially anechoic environment. The equation is

$$L_{\text{comb}} = 20 \log$$

$$\sqrt{(10^{(L_1/20)})^2 + (10^{(L_2/20)})^2 + 2((10^{(L_1/20)})(10^{(L_2/20)})(\cos a_1 - a_2))}$$

where

L_{comb} = combined sound level of two signals in dB,

L_1 = sound level of first signal in dB,

L_2 = sound level of second signal in dB,

a_1 = phase angle of L_1

a_2 = phase angle of L_2

The following illustrates the effect of phase on equal-level signals. A 6-dB addition signifies a dominant direct sound field at a precise point. It does not signify a sound power increase over a given area.

$L_1/$ [deg]	$a_1/$ [deg]	$L_2/$ [deg]	$a_2/$ [deg]	$L_{\text{comb}}/$ [dB]
90	0	90	0	96.02
90	0	90	10	95.99
90	0	90	20	95.89
90	0	90	30	95.72
90	0	90	40	95.48
90	0	90	50	95.17
90	0	90	60	94.77
90	0	90	70	94.29
90	0	90	80	93.71
90	0	90	90	93.01
90	0	90	100	92.18
90	0	90	110	91.19
90	0	90	120	90.00
90	0	90	130	88.54
90	0	90	140	86.70
90	0	90	150	84.28
90	0	90	160	80.81
90	0	90	170	74.83
90	0	90	180	0.00

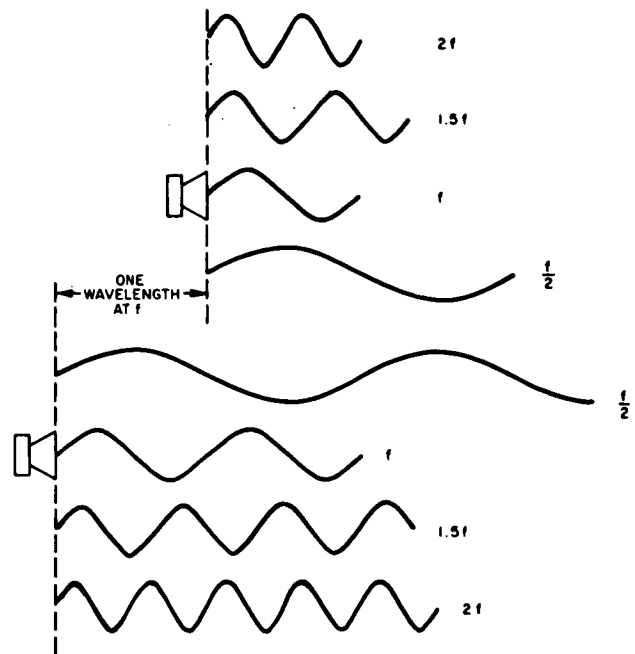


Fig. 2. Calculating addition and cancellation of signals.

familiar. This allowed for the rapid assessment of the probable auditory results which each type of response anomaly was likely to evoke. The very early broad-band, often high-level, reflections instantly inform the trained listener that he or she is in a small enclosure by generating a very short initial time-delay gap, thus ruling out the opportunity to hear the initial time-delay gap of the studio itself. This “masking” of the studio’s initial time-delay gap is undesirable. These effects were demonstrated by means of TDS, and the subjective results of avoiding very early reflections (less than 1–3 ms) were further elaborated upon. The first complete treatment of the concepts was published in [5].

3 SERENDIPITY

By 1978 April one of the authors had rebuilt his control room to conform to LEDE™ principles. Upon hearing the results of this approach, several pleasant surprises occurred. First we had specified a very hard surfaced, very diffuse live end for the entire room behind a plane across the mixer’s head position. In front of this plane the entire room was made as absorptive as practical (Figs. 4 and 5).

When the very hard reflective rear wall was faced by the listener, even with cupped ears, no sound was perceived as coming from the hard reflective walls. We realized that we were standing in the so-called Henry, Fay-Hall, Haas, etc., zone [roughly 20 ms or 22.6 ft (6.89 m) of path-length difference between the direct sound and the first reflection] [7]–[9] (Figs. 6 and 7).

Kuttruff had pointed out in 1973 [11] that if there is a hot, hard reflection within the first 20 ms, the masked shadow zone can be extended past 30 ms.² In fact, given the correct sequence of reflective energy from the rear and side walls,

² The authors are indebted to Ted Uzzle of Cambridge, MA, for researching and sharing this particular reference with them [11], [12].

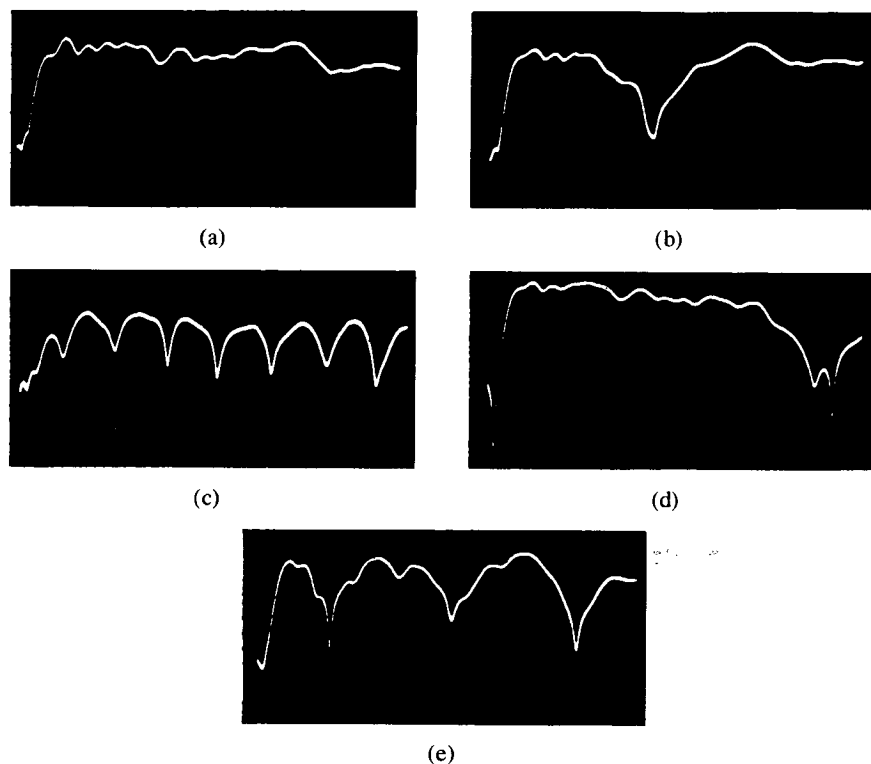


Fig. 3. Generation of response anomalies and their measurement by TDS. (a) Direct sound field of studio monitor. Response is from 0 to 10 000 Hz. (b) Total distance between direct and reflected sound field is 4.52 inches (114.8 mm). (c) Direct and reflected sound fields of studio monitor. Total distance between direct and reflected sound fields is 13.56 inches (344.4 mm). (d) Total distance between direct and reflected sound fields is 1.94 inches (49.3 mm). (e) Total distance between direct and reflected sound fields is 6.78 inches (172.2 mm).

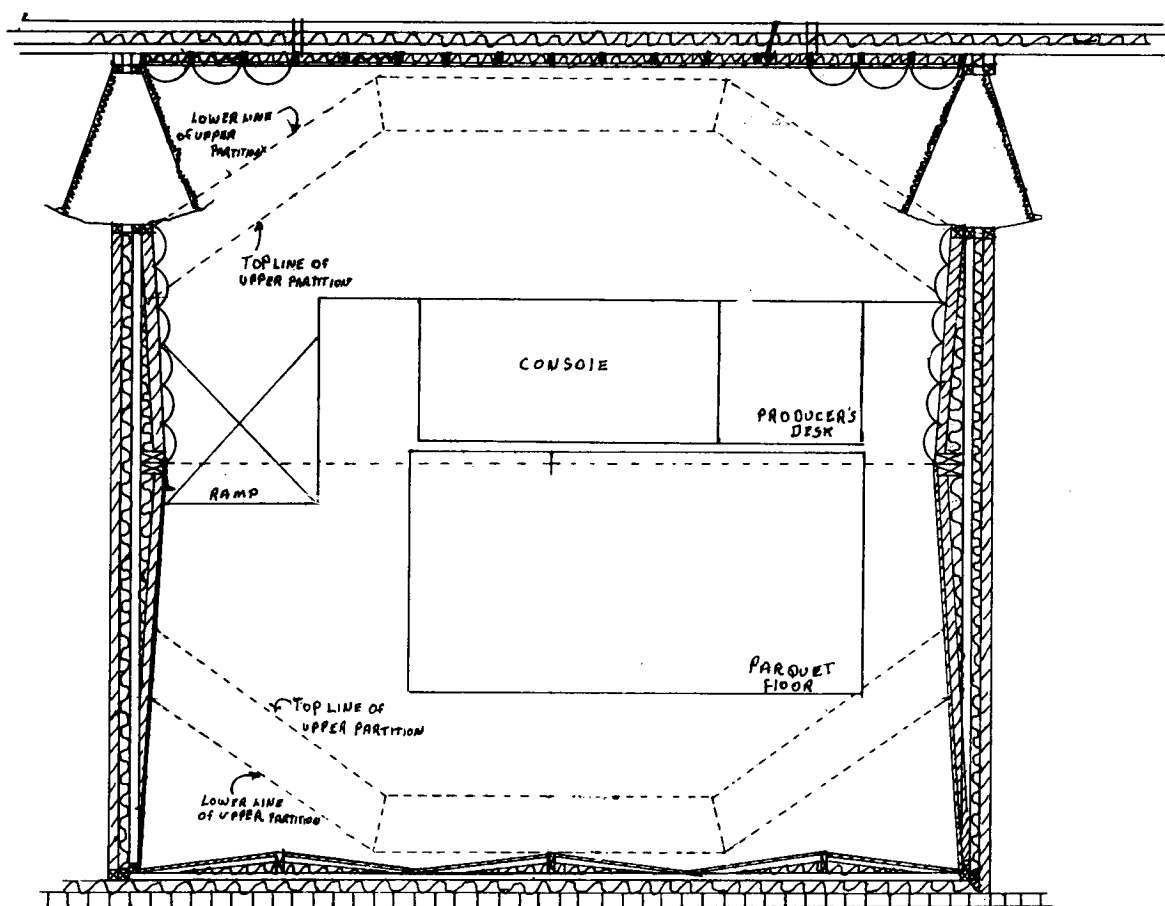


Fig. 4. Plan view of an LEDE™ control room now in service [6].

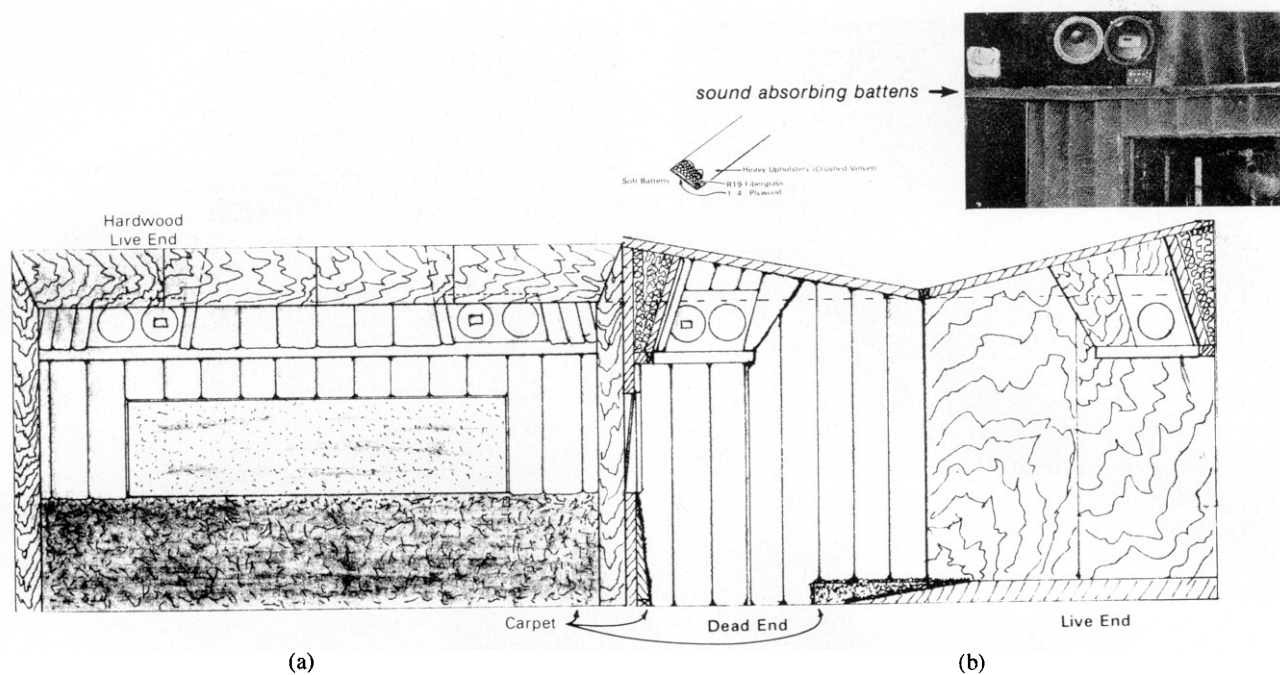
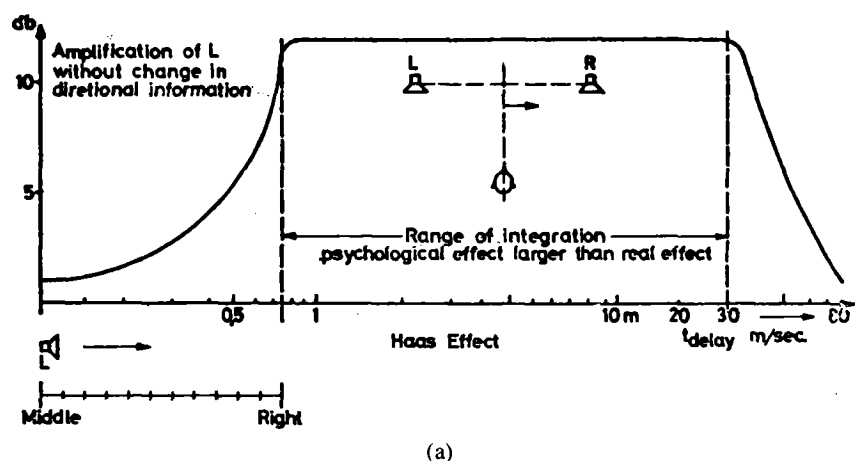
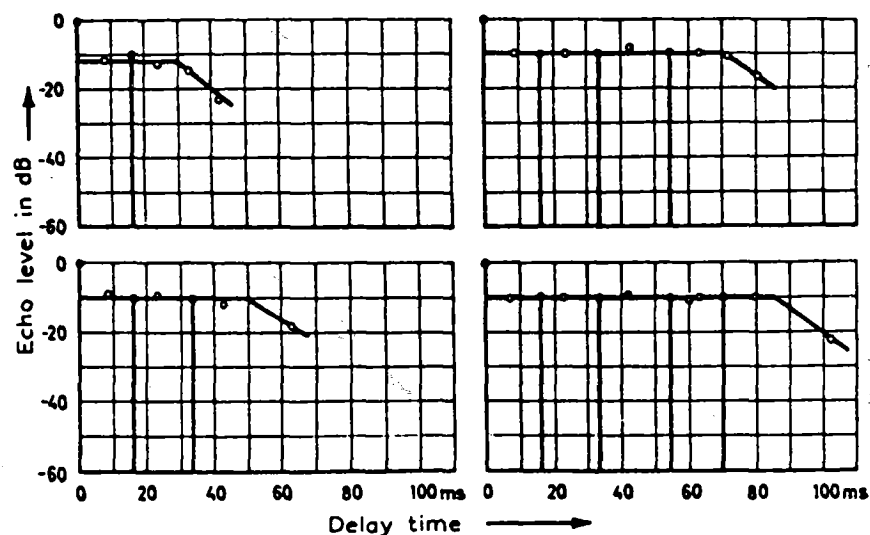


Fig. 5. Section views of an LEDE™ control room showing side walls and front wall [6]. (a) Front view. (b) Side view.



(a)



(b)

Fig. 6. (a) From [10]. (b) Absolute threshold of perceptibility of a delayed signal (reflection) being added to a sound field consisting of direct sound plus one, two, three or four reflections at fixed delay times and relative levels, which are denoted by vertical lines. The original sound signal is speech. All sound components arriving from the front [11].

excellent masking can occur out to 50 ms for second- and third-order reflections. Ando and Gottlob pointed out in a Letter to the Editor that "the delay time of the first reflection is not as important as the delay time of the strongest reflection" [13].

One of the primary functions of diffusion is to provide a vast number of narrowly spaced comb filters which combine to provide a high density of closely spaced reflections at the mixer's ears minus any major voids generated by isolated, very short path differences. The diffuse rear wall assists materially by ensuring that any inadvertent combinations with slight time differences are at a low level compared with the direct sound or the total sound from the walls.

Second, by significantly diminishing the energy of very early reflections and significantly enhancing diffuse energy from the rear portion of the room, we create an initial time delay (Beranek's definition in [14]) of 20 ms which now provides the acoustic conditions of a large space in a physically small room. The psychoacoustic effect of the LEDE™ design technique is to give the mixer's ears the acoustic clues of the larger space, thus allowing the perception of hearing the studio rather than the control room.

Note that the initial time-delay gap can be adjusted with ease over a relatively wide time interval, from about 5 to 6 ms to almost 40 ms, by the simple expedient of moving the console closer to or farther from a hard reflective diffuse rear wall.

Schroeder's work on quadratic-residue sequences, which manifested as surfaces containing wells of varying depth, promises remarkably broad-band control of diffusion [15].

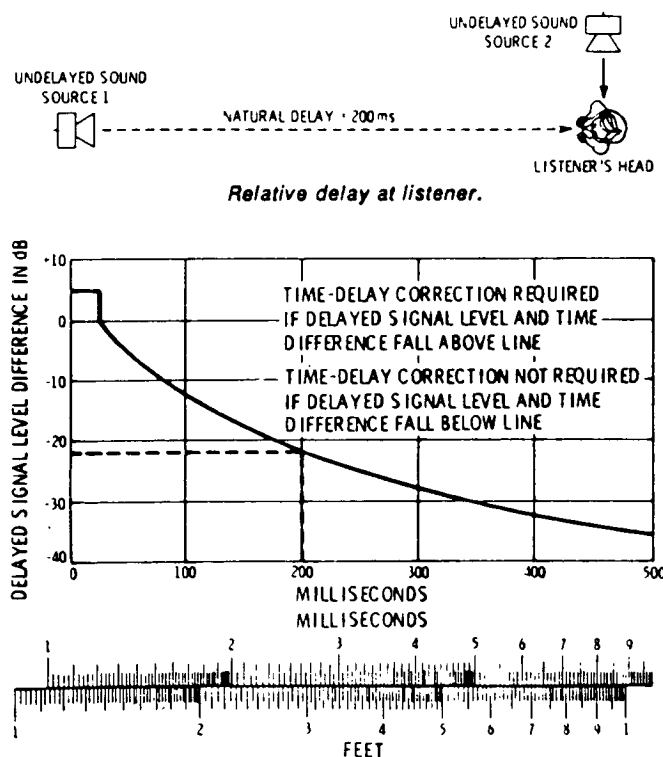


Fig. 7. Doak and Bolt delay versus level criteria. Note that in an LEDE™ control room the delay of the first significant reflection is typically at least 12 ms behind the direct sound and 20 dB or more below the direct sound level.

4 HOW "LIVE" A LIVE END?

First of all it is wise to observe that in many contemporary control rooms the level of the reverberant sound field is at or even below the ambient noise level (Fig. 8). This is especially true of control rooms using very absorptive rear walls and ceilings.

Fig. 9 illustrates the modifier of the critical distance introduced by such a technique in a contemporary control room. A room of this type will appear normally live if the mixer claps his hands but anechoically dead when the sound is emitted from the monitor loudspeaker [16].

Peutz, who developed the most widely used articulation equations, has shared a most useful equation with us [17]:³

$$\Delta dB = 0.221 \frac{\sqrt{V}}{h \cdot RT_{60}}$$

where ΔdB is the change in level, in decibel, for any doubling of distance beyond the calculated critical distance D_c ; V is the internal volume of the room in cubic foot (metric constant is 0.4); h is the ceiling height in feet, and RT_{60} is the time, in second, that it takes the reverberant sound level to decay 60 dB after the cessation of power input.

A useful conversion is

$$RT_{60} = 0.221 \frac{\sqrt{V}}{h \cdot \Delta dB}$$

We believe that ΔdB of 3–4 dB could be desirable in terms of creating a useful semireverberant environment in control rooms that reach sufficient volume.

The early form of Fitzroy's equation [18] is a useful

³ It is the authors' opinion that this paper is one of the most important papers on the subject of sound reinforcement ever presented to the Audio Engineering Society. It has never received publication in the *Journal*.

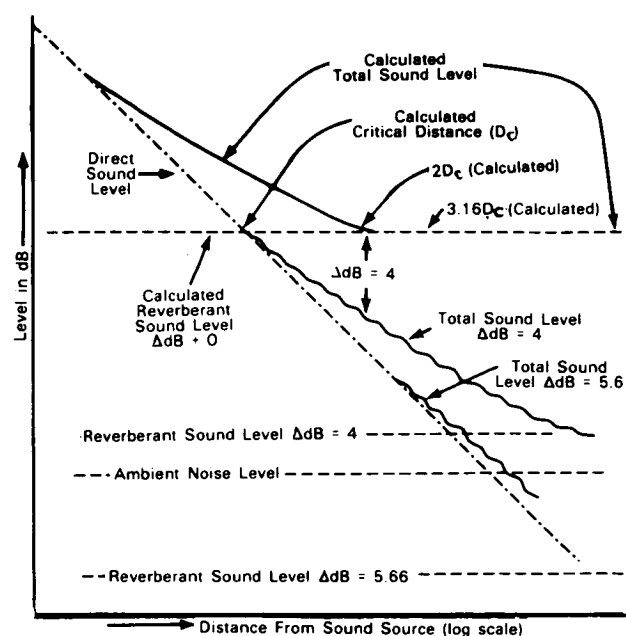


Fig. 8. Example of deviation from calculation of reverberant sound field.

approximation of energy decay, though all classical reverberation equations fall into the area so well described by Joyce's outstanding work [19], [20].

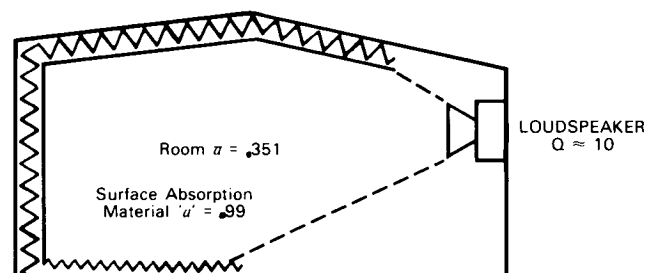
5 MEASUREMENT TECHNIQUES

We are currently measuring L_T with both the one-third-octave real-time analyzers by Crown and Ivie and the UREI level recorder utilizing warble tone with bandwidths as narrow as one-tenth octave proving useful. L_D is measured with the TDS analyzer. L_R is calculated from the power subtraction of L_D and L_N from L_T .

The properties of diffusion and the quality of the early sound field are measured using proprietary equipment designed by Heyser and utilizing a technique he has named energy time curve measurements [21]–[23]. This is a truly remarkable technique, which is considered an integral part of the TDS patent and displays on the screen of the fast Fourier transform spectrum analyzer the energy density (vertical scale) versus time (horizontal scale).

The energy density versus time technique provides for each sweep of the TDS analyzer a display on the fast Fourier transform analyzer which is the equivalent of 10 000 pulse photographs divided into 400 measurements of amplitude for each of the photographs.

The power of this instrumentation in the study of the early sound field is self-evident, and experience with it has offered some exceptional insights into the behavior of boundaries. The apparatus used is shown in Fig. 10. Fig. 11 shows the measurement via TDS of simple diffusers.



$$Mu = \left(\frac{1 - .351}{1 - .99} \right) = 64.9$$

Fig. 9. Effect of a dead rear wall [6]. If total $S\bar{a} = 440$, then loudspeaker will generate a reverberant sound field level comparable to that which would appear in a space having $440 \times 64.9 = 28\,556 S\bar{a}$.

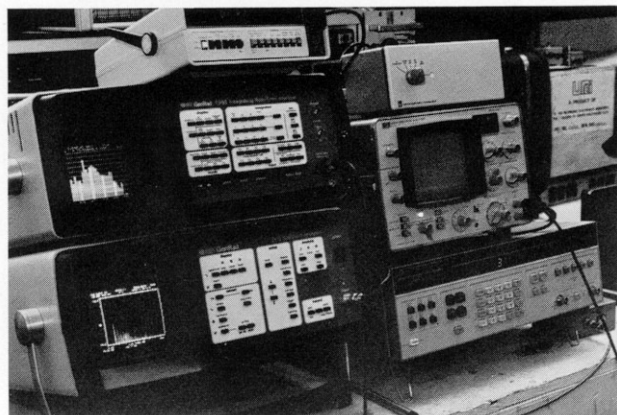


Fig. 10. Equipment used in ETC measurements.

6 THE ROLE OF THE INITIAL TIME DELAY

Beranek pointed out the importance of the initial time delay (ITD) to the perceived size of an acoustic enclosure in his monumental work [14]. In the LEDE™ approach the direct sound level L_D arrives at the mixer's ears unmarred by the control room surfaces because, at geometrical acoustic frequencies, it has passed through an essentially anechoic space. Thus the directional information coming over the reproducing channels is still intact.

We have the capability to adjust the ITD of the control room, as perceived by the mixer's ears, over a relatively wide range (1–2 ms to approximately 40 ms, depending on the size of the control room). Therefore it is now possible to take advantage of this parameter and make the ITD of the control room greater than the ITD of the studio. This allows the mixer to hear the ITD of the studio first (Fig. 12).

The ITD of the control room is now made not only longer but lower in level (through diffusion, not absorption). Therefore both the Haas effect and the Doak and Bolt criteria [30] ensure its inaudibility as a directional clue or as an ITD. The energy continues, however, its contribution to the overall sound field L_T as well as its tonal coloration (Figs. 13 and 14).

Snow's succinct remarks regarding such coloration is worthy of review by contemporary control room designers:

As long as there are reflections in the boundaries, this effect . . . will persist and there will be differences in response at various parts of the room. It should not be considered a fault — people like to listen to music in rooms, and this is a normal characteristic of that kind of listening . . . The proper procedure is to control the effect [24].

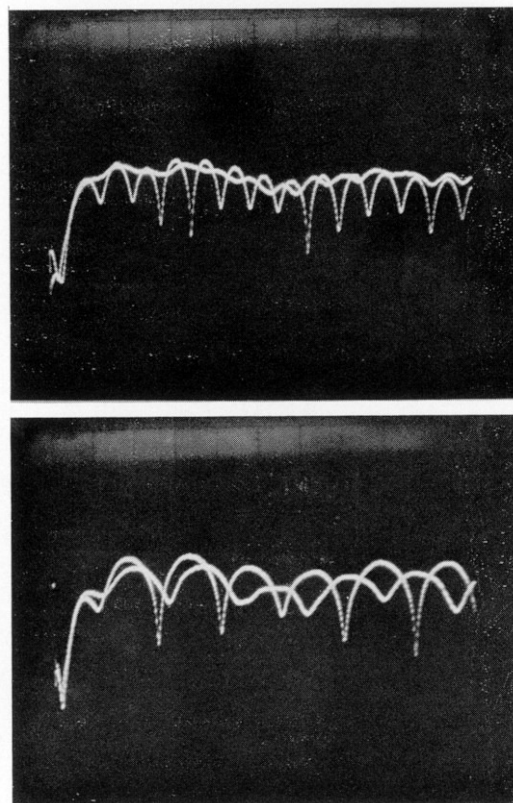


Fig. 11. TDS measurements of diffusion. Figs. 2 and 3 show how the comb filters were generated.

It is interesting that by the time one has established an ITD in the control room which is greater than the ITD in the studio, and has spaced the significant early reflections according to Kuttruff by means of a highly diffuse rear half of the control room, one finds that the contribution of this energy to the L_T is both useful and highly pleasing. The logic of the design lends a certain elegance to the progression of the development of the early sound field in the control room.

7 EARLY EARLY SOUND FIELD

Our energy time curve (ETC) measurements in control rooms have revealed signals that arrive ahead of the direct sound. While our first inclination was to dismiss these signals as equipment aberrations, subsequent investigation revealed that, on occasion, loudspeakers not thoroughly isolated from the structure they are mounted in can and do transmit sound via the structure to be reradiated from another structural surface (the ceiling, for example). Because of the higher velocity of sound through the structure, the signal actually arrives ahead of the direct sound through the air.

These signals are very low level; however, their uniqueness makes them unduly audible (much as preecho can be 60 dB down and very audible due to its unnaturalness —

who is used to hearing echos first in nature?). It is a possibility that some room's characteristic "sound" might spring from this effect.

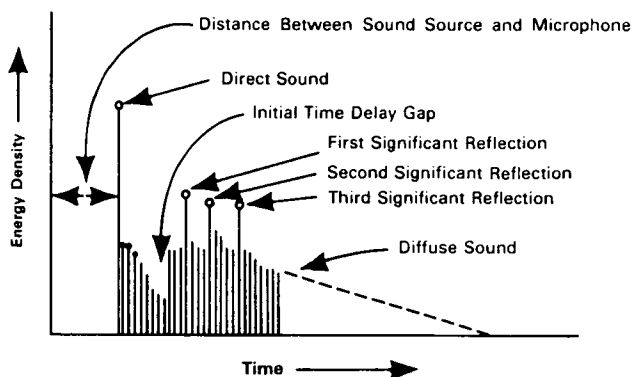
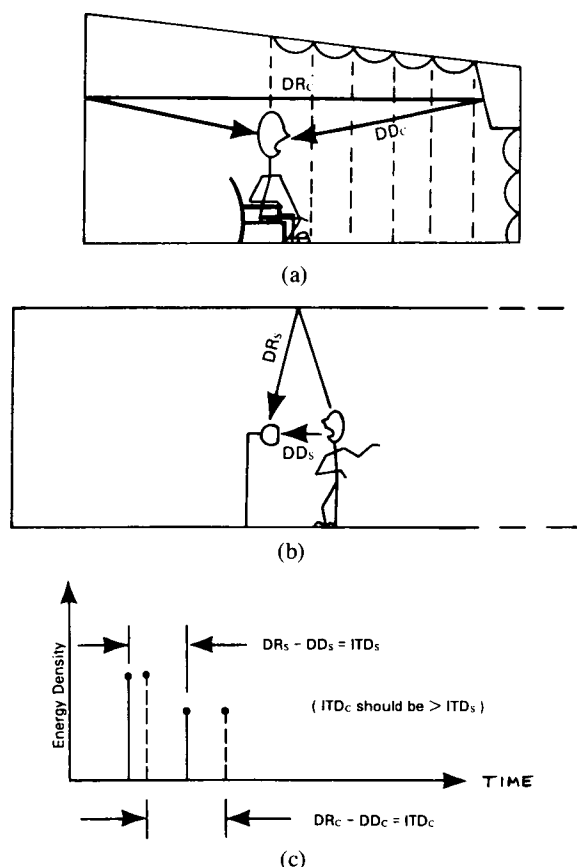
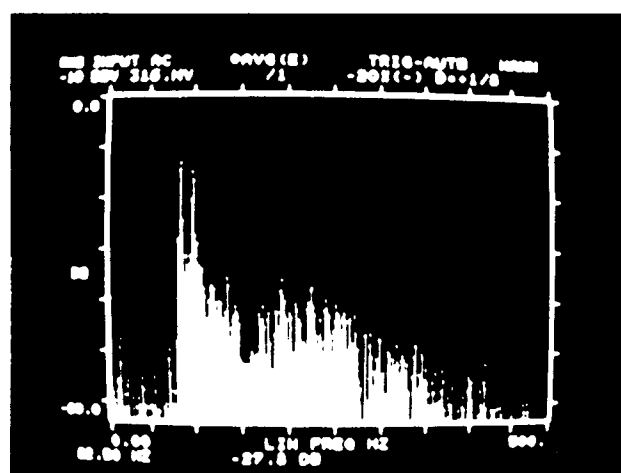


Fig. 13. Energy density versus time curve (ETC) for an LEDE™ control room.

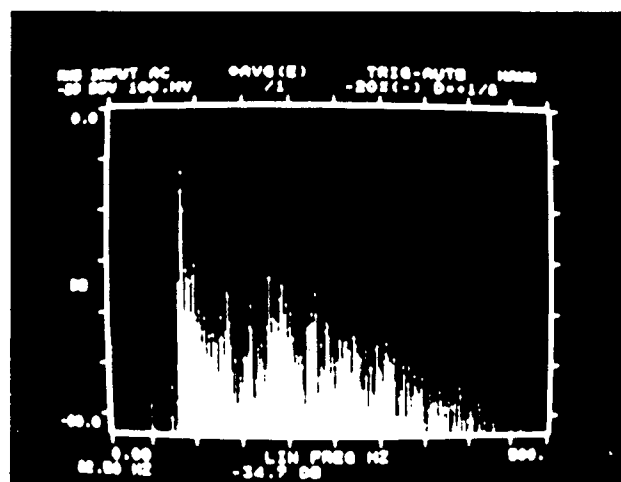


DR_s = Distance (or time) first reflection travels in studio
 DR_c = Distance (or time) first reflection travels in control room
 DD_s = Distance (or time) direct sound travels in studio
 DD_c = Distance (or time) direct sound travels in control room

Fig. 12. (a) ITD in an LEDE™ control room. (b) ITD of a studio. (c) Measurement of ITD of studios and control rooms.



(a)



(b)

Fig. 14. (a) Actual ETC measurement in an LEDE™ control room. Note that the second reflection is the recording console face. (b) Effect of an acoustic hood over the console.

8 LEDE™ AND THE LOW END

The birds and the bees of audio are clearly stated by Morse and Ingard [25] regarding when to use wave acoustic techniques versus geometric acoustic techniques. In its greatest simplification the symmetrical LEDE™ room should gradually begin to become transparent to frequencies below

$$\frac{3 (\text{velocity of sound})}{\text{room's smallest dimension}}$$

Since the low frequencies will ignore the type of materials used to create the LEDE™ effect, the outer shell of the room should then take over and be a solid, rigid, asymmetrical boundary. Helmholtz resonators (bass traps) may on occasion be necessary when the overall volume of the room is too small.

9 COMMENTS REGARDING CONTEMPORARY MEASUREMENT SYSTEMS

Point-wave duality is an intrinsic property of the Fourier transform map. What appears as a point in one description will show up as a wave in the alternative description. Therefore anything that happens in a restricted interval in one description will show up as broad wavelike smears in the other description.

We have all lived with the desire for a short time interval impulse that would allow a high-resolution spectrum to be observed, sometimes without realizing that it is not the apparatus that is at fault, but the physics. Even more devastating for the serious investigator is the fact that the Fourier transform only works for linear systems.

The signal used in TDS has a constant total energy density and a uniquely defined partition into potential and kinetic energy densities. Using the TDS wave analyzer as the "front end" of our measurement system and the fast Fourier transform as a predictable "storage bin" has resulted in measurements that are several orders of magnitude better in resolution than more orthodox contemporary fast Fourier transform impulse techniques [26].

10 THE TEF™ MEASUREMENT TECHNIQUE

The use of a TDS analyzer in conjunction with a fast Fourier transform as a demodulator and "storage bin" to obtain energy density versus frequency curves, energy density versus time curves, and frequency versus time curves have been packaged into what is called the time, energy, frequency (TEF™) measurement system, under license from the California Institute Research Foundation. Figs. 15 and 16 illustrate the concepts.

11 HARDWARE PITFALLS

We would be unfair if we did not caution control room designers with regard to the masking effects that mis-designed loudspeakers, microphones, and amplifiers exert on the ability to hear the benefits of LEDE™ acoustic control. Fig. 17 shows the direct sound level output from a widely used monitor loudspeaker. This coloration has, in

each case where we have encountered it, been blamed on the room prior to its exposure by TDS measurements [6]. Since Trott has shown that the "acoustic center" of a cone loudspeaker can vary up to 0.9 of its radius in back of its physically apparent center [27], in addition to the recognized phase difficulties associated with crossover networks, it is sobering to think of the consequences of not Time Aligning™ a critical-use monitor loudspeaker [28]. Pressure zone microphones, once heard, illustrate more clearly

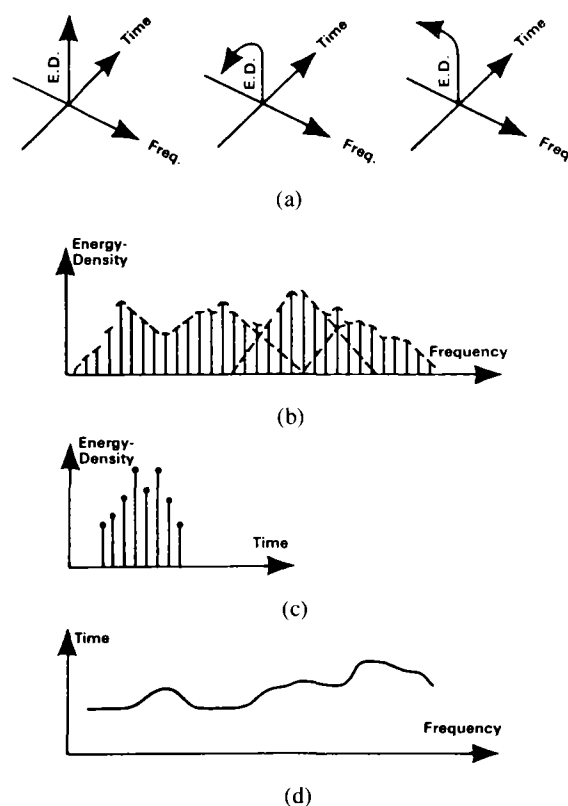


Fig. 15. (a) Energy density shifting in time and in frequency with change in level. (b) Display of energy density versus frequency curve (EFC). (c) Display of energy density versus time curve (ETC). (d) Display of frequency versus time curve (FTC).

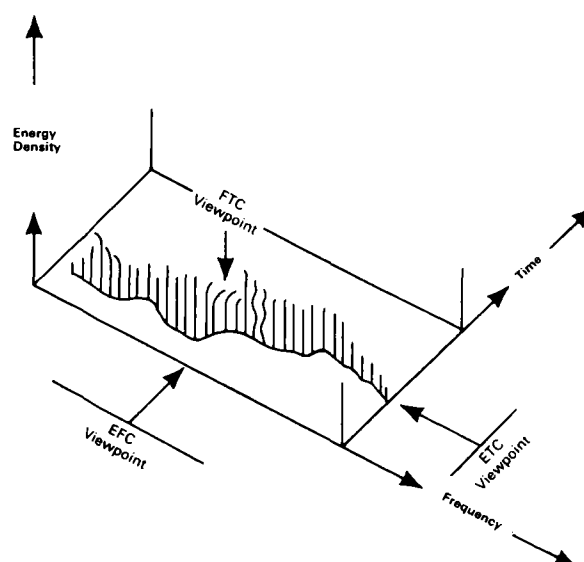


Fig. 16. Conceptual view of the three methods of measuring the frequency response in Fig. 15.

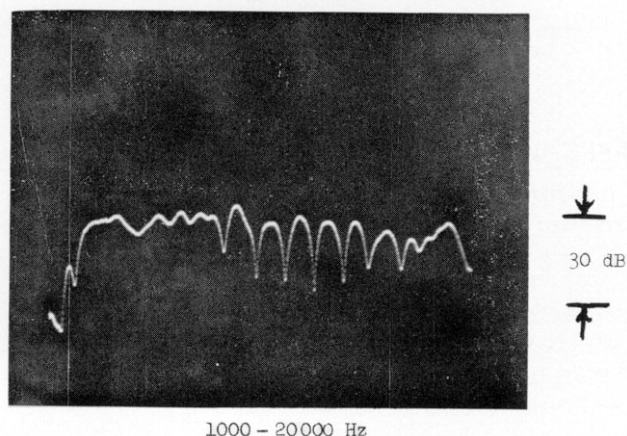


Fig. 17. A well-known monitor severely out of alignment.

than any description the gross coloration that uncontrolled response anomalies generate. Response anomalies developed between artist, musical instrument, and the acoustic environment they are in are all part of the "tonal color" of the performance in that space. Not part of the performance are response anomalies created by the presence of the microphone being near a reflective surface. The complex amplitude-phase transfer function of the performer-instrument-room arrives at the typical microphone setup only to be grossly reshaped by a series of room anomalies associated with microphone placement. (The role of multi-track recording techniques is to further destroy this information, but this subject is worthy of many separate discussions.)

Pressure zone microphones (PZM™) completely overcome this problem by placing an extremely small-pressure-calibrated high-quality microphone within 0.004–0.005-inch (0.102–0.127 mm) of a hard surface so that the microphone diaphragm and the hard surface form the top and bottom of a cavity whose side wall is the air itself [29].⁴

We have found startling differences between amplifiers but almost zero correlation between what we hear and the specifications used. Critical points seem to be their sensitivity to real-life loads and the regulation of their power suppliers. There is a preferred polarity between an amplifier and a loudspeaker. Reversing polarity to a loudspeaker (on a single-channel system) results in a discernible audible change, often great enough to exceed that caused by the difference between two amplifiers being compared.

12 CONCLUSIONS

In addition to the usual concerns for controlled ambient noise levels, isolation of both airborne and structure-born interferences, and other standard precautions relevant to recording facility construction, the following checklist of LEDE™ priorities can prove useful:

- 1) Determine the ITD for the studio with which the control is to be associated.
- 2) Choose a volume as large as possible for the outer

acoustic shell of the control room (low-frequency boundaries) and make it as asymmetrical as possible while maximizing mass, stiffness, etc.

3) Using the smallest dimension, calculate "crossover" frequency for gradually changing from a symmetrical geometrical acoustic interior shell to the asymmetrical wave acoustic outer boundary.

4) Design a control room ITD > studio ITD.

5) Arrange the rear wall, rear side walls, and rear ceiling for time-spaced diffused controlled-level early reflections that fall in the Haas effect zone.

Much nonsense has been written on occasion regarding the coherent addition of sound pressure at a point, as if it were a summation of four times the acoustic power. The coherent addition of two signals signifies an anechoic space with the measurement point precisely equidistant from the two sources. It is a meaningless measurement in a control room in terms of relevant criteria. The TEF™ measurement approach permits the objective measurement of meaningful parameters. Such measurements lead to better solutions to old problems. The LEDE™ control room is, we believe, such an example.

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⁴ Ed Long and Ron Wickersham developed the pressure recording process of which PZM™ is a licensed process.

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APPENDIX

LIVE END–DEAD END (LEDE™) CONTROL ROOM CRITERIA

The term LEDE™ is applied to a control room when the following criteria have been satisfied:

1) There is a low-frequency asymmetrical outer shell, free of pronounced resonances at low frequencies. This shell is large enough to allow the development of bass frequencies.

2) There is a symmetrical inner shell. The crossover frequency between the outer bass shell and the inner geometric frequency shell is

$$f_x = \left(\frac{3(\text{velocity of sound})}{\text{smallest room dimension}} \right)$$

3) There is an effectively anechoic path between the monitor loudspeakers and the mixer's ears which extends for at least 2–5 ms beyond the studio's initial time-delay gap.

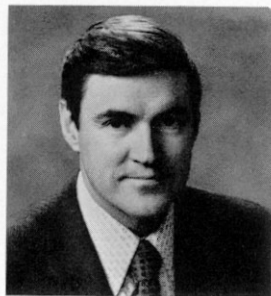
4) There is a highly diffused (at geometrical frequencies) sound field present during the initial onset of the so-called Haas effect.

5) The monitor loudspeakers, microphone technique, and mixing console do not "mask" the desired anechoic path from the monitors to the listener, including the period beyond the monitor to the ear's physical distance (studio ITD + 2–5 ms)

6) No early early sound (EES) is present. This is sound that arrives at the mixer's ears ahead of the direct sound traveling through the air. EES occurs when monitor loudspeakers are not shock mounted and therefore radiate through the structure and reradiate in the air, usually from the ceiling, near the listener.

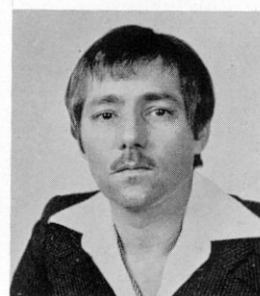
7) The hard-surfaced rear wall, rear side walls, and rear ceiling are so spaced temporally as to provide interwoven comb filter patterns which become a high-density early sound field without measurable anomalies.

THE AUTHORS



D. Davis

Don Davis is president of Synergetic Audio Concepts and is active in teaching seminars in acoustics and electroacoustics. He also is an international consultant on acoustics.



C. Davis

A former vice president of Altec and of Klipsch and Associates, Davis has authored three books, *Acoustical Tests and Measurements*, *How to Build Loudspeaker Enclosures*—written with the late Alexis Badmaieff (over

200,000 in print), and *Sound System Engineering*, coauthored with his wife, Carolyn. He is the author of over a hundred technical articles on audio subjects, and the inventor of the Acousta-Voice method of room-sound system equalization as well as the Live End–Dead End™ (LEDE™) control room design technique. Mr. Davis' firm, Synergetic Audio Concepts, is the appointed agent of the California Research Institute foundation for the licensing of time, energy frequency TEF™ measurements under the Heyser-Cal Tech patents.

The recipient of numerous awards in the field of acoustics and electroacoustics for his work in the design of sound systems and their acoustic environments, Mr. Davis has in the past three years concentrated on the interface of electroacoustic transducers to acoustic environments requiring nontraditional analysis techniques (IE control rooms). He is a fellow of the AES, a senior member of the IEEE, a mem-

ber of the SMPTE, and a member of the National Council of Acoustical Consultants.

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During his career in audio, Chips Davis has worked with such entertainers as Joey Heatherton, Andy Williams, Steve Lawrence and Eydie Gormée, Paul Anka and Wayne Newton. He is president of Las Vegas Recording and maintains a heavy work schedule as a studio designer, audio consultant, and instructor for special classes on Time-Energy-Frequency™ measurements. He graduated from Southern Nevada Institute of Technology with a degree in electronics and served in the U.S. Navy from 1962–1966 as ETR-2 radar and electronic countermeasures; microwave. The world's first LEDE™ control room was built by Chips for Las Vegas Recording and he has since designed a number of advance rooms employing the criteria outlined in this paper.