THE MEASUREMENT OF TIME/FREQUENCY RESPONSES IN SMALL ROOMS.

R. Walker Consultant.

1 INTRODUCTION.

In the relatively recent past, significant attention has been paid to the effects of early reflections in control rooms and listening rooms[1]. The need has arisen to measure a number of parameters of the acoustic signal in short intervals of time. Many years ago (1945-46), Shorter used time domain gating followed by conventional frequency response analysis to measure the short-term responses of loudspeakers[2]. More recently, but still some years ago (1978-79), the present author used a similar method to measure the time-frequency responses of rooms[3]. In both cases, the methods were clumsy, inflexible and extremely laborious to apply, to the extent that no great use was made of them in either case.

Modern methods are simple to apply, using commercially available equipment. Some proponents of such measurement methods have succeeded to the point where their methods are accepted essentially as *de facto* standards for such measurements[4,5].

The conventional way of describing the characteristics of acoustic or audio systems is as functions of frequency. However, all real signals, whether acoustic or otherwise, are implicitly functions of time – in the sense that the signal only exists at all as some measure of a physical attribute, which may or may not vary with time. The concept of frequency and the frequency domain is a mathematical abstraction, with no physical existence. It can be derived from other representations by appropriate mathematical operations. It is well known that the impulse response of a linear, time-invariant system theoretically contains all of the information necessary to specify the system response fully. However, the impulse response is a time domain function. In practice, it may or may not effectively be limited in the frequency domain, depending on the equipment used to measure it, but there is no information in the impulse response to indicate the frequency content directly.

For all of these reasons, it is important to understand the transformations from time domain to frequency domain, and in particular the inherent resolution limits of the joint frequency-time space.

2 THE FOURIER TRANSFORM.

All of the practically meaningful interpretations of time domain events in the frequency domain (and indeed, by implication, all frequency-domain aspects of analogue circuit theory) are based on the Fourier transform. That provides a means of translating between the time and frequency domains. The basis functions for the Fourier Transform are the sines and cosines.

In principle, any orthogonal set of basis functions could be used in the same way, and indeed are in other applications. It so happens that the sine and cosine eigenfunctions occur naturally as solutions for real-world systems, for example mass/spring or voltage/current resonant systems. This author has, in the past, also been involved with systems in which the natural decomposition was in the form of discrete dyadic functions (Walsh/Hadamard), which better suited that domain[6].

The theoretical basis of the Fourier Transform can be found in most text books on circuit theory or signal processing. It is not repeated here. In principle, a time record of infinite length is convolved with an infinity of in-phase and quadrature sinusoidal components to produce an infinite series of (complex) components. It can easily be shown that the original time signal and the summation of the sinusoidal components are identically equivalent representations[*].

In the limit, the signal must exist, and be available to the analysis, for all time (*–inf* to *+inf*). Conversely, an infinitesimal time event carries no frequency information at all. In the real world, such theoretically perfect signals cannot exist. Even a simple measurement of the frequency of a sinusoidal waveform from an oscillator would require an infinite time. In practice, it is clearly sufficient to limit the time domain record to some reasonable length, such that the effects of the truncation are acceptable, in the context of the measurement. (In the above example of the frequency measurement, even by simple cycle counting, the time-gate would have to be long enough, or the number of averages large enough, to produce the desired resolution.)

In effect, the product of time and frequency resolutions is a constant, approximately equal to unity (in seconds and Hz.). Intuitively, it is obvious that information must be available for a reasonable fraction of a full cycle before anything reliable can be inferred about the frequency or phase of a sine wave. Thus, to make statements about time intervals of the order of one or two milliseconds implies frequency resolutions of not better than 0.5 - 1kHz.

The usefulness of the Fourier transform for the description of system responses lies in the fact that the Fourier transform of the impulse response gives the frequency domain response called the Transfer Function – equivalent to the response of the system, in amplitude and phase, to excitation at each frequency separately by a sinusoidal waveform which has existed, and been applied to the system, for all time. That is what is commonly understood to be the steady-state frequency response.

3 DISCRETE FOURIER TRANSFORMS.

The theory outlined above applies to a time domain signal of infinite duration and which is continuous, that is it is defined at every instant. The resulting transform has an infinity of components with infinitesimal frequency domain spacing. None of those aspects is very practical.

To make practical use requires some simplifications and potential approximations. The time domain record length can be restricted if the signal is assumed to be repetitive. Then the resulting frequency domain components exist only at multiples of the cyclical frequency (i.e. at multiples of 1/*sample length*). They still extend over the whole, infinite frequency range but the higher ones, i.e. those above the required system frequency limit and the negative ones, may be ignored in practice. In a real, band-limited system they would usually be negligible anyway. Theoretically, some additional complications will arise if the time domain amplitude distribution is not continuous but is quantised, as in digital systems. For most practical purposes, that simply introduces additional noise and can be ignored if the resulting dynamic range of the measurement system is adequate for the purpose. Most modern digital measurement systems are equivalent to at least 12-

---

[*] One aspect which often causes confusion is the concept of 'transients' in audio signals. Most people who are familiar with audio recognise that an electronic circuit has to respond faithfully to the rapidly-changing time-waveforms of real audio signals. Most also understand the concept of an audio bandwidth. Relatively few recognise the inescapable link between them. Some years ago it became fashionable to consider a certain type of distortion in audio amplifiers which, in any reasonable amplifier, can only be caused by applying transient signals outside the working range of the amplifier and which can be avoided entirely by a suitable low-pass filter on the input. Another widely misunderstood example is the purpose and behaviour of loudspeaker crossover filters. In that case, the complicated time-function of the audio signal is split into different frequency bands - that is, operated on by a kind of real-time, analogue Fourier transform processor - to be recombined acoustically. The objective, in principle, is to reproduce the original time-domain waveform as an acoustic pressure signal. Even some audio professionals think that the low-frequency path (that is, the 'woofer') has to respond rapidly to transient signals and are puzzled as to how that can be achieved, given the obviously great mass and slow response time of the low-frequency drive unit.

bit resolution, corresponding to reasonably accurate representation of components down to about –60dB.

If the time axis is also quantised by some form of periodic sampling process, as it will be in any measurement system based on digital processing, then a further restriction applies to the data and its Fourier transform, which then becomes a Discrete Fourier Transform (DFT). According to the normal (Nyquist) sampling theory, the highest frequency component that can be represented correctly is equal to one half of the sampling frequency. If the continuous signal is not pre-filtered, higher frequency components will be aliased into the baseband and will not be distinguishable from the real baseband signals. That is not usually a problem because anti-alias filters are always included in any realistic measurement system (though computer audio cards may not always include the necessary pre-filtering, especially for reduced sampling frequencies). Minor difficulties may be encountered with the residual alias components in some cases, especially if the time signal contains constituents which would otherwise transform to high levels at frequencies outside the range of immediate interest. That is especially true in the region close to the Nyquist frequency.

Despite these limitations, it is generally accepted that sampled and quantised versions of real-world signals are reasonably accurate and useful representations of the originals, provided that the resolutions in both amplitude and time domains are sufficient for the purpose.

4 TRANSFORM WINDOWS.

A continuous, nonrepetitive signal (for example, the whole or part of an impulse response) can be treated as repetitive, and therefore practically amenable to Fourier transformation to the frequency domain, if part of it is selected and then assumed to be repetitive. In the frequency domain, the associated loss is that of frequency resolution. The transform will only contain information about frequencies harmonically related to 1/*sample length*. The manner of selection also has important consequences. The weighting function used to select the data in the time domain is usually called a 'window'.

The theory of data truncation and windows can be found in almost any textbook on signal processing. (Reference 7 is a particularly comprehensive review and analysis of 44 different windows.) In brief, the frequency domain response obtained from the transform includes convolution with the transform of the window. In general, the transform of the window function will have a main response of finite width and a sidelobe response perhaps extended to frequencies quite remote from the centre. As an example, Fig. 1 shows the Fourier transform of a rectangular window. It has a first sidelobe amplitude of -13.3dB and subsequent sidelobes which reduce at a rate of 20dB per decade. That response means that each individual frequency component in the output transform (sometimes called a 'bin') may contain quite large contributions from nearby frequencies and some contributions from quite remote frequencies, depending on the shape of the window in the frequency domain. That effect is commonly known as 'leakage'. A large number of windows have been developed over a period of many years to optimise the response for different purposes. Table 1 shows the main characteristics of some of them[7,8].

If a signal really does exist in time for only a short period and is isolated from other signals then it is a comparatively simple matter to decide which part to select. The window can be rectangular, with a finite value spanning the signal, zero-valued outside and infinitely sharp transitions between the two regions. Apart from the inevitable limitation of frequency resolution caused by the shorter time interval, nothing else is lost (or gained).

When the signal also exists outside the time interval of interest then the application of a rectangular window would create alias components because of the abrupt transitions at the edges of the window. The alias components would contaminate the wanted response because of the leakage corresponding to the sidelobe properties of the window. Thus, some form of tapered transition is necessary to reduce such effects – to make the input signal effectively zero at the ends of the window and thereby obtain a better compromise between frequency discrimination and spurious responses.

Many forms of gradual transition windows have been developed for different purposes. As shown in Table 1, those window functions have different properties, some producing high resolution at the expense of leakage and others giving high rejection of sidelobe response, at the expense of poorer frequency resolution.

5 MEASUREMENT OF ACOUSTICAL TIME-FREQUENCY RESPONSES IN ROOMS.

The human hearing system is a complicated signal processing system, especially in the context of the effects involved in a stereophonic audio illusion. The following is a very brief summary of a great deal of psychoacoustic research, quoted without references because of the very large (and sometimes contradictory) body of evidence. It is included only to illustrate the measurement system requirements.

In the interval after the arrival of the direct sound up to about 5ms, any delayed sound is integrated with the direct sound to form some impression of the sound 'quality'. The sense of direction is governed by the arrival of the direct sound.

From about 5ms to about 20ms information is extracted about the direction of a sound source. After about 20ms the extracted information is largely about the surrounding space rather than the source itself, until about 50–80ms when the sound becomes distinctly reverberant.

For reflections in a small room, the interval up to about 15–20ms is the most important. Although in the period up to 5ms reflections are not perceived directly, they can have an important influence on the sound quality. For example, a signal with a relative time delay of 1ms could produce strong cancellations, at odd multiples of 500Hz, by interference with the direct sound. In most cases of very short delays, the surface would not be large enough to cause a strong reflection at 500Hz, but it might be at 1500Hz and higher harmonics. Such cases are frequently encountered in studio control rooms, where the top surface of the mixing desk usually forms an efficient reflector. The (potential) reflection from the mixing desk is likely to arrive at the listener about 0.8 to 1.2ms after the direct sound from the loudspeakers.

Reflections later than 5ms can disturb the subjective stereophonic imaging process, causing mislocation of images. Individual early reflections from room surfaces are likely to occur at about 3ms (from the ceiling), 5–8ms (from the side walls) and 15–20ms (from the rear wall). Thus, for the measurement of early reflections in control rooms, it is desirable to be able to resolve time differences of the order of 1ms. It is also desirable to obtain as high a frequency resolution as possible, implying longer time records, in order to obtain some idea of the frequency characteristics of any reflections. These are clearly conflicting requirements.

Fortunately, the stereophonic illusion process involves mostly the higher frequencies. Although some sense of spaciousness is conveyed by frequencies between about 300 and 500–1000Hz, the main image-forming frequencies are those from about 500Hz upwards.

These factors lead to measurement processes based on time resolutions of about 1–2ms, resulting in frequency resolutions of the order of 0.5 to 1kHz. It is, as a result, conceptually possible to identify and measure reflections with time and frequency resolutions high enough to be useful for the investigation of stereophonic systems in relatively small rooms. (However, it is also likely that the actual performance of the human hearing system exceeds anything that can currently be implemented by instrumentation.)

There are two particularly useful measures of time domain responses – the so-called 'Energy-Time' response (ET) and the 3-dimensional Energy-Time-Frequency response (ETF). The first of these, the Energy-Time curve, is the square of the complex system impulse response. It is taken to represent 'instantaneous' energy. Although the precise, theoretical nature of the response has been the subject of some discussion[9], it does present a view of the time domain response in which representations of discrete reflections can be observed. The measurement may include some form of frequency-domain pre-filtering and weighting of the raw impulse response in order to highlight particular features. In some implementations, that frequency response processing is, at best, unclear. It can have a pronounced effect on the results obtained, and its investigation was one of the main objectives of the work described in this article.

The second useful measure of response is the 3-dimensional, 'waterfall' plot. For this, the time domain impulse response is translated to the frequency domain by discrete Fourier transform, with a relatively short time window. The location of the window and the transformed block in the time domain is progressively shifted to later times, to produce a series of frequency responses calculated for different time intervals. The frequency resolution is limited by the windowing process to some proportion of the inverse of the window length (depending on the type of window). It is also limited in time resolution by the length and shape of the time-domain window. Despite these limitations, a useful display indicating approximate times and frequency responses of reflections can be obtained. It can also clearly be seen that the time and frequency resolutions are defined by the window function in exactly the same predictable way as for any other Fourier transform process.

6 INSTRUMENTATION.

One convenient instrumental system that greatly simplifies the measurement and post-processing of impulse response functions is MLSSA (Maximum Length Sequence System Analyser)[4]. It comprises a source signal generator and a response recorder which works not on the impulse response directly but by using a comparatively long-duration, noise-like test signal. The actual impulse response is obtained by cross-correlation of the measured response with the original signal. That results in a significant degree of noise rejection and a more reliable measurement. The system also includes a wide range of post-processing functions.

Many other measurement methods are available. In some, the time and frequency weightings and resolutions that are effective for any particular condition are clearly evident. In others, the method of operation tends to obscure the effective settings. This article is not concerned with any particular, proprietary method of measurement. It does, however, rely on measurements carried out using a MLSSA system to illustrate general issues.

References 1 and 10 give some examples of a large number of measurements carried out on early reflections, in a number of implementations of the Controlled Image Design principle. In those real cases, significant irregularities in the frequency responses of most reflections gave rise to the apparent discrepancies between the ET and the EFT responses, which are the main subject of this article.

7 EXPERIMENTAL RESULTS.

In all of the following measured results, unless it is otherwise made clear, the response amplitudes are given in decibels relative to the level of the direct sound and the time delays are given relative to the arrival of the direct sound at the measurement microphone.

7.1. Overview and Earlier Observations.

In the initial stages of the work on early reflections in rooms[1] it was observed that the selection of filter, window, transform lengths, etc. had a pronounced influence on the apparent amplitudes of the results. For example, in some cases where the ET response apparently showed reflections of the order of –20dB or lower, the EFT response showed narrowband reflections higher than –10dB at some frequencies. Figs. 2 and 3 show an example from Reference 9. In that case, the reflection at about 7.5ms appears in the ET response to be at about –22dB, whereas the EFT response shows it to be at about –10dB at 3800Hz (the overall system gain was such that the direct sound measured approximately +3dB). The ET response was calculated with a Blackman-Harris window rejecting just the extreme high and low frequencies, leaving most of the middle frequency range evenly weighted. The EFT response was measured with a 4.26ms transform length, with a half-Hann window, giving an effective frequency resolution of about 300 – 400Hz.

Figs. 4 and 5 show an even more extreme example, taken from other work not related to the design of rooms with very low early reflection levels. In that case, a very severe and irregular reflection again gave rise to large differences between the ET and the EFT responses. The level for the reflection at about 3.6 ms in the ET response, Fig. 4, appears to -6.75 dB. In the EFT response, Fig. 5, the same reflection appears to be at -0.6dB at about 4400Hz. Actually, the direct sound level at the same frequency also measured -0.6dB so that the difference was actually 0dB! In both of those cases, the same impulse response data was used for the two types of analysis. Clearly, significant discrepancies can arise between results given by different analysis methods.

The reason for these differences lies in the difference in the effective bandwidths of the two measurement processes. In general, any measurement system will produce a result that is some kind of weighted average of all of the information falling within its scope. The wideband ET response produces a value for the reflection amplitude that is the square of the sound pressure, averaged over the whole effective frequency range. With an uneven frequency response, the amplitude of any particular, higher-level components will be reduced by the inclusion in the averaging of lower-level components. In contrast, for the relatively narrow-band ETF analysis the different amplitudes of the constituent components will be better represented. The same effect is also observed if the ET response is restricted to narrower frequency bands. Fig. 6 shows the same ET response as in Fig. 4, but with a half-octave wide filter centred on 4400 Hz. The apparent ET reflection amplitude was changed to 0.0 dB by the change to the narrower-band analysis.

All of these effects are, in principle, easily predictable. What is less immediately obvious, and may cause practical difficulties, is the more or less obscure way in which some of the available measurement systems operate and their effective time and frequency resolutions. In some cases, the measurement system responses are so heavily weighted to the higher frequencies that they are only effectively measuring the extreme upper end of the spectrum. That may even be outside the normal audio range.

7.2. Experimental Synthesis of Reflections.

In order to illustrate the effects of measurement bandwidth under controlled conditions, an experimental simulation was set up of a room with a single reflection. Fig. 7 shows the schematic arrangement. The system under test was a direct connection, with the addition of an electronically delayed signal. The delay was set to 5ms and a mixture consisting of the input test signal and the delayed signal at about -6dB was taken as the output signal. A bandpass filter, of nominally 4kHz to 6kHz, could be inserted into the delay path. The filter actually produced a passband gain of approximately +2dB, making the filtered delayed signal about -4dB relative to the direct signal. (Because of the simple passive mixer, there were also other signals corresponding to recursive passes through the system, at multiples of the delay length.)

The sampling frequency was 30 kHz and the overall measurement system bandwidth was 10 kHz.

7.2.1. Broadband delay.

Fig. 8 shows the impulse response of the test system without the filter in the delay path. The 5ms delayed response is clearly visible. The later responses are also visible, up to about the fourth order.

Fig. 9 shows the Fourier transforms of the main and the delayed signals. They were obtained using the maximum possible time window, just less than 5ms, with half-Hann weighting. The lowest reliable frequency and the frequency resolution are indicated by the bar at the bottom of the graph (just over 250Hz). The results have not been equalised for the inherent response of the measurement system so they include all of the measurement system effects. The frequency response of the delayed signal was reasonably, but not exactly, uniform. That was essentially the frequency response of the artificial reverberation generator used to produce the delay. The response irregularities at low frequencies were caused by a small dc offset in the response in combination with the windowing process (actually, a small dc recovery effect because of the limited or imperfect low frequency system response). The delayed signal response can be seen to be between 5 and 7 dB below the direct signal response.

Fig. 10 shows the ET response obtained using a Blackman-Harris window. The first peak, at 5ms, was measured to be very close to –6dB relative to the direct sound, as expected. Fig. 11 shows the ETF response, obtained with a 4.26ms half-Hann window. That produced an effective frequency resolution of about 400–500 Hz. The delayed signal was measured as -6dB relative to the direct signal.

In that case, the delayed signal had an essentially uniform spectrum. Differences in effective bandwidth or spectrum averaging in the measurement systems would not be expected to cause differences in the measured results.

7.2.2. Band-limited delay.

Fig. 12 shows the impulse response of the test system with the bandpass filter in the delay path. The 5ms delayed response is again clearly visible.

Fig. 13 shows the Fourier transforms of the main and the delayed signals, obtained as before. The delayed signal can be seen to be centered on 5kHz, with a relatively broad response, 3dB down at approximately 4kHz and 6kHz. The measured value of the delayed signal at 4904Hz was 4.48 dB below the direct signal - within 0.1 dB of the measured steady-state level difference. The response irregularities at low frequencies and the trend to a dc level of about -20dB were again caused by the low-frequency transient recovery effect.

Fig. 14 shows the ET response obtained using a Blackman-Harris window. The first peak, at 5ms, was apparantly at a level of –12.8dB relative to the direct sound. This was a measure of the average ‘energy’ levels of the two signals over the effective bandwidth of the measurement.

Fig. 15 shows the ET response obtained with pre-filtering of the impulse response, using a half-octave wide filter centred on 5kHz. In that case, the filter bandwidth, of nominally 4200Hz to 5900Hz, just encompassed the width of the delayed signal response in the frequency domain. The measured apparent level difference of 4.49dB corresponded closely with that obtained from Fig. 13. The significantly poorer time resolution of the narrower band filter is also evident.

Figs. 16 and 17 show intermediate conditions, for a one-octave and a two-octave filter respectively. The progressive change in the apparent ratio of the 5ms reflection to the direct sound is clear, with progression to -5.8 and -9.8dB respectively.

Fig. 18 shows the ETF response, obtained as before. It shows the delayed signal at -4.45dB relative to the direct signal at 4934 Hz, in close agreement with the true level. In that case, the effective ±3dB bandwidth was about 340Hz (= $\pm 1.44\Delta f$ for a 128 sample record at 30kHz sampling frequency). It was clearly narrow enough to represent just the passband component of the delayed signal at any one centre frequency. Clearly, there was no significantly inclusion of remote frequencies in the measurment process.

8 DISCUSSION OF RESULTS.

For any type of measurement it is self-evident that the final result will be some form of weighted average of all of the data which falls within the measurement scope, in time or in frequency or a complex combination of both. From the experimental model of a single room reflection, it has been shown that the measured results of ET and ETF responses were closely in accordance with the expectations, averaged over whatever frequency band was in effect at the time.

Thus, if a measured reflection amplitude is obtained from either ET or EFT responses, the effective ratio obtained for reflected to direct energy depends on the bandwidth of the measurement and on the signal frequency responses.

The nominally unfiltered (wideband) ET response will produce an overall value for the ratio of the two signals, averaged over essentially the whole frequency range. For a reflection (or a direct signal) which has pronounced frequency variations, the amplitudes of the highest components may be underestimated. Narrower frequency bands may be used by effectively applying a frequency domain bandpass filter before calculating the response.

The EFT response will always produce a relatively high frequency domain resolution (at least in comparison with the unfiltered ET response), giving the true response at each frequency, within the limitations of the Fourier and Nyquist theories.

It may be argued that the time resolution provided by either an ETF or a band-limited ET response is too poor to be of much practical use. However, theoretical considerations limit the obtainable resolutions to about the equivalent of 1ms/1kHz. That is inherent in the principle of time-frequency analysis. Any system which appears to offer much higher time resolution is inevitably restricted to high frequencies only, or at least to include high weighting factors for the higher frequencies.

In the examples shown, differences in measured levels of between 6 and 10 dB between wideband and narrowband analysis methods have been presented. Whilst those are large enough differences to be of significance, they are still less than can be observed using some types of instrumentation. In the examples given in this article, the overall system bandwidth was limited to 10 kHz. The differences between the analyses would have been greater if the overall system bandwidth had been greater. If the overall bandwidth had been, say, 30 kHz (which appears from published results to be a common setting for some swept sine-wave instrumentation) then the differences might have been 5 dB greater still, based just on the difference in effective bandwidths. Other factors, for example non-uniform frequency-domain weighting functions, may make the differences larger still.

9 CONCLUSIONS.

The fundamental resolution limits of time-frequency measurements have been described. It has been shown that the practical limits of time and frequency resolutions which can be obtained are adequate to quantify the early reflection patterns in small rooms. These limitations are close to, but probably just about adequate to describe those parameters important in the perception of the stereophonic illusion.

Experimental measurements, using a single electronically-simulated room reflection, have demonstrated that the results obtained correspond with the theoretical expectations.

It has also been shown that a reasonably accurate knowledge of the effective measurement system bandwidths and resolutions is essential before the results can be interpreted properly. With appropriate instrument settings, the results obtained for the responses of relatively short time events can accurately represent the actual physical conditions - within the inherent limitations of time and frequency resolution.

10 REFERENCES.

1. Walker, R., A new approach to the design of control room acoustics for stereophony. Paper G1-1,Preprint #3543, 94th AES Convention, Berlin, March 1993 (1993).
2. Shorter, D.E.L., Loudspeaker transient response. BBC Quarterly, I, 3, October 1946 (1946).
3. Walker, R., A preliminary investigation into the measurement of time and frequency response of listening rooms and control cubicles. BBC Research Department Report No. 1979/9 (1979).
4. D'Antonio, P. and Konnert, J., Complex time-response measurements using time-delay spectrometry. J.A.E.S., 37, 9, September 1989 (1989).

5. Rife, D.D., Transfer-function measurement with maximal-length sequences. J.A.E.S.,37, No. 6, June, 1989 (1989).
6. Walker, R., Hadamard transformation: A real-time transformer for broadcast standard television. BBC Research Department Report 1974/7 (1974).
7. Harris, F.J., On the use of windows for harmonic analysis with the discrete Fourier Transform. Proc. I.E.E.E., 66, No. 1., January 1978, pp. 51-84 (1978).
8. Gade, S and Herlufsen, H. Use of weighting functions in DFT/FFT analysis (Part 1). Bruel and Kjaer Technical Review, No. 3, 1987 (1987).
9. Vanderkooy, J. and Lipshitz, S.P. Uses and abuses of the energy-time curve. J.A.E.S., 38, 11, November 1990, p. 819 (1990).
10. Walker, R. “Early reflections in studio control rooms: the results from the first controlled Image Design installations”. 96th AES Convention, Amsterdam, 1994, Paper P12-6 (preprint no. 3853) (1994).

11 ACKNOWLEDGEMENTS.

This paper presents work originally carried out at the Research and Development Department of the British Broadcasting Corporation and is published with their authority.

12 APPENDIX - FREQUENCY DOMAIN PARAMETERS OF SOME WINDOW TYPES[6].

Window type	**Noise Band-width**	**3dB Band-width**	**Ripple dB**	**Highest sidelobe dB**	**Sidelobe falloff dB/dec**	**60dB Band width**
Rectangular	Δf	$0.89\Delta f$	3.92	–13.3	–20	$665\ \Delta f$
Hann	$1.5\Delta f$	$1.44\Delta f$	1.42	–31.5	–60	$13.3\Delta f$
Kaiser-Bessel	$1.8\Delta f$	$1.71\Delta f$	1.02	–66.6	–20	$6.1.\Delta f$
Flat-top	$3.77\Delta f$	$3.72\Delta f$	0.01	–93.6	0	$9.1\ \Delta f$
Blackman-Harris[7]	$2.00\Delta f$	$1.90\Delta f$	0.83	–92.0	–6	$\approx 6\Delta f$

where Δf is the frequency domain line spacing (= 1/*period*)

Definitions:

Rectangular: $w(t) = 1$

Hann: $w(t) = 1 - \cos 2\pi t/T$

Kaiser-Bessel $w(t) = 1 - 1.24 \cos 2\pi t/T + 0.244 \cos 4\pi t/T - 0.00305 \cos 6\pi t/T$

Flat-top
$$w(t) = 1 - 1.93 \cos 2\pi t/T + 1.29 \cos 4\pi t/T$$
$$- 0.388 \cos 6\pi t/T + 0.0322 \cos 8\pi t/T$$

Blackman-Harris[7]
$$w(t) = 0.35875 - 0.48829 \cos 2\pi t/T$$
$$0.14128 \cos 4\pi t/T - 0.01168 \cos 6\pi t/T$$

The functions are defined on the interval $0 <= t < T$, and $w(t) = 0$ elsewhere.

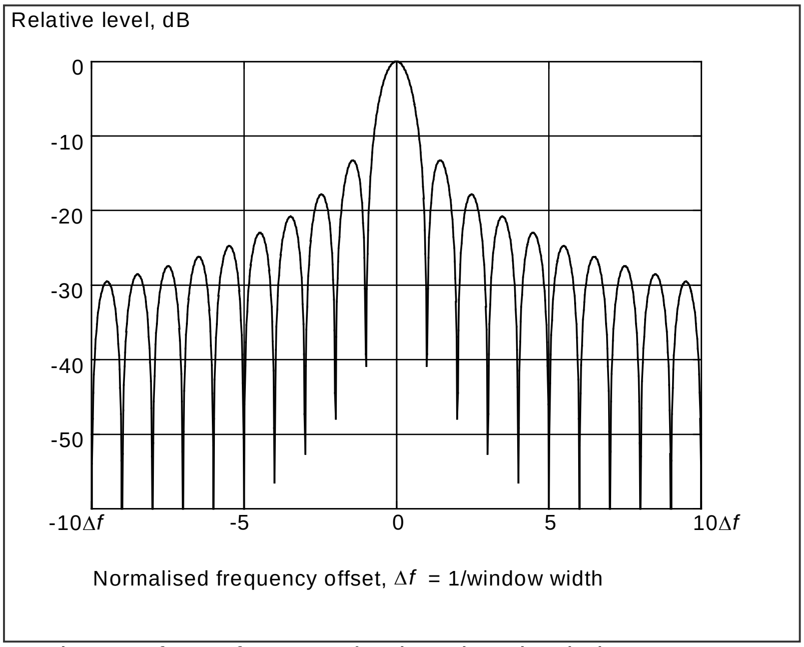

Fig. 1 Fourier transform of rectangular time-domain window.

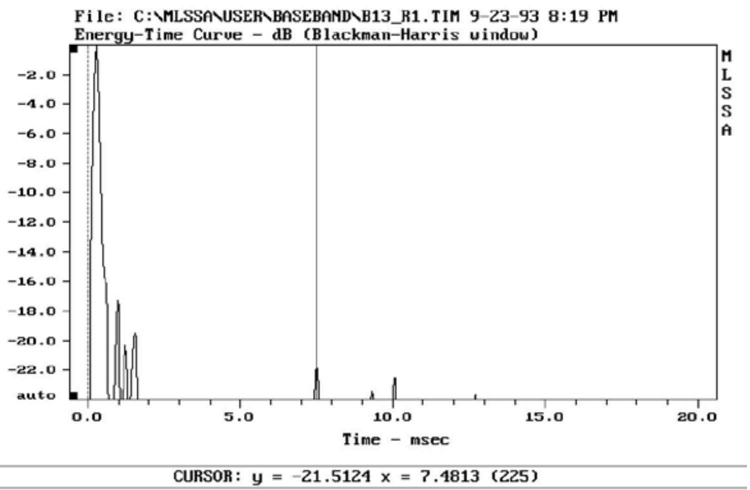

Fig. 2 ET response for controlled reflection room, showing reflection amplitude of <-20dB at ≈7.4ms delay

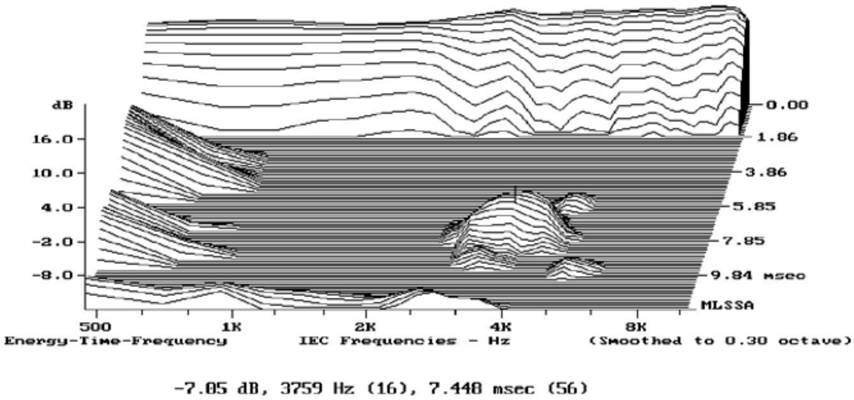

Fig. 3 EFT response for controlled reflection room, showing reflection amplitude ≈-10 dB at 7.4 ms delay and 3800 Hz.

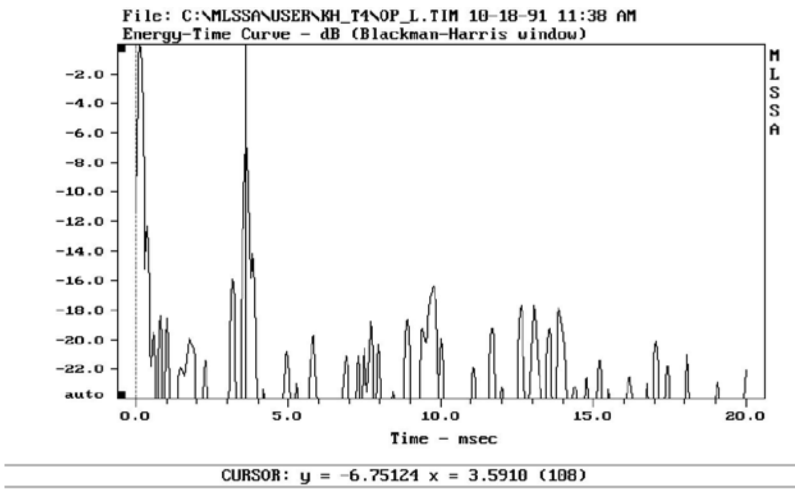

Fig. 4 ET response for conventional control room, showing -6.75dB reflection amplitude at 3.6ms delay.

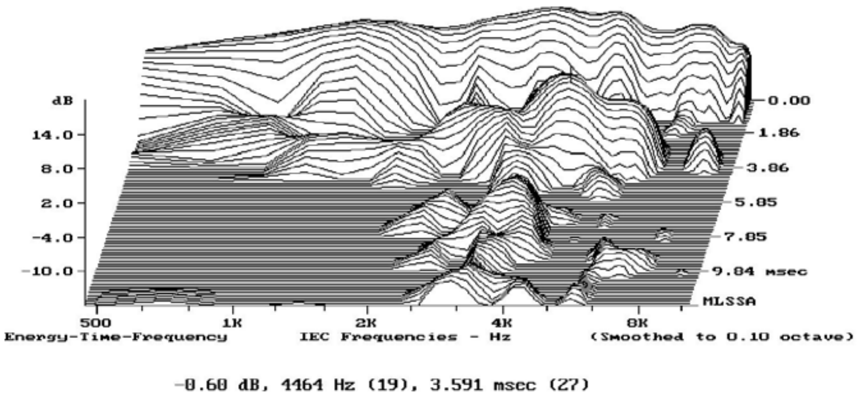

Fig. 5 EFT response for conventional control room, showing -0.6dB reflection amplitude at 3.6ms delay and 4500 Hz.

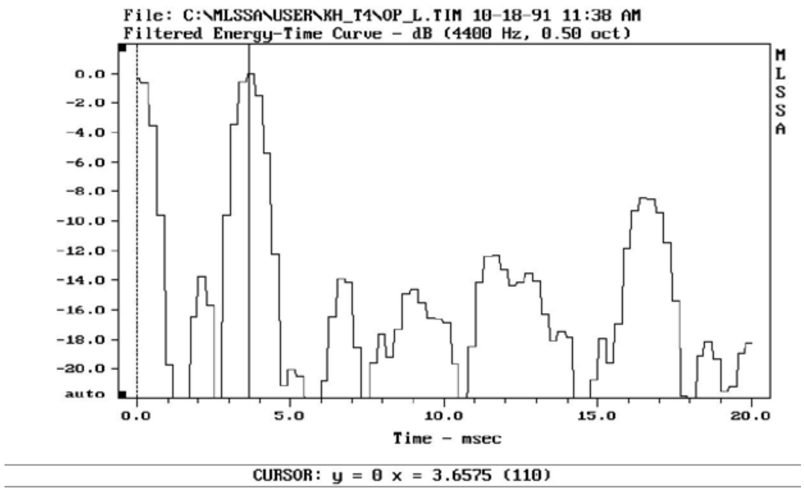

Fig. 6 Filtered ET response for conventional control room with 0.5 octave filter, showing -0.0dB reflection amplitude at 3.6ms delay.

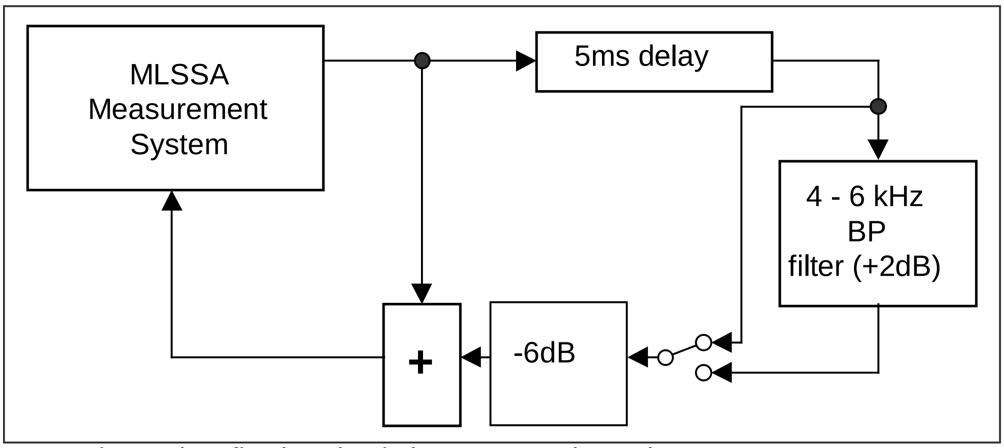

Fig. 7. Electronic reflection simulation system schematic.

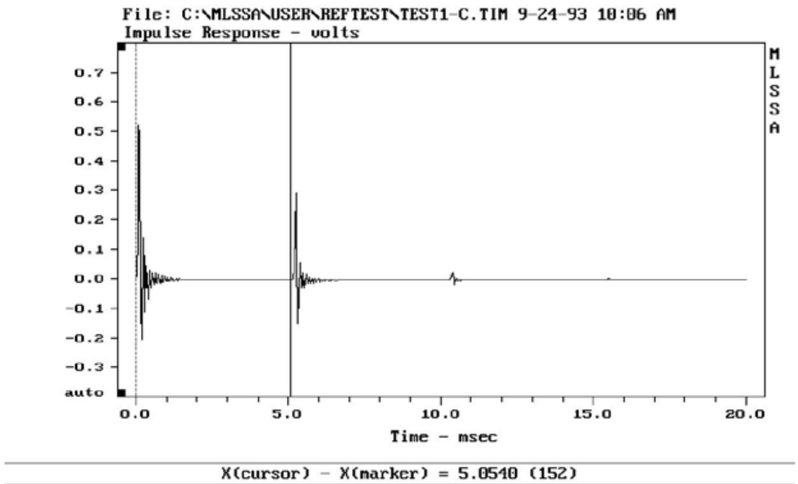

Fig. 8 Impulse response of unfiltered electronic reflection simulation system.

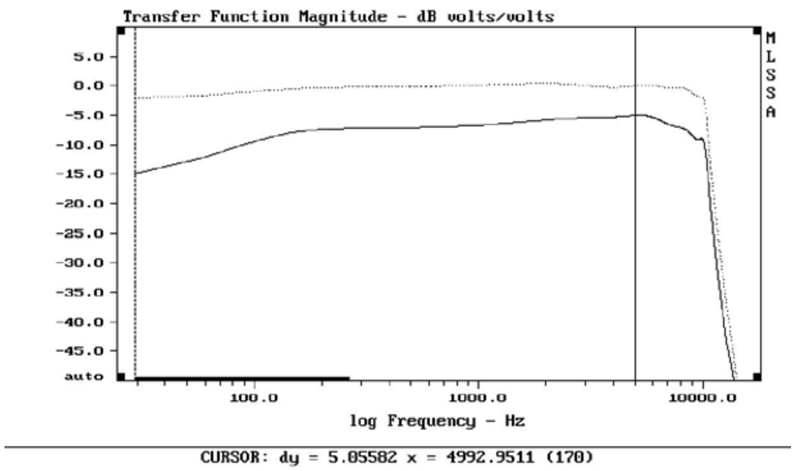

Fig. 9 Fourier transforms of direct and delayed signals for unfiltered electronic reflection simulation system, 4.8ms half-Hann window.

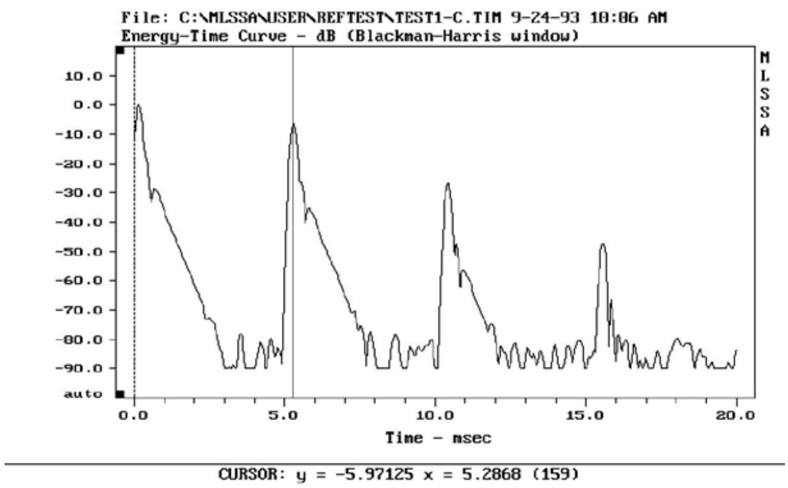

Fig. 10 Wideband ET response of unfiltered electronic reflection simulation system, showing -6.0dB reflection at 5.3ms.

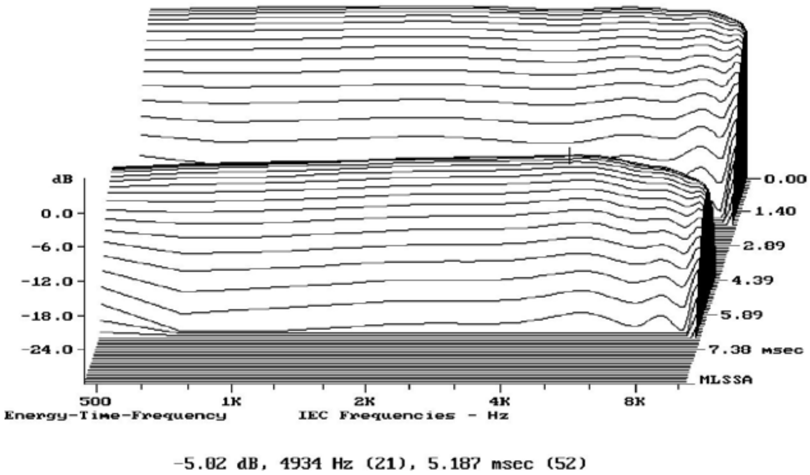

Fig. 11 EFT response of unfiltered electronic reflection simulation system, showing -5 to -7 dB reflection at 5.0ms.

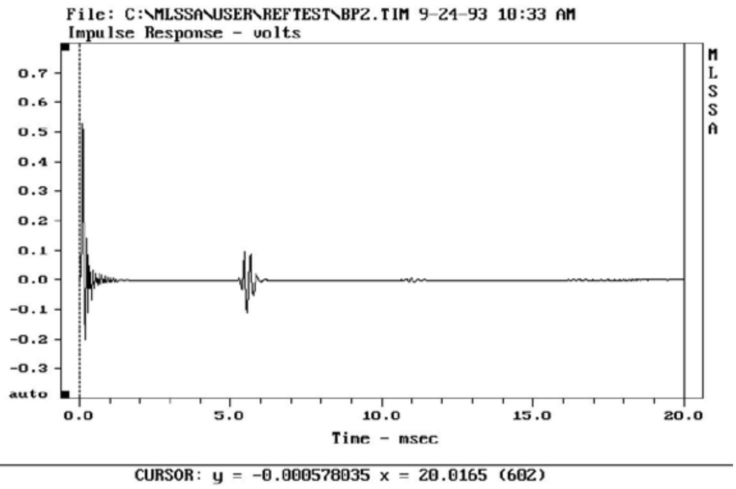

Fig. 12 Impulse response of bandpass filtered electronic reflection simulation system.

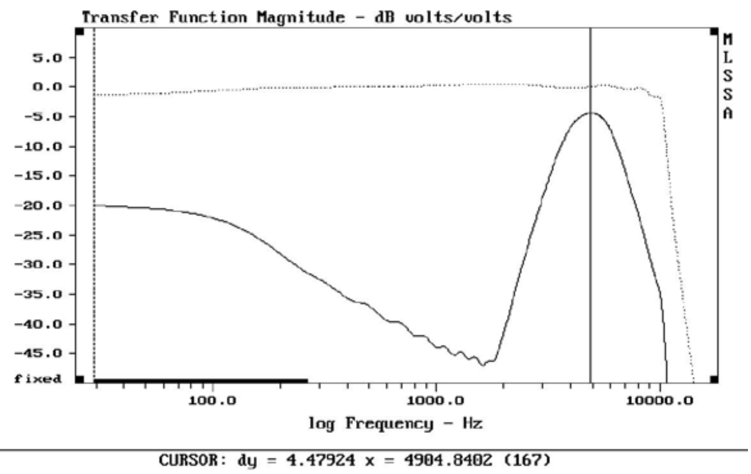

Fig. 13 Fourier transforms of direct and delayed signals for bandpass filtered electronic reflection simulation system, 4.8ms half-Hann window, showing -4.5dB response at 4904Hz.

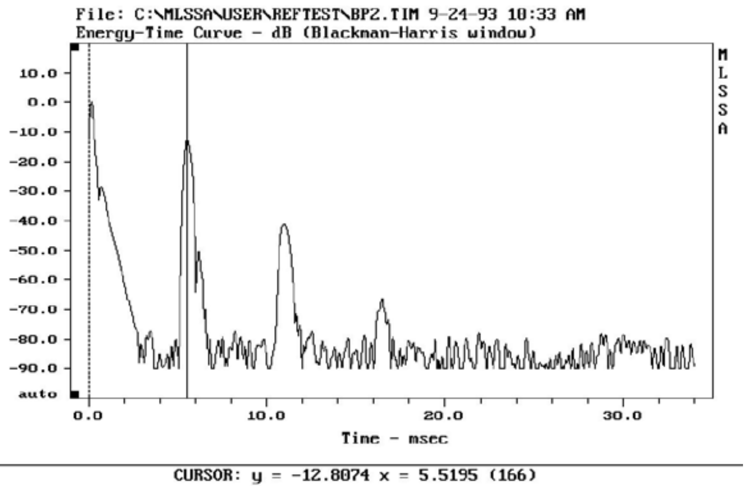

Fig. 14 Wideband ET response of bandpass filtered electronic reflection simulation system, showing -12.8dB reflection at 5.5ms.

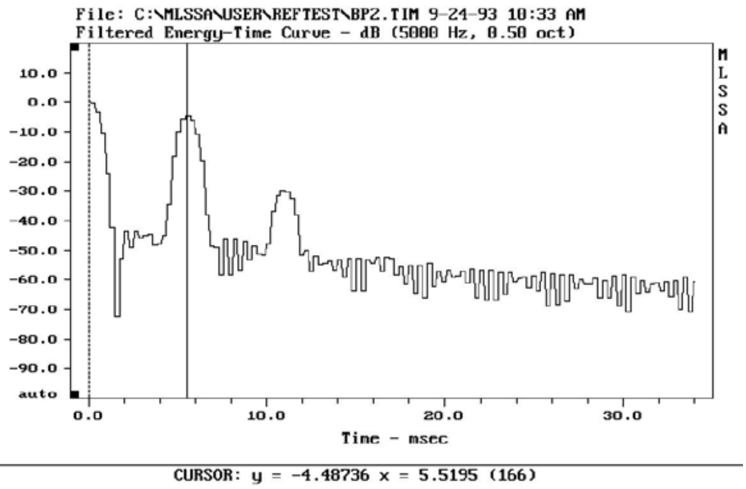

Fig. 15 ½-octave ET response of bandpass filtered electronic reflection simulation system, showing -4.5dB reflection at 5.5ms

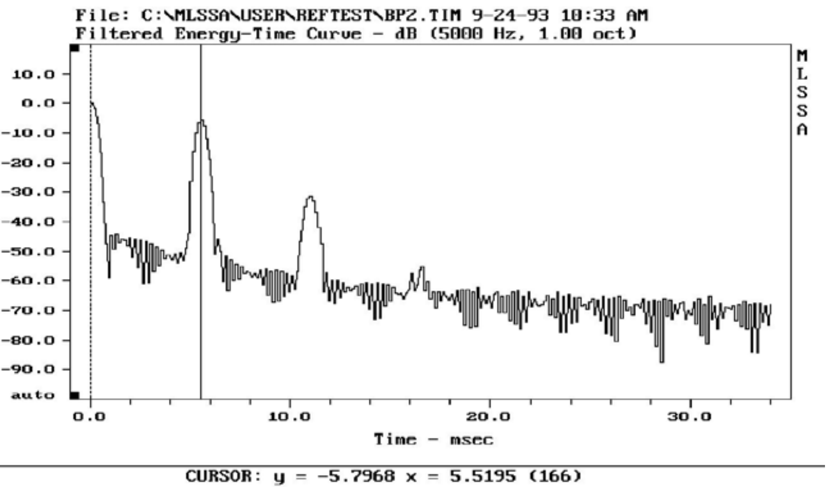

Fig. 16 1-octave ET response of bandpass filtered electronic reflection simulation system, showing -5.8dB reflection at 5.5ms

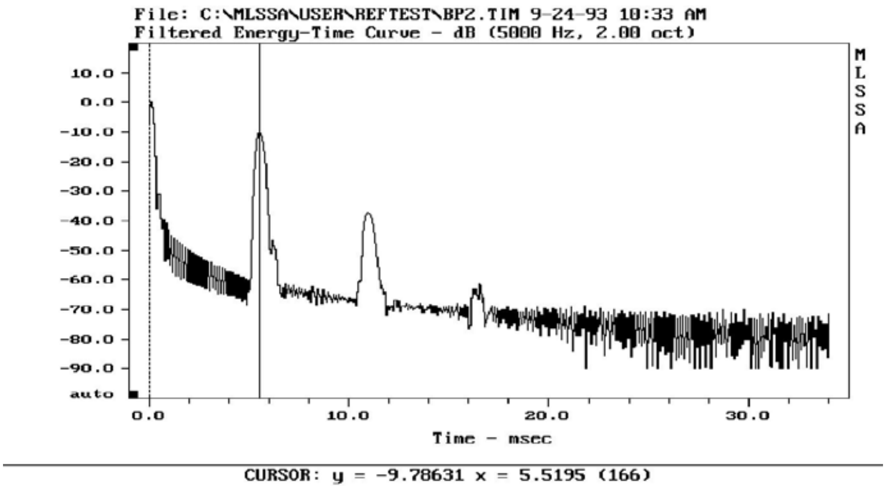

Fig. 17 2 -octave ET response of bandpass filtered electronic reflection simulation system, showing -9.8dB reflection at 5.5ms

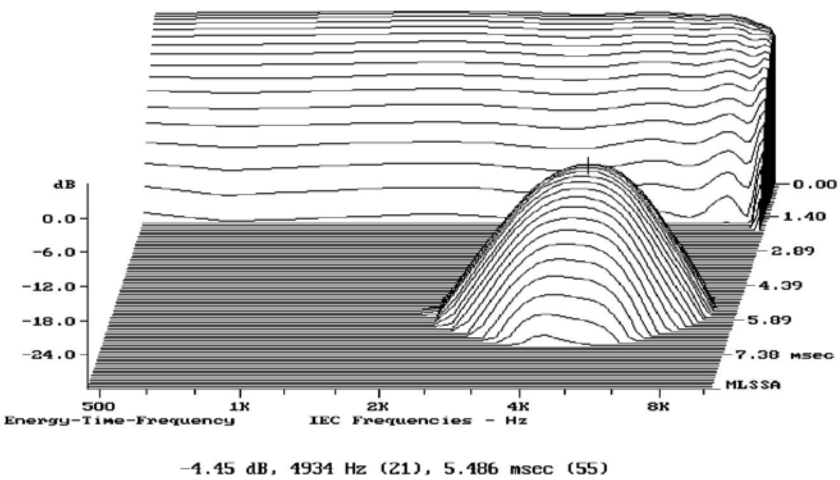

Fig. 18 EFT response of bandpass filtered electronic reflection simulation system, showing -4.5dB reflection at 5.5ms.