COMPARATIVE ACOUSTIC MEASUREMENTS. THEIR APPLICATION IN THE MEASUREMENT OF A CONTROL ROOM DESIGNED FOR GRANADA TELEVISION.

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Introduction

There has been a great deal of controversy in the last five years between the protagonists of T.E.F measurement and followers of more familiar techniques such as Impulse and gated noise. This argument has mainly raged in the various acoustic journals published in the U.S.A. In particular Don Davis (1) went so far as to discount FFT measurement altogether as a valid means of obtaining meaningful date in anything other than a noiseless, linear, time invariant system.
His arguement relied on the fact that to measure with a frequency bandwidth of 20KHz, a 1024 line analyser would measure an input frame length of:

$$Tw = \frac{\text{FFT size}}{2.56 * F} \quad \text{therefore } Tw = \frac{1024}{2.56 * 20,000} = 0.02 \text{ s}$$

If the test signal were an impulse then for a bandwidth of 20KHz

Impulse t = 1/ 20,000 = 0.00005 s = 50 us.

This would mean that the analyser input frame is set for 20ms with a frequency resolution of 1/.02 or 50Hz to measure an impulse of 50 us.
The result is that the analyser sees useful signal of necessarily restricted amplitude for only 0.25% of the time it is receiving data. In normal acoustic environments the problem of signal to noise ratio seems insurmountable.

It is important to note the general constraint of Tr * Fr = 1.

This makes meaningful measurements below 50Hz extremely suspect as far as control rooms are concerned. The claim of one designer to create monitoring environments flat to 5Hz begs the question of measuring the room decay time with a resolution of 200ms.

T.E.F

The Heyser process of time delay spectrometry (2) is now well enough categorised to make further description of the basic principle superfluous. There is however a basic uncertainty in the transfer function created by the bandwidth and phase of the tracking filter employed. This does not invalidate the method but is merely one example of mathematics being unable totally to represent the analytical signal.
In general the Energy Time Curve does not exhibit causality

without some mathematical processing. (3)
The use of windowing functions enables the known, useful factors of the ETC. to be displayed at the expense of time resolution.
The mathematical analysis of real data may yield unreal results, events that precede causes. Data interpretation through judicious windowing techniques can produce greatly improved resolution of the wanted signal data.
In the TEF analyser the ETC. is produced independently of the EFC due to the essentially different configuration of the hardware.
Unfortunately this makes it impossible to derive the transfer function from the impulse response in order to test its validity as a transform. However comparative measurement do yield acceptable correlation.
As the ETC is the magnitude of the complex (quadrature) sum of the impulse response and its Hilbert transform, the real and imaginary parts may be analysed separately to cross-correlate the component forms.
The TEF has several distinct advantages together with corresponding limitations.
One is the inability of the system to post process data using digital filter techniques. The other is the change of time window with ETC bandwidth. This makes life extremely difficult when attempting to overlay impulse responses of different bandwidth.
I have already mentioned the inability to transform directly from the time domain to the frequency domain.
In many ways these limitations are merely inconvenient. It is necessary for example to perform at least eight ETC measurements of octave bandwidth to obtain full reverberation data. Of these the 50Hz bandwidth display is of a window 400/50 = 8 s
With a 30 dB display of a small control room the base display occupies only 0.1s of screen or 1.25%.
In the frequency domain a minimum of three, decade sweeps are required to display the amplitude response with appropriate frequency resolution.
The time offset in the TDS measurement is also and therefore post processing of the phase response cannot be done.
To summarise TDS has huge potetial but when implemented in the TEF analyser the contraint of predetermining the data required can be both time consuming and frustrating.

MLSSA

There is another approach to two port measurement which combines the inherent advantages of TDS with the fundamentality of an impulsive measurement which captures the entire transfer function without the previously discussed problem of noise and dynamic range which inevitably leads to non linearities and invalidation of data.
In a general discussion of cross-correlation methods (4) J. Vanderkooy has stated that TDS as a method may be extended and enhanced in its application by the use of digital filtering with minimal spectral bias.
Indeed the entire process of practical system measurement has advanced more with the advent of fast, inexpensive data

processing than any radical change in thinking with regard to theory.
In order to optimise the cross correlation of data in a two port measuring system it is necessary to generate a signal source which is at all times both synchronous and repeatable. This may be achieved by using a binary Maximum Length Sequence or MLS.
MLS. analysis was proposed by Manfred Schroeder (5) and developed as a method by Boorish and Angell (6), culminating in the release in 1988 of MLSSA by Doug Rife of DRA Laboratories.
As the excitation signal is deturministic the measured impulse response as applied to known systems is practically perfect. In actual application however there may be some uncertainty due to noise. Averaging will eliminate this factor in all but the most erratically noisy environment.
Because the measurement exists in discrete time, the temporal resolution is limited by the sample period. Consequently there is always a small uncertainty in the Group Delay. There also may be uncertainty due to the limitation of the sequence length which defines the impulse measurement length. However, as long as the system measurement is band limited to the Nyquist frequency, the intermediate values are available by interpolation.
This condition is set automatically for the MLSSA algorithm using the onboard, programmable, digital filter.
The time domain window may be extended to 32768 points which is half of the 2^16 binary limitation of the present system running in an IBM PC and MSDOS computer environment.
This gives potentially cross-corrolated frequency resolution for a measurement of 10KHz bandwidth of:

$$\frac{1}{MLS} * 2.56 * F = \quad 0.78\text{Hz}$$

To obtain this degree of resolution in the frequency domain requires an FFT of comparable size to the impulse length. MLSSA will compute FFTs up to 32768 points which with a sampling rate of 33Hz would give an Fr. of 1Hz.
It should be noted that Fr is always limited to 1/Tr where Tr is defined as the length of the impulse to which the FFT is applied.

Step Functions

The use of a programmable step function source is extremely useful in comparative measurement when noise is not a problem. It has yielded excellent resolution at lower frequencies which tend to be ignored in impulse response displays due to the linearity of the original signal. To have both sources available within the same portable computer has made measurements possible which previously posed considerable transportation problem.
The 6dB / Octave transfer of the step function gives an accurate indication of the energy transfer characteristic in terms of subjective audibility.

Measurements

The original purpose of my project with Granada TV was to design and build a control room with complete freedom as to acoustic treatment but within the confines of an existing control room.
This room also happened to be of the same basic dimensions and structure as another room which had been previously measured.

The measurements of two existing control rooms provided sufficiently different results to know that there would be a suitable comparison available at the end of the project.
As over 100 measurements were taken it is necessary for the purpose of this paper to summarise the results:

Studio 12
A production control room of almost identical arrangement to the room to be redesigned, this room had been acoustically treated in the conventional manner using mainly porous wall panels, ceiling tiles and a carpeted floor.

Audio Dubbing
A purpose built room for audio post production which was generally regarded as acceptable in terms of accuracy. The layout of the room was more conventional than Studio 12 in that the mixing axis was parallel to the longer walls but was not on the centre line.

Studio 8
A production control room which was refurbished according to a design brief, given by Granada TV, to create an optimum stereo monitoring environment within a typical small control room, given all the constraints that that implies.

The process of designing for stereo listening requires an understanding of those factors which most influence the perception of the image. These factors must not override the more fundamental requirements of a listening room. It is also important to recognise the difference between monitoring requirements and those of more general sound reproduction.
The general design for Studio 8 is shown in Fig.1 and follows in principle a geometric model which minimises reflective energy arrving at the mixing position up to 10 ms after the direct sound from the monitors. In orde to achieve this aim it was necessary to mount the loudspeakers in an boundary plane so that the low frequency energy is radiated in phase with the more directional mid and high frequency energy. Under these conditions two salient factors emerge:

1) The LF response of the speaker must be optimise for a radiation plane which may be less than 2pi in small rooms.
The Q of the speaker will depend on the number of drivers, their diameter and the boundary loading. Fig.2 shows the effect on the quasi free field response of a Rogers monitor in Studio 8. A 6dB increase in SPL was accurately anticipated at the design stage

and a contingency was allowed for the installation of a optimised monitor system with appropriate port tuning.

2) Under such loading conditions the monitor system must be placed symmetrically about the room axis and the radiation geometry should be identical for both speakers. This will create Phase coherence for all frequencies and corresponding stability of the stereo image.

There is one subject upon which there is broad agreement and that is the decay time of small rooms and the fact that it should be short (in the order of 0.2s to 0.3s). Unfortunately there are not enough reflections in a small dead room to create a statistically diffuse sound field and so rooms with similar RT60 values may exhibit quite different subjective characteristics. This situation may be improved by analysing the sound decay with greater regard to the ETC.
The effect of energy arriving in the first 5ms (taking the direct sound arrival at the mix position as 0ms) is to define the stereo perspective and the fundamental timbre of the system. Early reflections cause deep comb filters of linear bandwidth and therefore their effect is more destructive at low frequency. In the region between 50 and 100 Hz destructive reflections are created by the room boundary conditions and therefore well distributed damping is required. This will also improve the modal control of the room which will describe the decay time.

Final Tuning

After the initial reconstruction of Studio 8 a series of listening tests were carried out using the EBU test CD and other familiar material.
It was generally agreed that the room was too bright and that the monitor system exhibited excessive LF output. The stereo imaging and balance was judged to be excellent.
The brightness was found to be caused by resonance of the diffusion system employed in the ceiling, side and rear wall treatment and was eliminated by the use of neoprene strips paced behind the open slots of the diffusing panels. A total area of 1.5 sq.m effected a solution!
The ETC measurement for the room did not show a great change as result of this modification as the resonance was of relatively narrow band width.

Postscript

The project started as a design exercise with related measurement but in the final analysis became a fundamental question of the validity of measurement itself.
The arrival of MLSSA added a further dimension to the whole exercise which has yet to be fully appraised.
Several facts emerge which are worthy of mention:

1) The early soundfield is absolutely dominant in the perception of stereo material and to this end both impulse response and step

functions should be used to analyse the initial response. Figs 3-5 show the effect of bandpass filtering on delay in the system.

2) The use of T60 as a final judgement is ineffective but T60 must be within the prescribed limits for more subtle analysis to be of any value. The modification to the diffusion of Studio 8 did produce a reduction of 20% in the T60 at 500Hz (octave band) which was clearly audible but the original measurement had not caused particular concern.

3) The general requirement for television sound production monitoring is control rooms with extremely low coloration and the minimum of early reflective energy. This will create a time window in which the sound produced in the studio can be appraised without interference.
High resolution measurement is an essential tool in the development of such rooms but special regard and a healthy level of intuition is required to make the correct judgement as to which dimension contains the key to any particular door.

Diagrams

Fig 1. Control room studio 8. Design scheme as agreed.
2. LF loading of Rogers LS and electronic correction.
3. M3 monitor output as bandsplit overlay of step function
4. Midrange response of driver with overlay of response through crossover set at 600 & 5KHz.
5. Transfer function of (4).
6. ETC of studio 8 indicating reflected energy V time.
7. TEF waterfall of studio 8. Comb filter produced by early reflection from low ceiling. The geometry of the design could not be changed.

References

1) D. Davis Syn Aud Con Vol 12, no.2 1985
2) R. Heyser Time Delay Spectrometry J.A.E.S. Oct.1967
3) A. Duncan The Analytical Impulse J.A.E.S. May.1988
4) J. Vanderkooy Another Approach to TDS J.A.E.S. July.1986
5) M.Schroeder Integrated Impulse Method J.ASM Aug.1979
6) Borish & Angell An efficient algorithm for the Impulse Response using Pseudorandom Noise. J.A.E.S July.1983

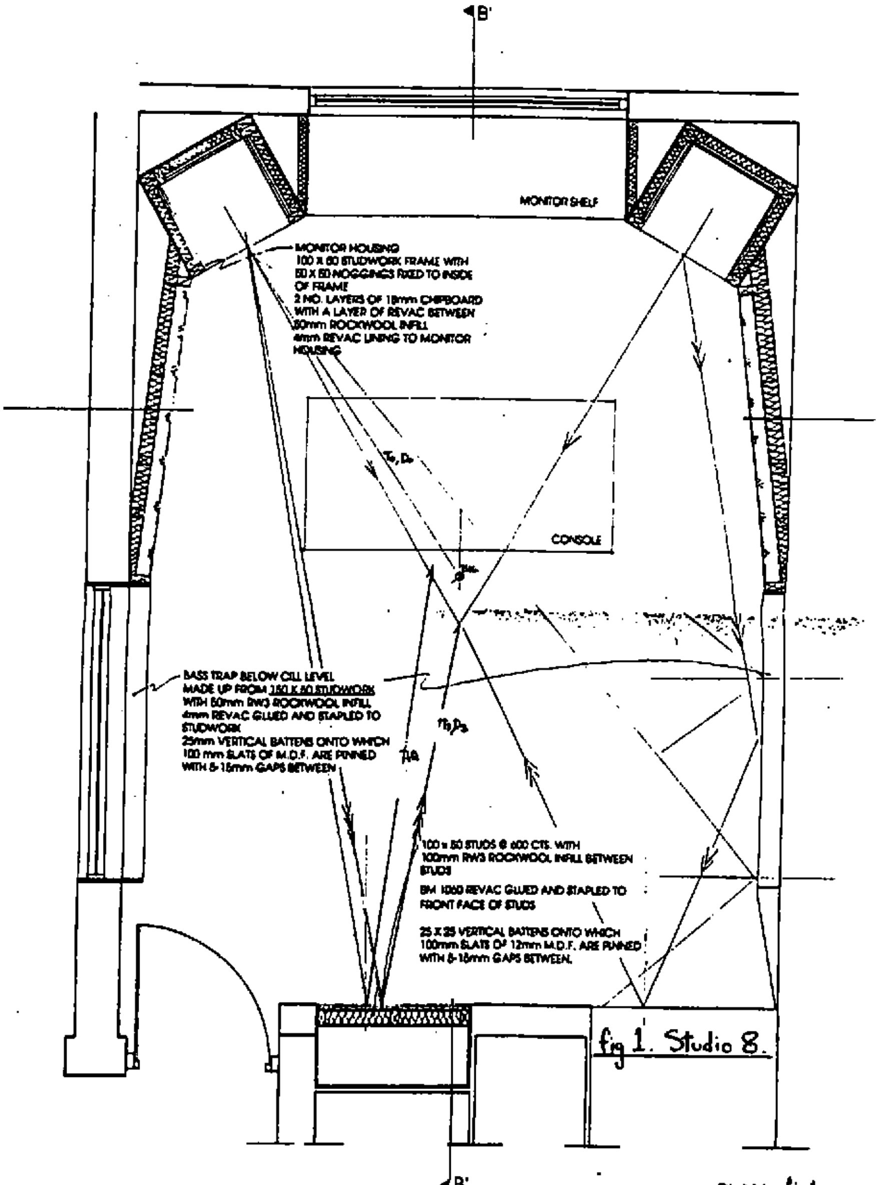


fig 1. Studio 8.

PLAN fig 1
GRANADA TV
STUDIO 8
A S MUNRO 1985

Fig 2 . LF LOADING EFFECT.

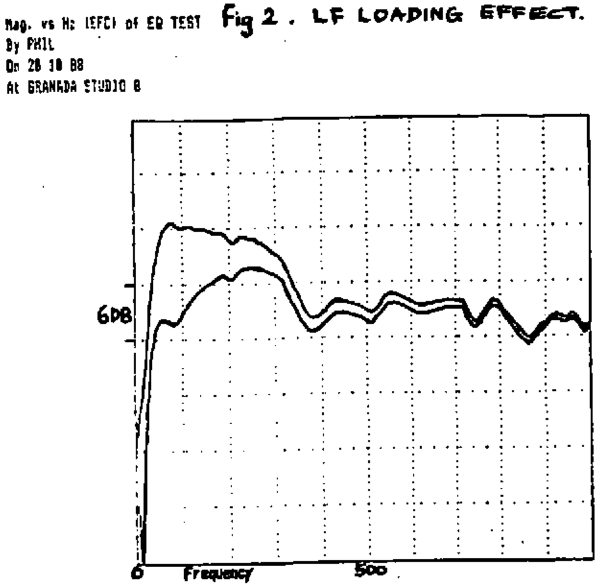
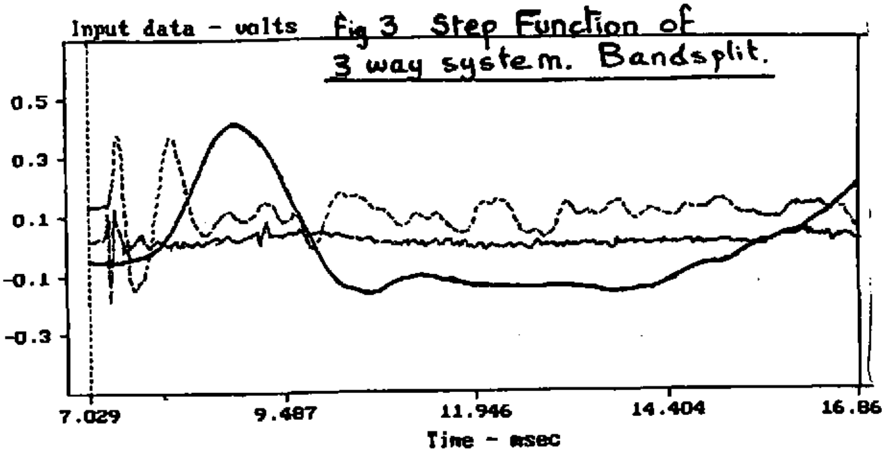
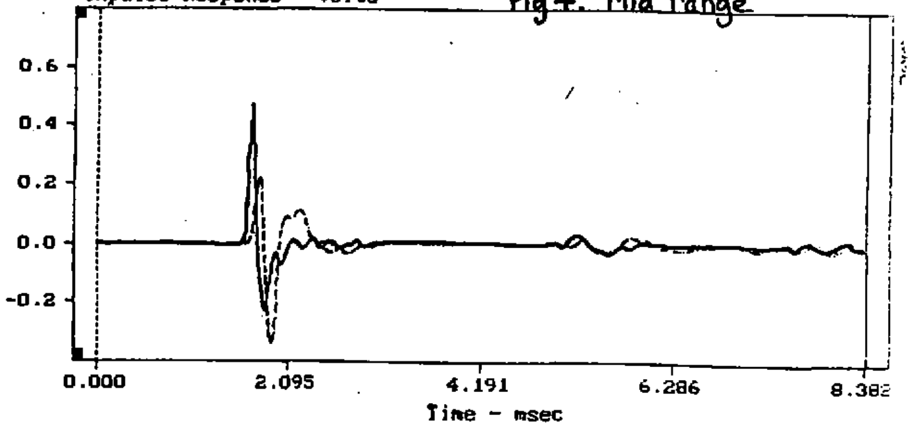


Fig 3 Step Function of 3 way system. Bandsplit.



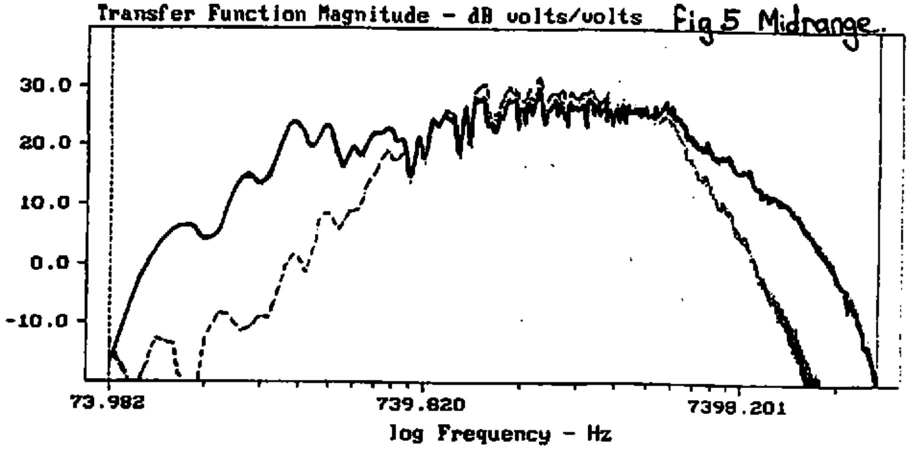
File: D76A.TIM 10-20-89 11:59 AM
Impulse Response - volts
Fig 4. Mid range



D76, driven direct (solid), thru crossover (dot)

10-20-89 1:45 PM MLSSA: Time Doma

File: D76A.FRQ 10-20-89 11:59 AM
Transfer Function Magnitude - dB volts/volts
Fig 5 Midrange.



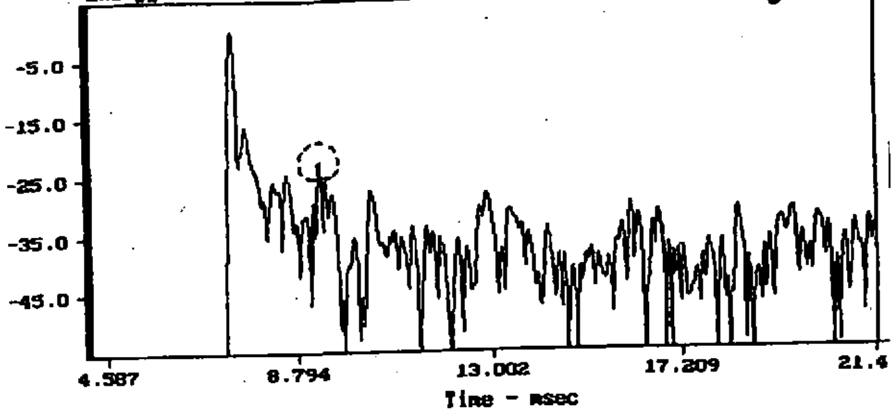
CURSOR: dy = 24.9961 x = 20004.7347 (1352)

D76, driven direct (solid), thru crossover (dot)

10-20-89 1:50 PM MLSSA: Frequency Doma

File: GRANBR.TIM 8-9-89 3:19 PM
Energy-Time Curve - dB (Blackman-Harris smoothing)

Fig 6.



3-D of MODIFIED CONTROL ROOM
By PM
On 12/11
At GRANADA STUDIO B

Fig 7