SOME EFFECTS OF EQUALISATION ON SOUND SYSTEM INTELLIGIBILITY & STI MEASUREMENT ERROR

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1 INTRODUCTION

Sound System equalisation dates back 50 years and real time analysers and 1/3 octave equalisers around 35 years [1]. The ‘graphic’ equaliser has also been around for over 25 years and DSP based equalisers for almost a decade with the last 5 years seeing a profusion of commercially available devices. However there would appear to be no study cited in the literature that relates the equalisation of a sound system and the potential improvements obtainable in speech intelligibility. Yet there is much anecdotal evidence and the author for one has been using and relying on the technique for the past 20 years. Sound system equalisation remains one of the most hotly debated and controversial topics with many differing and opposing opinions, which perhaps suggests that it is not fully understood – even after nearly half a century ! Whereas the underlying physics of the loudspeaker-room interaction has not changed, our ability to measure, analyse and visualise the frequency response of a sound system has improved beyond all recognition. The invention of Time Domain Spectrometry by Dick Heyser in 1967 [2] and the subsequent development of FFT and MLS techniques have given rise to many new insights into sound system behaviour.

Much of the original research into intelligibility and particularly its measurement related to the performance of communication channels, initially for the Telephone (eg by Harvey Fletcher at Bell Labs in the 1920s) and then later for wireless systems and military communications. This research almost exclusively related to frequency response and bandwidth considerations, with little interest in the time domain, yet speech intelligibility can be degraded by distortions in both the time and / or frequency domain. However, at about the same time, there was considerable interest and research into auditorium and cinema acoustics and research into time domain aspects of intelligibility began eg by Knudsen in the 1930s. (Who had already been predated to a certain extent by Henry and Sabine in the 1850s and 1890s respectively). The two areas of research rarely crossed as reverberation was not a concern telephone or radio communications and the frequency response of the human talker was of little interest and certainly not a parameter under the control to the architectural acoustician. It was therefore not until the widespread use and installation of sound systems occurred that these two domains were brought together. Whereas it was appreciated back in the 1960s that equalisation could improve the gain before feedback, naturalness and overall sound quality of a sound system eg Boner, Davis, Catania [3, 1, 4] rarely is mention made of the effect on intelligibility. A rare exception to this is the brief comment Boner makes in reference [5].

Despite the benefits of sound system equalisation being known of for over 40 years, it is surprising to find the large number of systems still being installed without appropriate equalisation. Ironically, this would particularly seem to apply to emergency paging systems operating in difficult acoustic environments where intelligibility is of the utmost importance. However, the growing trend of employing DSP based control, routing and mixing systems, where equalisation facilities are normally an inherent part of the system, appears to be slowly changing this. Interestingly, none of the equaliser manufacturers surveyed during the research for this paper mentions intelligibility within their literature.

2 FREQUENCY RESPONSE ANOMALIES

Conventional equalisers operate in the Frequency Domain with the Time Domain aspects of a system remaining effectively fixed (assuming that signal delays are not introduced). It would therefore seem reasonable to expect that any improvements in perceived intelligibility would be spectrally related. The in-situ performance data of around 30 sound systems was analysed and a several of the underlying mechanisms that can give rise to either poor or modified frequency responses established. These include :

- Loudspeaker-Boundary interactions.
- Loudspeaker-Loudspeaker interactions.
- Room Mode excitation.
- Unbalanced reverberant room gain.
- Non uniform sound power radiation of loudspeaker
- Off axis radiation effects / responses
- Excess air attenuation, and temperature & humidity effects.
- Grazing incidence effects eg transmission over seating.
- Cavity resonances
- Transmission path disruption / impedance changes eg under-balcony slot effects
- System microphone proximity effects.

Although the on axis frequency response curves for almost all the loudspeakers involved in the survey were reasonably flat (with the exception of re-entrant horns), the resultant in-situ response curves often were not. Some typical examples are shown in figures 1- 4.

Figure 1 shows the effect of a local boundary on a high quality monitor loudspeaker's frequency response. (In the absence of the wall, the response was within +/- 2dB from 100 Hz – 20 kHz).

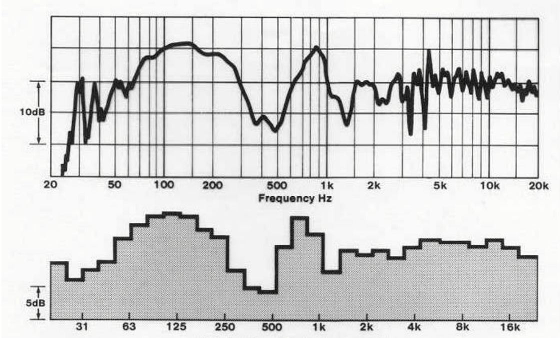


Figure1 Loudspeaker – Boundary Interactions

The comb filtering effect that boundary reflections produce is clearly demonstrated in figure 2. The solid line shows the loudspeaker's response in the absence of the boundary.

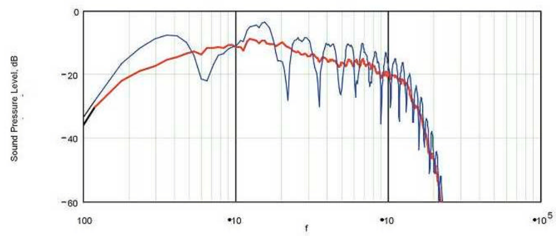


Figure 2 Loudspeaker Interference Comb Filtering

Figure 3 shows the effect of mounting a short column loudspeaker in corner – again the on axis anechoic frequency response was reasonably flat.

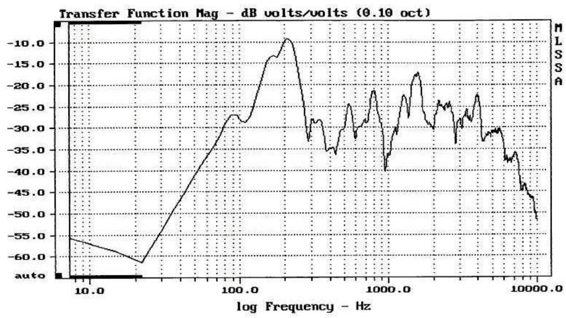


Figure 3 Corner Mounting Loudspeaker

Figure 4 shows the measured response of a distributed sound system in a reverberant concourse employing high quality sound projector loudspeakers. Again their anechoic response is satisfactory. In this case, the 10 dB broadband low to mid frequency peaked response is caused by a combination of factors including, boundary interactions /

loading, non-uniform sound power radiation and frequency dependent room gain.

Figure 5 shows the result of mounting (arraying) or stacking loudspeakers in close proximity to each other.

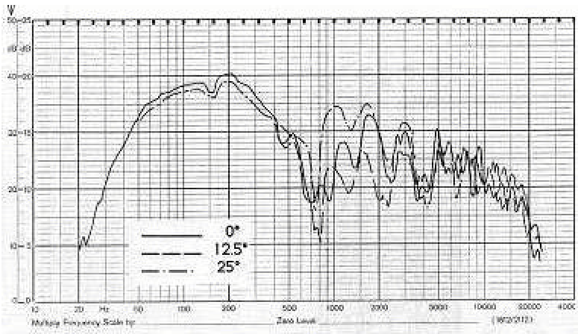


Figure 5 Loudspeaker Array Interactions

The effect on high frequency performance by excess air attenuation over distance is shown in figure 6. In terms of sound quality / spectral balance alone, the case for equalisation would seem to be overwhelming let alone the other benefits such as improved gain before feedback.

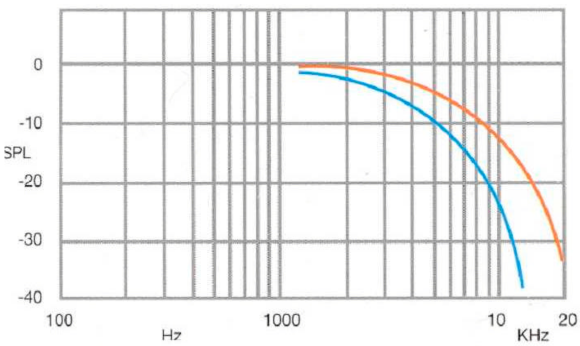


Figure 6 Excess Air attenuation - HF Losses

3 MEASURING THE RESPONSE OF A SOUND SYSTEM

One of the problems of measuring the frequency response of a sound system, is that the measurement microphone is rarely immune to the effects of local reflections either from seating or boundaries. The latter effect can occur even in the centre of large spaces well away from peripheral boundaries due to the floor bounce effect. In this very common situation, a reflection from the floor arrives at the measurement microphone within a very few milliseconds of the direct sound and causes interference resulting in comb filtering. The effect is effectively as shown in figure 2. This measurement error will then modify or obscure the actual system response. A number of techniques can be adopted to try and overcome the problem, such as using sound absorbing material to reduce the bounce or using a Pressure Zone microphone. Both techniques have their advantages and disadvantages. For example, figure 7 shows the effect of introducing a highly efficient absorber (150mm thick Sonex) on the floor to overcome the reflection effect. Figure 7(a) shows the effect on the measured frequency response of the high quality monitor loudspeaker used for the test.

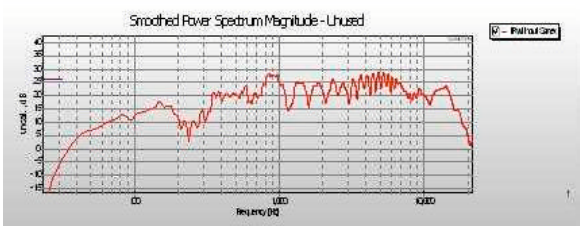


Figure 7 (a) Floor Bounce Interference

The Comb filtering due to the floor reflection can clearly be seen, particularly at high frequencies. The effect of introducing the 150 mm Sonex can be seen in figure 7(b).

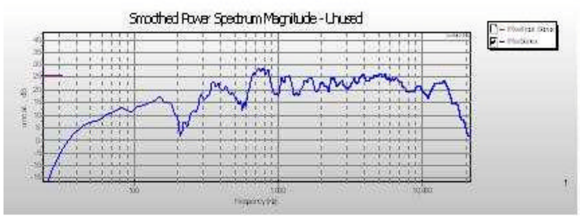


Figure 7 (b) Floor Bounce with 150mm Sonex

Whereas the high frequency ripple has been largely overcome, it is interesting to note that a significant new dip in the response was introduced at around 600 – 700 Hz. In practical terms it would be difficult to find a more efficient transportable absorber and so a compromise still has to be reached.

Floor mounting a PZM microphone will certainly overcome the reflection effect, but this frequently puts the microphone either out of the coverage pattern of the loudspeaker or in a totally unrepresentative location. (How many people in practice listen to sound systems with their heads nailed to the floor ?). Whereas modern acoustic analysers enable time windowing to be readily performed, the floor reflection typically arrives within around 1-5 ms of the direct sound, so windowing it out, also drastically reduces the resultant frequency resolution. Furthermore, the type of measurement window can significantly affect the apparent frequency response. It is quite possible for example to entirely miss information all together. The other question that time windowing raises is how wide should the window be ? Whereas for more than 30 years, the real time analyser has allowed us to look at the steady state response of a system, TDS, FFT, MLS and more recently swept sine techniques allow us to effectively choose any temporal element or fraction of the signal. But which part or time period relates to our perception of hearing the embedded information and formation and integration of the perceived spectral balance ?

There is considerable debate amongst the audio and acoustic fraternity concerning the role of early reflections and their integration time. Architectural acoustic research relating to the integration of early reflections in auditoria and rooms repeatedly suggests a time of around 35-50mS and this is accepted by most working in the field. However, many members of the audio fraternity disagree and suggest 20mS or less if indeed there be any integration at all ! A similar debate also exists with regard to equalisation. There is a school of thought that propounds that only the 'direct sound' spectrum should be equalised and carrying this to its logical extreme have been known to argue that a loudspeaker / loudspeaker system can be completely equalised prior to installation. The examples presented in figures 1-7 clearly show that this can not be the case. Whereas it is true that a conventional frequency domain equaliser effectively adjusts the direct sound radiated by a loudspeaker, it can however be adjusted so that overall resultant frequency response, including the early reflections and reverberation are affected and improved – though not individually of course.

4 EQUALISING FOR INTELLIGIBILITY

Over the years, a number of preferred curves for system frequency responses have been proposed. A typical one is shown in figure 8.

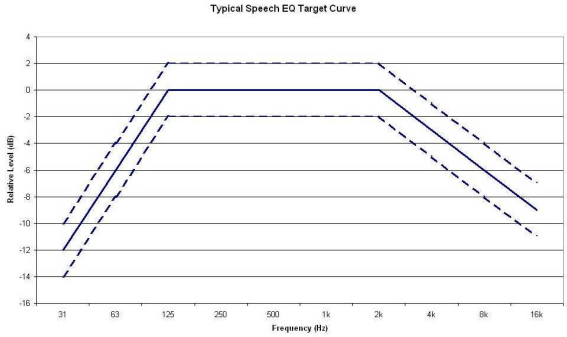


Figure 8 Typical Target EQ Response Curve

This generally works well for distributed sound systems, but is by no means universal. For example, figure 9 shows the response of a system designed by the author for a medium sized room, which shows the response to be essentially flat out to 8 kHz.

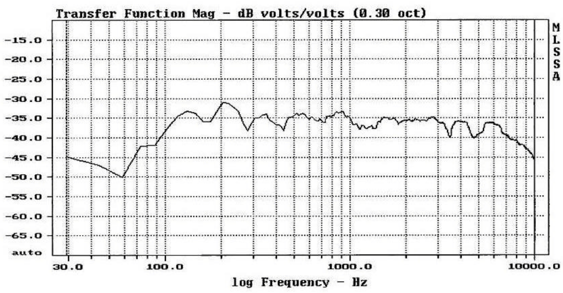


Figure 9 In room response – high DR system

This worked perfectly well and was not overly bright, which had this response been adopted in a more reverberant space it most certainly would have been. This is because the Direct to Reverberant ratios of most sound systems are in fact negative. This comes as a surprise to many but in the author's experience only rooms with low reverberation times or where relatively high Q devices are employed and / or where the distance between the listener and loudspeaker is small, disobey this rule. Under such conditions it is possible to employ a flat response out to much higher frequencies, as it is the direct sound that dominates the situation. However, in the majority of systems, this is not the case and the reverberant or reflected field is greater and effectively equalisation of the power response is required. Figure 10 puts this into context presenting the calculated critical distance for seven medium to large sized spaces. As can be seen, the critical distance is highly frequency dependent due not only to the acoustic characteristics of the space but also due to the frequency dependent directivity of the loudspeakers.

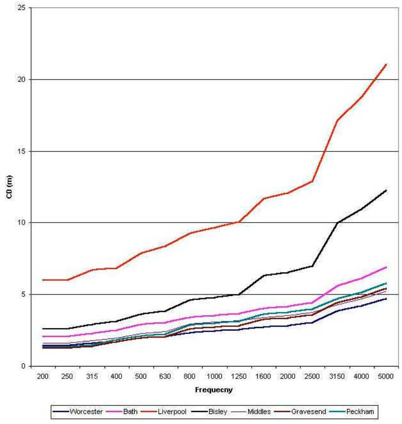


Figure 10 (a) Critical Distances for 4 inch LS

Figure 10 (a) plots critical distance versus frequency for a simple 4inch cone loudspeaker. The bottom five curves relate to medium size spaces of around 600 m³, with reverberation times of between 0.9-1.5 seconds, whereas the upper three curves relate to a large church, small arena and large Cathedral, with RT values ranging from 2 to 9 seconds. As can be clearly seen, at 1000Hz, the critical distance for the medium sized rooms is only around 3.0 metres, but increases as the volume of the space increases. However, it should be remembered that this is for just one loudspeaker. In an actual system, with several sources, the critical distance would decrease down to a metre or less. Figure 10 (b) shows the effect on critical distance of using a short column (line source) loudspeaker.

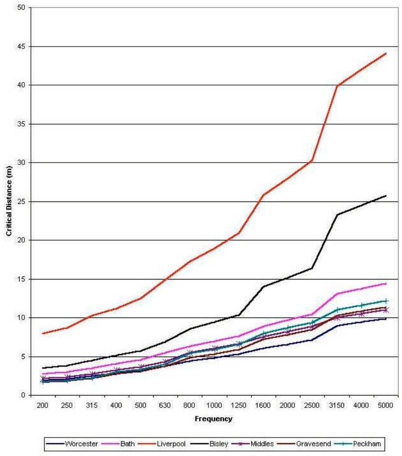


Figure 10 (b) Critical Distances for 750mm Column LS

This increases the critical distance to around 5.5 metres but again in an actual system this would typically decrease down to 2 –3 metres. To further put the above figures into context, measurements on many home hi-fi systems shows that the critical distance in most domestic rooms is rarely greater than 2-3 metres.

As noted above, beyond critical distance it is essentially the power response of the loudspeaker than is equalised. This will also include any boundary effects and source interactions.

Returning to figure 4, the response of a sound system measured in a reverberant concourse, it is interesting to compare this graph to the sound power response of the loudspeaker itself as shown in figure 11. A 10 dB peak in the power response occurs at around 500 Hz, correlating well with the peaked response shown in figure 4. The equalisation filter curve required to correct the response is shown in figure 12 and the final 'in room' response after equalisation is shown in figure 13.

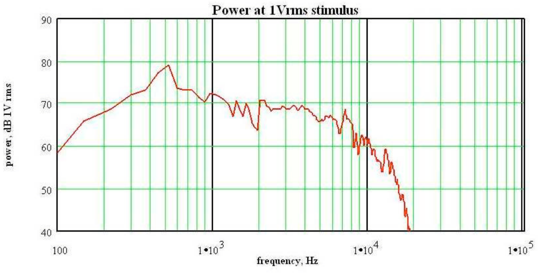


Figure 11 Concourse LS Sound Power Response

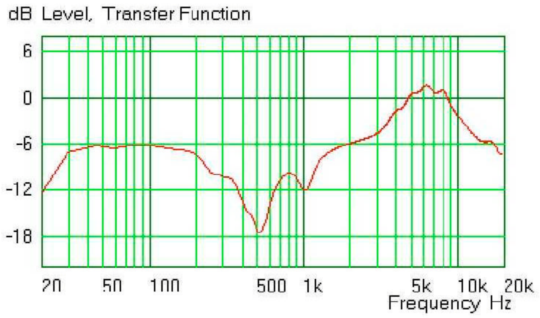


Figure 12 Concourse LS – Equaliser Filter Response

Figure 13 Concourse System Response after EQ

Not only after equalisation did the sound quality improve dramatically but so too did the intelligibility of speech broadcast over the system. Interestingly, the measured intelligibility performance in terms of % Alcons, RaSTI, STI and STIPa did not improve or indeed effectively alter within the bounds of the measurement tolerance. Unfortunately it was not possible to carry out a word score test under the prevailing site conditions but subjective opinion was very clear about the resultant improvement to clarity and intelligibility.

During the survey, it was noted that short column loudspeakers, when operating in highly reflective and reverberant environments were, without question the most difficult devices to successfully equalise. This would appear to be due to the inherent discrepancy between their axial frequency response and acoustic power response. Figure 14, a power response measurement made on a 750mm long column loudspeaker clearly shows this, with a broad band peak of 10 –15 dB occurring over the range 400 Hz to over 1 kHz.

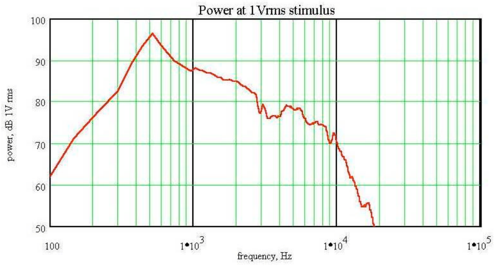


Figure 14 Sound Power Response 750mm Col LS

Figure 15 shows the smoothed response of a sound system measured before and after equalisation in a church with a 2 second reverberation time employing short column loudspeakers. On this occasion, word scores were carried out and a 21 % improvement in intelligibility was recorded after equalisation. However, carrying out STI and percent Alcons measurements showed there to be no apparent improvement.

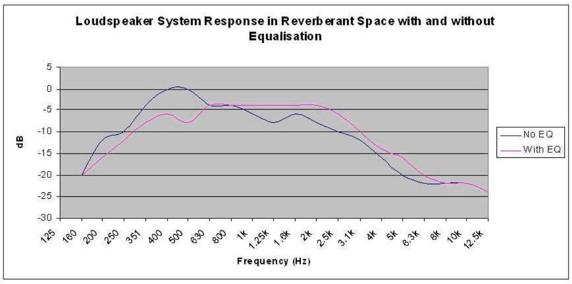


Figure 15 Church System Responses

In the author's experience of measuring and equalising literally hundreds of sound systems, rarely is the subjective impression of intelligibility and word score improvement confirmed by the currently available intelligibility metrics, when operating in high signal to noise ratio reverberant spaces. A good example of this shown in figure 16, which shows the response of a reverberant system (RT 1.8 seconds) with a high frequency roll off deliberately introduced. STI & STIPa measurements on the system showed this to be 0.46 STI or approximately

15% Alcons. The same result was obtained for both sets of curves, although subjectively it was very clear that one was very much less intelligible than the other. Comprehensive word score testing was undertaken and a score equivalent to 0.46 STI was recorded for the flat response. The word score however for the condition with the high frequencies rolled off was equivalent to just 0.3 STI and in far better agreement with the subjective impression.

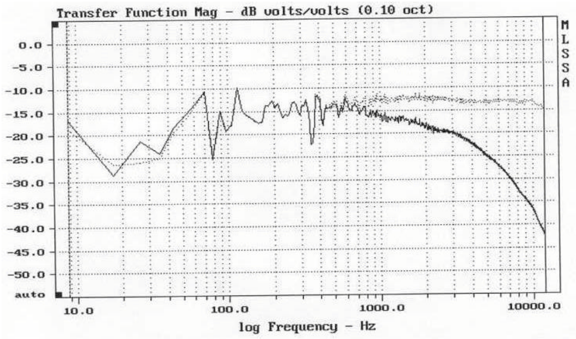


Figure 16 Reverberant System Responses

Although STI (an effectively STIPa) operate over the range 125 to 8 KHz, they are currently incapable of accounting for psychoacoustically significant variations in the overall response. To understand why this should be so is beyond the scope of this current paper, but will be reported upon at a later date. However, of related interest and more pertinent to the current issue is to understand the mechanisms by which equalisation would appear to improve intelligibility.

Logically, as equalisation relates to adjustments made in the frequency domain, the effect must be spectrally based * and would appear to relate to critical band frequency masking. It is well known that lower frequencies mask, lower level, higher frequency sounds as illustrated in figure 17.

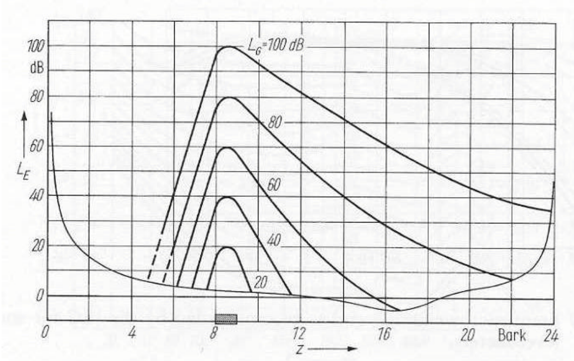


Figure 17 Frequency Masking Curves

Figures 3-5 and 15-16 illustrate situations where the high frequencies are attenuated and intelligibility suffers. Restoring the high frequencies or at least improving the high to low frequency balance with equalisation, also reduces the potential masking – particularly when the speech signal spectrum is incorporated (figure 18).

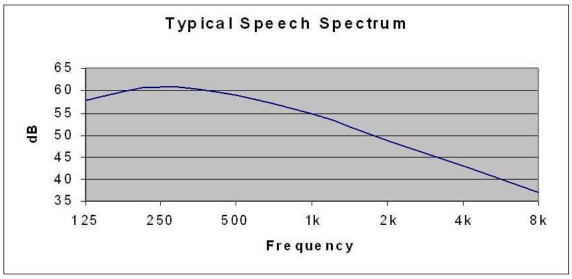


Figure 18 Speech Spectrum

An example of the effect can be seen in figure 19, which shows how critical frequency bands are masked. Although the more recent specifications for STI do include a masking effect, a survey, carried out by the author of the currently available programmes and instrumentation offering STI capability, shows that several of these do not incorporate this. Furthermore, extensive measurement and testing using the products that do, still shows there to be a discrepancy between subjective impression the word score test results and the STI.

The explanation for this is complex, but essentially it would appear that the masking algorithm applied within the STI measurement is not appropriate for speech in reverberant conditions. This is illustrated by the masking curves shown in Figure 20

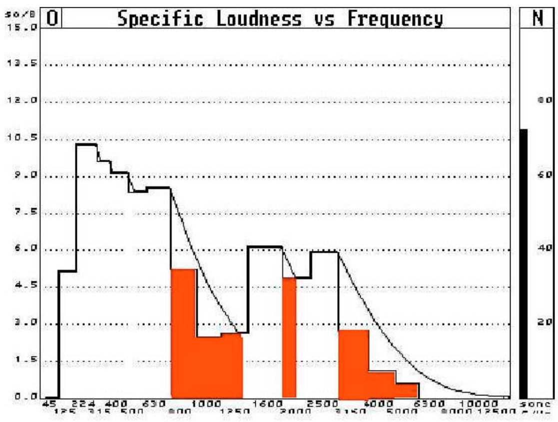
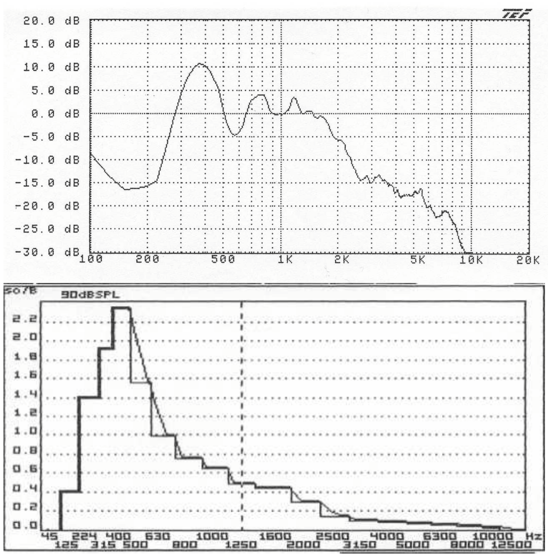


Figure 19 Masked System Responses



Figures 20(a) & (b) Showing poor system intelligibility response and associated masking.

As figure 20 (b) shows, although some masking is shown to occur, this is at 500 & 600 Hz and not in the high frequency consonant region – yet the upper frequency response graph 20(a) clearly shows these frequencies to be severely attenuated. (This being one of the primary causes for the poor intelligibility of this system, which equalisation significantly improved).

This raises the question as to whether standard masking curves can be (a) applied to speech signals which are known to be perceived differently to pure tones or noise or (b) Whether the curves are applicable to reverberant conditions. Research is continuing to further explain this problem and derive appropriate correction factors to take account of speech frequency masking within STI

5 SUMMARY & CONCLUSIONS

Although sound system equalisation has been available as a technique for over 50 years and 1./3 octave equalisers and Real Time analysers have been around for over 30 years, a surprising number of systems still do not incorporate appropriate equalisation.

Although it is well known that equalisation can affect and improve intelligibility, a comprehensive review of the literature found little published information regarding this.

It is speculated and corroborative evidence presented to show that the majority of sound systems probably operate with the listeners located beyond critical distance and that the reflected and reverberant fields therefore dominate.

Considerable anecdotal and also pilot study and word score information has been presented that suggests that the current intelligibility measurement metrics do not correctly account for the intelligibility improvements that can be achieved by appropriate system equalisation.

It is suggested that the underlying mechanism relates to spectral masking, though the well established masking curves do not appear to show this effect.

It would seem that a more complex masking criteria is required that simultaneously takes into account both temporal and spectral attributes.

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