Towards an active reverberant room

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ABSTRACT

A laboratory reverberant room is used for sound power, sound absorption, and sound scattering measurements, and needs to meet certain volume and absorption criteria in order to make good measurements. This paper examines the possibility of creating or enhancing a reverberant room through a system of electroacoustic feedback. For such a system, signals from a set of microphones could be distributed to a set of loudspeakers, with the system's transfer functions between microphones and loudspeakers (i.e., the gain, delays, and detailed spectrum) established so as to create desired exponential decay of the total system as sound feeds back through the combined physical space and electroacoustic system. Such an approach might increase the spatial diffusivity of the reverberant soundfield, increase the modal density of a room (by introducing new modes that have an electroacoustic component), and increase reverberation time. Experimental results using a simple feedback system in a rectangular reverberant room yielded potentially large changes in reverberation time, and, with careful tuning, allowed the room's magnitude response to be increased at frequencies where its natural response is weak.

INTRODUCTION

A reverberant room is used for laboratory measurements where a diffuse soundfield is desired – such as the measurement of sound source power, random incidence absorption coefficients, and scattering coefficients (usually with a scale model reverberant room). However, below a certain frequency, the modal density is insufficient for statistical soundfield assumptions, and the assumptions behind the acoustic measurements (for example, of sound source power) will not be met. Various approaches are taken to increasing the diffusivity of soundfields in reverberant rooms (such as the design of the room's form, the introduction of scattering surfaces, and the use of large moving elements), and this paper considers the viability of an electroacoustic solution to directly increase modal density in the low frequency range. The concept is to introduce artificial resonances that are in many ways similar to room modes through an electroacoustic feedback system.

There are some similarities between electroacoustic feedback and room resonance. In both, the regular superimposition of multiple sound waves produces a spectral interference pattern, with potentially strong peaks at points of maximum constructive interference. In both, these resonances have exponential decay functions (if the electroacoustic feedback is stable). There are differences too: a conventional electroacoustic feedback circuit has the wave circulating in one direction, rather than equally in both directions (the loudspeaker and microphone ensure that the wave circulates in one direction, at least in the electronic part of the circuit).

In this project we wished to produce a feedback system that behaves similarly to the low frequency room modes of a reverberant room. Possible configurations of feedback systems were tested in a bare rectangular reverberant room, but none were particularly successful, as the room response dominated the combined system response. This paper reviews the background theory of this problem, and presents selected experimental findings.

Review of rectangular room mode theory

Room resonance theory, at least for simple rectangular rooms, is well defined. Classic texts such as Cremer and Müller's *Principles and Applications of Room Acoustics* (1978) and Kuttruff's *Room Acoustics* (1991) provide expositions of this theory.

The frequencies of the normal modes of the room are given by equation 1, where n_x, n_y and n_z are integers defining the mode numbers, L_x, L_y, and L_z are the rectangular room dimensions, and c is the speed of sound.

$$f_{(n_x,n_y,n_z)} = \frac{c}{2}\sqrt{\left(\frac{n_x}{L_x}\right)^2 + \left(\frac{n_y}{L_y}\right)^2 + \left(\frac{n_z}{L_z}\right)^2} \quad (1)$$

The spectral distribution of these mode frequencies is an important consideration in the usefulness of rectangular reverberation room measurements. There have been several approaches to optimising the low frequency distribution by selecting particular room dimensions (most obviously, to minimise modes that coincide in frequency). Bolt (1946) suggested evenness in the average mode spacing as a method for identifying favourable room dimensions but it is now known that other methods, such as using the standard deviation of mode spacing (Louden, 1971), provide better results. Bonello (1981) developed a model from observation of a large number of real rooms. His criterion is based on increasing mode densities in successively higher third octave bands and an absence of double modes in a third octave band except where mode densities in a third octave band are greater than five. More recent work by Walker (1996) and particularly Cox and D'Antonio (2001) has improved computational methods for predicting mode distributions and suggests that a much wider range of acceptable room ratios are possible than was previously indicated, essentially just avoiding the much smaller ranges of unacceptable ratios.

Assuming that the room dimensions produce a reasonably even fine distribution, the problem of mode density can be approached statistically. In a rectangular room, the number of modes up to a given frequency, f, is estimated from equation 2. Here, V is the room's volume, S is its surface area, and L is the sum of the lengths of the room edges.

$$N_f = \frac{4\pi V f^3}{3c^3} + \frac{\pi S f^2}{4c^2} + \frac{Lf}{8c} \quad (2)$$

The number of modes within a given bandwidth may be derived from this. We can approximately state that mode density increases with frequency by a power of three.

The damping constant, δ, of a room mode may be defined in terms of the mode's exponential decay rate, where $E(t)$ is the spatially averaged energy density of the room decaying in time.

$$E(t) = e^{-2\delta t} \quad (3)$$

The damping constant is related to half-power bandwidth of the mode, Δf.

$$\delta = \frac{\Delta f}{\pi} \quad (4)$$

The reverberation time associated with the mode is

$$T = \frac{6\ln(10)}{2\delta} \approx \frac{6.91}{\delta} \quad (5)$$

With some simplifying assumptions, the damping constant of a mode can be estimated in a rectangular room from the absorption coefficients of each of the six surfaces. In equation 6, α' is the absorption exponent, defined as $-\ln(1-\alpha)$ for each of the six surfaces. Theta is the angle of incidence of the wave on each surface.

$$\delta = \frac{c}{4}\left[\frac{\cos\theta_x}{L_x}\left(\alpha'_{x0} + \alpha'_{xL}\right) + \frac{\cos\theta_y}{L_y}\left(\alpha'_{y0} + \alpha'_{yL}\right) + \frac{\cos\theta_z}{L_z}\left(\alpha'_{z0} + \alpha'_{zL}\right)\right] \quad (6)$$

The implication of this is that, for evenly distributed absorption in a bare cubic room, axial modes have lower damping constants (longer reverberation times) than tangential modes, which in turn have lower damping constants than oblique modes. The introduction of diffusing panels into a reverberant room serves to reduce this contrast.

According to Schroeder (1954, 1996), the frequency above which mode behaviour can be assumed to be statistical, f_s, could be taken as that where the average mode spacing is one third of the half power bandwidth. This could be calculated by evaluating the equations above, but it is also common to estimate this frequency as

$$f_s \approx 2000\sqrt{\frac{T}{V}} \quad (7)$$

Here T is the measured reverberation time of the room (although this is difficult to interpret if the reverberation time varies greatly with frequency).

The modal component of the room's pressure transfer function between two points depends on the position of each point in relation to the standing wave pressure distribution, as well as the frequency and damping constant. The pressure distribution for each mode is given by equation 8. C is a proportionality constant. Hence the pressure function through the space is a cosine in three dimensions (with absolute value maxima in the room corners), and the extent to which pressure transducers engage with modes depends their position within the cosine pattern.

$$p_{(n_x, n_y, n_z)}(x, y, z) = C\cos\left(\frac{n_x \pi x}{L_x}\right)\cos\left(\frac{n_y \pi x}{L_y}\right)\cos\left(\frac{n_z \pi x}{L_z}\right) \quad (8)$$

The mode-related transfer function is given by equation 9, where the amplitude A_n is determined by source and receiver positions (as suggested by equation 8). Here ω_n is the nominal angular frequency of a room mode (from equation 1), and the calculation can be made over a range of frequencies, for the relevant modes.

$$H(\omega) = \sum \frac{A_n}{\omega^2 - \omega_n^2 - 2i\delta\omega} \quad (9)$$

A simple feedback loop

The classic papers by Nyquist (1932) and Bode (1960) provide basic theory for understanding feedback in circuits. In this section we give an overview of a simple audio feedback system, using theory presented by Schroeder and Logan (1961).

Perhaps the simplest way to introduce additional resonances into the room using electroacoustic feedback is to use a delay line. With a delay of τ and a positive amplitude gain of g, the closed loop transfer function is given in equation 10.

$$H(\omega) = \frac{e^{-i\omega\tau}}{1 - g e^{-i\omega\tau}} \quad (10)$$

The magnitude of the loop transfer function is a comb filter, with peaks spaced at $f = n/\tau$ (for n=0,1,2,3…). Hence, a large delay yields closely spaced peaks. The ratio of maximum to minimum magnitudes of the transfer function is

$$\Delta L = 20\log\frac{H_{\max}}{H_{\min}} = 20\log\frac{1+g}{1-g} \quad (11)$$

The bandwidth associated with each peak is approximately given by equation 12:

$$\Delta f \approx \frac{-\ln(g)}{\pi\tau} \quad (12)$$

The reverberation time associated with each peak is

$$T = \frac{60\tau}{-20\log(g)} = \frac{6\ln(10)}{2\pi\Delta f} \quad (13)$$

As can be seen by the relationships in the equations, the spectral and temporal characteristics of room modes, and of peak frequencies in a simple feedback loop, can be considered in the same terms. The harmonic series associated with the delay is like the harmonic series formed by each of the three sets of axial modes in a rectangular room.

In designing a feedback system that behaves similarly to room resonances, we could start by deciding on the peak density (number of peaks per unit frequency, which is the inverse of the delay time). If we then choose a target reverberation time (thereby choosing the damping constant and half-power bandwidth of each peak), the closed loop gain of the system can be calculated from equation 14.

$$g = 10^{-3\tau/T} \quad (14)$$

If we wished to match a room's reverberation time of 5 s, and we wished to have peaks spaced at 1 Hz intervals, we would need a delay of 1 s in the feedback loop and a gain of 0.25 (or -12 dB). The half-power bandwidth of each peak would be 0.44 Hz. Of course the disadvantage of this long delay approach is that the build-up and decay functions are clearly stepped. However, these are not necessarily problems for measurements of steady state soundfields. Closed loop gains required for a variety of reverberation times and peak frequency spacing intervals are shown in Figure 1.

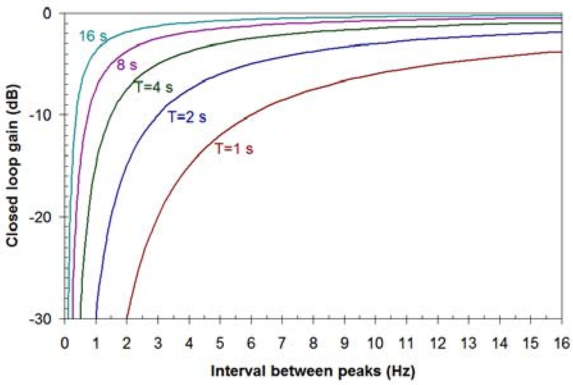

Figure 1. Closed loop gain required for reverberation times between 1 and 16 seconds for a range of peak frequency intervals (i.e., the inverse of τ).

A feedback system in a room

An electroacoustic feedback system involving transducers in a room is much more complex than the delay loop described above. The loudspeaker system (including filters such as might be used in a crossover system) is unlikely to have a flat magnitude response or linear phase response within the frequency range of interest. Even more complicated is the acoustic environment of the room, which combines many reflections with the direct sound path. Nevertheless, if we were able to assume that the magnitude response of these is approximately flat in the frequency range of interest, the unpredictable phase response should merely shift the peaks of the comb filter whilst maintaining approximately the same peak density as the purely linear phase feedback loop. Of course, the magnitude response is far from flat, and this poses a formidable challenge in integrating a feedback system into a reverberant room. It is theoretically possible to create all-pass feedback by creating a filter that matches the closed loop gain and phase for all frequencies of interest – but there are several practical problems with doing this (such as numerical errors in spectrum inversion, and the fact that the room system is not truly time-invariant). Changing the layout of a room (for example, introducing a loudspeaker or moving a test sample for the purpose of spatial averaging) also introduces time variance, which makes detailed optimisation of feedback undesirable, because such changes would render the system no longer optimal.

We wish to develop new modes (rather than reinforce natural modes), and so we would hope to have feedback peaks at frequencies between the natural room modes. The positioning of transducers can be helpful in this. In a rectangular room, the centre of the room is the location of a pressure node for every odd order mode. A point one quarter of the distance along each dimension has pressure nodes for the modes having an order of 2 in any dimensions (and indeed for every second even order mode). The combination of these positions for a pressure source and receiver yields a transfer function that is very sparsely populated by room modes, with the lowest frequency room mode being [n_x=4, n_y=0, n_z=0] (assuming L_x is the largest dimension). Figure 2 shows the theoretical modal responses for point pressure source-receiver pairs located at room corners (which are pressure antinodes for every mode), and in the centre and quarter positions. In the example of Figure 2, the lowest mode is around 27 Hz (not shown), but when the transducers are at the middle and quarter of the room the lowest activated mode is around 108 Hz. Hence if we wish to operate a single electroacoustic feedback system at frequencies other than the natural room modes, a pair of source-receiver positions such as this should be helpful.

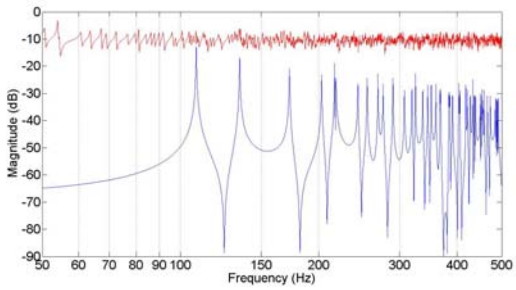

Figure 2. Calculated normalised magnitude of the mode transfer function for a rigid rectangular room 6.34 m x 5.10 m x 4.00 m with a uniformly distributed absorption coefficient of 0.026. The upper (red) function is for pressure transducers in the corners of the room, thereby activating all modes equally. The lower (blue) function is for a pair of pressure transducers, one in the centre of the room, and one a quarter of the distance along each of the three dimensions.

Another important issue is how much energy is transferred between the feedback system and the room (in both directions). Clearly, if the loudspeaker and microphone are very close or highly directional towards each other (so that the direct field dominates), there will be little coupling. Increasing the gain of the feedback system increases the coupling, as well as increasing its reverberation time, reducing the peak half-power bandwidth, and increasing the contrast between the peak levels.

EXPERIMENT SET-UP

We used a rectangular reverberant room to test the effectiveness of adding electroacoustic feedback. We did not add any diffuser panels, so that the room would be simple to model. The setup consisted of a measurement system organised into two configurations and a feedback system. The measurement system uses an omnidirectional loudspeaker source and eight randomly placed omnidirectional microphones. Rather than move this source to a different location in the room for a second configuration, a different omnidirectional loudspeaker was positioned at another location in the room, and used alternatively. Table 1 lists the coordinates of these microphones and Table 2 lists the coordinates of the sources.

Microphone	Type	s/n	X	Y	Z
1	4189	2534540	1.1	3.59	2.53
2	4189	2566074	1.85	2.87	1.71
3	4189	2534541	4.95	1.66	1.17
4	4189	2566073	3.24	3.56	1.7
5	4190	2639485	2.8	4.15	2.81
6	4190	2522008	2.16	1.42	1.41
7	4190	2491067	3.62	1.03	1.72
8	4190	2491066	3.36	2.15	1.21

Table 1. Eight measurement microphone coordinates, where the origin is a corner of the room

Source	Type	Make	X	Y	Z
1	4292	B&K	4.49	2.91	1.42
2	Globe	Outline	1.085	1.83	1.98

Table 2. Two measurement source loudspeakers coordinates

The eight Bruel & Kjaer Type 4190 and 4189 ½" free-field microphones were connected to a Bruel & Kjaer Pulse analyser system through Type 2669B pre-amplifiers into a Type 3560C front end with 24-bit analogue to digital convertors. This system is capable of measuring steady state response, reverberation time, narrow band frequency response (FFT) and CPB (1/*n*-octave) from each microphone simultaneously, or in groups without disturbing the set-up. Two separate configurations were installed to run alternatively by switching the connection from a Bruel & Kjaer Type 2716C audio amplifier to either a Bruel & Kjaer Type 4292 dodecahedron Omni-power source or an Outline Globe Source Radiator. The amplifier was configured in bridge mode to deliver 300 W at the output. The loudspeaker sources were placed in different positions on opposite sides of the room at the coordinates shown in Table 2.

The feedback system consisted of an omnidirectional microphone in the geometric centre of the room and two loudspeakers placed at one quarter distances on each dimension in opposite diagonal corners (following the concept of Figure 2). Table 3 lists the coordinates of the feedback system transducers and Figure 3 shows the room set up.

Microphone	Type	Make	X	Y	Z
1	4189	B&K	3.2	2.55	2
Speakers					
1	TA-500	Turbosound	1.6	1.275	1
2	TA-500	Turbosound	4.8	3.825	1

Table 3. Feedback system transducer coordinates

The electroacoustic feedback system was achieved using two Turbosound TA-500 three-way loudspeakers fed by two Labgruppen C 48:4 four-channel amplifiers rated at 1000 W each. Two of these channels bridged delivered 2,400 W into each woofer of the TA-500 systems from each amplifier. The transducer input for the feedback system was a Bruel & Kjaer Type 4189 free field omni-directional microphone connected through a Type 2669 preamplifier to a Nexus Type 2690A conditioning amplifier. The output of this was input to a MacPro computer through a Digidesign 003 analogue to digital convertor. The computer was set up to run MAX/MSP 4.6 software programmed to provide gain, delay, filtering and phase control of the fed back signal as needed before sending it back through the Digidesign 003 output to the Labgrubben amplifiers via a Dolby Lake Processor LP4D12 included to provide crossovers for distributing any signal in to the appropriate frequency bands for each of the three loudspeaker drivers (although we disconnected the mid- and high-frequency drivers from this – and only used the woofers).

Figure 3. Microphone and loudspeakers set up in the reverberant room

ACOUSTIC CHARACTERISTICS OF THE ROOM

Figure 4 shows the one-third octave band reverberation time of the room, averaged over the sixteen source-receiver combinations (two sources and eight receivers). The long reverberation times of the 25 Hz to 40 Hz bands are due to the lowest frequency axial modes (the 20 Hz band is below the room lowest room mode). The Schroeder frequency of the room might be taken as about 430 Hz (based on a reverberation time of around 6 s).

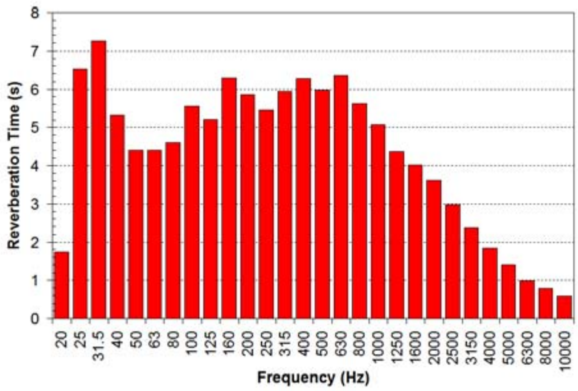

Figure 4. 1/3-octave band reverberation time of the reverberant room without the feedback system operating. Values are the average of measurements from each loudspeaker to the eight measurement microphones.

The power averaged transfer function of the room for all of the source-receiver combinations is shown in Figure 5. The lower part of the figure shows the theoretical response (which does not include the loudspeakers' frequency response). Although there is not a close match between the theoretical and measured responses, both exhibit large fluctuations below 130 Hz, with smoother responses above this (using 1/24th octave band resolution for the measured responses, and 1/24th octave smoothing of 0.1 Hz resolution theoretical response). This transition from rough to smooth response might be expected from the theoretical data on Figure 6, which shows Schroeder's criterion of average mode spacing being one third of the mode half power bandwidth in relation to the running average values. Hence a transition from statistical behaviour to behaviour in which individual room modes have identifiable influences would be expected below the initial Schroeder estimate, in the region below 200 Hz.

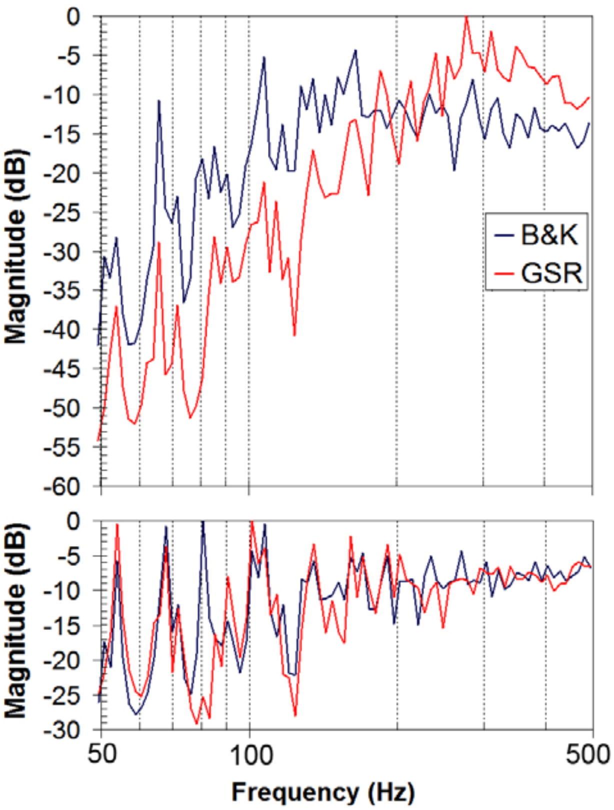

Figure 5. Power average of the frequency response of the room (without feedback) at the eight measurement microphone positions from the two measurement loudspeakers (Bruel & Kjaer Omnipower, and Outline Globe Source Radiator). The upper chart shows the measured sound pressure level, normalized by the maximum value. The lower chart shows the power averaged theoretical normalized transfer functions. The frequency resolution is 1/24th octave.

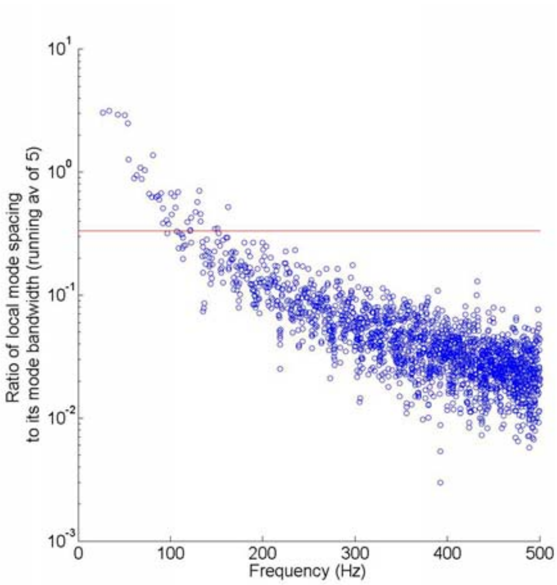

Figure 6. The ratio of running average mode spacing to bandwidth, showing Schroeder's criterion of 1/3 as the red horizontal line. Values are calculated from a uniformly distributed absorption coefficient of 0.026.

With the feedback transducers positioned in the modally sparse configuration illustrated by Figure 2, the measured open loop transfer function (Figure 7) is by no means in perfect agreement with the theoretical prediction. Discrepancies occur for several reasons, such as: the loudspeaker is not a point source, and may not be a pure pressure source; the volume of the loudspeaker is significant, and may affect the room response; the measured transfer function includes the pre-modal response (i.e., the direct sound and perhaps early reflections before the modes are fully formed). While the magnitude peak response is not as sparse as ideal, it is still much sparser than would be expected for other source-receiver positions. The highest peak, at 108 Hz, is mode (4,0,0), which is expected from the ideal model. The overall bandpass envelope of the magnitude spectrum reflects the response of the woofers (including the effect of the Dolby Lake loudspeaker processor).

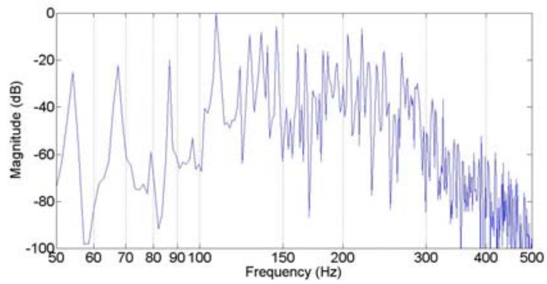

Figure 7. Measured gain of the open loop transfer function of the feedback system prior to equalization.

MEASUREMENT PROCEDURE

Before we could measure changes we needed to see what the steady state of the room was and how and to what detail we could rely on repeatability of the measurements. Background noise was measured in 1/24th Octave bands and averaged over the 8 microphone positions. Shown in Figure 8, peaks in background noise reveal excitation at axial room modes that correlate to peaks in the frequency response of the room of Figure 5. High frequency noise is the measurement microphone system self-noise.

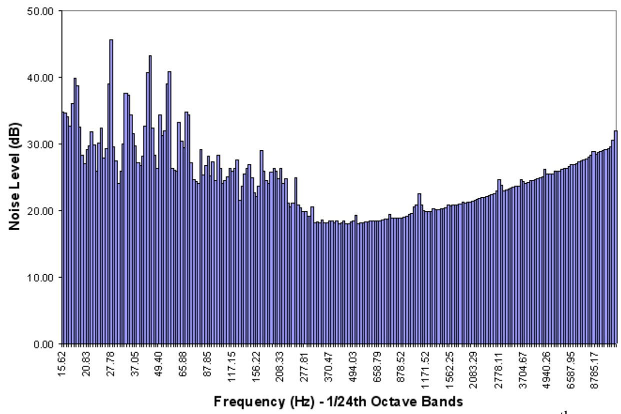

Figure 8. Background noise sound pressure level in 1/24th Octave Bands

The magnitude of the room's spatially averaged frequency response (including loudspeaker response) was measured by generating pink noise from a Bruel & Kjaer Type 1054 noise generator into source 1 and source 2 configurations alternatively. 1/24th octave band levels were repeatedly measured at each microphone position with the Pulse CPB analyser averaging on 60 seconds to give enough time to reach steady state with the long delay times involved. Standard deviation between repeated measured levels in 1/24th Octave bands was <1 dB above 40 Hz. We then measured levels at all microphone positions in 1/24th and 1/3rd octave bands and in octave bands without feedback, and power-averaged them to have a measured frequency response as a reference with which to compare for any changes caused by the introduction of the electroacoustic feedback loop into this measurement process. The feedback system was turned on and reverberation time was measured at each microphone for each of the two sources and averaged. The effect of varying gain and delay on reverberation time was observed.

We tried many approaches to implementing a feedback system, including a simple delay and gain loop (which is nonetheless complex due to the room and transducer response), conditioning the magnitude response of the loop by selective peak and notch filtering, and inversion of the loop's magnitude response in a linear phase filter. To begin with the simple delay system was too sensitive and prone to instability before we could put in enough gain to expect to make a difference. We were only interested in frequencies below the Schroeder frequency and so we modified the feedback system to work well below 200 Hz. With the tweeters and the midrange loudspeakers disconnected, the loudspeaker controller (Dolby Lake Processor) provided a -15 dB roll off to 500 Hz. We put in notch filters at 80 Hz, 108 Hz and at 126 Hz to tame the highest amplitude frequencies as recognised by the open loop transfer function measurement of the feedback system (these are due to natural room modes). These frequencies were becoming over-excited too easily causing premature instability. We put in a low pass filter and adjusted the *Q* and the bandwidth of the filters until we had a stable system that gave us workable control over gain and delay to begin to test with. We initially measured reverberation times with various gain and delay combinations especially examining effects with a delay of 1 s with reverberation times not much more than the natural reverberation time, evaluating the results using the 1/24th octave pressure levels from the noise generator compared to the levels measured without feedback.

Although we could easily achieve dramatic increases in reverberation time we initially had very little effect on steady state sound pressure level from the noise generator. We could not detect any new peaks in the steady state power spectrum being formed with our measurements in either narrow band FFT or in 1/24th octave analysis. Therefore we selected two frequencies that were very low in magnitude in the spatially averaged room response: 76.08 Hz and 95.77 Hz 1/24th octave band centre frequencies. We built two parallel filters centred on these frequencies so we could minimise instability caused by other frequencies as we balanced the gain and delay of the system to increase their amplitude through the feedback. We ran the measurement system with pink noise at the same level as for the room without feedback and found that a delay setting of 0.3 s (not including the delays in the room and in various components of the system) and appropriate closed loop gain achieved more than 6 dB increase in amplitude in the room at both of the chosen frequencies. Figures 9 and 10 show the measured open loop transfer function of the feedback system after these modifications.

The reverberation time at this setting is compared to the reverberation time without feedback in Figure 11. It is worth noting here that the feedback system was stable through these measurements but became unstable if the room door was opened. Measurements were conducted well above the noise floor of the room and the instrumentation.

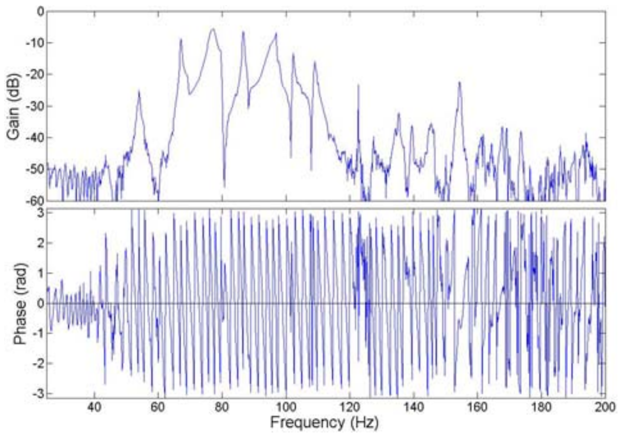

Figure 9. Measured gain and phase of the open loop transfer function of the feedback system including equalization.

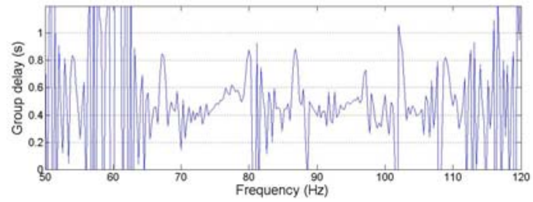

Figure 10. Group delay of the feedback system's open loop transfer function including equalization.

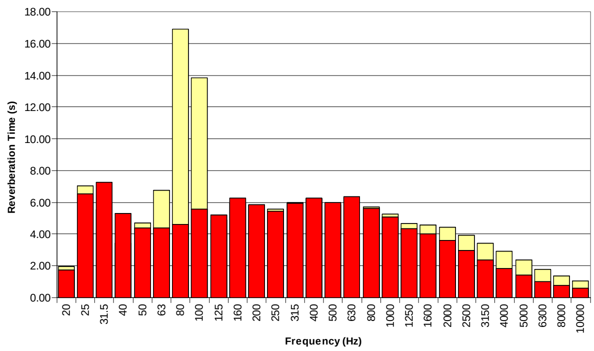

Figure 11. Reverberation time of the room with no feedback (red), and with the feedback system active (yellow).

Figure 12 compares the power averaged pressure level with and without the feedback system (as described above) for one of the loudspeakers, and Figure 13 shows the level difference introduced by this feedback system. Although the effect is largely restricted to two narrow bands, the feedback system is clearly boosting the spatially averaged transfer function at frequencies that have a naturally weak response.

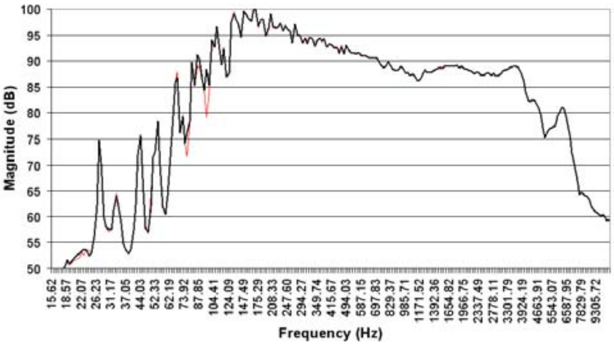

Figure 12. Power averaged pressure levels with no feedback (red), and feedback system active (black) for the Bruel & Kjaer Omnipower

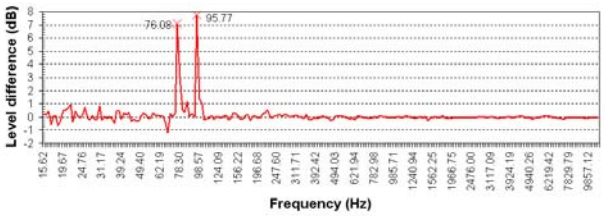

Figure 13. Difference between spatially averaged pressure levels with feedback and without, averaged for both omnidirectional sound sources

DISCUSSION

It is evident from the simple feedback theory presented in this paper that an electroacoustic feedback system, if implemented with sufficient spatial complexity, could turn an anechoic room into an environment similar to a reverberant room. However, there might not be a practical reason to do this, as it would require much more equipment and effort than used for precision measurements in an anechoic room. Introducing a feedback system into an already highly reverberant room was met with some success in our experimental work, the problem being that the room modes have a very powerful influence on the acoustic environment, which is difficult to compete with using electroacoustic feedback. The feedback system can substantially lengthen reverberation time, but it has only a modest effect on the steady state transfer function of the room, and so would need substantially more refinement to artificially increase modal density to the desired extent.

By targeting two frequencies and applying an electroacoustic feedback system tuned to boost those frequencies, we were able to increase the spectral density in that region by means of an amplitude increase in the nulls left by room modes. It should be possible then with a more elaborate feedback system, with many more parallel filters centred on other low amplitude frequencies, to similarly boost more of the low frequency spectrum.

CONCLUSION

It is conceivable that electroacoustic feedback could be used to enhance the room response of a reverberant test room, as the feedback is a similar phenomenon to room modes. This simple study has highlighted the difficulty in competing with the strength of room modes, but perhaps this could be overcome. For the measurement of sound power using the source substitution method, the introduction of artificial room modes should provide no difficulties, as the system retains its stable linear time-invariant properties (assuming that the electroacoustic system has sufficient dynamic range). However, if one was to use artificial modes in the measurement of sound absorption, the calculations would be much more involved than those used for a natural room. Nevertheless, there are probably much simpler and more effective approaches to enhancing a reverberant testing room's response than adding an electroacoustic feedback system.

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