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Improving the gain before feedback margin in Video-Teleconferencing and closed loop Electroacoustic systems

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ABSTRACT

Acoustic feedback stability is a fundamental limitation of all Public Address, Sound Reinforcement and Duplex Teleconferencing systems. Over the past 30 years, a number of techniques have been developed to help improve the gain before feedback margin. This paper reviews progress to date and demonstrates that a new class of loudspeaker, the Distributed Mode Loudspeaker inherently possesses a number of characteristics which potentially make it less prone to feedback. Initial experiments are reported which show a 4dB improvement in Feedback Margin without electronic assistance, gains comparable with most other current signal processing techniques.

INTRODUCTION

Acoustic feedback stability is a fundamental limitation of all Public Address, Sound Reinforcement and Duplex Teleconferencing systems. It is caused when a significant portion of the sound transmitted by the loudspeaker(s) is picked up by an active sound system microphone and re-transmitted round the electro-acoustic loop with increasing gain. Whereas in principle the howl point is calculable, given a knowledge of the transducer characteristics and room characteristics/acoustics, this can only be on a statistical basis. The problem is complicated by the complex transfer function that occurs between the loudspeaker and microphone due to the presence of room resonances and discrete specular reflections.

Over the past twenty years or so the theory of feedback pertaining to the real world situations as opposed to anechoic or reverberant chamber laboratory setups has grown and improved. Concurrently a number of techniques to improve the gain before feedback margin and stability of sound systems have also been devised [4-31]. Recently digital technology has been brought to bear with further potential stability improvements. Recently there has been an increasing trend to combine video and teleconferencing facilities in individual PC workstations. The workstation however, is a fairly hostile acoustic environment for an electroacoustic closed loop system to operate in. Strong local reflections are created by the interface between the system loudspeakers and the computer monitor and desktop for example. The process and resultant effects on the acoustic response are shown in Figures 1 & 2. Furthermore these same surfaces can create strong early reflections back into the microphone leading to strong comb filtering and further response anomalies. In more sophisticated arrangements, directional information and cues may be provided to listeners via complex signals based on HRTFs or via means of auxiliary loudspeakers. Strong local specular reflections can also influence and disrupt such signals.

Recent experiments with Distributed Mode Loudspeakers (DML) suggest that they undergo less boundary interaction response distortion [35] and may be less prone to acoustic feedback [36]. An additional 'passive' means of gaining improved feedback stability margin may therefore be available, effectively for 'free', as a by-product of the complex acoustic characteristics of the device.

Historical Perspective

Feedback is hardly a new problem with the first reports dating back to 1890! This, of course, was well before the first sound reinforcement systems came into being but instead related to the problem of the 'humming' telephone. This occurred if the separate transmitter and receiver of the early telephones came into close proximity with each other and created a feedback loop via the air-path between them. In 1901 it was discovered that changing the polarity of the connections to the earpiece changed the pitch of the feedback tone. Feedback research had begun. It was soon discovered that transducer characteristics and the phase lag in the system together with the air-pathlength were primary determining factors. In 1926 Harvey Fletcher [1] encompassed all the preceding work into a coherent theoretical picture and opened the way for the later in-depth studies into feedback in both electrical and electro-acoustic systems eg Nyquist (1932) [2].

FEEDBACK CAUSES and SOLUTIONS

Nyquist showed that if the loop gain exceeded unity then a system would go unstable and oscillate or 'Howlround' in the case of acoustic systems. It is often forgotten that this is in fact only the case when the phase shift is 0, or 360 degrees ie 2π or an integer multiple of this. At other phase shifts, gain greater than unity is possible. Thus two parameters are theoretically available to us to manipulate, ie gain (amplitude) and phase. However, in practice, the distance and hence transit time between the loudspeaker and microphone introduce a permanent phase shift and thereby limit this approach (ie at some specific frequency, this time/distance must correspond to 2π or a multiple thereof).

Figure 3 describes a simple acoustic feedback loop system. An acoustic signal is radiated by the loudspeaker to the listening position via path P1. A portion of the radiated signal also finds its way to the microphone via path P2. This latter path, together with the microphone, the amplifier and the loudspeaker, constitutes a closed loop through which the signal passes repeatedly.

The lower diagram in the figure represents the feedback mechanism in a more schematic form. Here the complex path P2 becomes $\overline{G}(w)$ which includes the transfer function of the loudspeaker. Similarly, complex path P1 becomes $G(w)$. The complex amplitude spectrum of the output signal (ie that at the listeners seat) is given by:

$$S'(w) = q.\overline{G}(w)\left[S(w) + G(w)\frac{S'(w)}{\overline{G}(w)}\right]$$

From this expression, the transfer function of the whole system including the effects of acoustic feedback, $G'(w) = \frac{S'(w)}{S(w)}$:

Can be calculated:

$$G'(w) = \frac{q.\overline{G}(w)}{1 - q.G(w)} = q.\overline{G}(w)\sum_{n=0}^{\infty}[q.G(w)]^n$$

This latter expression clearly shows that acoustic feedback is brought about by the signal repeatedly passing round the loop. The factor $q.G(w)$ which describes the amount of feedback is termed the 'open loop gain' of the system. If the gain is high, the spectrum $G'(w)$ of the received signal and hence the signal itself may be quite different from the original signal with spectrum $S(w)$. (For example when the loop feedback factor is high, >0.6 say, then response variations of approximately 12dB can occur (Benson [15])).

This model works well but the complex transmission path transfer function between the loudspeaker and open microphone will in practice not be known. However it is possible to determine the howl points of a system in a room by direct measurement of the loop gain and phase (Connor [13&19]).

Shroeder [4&6] in 1961 proposed the introduction of a frequency shifting network into the loop. By adding a small frequency shift into the feedback path (typically around 5Hz) each loop round trip would occur at a slightly different frequency, thus preventing the conditions necessary for oscillation to occur. Although theoretically improvements of up to 10dB should be attainable, in practice a 4-5dB improvement typically results before the process becomes subjectively unacceptable. The method is also limited to speech signals/systems as on musical signals, the effect is akin to adding significant and objectionable 'wow' to the signal. The process effectively averages the transmission response curve and thereby reduces the frequency peaks in the system transfer function. The concept was extended by Glueke and Broadhurst who devised a true phase shifting device. They report a potential gain improvement of 4dB. A more recent alternative to the approach was reported by Elko and Goodwin in 1992 [29]. In this case they employed a special loudspeaker array with modulated directivity to 'average' the response peaks in the transfer function by causing the modal excitation of the room to be time-varied.

Although Frequency Shifting is still in limited use (often erroneously called phase shifting) these techniques have tended to be generally replaced by frequency response equalisation and narrow band filtering.

In 1955, Snow [3] demonstrated that any small anomaly in a sound system frequency response was greatly exaggerated when the system was near to feedback. He suggested that notch filters could be installed to overcome particular peaks in the response and so reduce 'singing' as the critical feedback gain was approached. The Boners extended this concept in around 1965 [8&9] with their patented room tuning process which employed multiple, narrow band notch filters, each constructed from L-C networks in situ. They theorised that the normal modes, when excited by a sound system loudspeaker and sensed by a microphone, became self-oscillatory, as the system gain was increased. Their approach was to firstly flatten the overall system response and then null out the room mode effects with the notch filters. Sound system gain and sound quality were both reported to improve dramatically.

Davis [21] later found that he could obtain similar results by equalising the overall room mode envelope response rather than the discrete room modes themselves (Altec Acoustivoicing). This was accomplished by using a set of contiguous, combing filters spaced at 1/3 octave intervals – the precursor to the 1/3 octave graphic equaliser. The ease of use of the latter approach caused it to become the most widely adopted method of feedback control (eg Mapp 1985 [27]). By flattening the complete system response and bringing the overall level up to the feedback safety margin limit, the overall gain (level), spectral response and sound quality and intelligibility can be improved. Interestingly, the advent of modern, digitally controlled parametric filters and precision, high definition, time selective analysers, has brought about a new interest in narrow band notch filtering. (Particularly now it is understood that it is the early reflections rather than the room modes that generally are the cause of feedback in combination with the overall acoustic power response of the system). A combination of both 1/3 octave, broadband graphic equalisation together with narrow band notch filtering (approximately 1/20 or even 1/50 octave) is becoming popular [30]. However, a skilled operator and sophisticated Time/Frequency domain analysis equipment is required to tune the system and optimise the response. As digital parametric filters improve and become less expensive/more available, their improved accuracy is likely to cause them to supersede 1/3 graphic equaliser filtering in many fixed installations particularly with the advent of intelligent, self setting digital notch filters for automatic feedback suppression/gain optimisation. Depending upon the acoustic power response of the system components (the loudspeaker in particular) 5-8dB of improved gain before feedback margin can typically be obtained.

A new approach recently reported [31] uses modern DSP techniques to produce an adaptive filtering technique combined with a frequency shifter. The claimed improvement in feedback margin is 6-20dB, though this is highly dependent on the local acoustic conditions and is not suitable for use with music signals.

Unfortunately, such complex approaches cannot be afforded in the simpler workstation type tele/video-conferencing systems. However the appropriate selection of the microphone and loudspeaker are important and can have a significant effect. For example, directional microphones eg Cardioid / Supercardioid etc can have the ability to reject sound arrivals by approximately 15-20dB depending upon the angle of incident sound. However, this is only the case for direct sound components. The general reverberant or diffuse field will only be rejected by up to 6dB. (This is also usually highly frequency dependent, particularly with the cheaper forms of microphone often supplied as a part of a workstation teleconferencing systems. Furthermore, in the authors' experience, the directional advantage is frequently negated by reflections, either from the user or from the local surrounding surfaces, impinging directly into the front of the microphone. Directional loudspeakers of course may be used in certain sound reinforcement applications, but often in distributed sound systems or many conferencing/teleconferencing situations, the microphone has to be located within the loudspeaker's coverage pattern. Here, in the author's experience, it is generally advantageous to employ devices with smooth frequency responses and essentially flat power output characteristics (see also Queen [18]). The use of dipole sources (either as specific units or formed from combinations of loudspeakers in situ can also be effective (Jones [17]) and conferencing systems have also been developed using this technique.(Jaffe [ ])

From the foregoing brief review, the following points can be summarised:

- Feedback is caused when a significant portion of the sound is transmitted by the loudspeaker(s) is picked up by an active sound system microphone and re-transmitted round the electro-acoustic loop with increasing gain.

- Although both the amplitude and the phase of the feedback signal (loop gain) can theoretically be adjusted, in practice the mic-LS separation produces an initial non adjustable phase shift which will set the first potential feedback frequency.

- The response of the signal fed back to the microphone is complex, being modified by local specular reflections and room mode excitation.

- A flat acoustic power response will generally minimise potential feedback and reduce the reliance on corrective techniques.

- Directional Microphones and Loudspeakers can be employed to reduce direct component signal pickup. However, in many situations it is the diffuse field or discrete reflections that limit the overall gain.

- A number of electronic techniques have been developed to help overcome feedback and improve the potential feedback margin of a system. These include continuously Frequency and Phase shifting the signal or array beam dithering to smooth the peak response. Improvements of around 4-5 dB are possible but the techniques are generally not permissible with music signals. Employing a DSP adaptive filter in conjunction with frequency shifting can further improve the gain but is highly room acoustic dependent.

- Equalisation of the envelope response is by far the most common approach to feedback improvement and can be very effective when carried out in conjunction with careful notch filtering. The recent development of digitally controlled stable notch filters and automatic feedback frequency detection and filtering techniques is enabling improvements of 5-8 dB plus to generally be obtained. A skilled operator is generally needed.

Bearing in mind the above causes of feedback, it struck the author that Distributed Mode Loudspeaker technology might offer advantages over conventional types as a number of the inherent properties and characteristics would seem to automatically exhibit some of the features known to reduce or ameliorate potential feedback. These include non-coherent radiation and reduced boundary interactions, complex phase response, potentially flat power response, dipolar radiation characteristics and simple means to form cardioid or other directional radiation characteristics.

A number of initial investigations were therefore carried out to see whether DML technology could offer any inherent advantages. However, before describing these, a brief introductory review to the DML will be given.

INTRODUCTION TO DISTRIBUTED MODE LOUDSPEAKER

The distributed mode loudspeaker has recently been extensively described. (eg [32,33,34]) but may be summarised as follows :

The device is generally a thin (0.2 to 0.6 cm) planar panel of high stiffness and low mass, generally constructed as a composite, i.e. a lightweight core bonded to high tensile skins. Historic implementations for panel speakers have been non ideal and have suffered from restricted bandwidth, uneven power response with frequency, and low effective sensitivity. In addition modal distribution over the panel surface has been inadequate and has not been extended to the highest audible frequencies.

For the DML the research has concerned the development of an expanded theory for bending wave loudspeakers whereby the operable frequency is extended to well below the coincidence frequency by optimising the modal distribution. In addition the properties of the panels are specified to maintain this dense modal behaviour to the highest working frequencies. The equation sets have been developed for maximum modal density and define critical aspects for the panel design including geometry, stiffness, density, surface area and the precise location of the exciter(s). Interleaving of modal resonance series for several degrees of freedom is specifically demanded, while the location of the exciter(s) is also calculated for the most uniform modal drive. The lowest practical frequency limit is specified as significantly higher than the fundamental bending mode for a panel. Thus a 50Hz to 15kHz -6dB panel may have a first fundamental at 20 Hz.

In general it may be shown for an infinite panel / plate in a vacuum that the mechanical impedance is purely resistive, and that in a fluid medium e.g. air, this remains true at higher frequencies:

$$Zm = 16h^2\rho\sqrt{[Q/3\rho(1-s^2)]}$$

where for a simple plate h = the half thickness, ρ is the density, Q is the Young's modulus, and s is Poisson's ratio.

It is useful to consider the panel as a randomly vibrating area whose power depends on the square of velocity; i.e. for constant power with frequency a constant velocity is required from the exciter. This equates to constant force which is conveniently provided by an electrodynamic transducer which when acting on a resistive mechanical impedance results in the desired constant velocity drive.

A radiated acoustic power is obtained which is essentially independent of frequency, in marked contrast to the cone type speaker which has a uniform power response over just a rather limited range, termed the mass controlled region.

The DML, has the inherent ability to deliver an eight octave bandwidth in one pass, with an essentially constant, wide directivity with frequency. Figure 4 shows a typical DML polar response. Over the lower mid and mid frequency ranges the device can be seen to be Bipolar, with significant nulls at 90 degrees off axis. Above coincidence, however, the polar maintains a nominally uniform directivity and coverage. In addition the radiation is largely diffuse, appearing from a relatively large source. Recent analysis suggests that the DML can in fact be thought of as a 'diffuse point source'. Figure 5 shows a laser scan of the velocity of the surface in a typical panel which clearly illustrate the pseudo-random nature of the vibration.

Diffuse Radiation from a DML

It is possible to take a highly simplified view of the radiation properties which are likely to characterise the output of a DML. While a multiplicity of discrete modes are present at the micro level, at the macro level and with greater relevance in the practical far field or listening zone, the output of many modes will be summed at a given point. The result approximates to diffuse radiation with a much reduced phase content.

Some relevant analysis has been done on elastic plates which provides support for the concept of a wide angle output from a bending wave radiator including high frequencies.

Fiet [37], gives an accompanying relation for Zm, the mechanical impedance for the plate as:

$$Zm = Eh^3/[12(1-\delta^2)]$$

where δ is Poisson's ratio, E is Young's modulus, and h is the plate thickness. This is more frequently presented in the form:

$$Zm = 8 \cdot \sqrt{B \cdot \mu}$$

where B is the bending stiffness., μ is the density.

Azima and Mapp [35] have shown DMLs to react far less intensely with room boundaries and reflecting surfaces than conventional coherent loudspeakers. There is also evidence to suggest that a DML will excite room modes to a lesser degree. It is therefore reasonable conjecture that the peaks within the feedback loop response will be lower than with conventional loudspeakers and hence greater feedback margin should theoretically be achieved. Figure 6 shows the effects of a local boundary on a conventional small 2 way loudspeaker and a mid sized (approx 600 mm square) DML. The inherently smooth frequency response of the cone loudspeaker can be seen to collapse into severe comb filtering with resultant 6 dB response peaks. The DML however is clearly seen to suffer far less interaction.

DMLs also react far less strongly with each other than conventional loudspeakers. Figure 7 shows this reduced interaction by means of an interference spectrogram. The y axis denotes increasing frequency whilst the x axis denotes the tracking distance, normal to two nominally identical loudspeakers some 2m apart (i.e. the display shows the distance traversed by a measuring microphone, tracked across in front of two loudspeakers in an anechoic chamber). The intensity (colour) of the plot relates to the signal amplitudes. The upper trace shows the frequency interference in the spatial domain for a conventional (coherent) loudspeaker pair. A strong set of uniform peaks and troughs can clearly be seen. The corresponding interference spectrogram for a pair of DM loudspeakers however, clearly shows far less mutual interference to be occurring. (Again the implications in terms of feedback loop response peaks can be considered).

IMPROVEMENTS IN ACOUSTIC FEEDBACK MARGIN IN SOUND REIFORCEMENT & TELECONFERENCING SYSTEMS

Figure 8 shows how the wide dispersion and diffuse nature of the DML results in a very much more uniform sound field distribution within a room as compared to a conventional cone loudspeaker. Again the implications in terms of microphone pickup and the resultant acoustic distortion of the signal round the feedback loop are clear to see. For the above reasons and based on reliable but anecdotal evidence, it was thought that Distributed Mode Loudspeakers should inherently be able to offer an improved gain before feedback margin.

DML FEEDBACK MARGIN EXPERIMENTAL INVESTIGATION

Two different setups were investigated, both based on subjective experiences with DM loudspeaker operation. The first concerned the use of a large DML panel for Sound Reinforcement purposes, the second related to a Teleconferencing application.

Case History 1

The experimental setup to emulate a sound reinforcement system application is shown in figure 9. A live microphone was setup 300 mm in front of a wall mounted 1.4m^2 DML panel. A small loudspeaker was used as a 'voice' source for the system to reinforce. The gain of the system obtainable just before feedback occurred was measured at three different room positions and the results averaged. No equalisation or other processing was employed. The experiment was then repeated using a small 3.5 inch full-range loudspeaker in a tuned enclosure. This provided a wide dispersion source with a well controlled flat axial response. The average increased gain of the DML over the cone loudspeaker was 4.7 dB taken as a broadband signal. Figure 10 compares the overall gain margins for the 3.5 inch and DML panel in terms of 1/3 octave band measurements. Although very worthwhile and comparable with other feedback margin improvement techniques, the 4.7 dB achieved was lower than that based on the potential advantages discussed above. This is currently being further investigated and will be reported at a future date. The case of two or more loudspeakers is also being researched.

Case History 2

The second setup simulated a teleconferencing situation where duplex operation allows an open microphone to be in proximity to the system loudspeakers. The microphone is therefore able to pick up sound from the loudspeakers and re-transmit it back round the system. (The modern day counterpart of the 'Humming Telephone' of 1890 which started the whole feedback issue!)

Traditionally, Video Conferencing rooms have to be acoustically treated in order to reduce the reverberation time down to around 0.3 seconds and control early reflections. This is not the case with mobile or desk top computer based systems. The growing trend towards providing sound localisation information also means that there is little scope available in protecting the microphone from the loudspeaker signal.

The test setup is shown in figure 11. In this case the loudspeaker to microphone distances were 900 mm and 1500 mm. Both a single loudspeaker and dual loudspeaker setup were investigated. In this example a good quality commercial computer loudspeaker (3 inch cone) and a smaller DML unit (0.062 m^2) were employed. The closed loop gains for the two loudspeaker setups were measured. With the single loudspeaker setup and the live microphone at 900 mm in front of the active unit, the DML provided 3.7 dB more broadband gain than the cone loudspeaker. With two loudspeakers operating together (in phase) the gain before feedback margin for the DML did not change, however, the gain before feedback margin for the cone loudspeaker decreased by 3.7 dB producing a differential gain margin of 7.4 dB in favour of the DML. In fact it can be argued that an even greater advantage was gained by the DML as it was operating in an unbaffled state and so was providing twice the coverage of the cone loudspeaker. Subjectively, a louder signal was also perceived.

Figure 12 presents the resulting in-room frequency response curves for the two loudspeakers used both singly and together. In the latter case, the interactive effects can be clearly seen. The two test loudspeakers were not equalised or processed in any way as such optimisation control are rare in this form of application and a direct 'out of the box' form of comparison was aimed for. Another interesting advantage of the DML is that due to lower room boundary / modal interaction, the signals arriving back at the microphone sound less coloured and reverberant than with a traditional coherent source. The subjective quality of the resultant system signal is therefore improved.

CONCLUSIONS

The problems of feedback in electro-acoustic systems have been shown to have been with us for just over a hundred years. As the use of tele/videoconferencing and the need for improved communications and sound systems increases, so too do the practical limitations imposed by feedback. Although techniques for improving the gain before feedback margin have been around for 30 years, they suffer from a number of operational limitations. In the case of frequency / phase shifting, improvements are limited to 4-5 dB and are permissible only with speech signals. Equalisation can bring about slightly greater improvements but requires a skilled operator and time to set up. The emerging technology of adaptive DSP filters would appear to offer the greatest improvement but may need to be employed with system equalisation for optimal sound quality and clarity. Each channel may require separate filters.

DMLs have been shown to inherently offer comparable feedback margin improvements to traditional corrective measures. By providing a flat power response, they should require little or no additional processing for optimal sound quality. This has significant advantages in conferencing / teleconferencing or simple sound system applications. Their reduced room boundary interactions and modal excitation also suggest less closed loop interaction and improved potential subjective quality.

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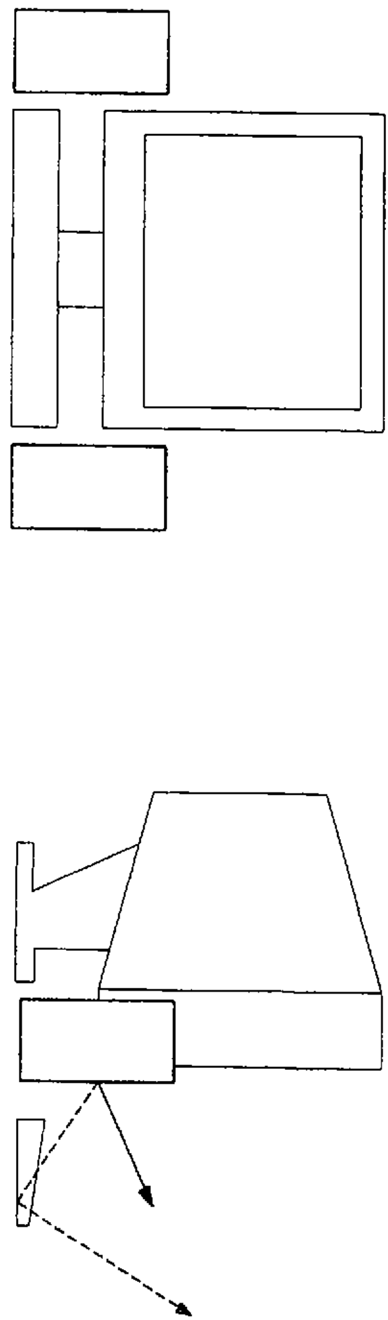
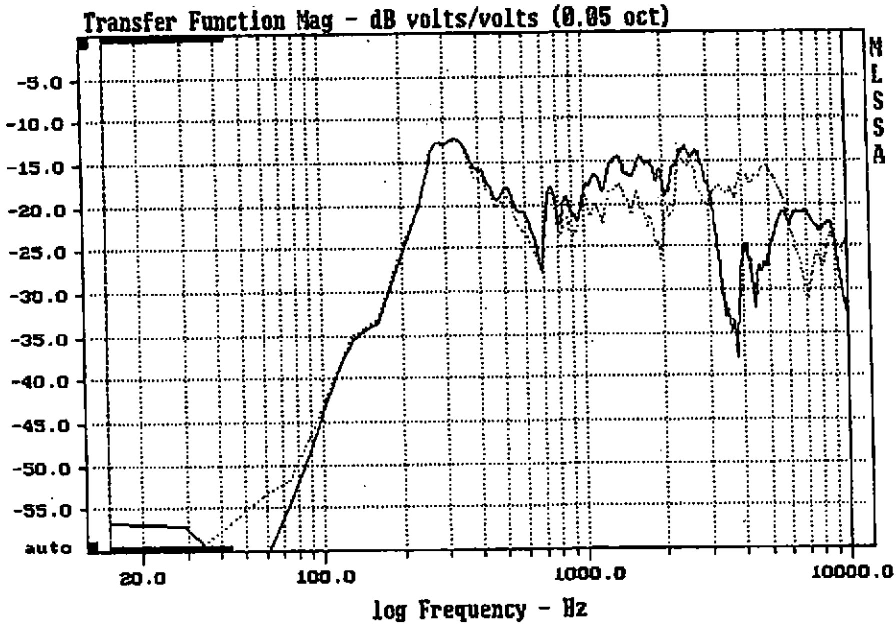


Figure 1 Typical computer workstation loudspeaker setup

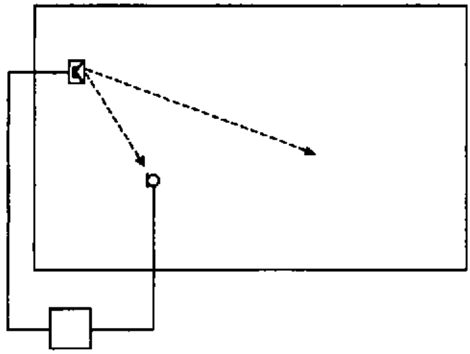


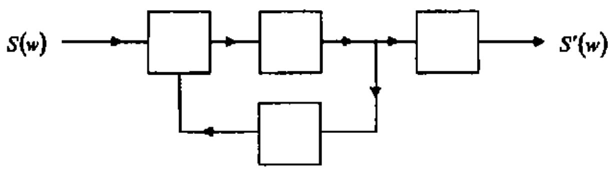
CAM LS DIRECTLY ON DESK & ON STAND (DOTTED)

MLSSA: Frequency Domain

Figure 2 Typical frequency response from in situ computer loudspeaker

Figure 3



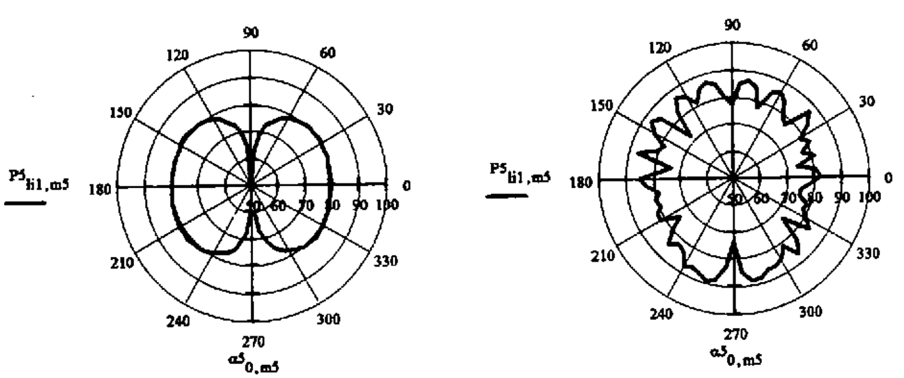


Simple Acoustic Feedback Loop System

Figure 4

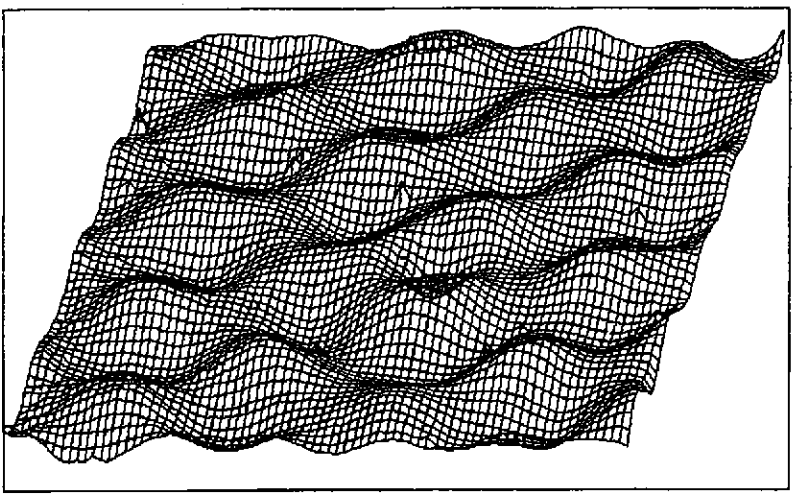
POLAR RESPONSE-DML Panel at 500Hz

POLAR RESPONSE-DML Panel at 4kHz



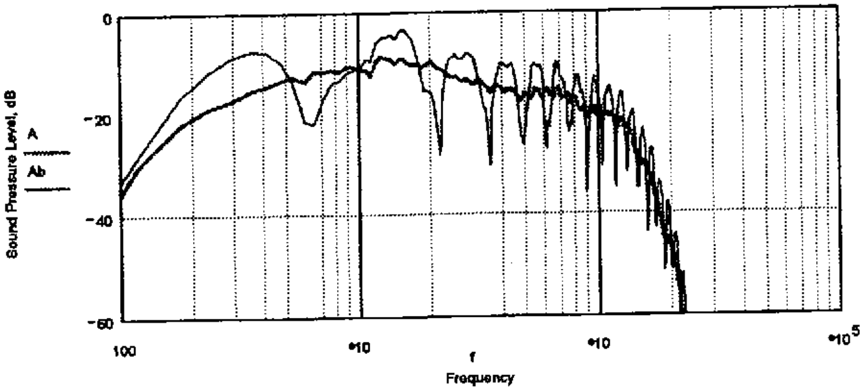
Typical DML Polar Response

Figure 5



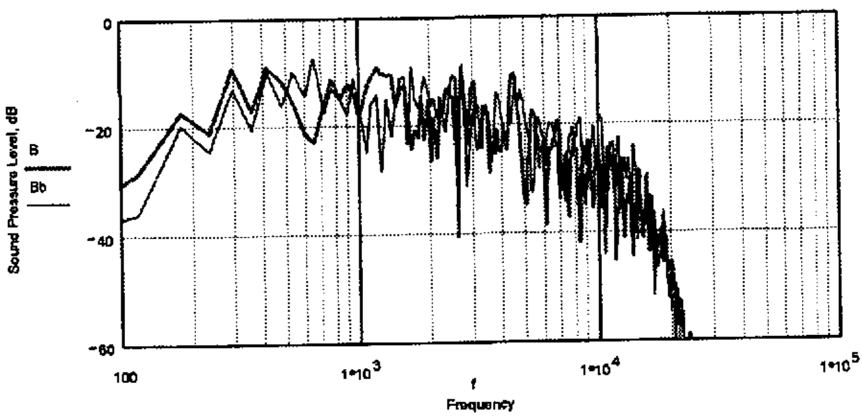
Laser Scan of DML

Figure 6



'A'
'A' + boundary

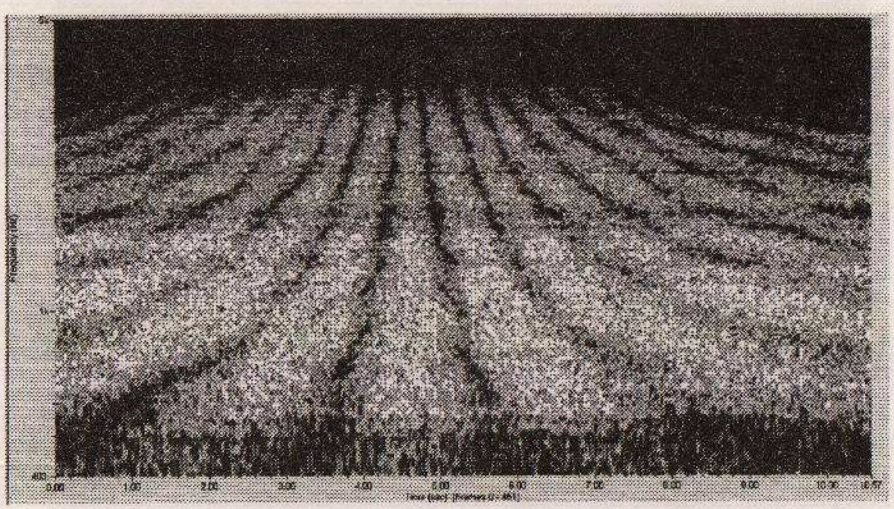
Radiator 'A' in Free-space and with boundary influence



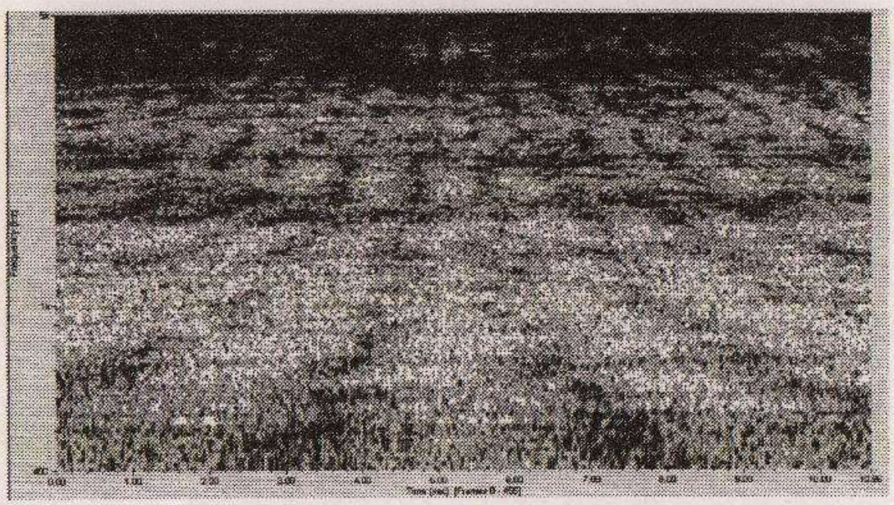
'B'
'B' + boundary

Radiator 'B' in Free-space and with boundary influence

Figure 7

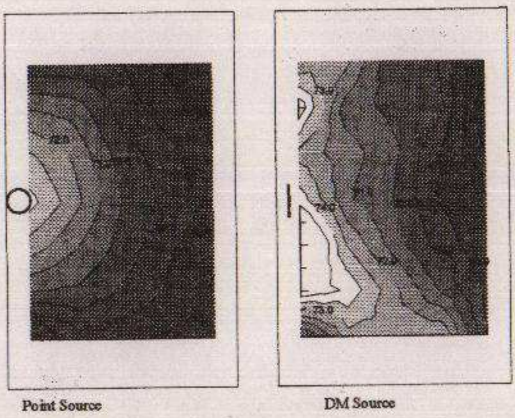


Interference Spectrogram of a Coherent Source



Interference Spectrogram of a DML Source

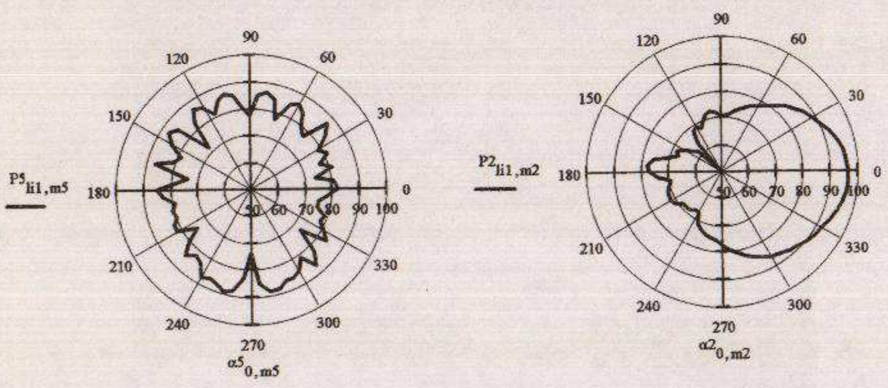
Figure 8



SPL contours, source firing across the room (f=8kHz)

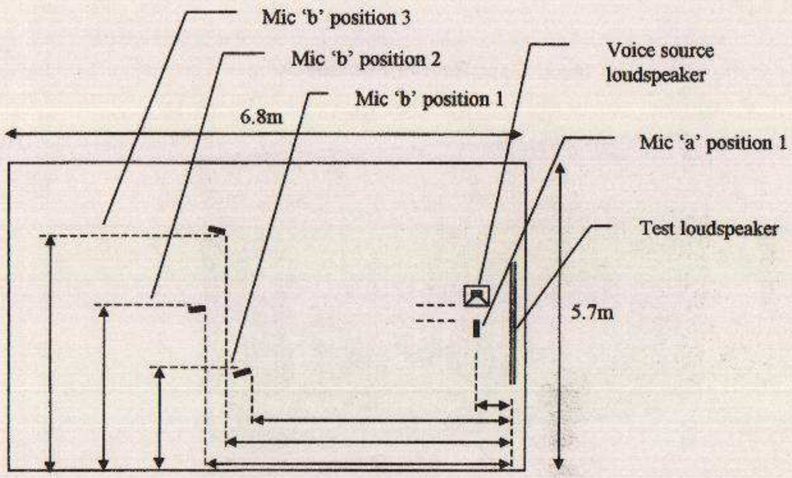
POLAR RESPONSE-DML Panel at 4kHz

POLAR RESPONSE-3.5" Test Speaker at 4kHz

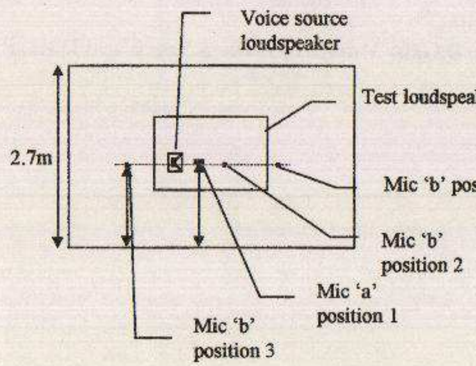


Dispersion and Diffuse Nature of the DML Verses a Coherent Speaker

Figure 9



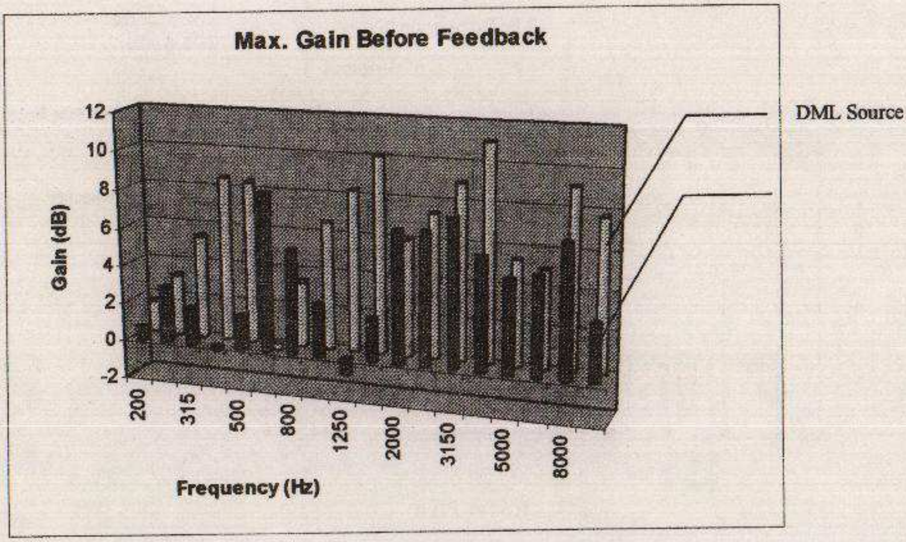
Room Plan



Room End Elevation

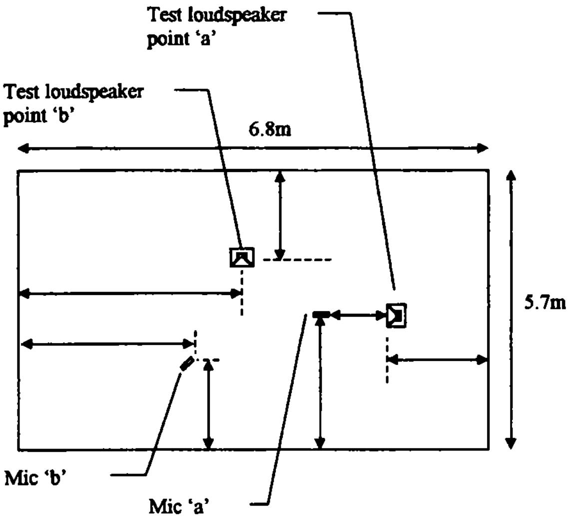
Setup to Emulate a Sound Reinforcement System

Figure 10

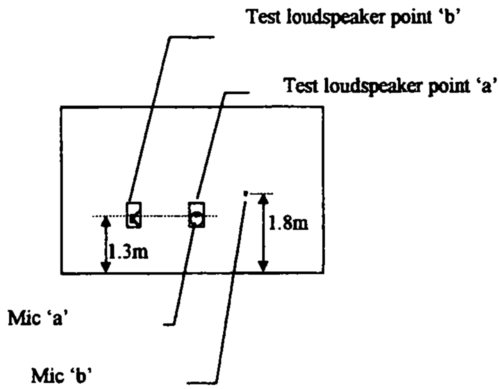


Compares the Overall Gain Margins for the 3.5 inch and DML Panel

Figure 11



Room Plan



Room End Elevation

Diagram of Test Setups

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