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BOARDROOM AND CONFERENCE SYSTEMS - A REVIEW OF BASIC PHILOSOPHIES

V.W. Dobbs (1) and P.W. Barnett (2)

(1) Spectrum Audio Ltd.
(2) AMS Ltd.

INTRODUCTION

Boardroom and Conference Systems may be sub-divided into many groups, types and indeed, principles of operation. However, within the family there are essentially two distinct approaches. They may be conveniently labelled as overt and covert systems; the group names refer not only to the visual aspect but also the acoustic performance.

Overt systems generally comprise a microphone and loudspeaking station which are placed on a table or built into seating. The microphones are often mounted on gooseneck extensions and are clearly visible; in addition they rely upon microphone-talker distances of less than 500 mm to ensure that an adequate feedback margin is maintained. In electronic terms overt systems generally operate on a daisy-chain principle and may either be manually-controlled or voice-activated.

Our main interest is in covert systems. For a system to be truly covert the microphones and loudspeakers are necessarily concealed and acoustically the system has to perform unobtrusively which generally necessitates a comprehensive electronic package as well as high performance transducers. In addition, these systems not only have to be custom-built but they generally start with the disadvantage that the microphone-talker distance is greater than 500 mm.

OPERATIONAL CONSIDERATIONS

There are two basic philosophies that can be employed:
1) Managed.
2) Automatic.

Managed Systems

Managed systems as the name implies requires manual intervention to select which microphone is to be 'live' and in general terms only a single microphone is live at any given time.

With covert systems this is of prime importance since feedback is always an attendant problem (caused by the large microphone-talker distances). The reduction in gain prior to feedback is at best $10\text{Log}_{10}M$ (where M = the number of open microphones). Clearly this expression relies upon the acoustic space acting as a continuum; whilst in practice it would be modified by the permutations and calculations of loudspeaker-microphone paths thereby providing a dramatic and often unexpected reduction in gain.

The managed aspect of the system is a matter of the electronics and of the client's requirements - most tend to be software-controlled and are therefore not only adaptable but allow queuing, priorities and identifying signals for voice-logging purposes.

lanaged systems tend to be popular under conference conditions but unpopular in he 'cut and thrust' of the boardroom. Acoustically, managed systems are the asiest to design and are able to provide the greatest degree of enhancement.

utomatic Systems
hese systems have no user-controls and are voice-activated. There are many ystems available involving a wide degree of sophistication. Whatever the ystem employed it is important to understand the restrictions imposed. Firstly e should be aware that the basic principle, in spite of manufacturer's claims, emains the same - to restrict the number of simultaneous 'live' microphones. lost manufacturers make claims that the number of live microphones is unlimited nd that the system automatically compensates for this effect. The reality of he situation is that as the number of open microphones increases, the level of oice enhancement decreases.

n addition to gain compensation, an unfortunate necessity, automatic systems uffer from a wide range of attendant problems especially when the microphone-alker distance is large. Firstly, we should understand that speech has a airly wide dynamic range (>30 dB) and as such hysteresis has to be applied to he system to avoid gating of quiet syllables. Secondly, the system has to be nsensitive to self-activation; if the loudspeaker output activates adjacent nicrophones, then the system automatically compensates for the additional gain y reducing the gain of each individual channel.

lanufacturers have been to considerable pains to reduce these effects; one nethod is to sum the output of all microphones and by use of comparitors allow only those channels that exceed this average level by a preset amount. Another nethod is to employ two microphones positioned back-to-back. Voice activation is only affected when the output of the microphone on the talker side exceeds the other by a preset amount.

Finally, even the most sophisticated and highly engineered systems are prone to switch off at the ends of words or after a long pause; when the direct component of the voice is present this is generally not noticeable. However if this signal is used to relay sounds to adjunct spaces or for voice-logging this often becomes a problem.

AESTHETIC CONSIDERATIONS

Without a doubt the most difficult problem is placement of the microphone. Although unpopular they need to be sited in the conference or boardroom table. To avoid the clutter of paper and to allow writing space, the microphone-talker distance invariably is greater than 1m.

Considerable care has to be taken to ensure that any decorative covering is as acoustically transparent as possible and that adequate equalisation is available to compensate for any adverse effects. In addition, attention is required to provide adequate vibration isolation. Fig. 1 shows alternative approaches. In voice-activated systems, it is often necessary to include a high pass filter to avoid low frequency vibration activating the microphones.

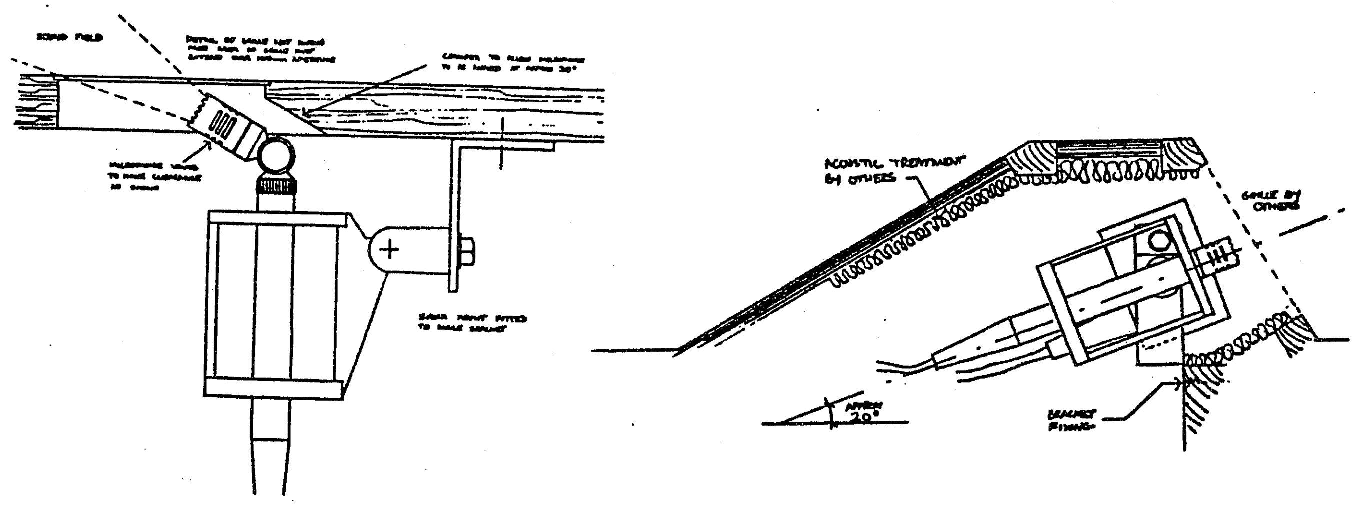

Fig. 1

The loudspeakers are less of a problem, these may either be housed in the table or positioned overhead. Care has to be taken to ensure that the loudspeakers have the required directional characteristics to improve the feedback margin.

ACOUSTIC CONSIDERATIONS

There are four basic requirements:
(i) To determine the required enhancement.
(ii) To optimise the gain prior to feedback.
(iii) To optimise the signal-to-noise ratio.
(iv) To optimise the arrival time of the reinforced signal.

Enhancement Level

The enhancement level depends upon many factors, however, in general we would expect to reduce the acoustic distance between talker and listener to around 1-3 m. In most cases we would not expect to provide for reinforcement to listeners within this distance and therefore to improve the gain stability it is usual to turn off, or preferably attenuate, the output of these local loudspeakers.

In some instances, especially where the acoustics of the space are well designed it may only be necessary to provide a high frequency element to improve the intelligibility.

Gain Prior to Feedback

Our feedback calculations are based on the premise that the sound pressure level at the microphone caused by the loudspeaker results from a large number of reflections and therefore may be approximated by the reverberent sound pressure level.

Although this model is necessarily simple, our experiences have shown that providing adequate parametric equalisation is applied to the system, feedback calculations which of necessity must ignore the effect of room modes, give surprisingly good correlation with measurements.

In addition it does provide useful data on which to base the system design and indeed provides the best indication possible regarding the performance and feasibility of the system.

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We adopt the following procedure:
Calculate the sound power level of each source to produce the required sound pressure level at the listener. Modify the Pwl to take account of local absorption (making allowances for spillage etc.) and hence calculate the total sound power output of the system.

From the system sound power output the expected sound pressure level at the microphone may be calculated. The calculated system sound pressure level is then compared with the expected talker level at the microphone. The difference is the gain prior to feedback.

Fig. 2 shows the expected typical sound pressure level of a normal talker at 1m.

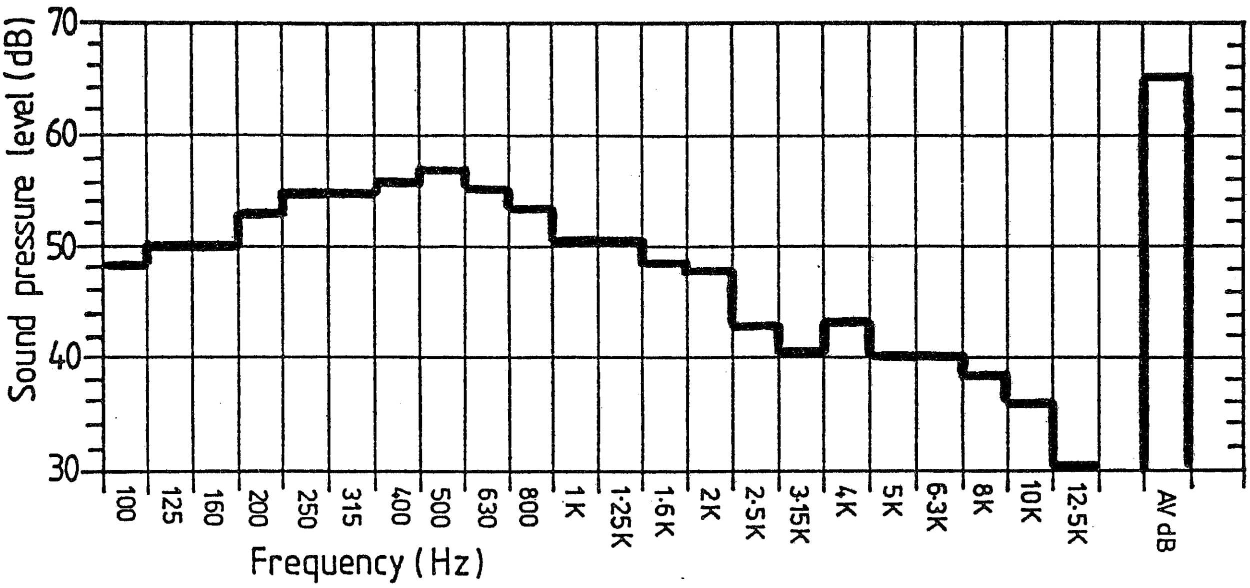

Fig. 2

It is worth noting that ideally the feedback calculations should be carried out in ⅓ rd octaves.

<u>Signal-to-Noise Ratio</u>
Under most circumstances signal-to-noise ratio is not a problem. However, if a system is required as a result of a high ambient noise level then it must be remembered that the ultimate signal-to-noise ratio attainable cannot be better than that at the microphone.

It is of course necessary to inspect this with due regard to the spectral content of both the signal and the noise.

We would advocate that for boardroom and conference systems the signal-to-noise perceived by the listener should not be less than 10 dB in any ⅓ rd octave (200-4 kHz) preferably it should be greater than 15 dB.

The signal-to-noise ratio should be calculated at the listener in which case the total system noise output is summed with the ambient. It should be noted that the total system noise output is the sum of the amplified ambient and any electronic noise.

In most cases a 10 dB signal-to-noise ratio is not a problem, however considerable care has to be exercised in automatic systems as it is possible to quickly degrade this parameter if the system is not correctly adjusted.

For example, as the number of 'live' microphones increases the overall gain and hence the noise output remains constant. The maintenance of gain parity is at the expense of individual channel gain and therefore the signal level will be reduced.

Arrival Time of Reinforced Signal

Clearly the arrival time and amplitude of the reinforced signal is important to ensure that it falls within the Haas criteria.

This is particularly important if the reinforced signal has been spectrally modified to improve the intelligibility.

ELECTRONIC CONSIDERATIONS

It is of paramount importance to ensure that both input and output transducers are of good quality and that their characteristics are well known and understood.

There are two basic principles that may be applied to the operation of the system:

(i) Routing
(ii) Bussing.

Routing

Routing is generally carried out with software-controlled switches; in this case the input signal may be directed to the required outputs at the required amplitude in the required time. Fig. 3 shows an example of a system utilising routing.

Bussing

Bussing is a fixed matrix of inputs and outputs. Signal paths may be amplitude and time conditioned.

In general terms bussing is less flexible and requires individual signal processing on each channel.

Fig. 4 shows the block diagram of a bussed system installed in the boardroom of the Hong Kong and Shanghai Bank.

BOARDROOM AND CONFERENCE SYSTEMS - A REVIEW OF BASIC PHILOSOPHIES

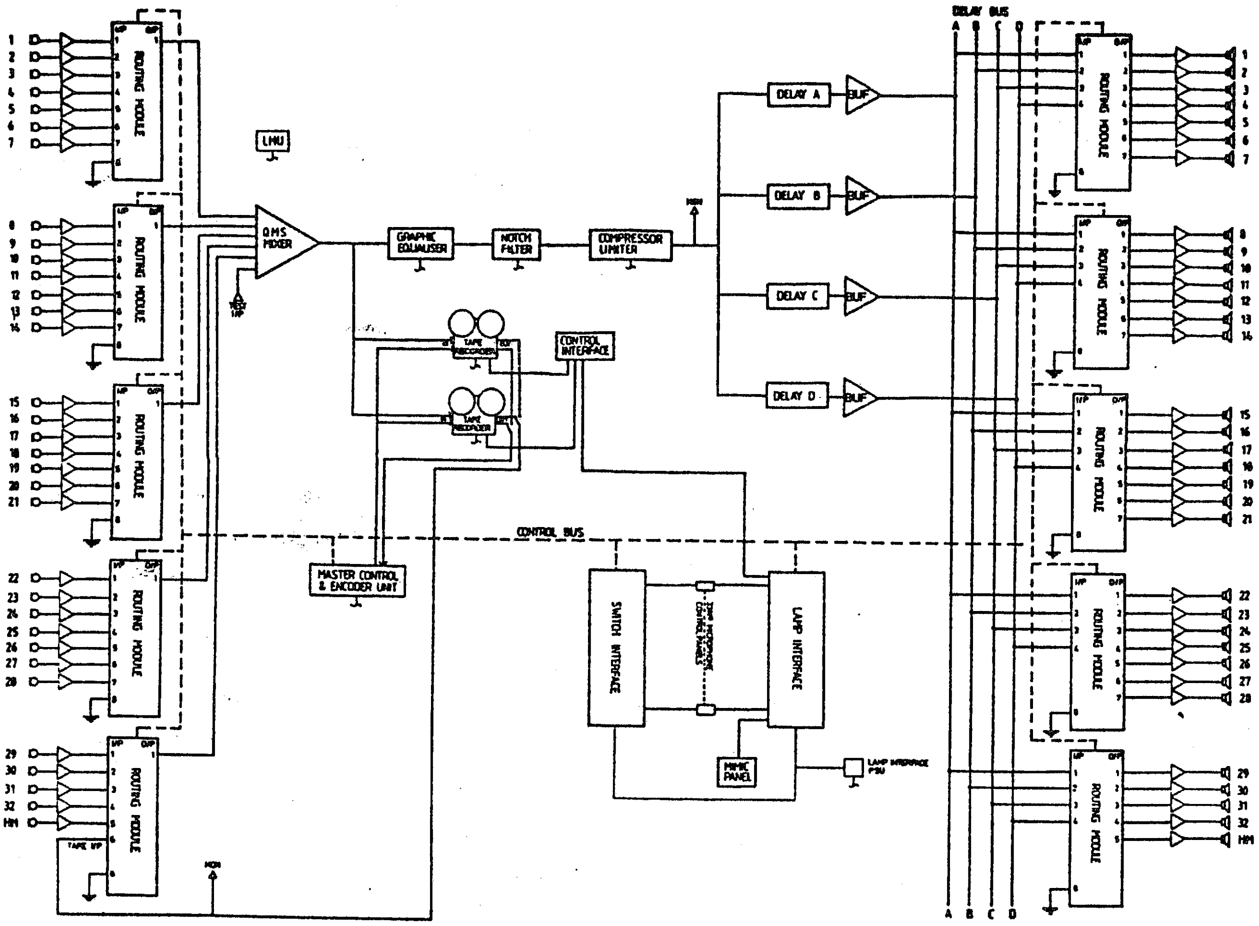

Fig. 3

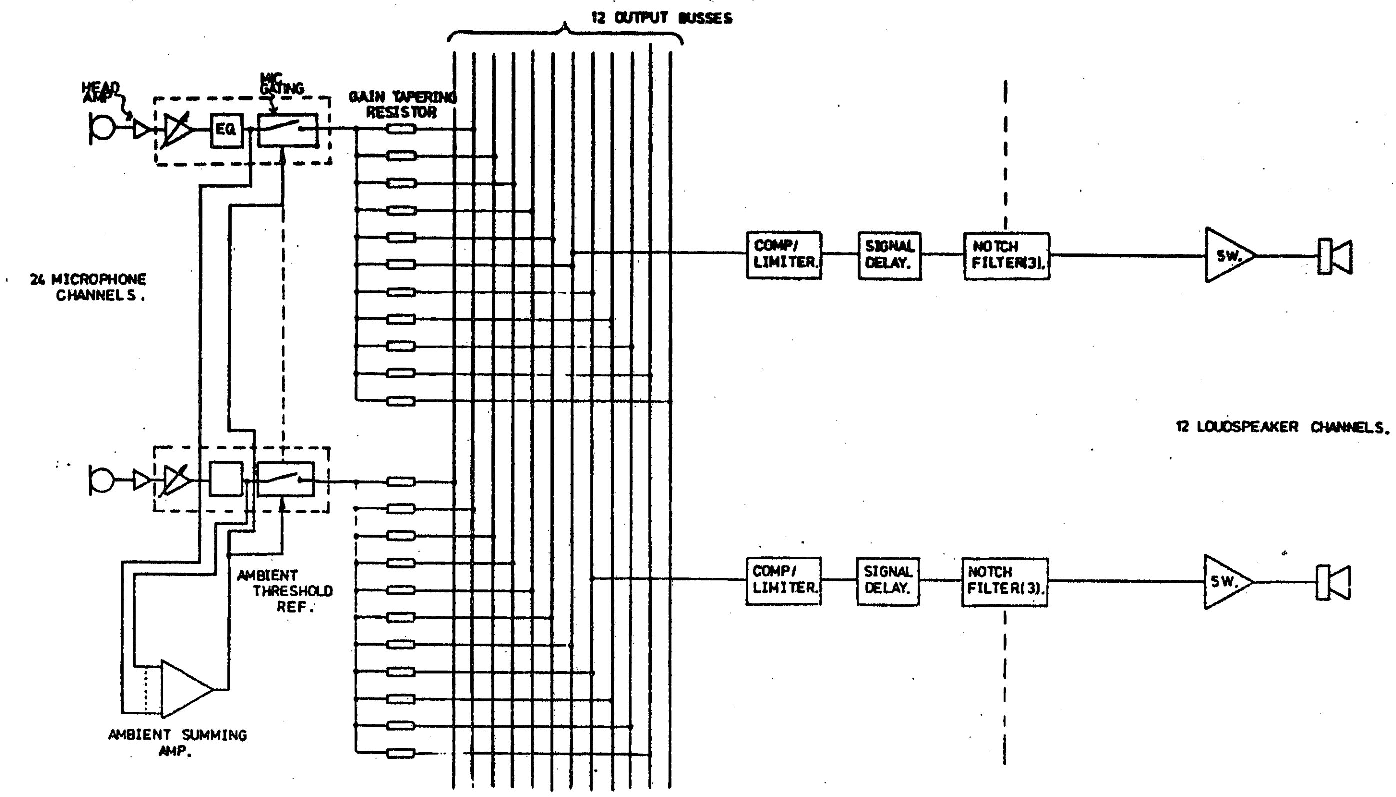

Fig. 4

CONCLUSIONS

The requirement for such systems is increasing - interior designers and architects are unwilling to settle for systems whose components may detract from the aesthetics of the space.

Unfortunately, each space and hence each system is different, techniques and technology change to suit the challenge. Experience cannot therefore be indescriminately extrapolated and considerable reliance must be placed upon theory and calculations.

We are all too aware of the limitations and assumptions of the attempts to quantify the salient design parameters. Our endeavour is to reduce this uncertainty by conducting rigorous tests on the system transducers and comparing design calculations with measurements made on the installed systems.

Our concern and possible preoccupation with theory and attention to detail results from the fact that these systems are installed in some of the worlds most prestigious buildings and are used by persons who may not be sympathetic to failure.

ACKNOWLEDGEMENT

The Hong Kong and Shanghai boardroom system was specified by Arup Acoustics.

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