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LOUDSPEAKER DESIGN FOR SPEECH INTELLIGIBILITY IN REVERBERANT SPACES

by

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Introduction

Intelligible speech reinforcement in Churches and Cathedrals has always been a daunting task for the acoustic engineer. The excess reverberant energy that predominates in such places results in discrete sounds becoming confused and tangled together, making it impossible to distinguish between the individual syllables that constitute normal speech. Speech comprises syllables of some 200ms in length, and it is necessary that the energy from one syllable falls to a low level before the onset of the next syllable if the ear is to distinguish each one successfully. Practical experience has shown that if speech is to be reasonably intelligible then the level of sound from each syllable must fall by at least 10dB before the onset of the next syllable. In anechoic space the peak signal to reverberant energy difference can approach 50dB, but in very reverberant spaces it would be more usual to find peak signal to reverberant energy relationships of only 5dB or so. Such long reverberation times are not normally encountered in practice, but there is one special case where reverberation times of this order are not uncommon, and that is in Churches and Cathedrals.

Due to the very nature of their construction Churches and Cathedrals have to be considered hostile from the sound reinforcement viewpoint. The complete absence of any effective sound absorbent within such places results in reverberation time/frequency response characteristics that often exceed 7 seconds at low frequency and rarely fall below 3 seconds at high frequency. Quite often the ornate stonework and carvings and the sheer volume of the enclosed space results in a reduction in the reverberation time at high frequency, but the overall reverberant energy in such places is still very high. Two typical Cathedral type reverberation time characteristics are shown in Figure 1, from which it can be seen that the mid-frequency reverberation time at Canterbury Cathedral is some seven seconds and at Westminster Abbey is six seconds. By comparison the mid band reverberation times of even the largest concert halls rarely exceed 2 seconds.

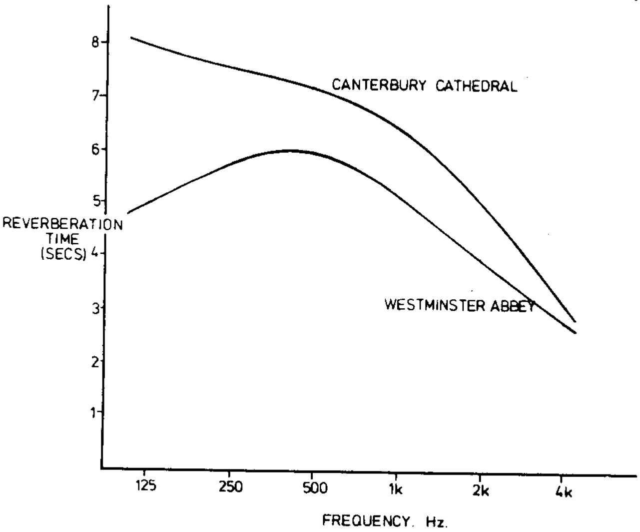

Figure 1
Reverberation Time/Frequency response characteristics in the Nave at Canterbury Cathedral and Westminster Abbey

To minimise excitation of the reverberant space it is usually arranged that any speech reinforcement loudspeakers used are fairly directive and that they radiate sound over a seating area that has a high absorption coefficient.

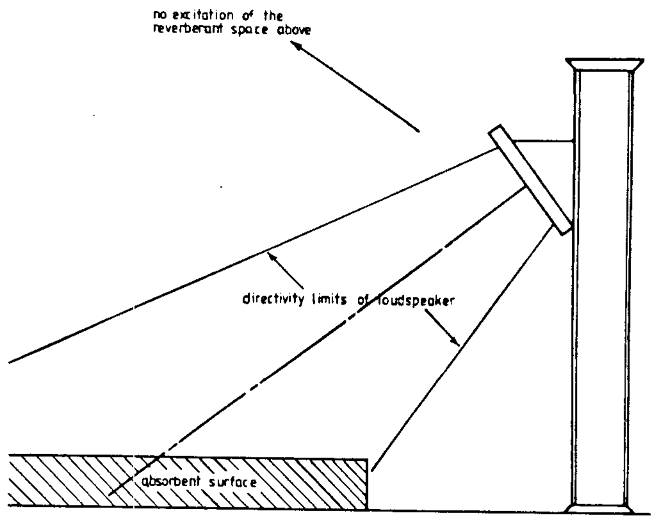

Figure 2
Accepted Method of Positioning Column Loudspeakers to minimise excitation of reverberant space

It is normal under these circumstances to use column loudspeakers, the theoretical directivity characteristics for a typical example being illustrated in Figure 3. Because the vertical sound source dimension of the column is large the directivity characteristics in that plane are fairly narrow, whilst in the horizontal plane the loudspeakers are reasonably non-directive, the lateral response usually being

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controlled by prudent choice of drive unit. For most normal column loudspeakers the vertical side lobe intensity is generally only 10-12 dB lower than the major lobe intensity. Further reduction in side lobe intensity can be achieved by using a taper feed column (increasingly smaller levels of signal being applied to the drive units the further they are from the centre unit) though this technique is limited by dimensional and economic constraints.

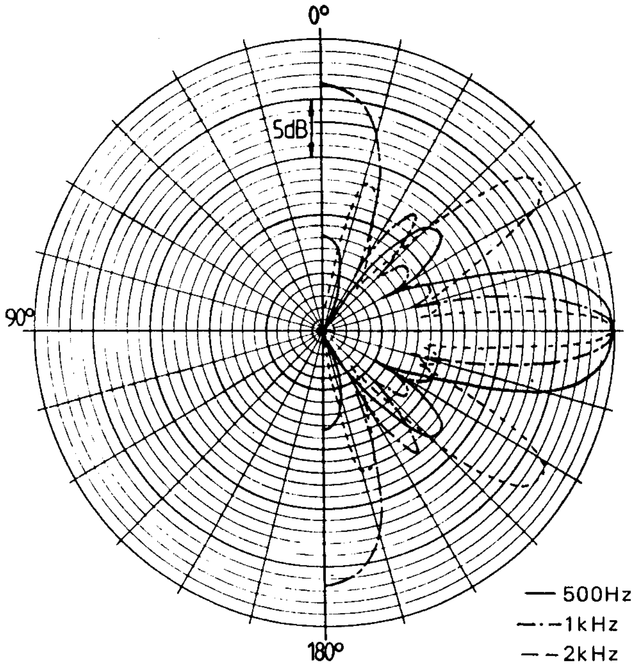

Figure 3
Theoretical polar response of a normal Column Loudspeaker at 500 Hz, 1kHz and 2 kHz (5 drive units at 12" spacings)

As column loudspeakers are deliberately directive in the vertical plane it is necessary to angle the loudspeakers downwards towards the listeners to achieve reasonable acoustic coverage and though this is technically adequate such positioning is often criticised by the user and is often thought to spoil the aesthetic quality of the Church or Cathedral. Moreover the large side lobes of a standard column result in substantial excitation of the reverberant space in the quadrant above the normal plane.

Origin and Operating Principle of the Abbey Mark X Loudspeaker

Some years ago this problem was examined by the scientists at the Building Research Station, with particular emphasis being placed on a new speech reinforcement system that was to be installed in Westminster Abbey. Taylor and Keeler proposed and implemented a novel and very elegant solution to the problem. By carefully controlling the amplitude and phase of the electrical signal fed to each of a row of drive units in a column loudspeaker the directivity in the vertical plane can be adjusted to provide a downward radiating lobe without the need to angle the loudspeakers. Briefly the design calls for a line of equally spaced loudspeakers, the signals to which are decreased progressively from the centre unit in the series of ratios 1, 1/3, 1/5, 1/7 etc with the units in the lower half of the line being connected out of phase with the units in the upper half of the line. In addition a central drive unit is excited in positive phase quadrature with respect to the equally spaced units and is fed with a signal some 4 dB greater than the signal fed to the two adjacent central drive units. Additional drive units are also used to complete the array, the amplitude and phase shift specification for which are discussed in more detail later in this article.

The theory and operation of the loudspeaker may be considered analagous in many ways to Fraunhoffer diffraction. Imagine a row of loudspeakers positioned in a line with each loudspeaker separated from its neighbour by an identical distance, and imagine a listener positioned a large distance from the line of loudspeakers at an angle θ from the normal (Figure 4).

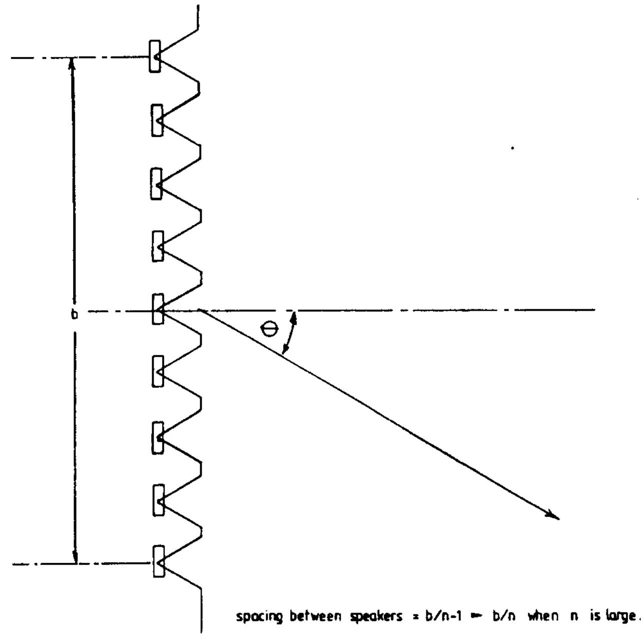

Figure 4
Theoretical line source loudspeaker array

The path difference at the receiving point between the signals arriving from the two adjacent loudspeakers is given by

$$\frac{b}{n}\sin\theta \qquad \text{..........(1)}$$

and the phase difference (ψ) is given by

$$\psi = \frac{b}{n}\sin\theta \times \frac{2\pi}{\lambda} \qquad \text{..........(2)}$$

At the receiving position the intensity of the perceived signal is equal to the vector sum of all the different signals arriving from each of the loudspeakers. If for example the phase of the signal fed to loudspeaker No. 5 is delayed by α then the phase difference of l/s'5 relative to its neighbour is given by:

$$\frac{2\pi\frac{b}{n}\sin\theta}{\lambda} + \alpha \qquad \text{..........(3)}$$

and if the amplitude of the signal fed to loudspeaker 5 were A_5 then the final relative sound pressure level at angle from the loudspeaker would be:

$$= A_5 \cos \frac{2\pi\frac{b}{n}\sin\theta}{\lambda} + \alpha \qquad \text{..........(4)}$$

and the final sound pressure level from the combined loudspeakers would be:

$$A_\theta = \sum_{n=1}^{n=n} A_n \cos \frac{2\pi\frac{b}{n}\sin\theta}{\lambda} + \alpha \qquad \text{..........(5)}$$

This equation is usually written in the form:

$$A_\theta = \sum_{n=1}^{n=n} A_n \cos \frac{2\pi\, b_n \sin\theta}{\lambda} + \alpha \qquad \text{..........(6)}$$

where b_n is the distance of the nth loudspeaker from the reference point, usually taken to be the central loudspeaker of the array.

By adjusting the amplitudes and phases of each of the signals fed to the individual drive units as described earlier it is possible to dramatically tailor the vertical polar diagram of the loudspeaker. Taylor (ref 1) has shown that if the amplitudes and phases of the drive units are as shown in Table 1 below then a strongly downward directed lobe will appear in the lower quadrant of the loudspeaker.

Table 1
Amplitude and Phase shift data for the Mk X Loudspeaker

LS No.	Relative Amplitude	Phase of Signal	Position (ins)
1	1	+180°	-24
2	2	+ 90°	-20
3	3.33	+ 90°	-12
4	10	+ 90°	- 4
5 (Central drive unit)	16	0°	0
6	10	- 90°	+ 4
7	3.33	- 90°	+12
8	2	- 90°	+20
9	1	+180°	+24

Using equation 6 with the parameters listed in Table 1 it is possible to produce computer predictions of the resulting distribution of sound in the vertical plane of the loudspeaker, the results being shown in Figure 5.

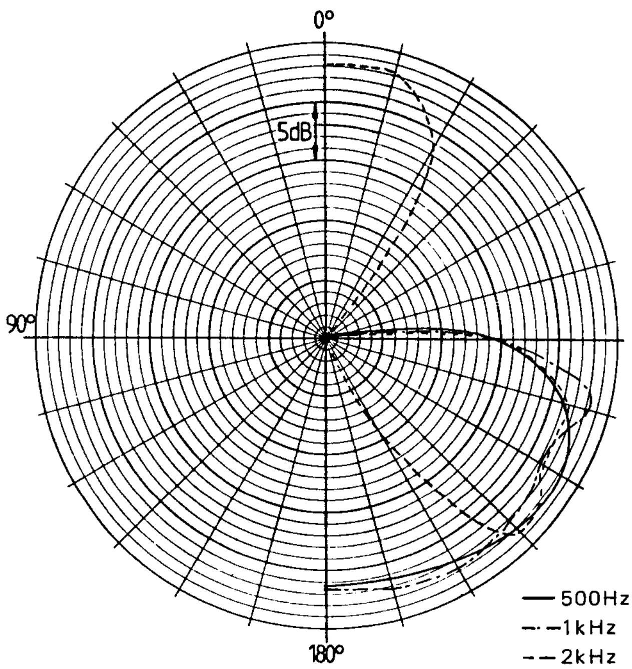

Figure 5
Theoretical polar responses of Abbey Mk X loudspeaker at 500 Hz, 1kHz and 2 kHz

Theoretically the directional properties do not extend beyond 1.6 kHz as the spacing between each drive unit becomes large in relation to the wavelength of sound being radiated. As noted earlier however the theory assumes (incorrectly) that the drive units radiate uniformly in all directions and at all frequencies. Theory would predict a very strong upper lobe at 2 kHz but because the output from the drive units is substantially down at such acute angles to the normal the upper lobe is not fully realised, resulting in some useful directivity being achieved to beyond 3 kHz. In fact the reproduction of sound at very high frequencies is of little significance in the practical instance as it has been found that the overall bandwidth has to be limited to 300 Hz - 5 kHz for best speech intelligibility. The excessive low frequency reverberant energy that could be produced if bandwidth limiting were not used would impair intelligibility and a balanced overall spectrum around 1 kHz is always to be preferred once a decision favouring a high pass filter has been made.

Description of the Production Abbey Mark X Loudspeaker

AIRO has recently completed a thorough re-appraisal of the Abbey loudspeaker systems, the latest design now incorporating a number of revisions. A need was felt for the latest drive units to have improved power handling capability and a smoother frequency/phase response. The design requires that the three central drive units radiate the majority of acoustic energy, the remaining drive units altering only the polar diagram of the loudspeaker and handling between them only a fraction of the total electrical power fed to the transducers. An economic solution might have been to use good quality drive units for the central three speaker positions and 'economy' drive units for the remaining positions. However this is not a practical solution as the theory required that all drive units provide an identical performance in respect of frequency response, phase response and directivity, thus similar units have to be used throughout the system. A loudspeaker will of course have its own inherent phase response which can vary dramatically with frequency and there is no guarantee that the radiated signal from the loudspeaker will be in phase with either the driving current or the voltage across the loudspeaker terminals. However this is of little significance in this instance as it is the relative phase response between drive units which is of importance. In the end it was decided to use expensive but good quality circular drive units having an excellent frequency response, low distortion and good power handling capability.

Another problem was to devise some means by which the electrical signal could be shifted in phase by 90° without altering the frequency response of the signal. The theory called for two audio signals, each with the same frequency response but with a phase difference over the usable frequency range that remains constant at 90° . The problem was solved by using the electronic circuit shown in figure 6.

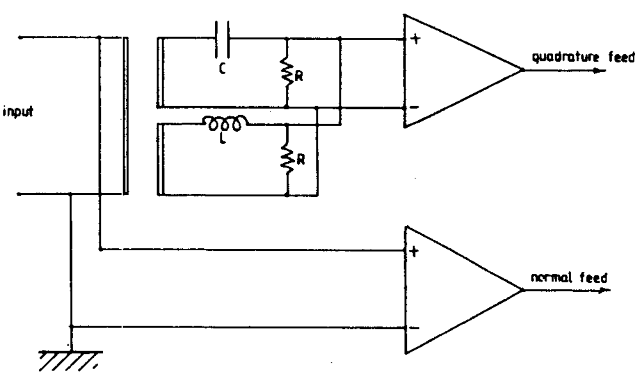

Figure 6
Electronics network used to derive a constant 90° phase shift signal

The audio signal is fed to a transformer, the two secondaries of which are connected to a resistor and inductor in series and a resistor and capacitor in series respectively. The voltage appearing across the inductor rises proportionally with frequency whilst that across the capacitor falls with frequency. The signals across the capacitor and inductor are 90° out of phase with the signal fed to the transformer and 180° out of phase with respect to each other. If the circuit is connected as shown in figure 8 to a differential amplifier then the signal at the output of the amplifier is essentially the same as the signal fed to the transformer but with a constant 90° phase shift at all frequencies of interest. Some errors in the amplitude response do appear at extreme high and low frequency but are not significant as the frequency response is bandwidth limited. With careful design the quadrature response can remain esssentially flat(±1dB) from 300 Hz to 3kHz with a phase response that is within ±10° of the phase shift required.

The measured polar responses of the final design in the vertical plane are shown in Figure 7 from which it can be seen that the response mimics closely the predicted responses shown in Figure 5 except at high frequency (2kHz) where the upper quadrant lobe is usefully suppressed.

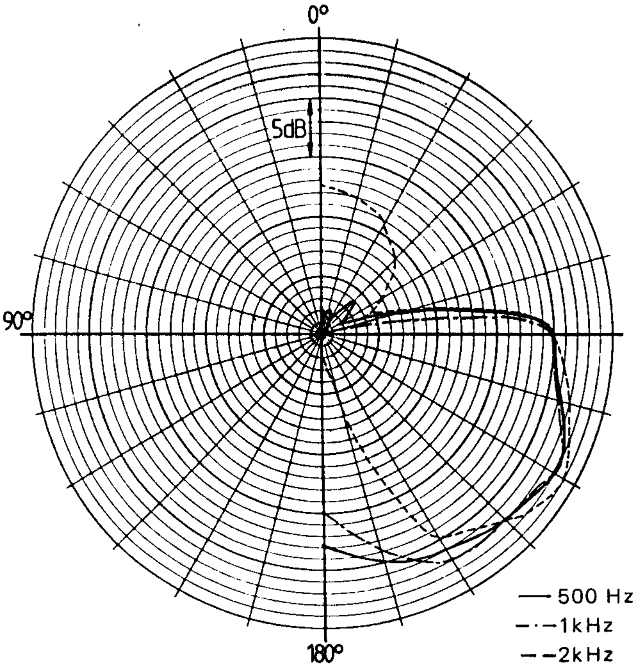

Figure 7
Measured polar responses of Abbey Mk X loudspeaker at 500 Hz, 1 kHz and 2 kHz

Some reduction in upper quadrature rejection is inevitable due to phase errors in the quadrature signal feed to each loudspeaker and to small variations in the performance of the drive units in respect of sensitivity, frequency response and phase response. Such variations have to be kept to a minimum by careful choice and selection of the drive units, another reason why the latest circular drive units were preferred to the original elliptical units.

The final design can achieve sound pressure levels in excess of 100 dB in the usable frequency range at a distance of 1 metre from the face of the loudspeaker on the major lobe. All the phase shifting and power amplification electronics are fitted in an on-board electronic package and so both DC power and an audio signal feed has to be routed to each loudspeaker using twin core screened cable. It is, however, theoretically possible to superimpose the audio signal on the DC line if only normal 100V line loudspeaker cable is available.

Concluding Observations

To summarise, the main advantages of the Abbey Mk.X column loudspeaker can be listed as follows:-

a) The loudspeaker can be fixed vertically against the supporting wall or column without tilt, resulting in a more visually acceptable installation.

b) Providing the column is visible to the audience its mounting height above ground level is not particularly important. It is usual to mount it some 5 metres or so from the ground to ensure reasonable sound propagation and keep it out of reach of prying hands.

c) A large proportion of the total sound energy radiated is confined to the lower quadrant, thus minimising excitation of the reverberant space above.

d) The setting up time for portable columns is minimal compared with the time consuming procedure required to optimize the height, position and tilt of a conventional portable column.

Figure 8
Abbey Mk X Loudspeaker at Canterbury Cathedral

The loudspeakers are very discreet in appearance, the units being less than 3 inches thick and only six inches wide, and are normally supplied covered in an open weave fabric material similar in colour to the stonework on which the loudspeaker is to be mounted. Together with their slim dimensions and discreet coverings the loudspeakers can be quite unobtrusive, a consideration that can be of great importance in avoiding the disruption of fine architectural lines which feature in many Church and Cathedral interiors.

References

1. Taylor, P.H. Patent Spec. No. 1456790 'Improvements in sound radiating apparatus and systems' February 1975

2. Bailey, I. BRS Note 'New Designs for Column Loudspeakers' 122/75