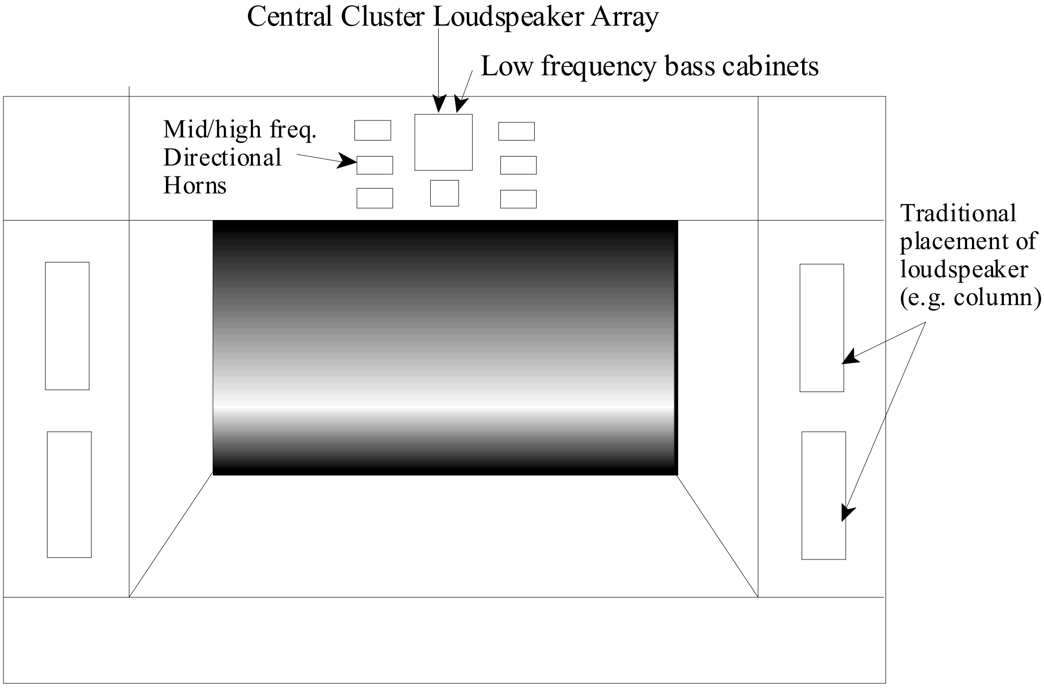
Level 3 Module AGA03304 Electroacoustic System Design
Solution to Examination Questions January 2000

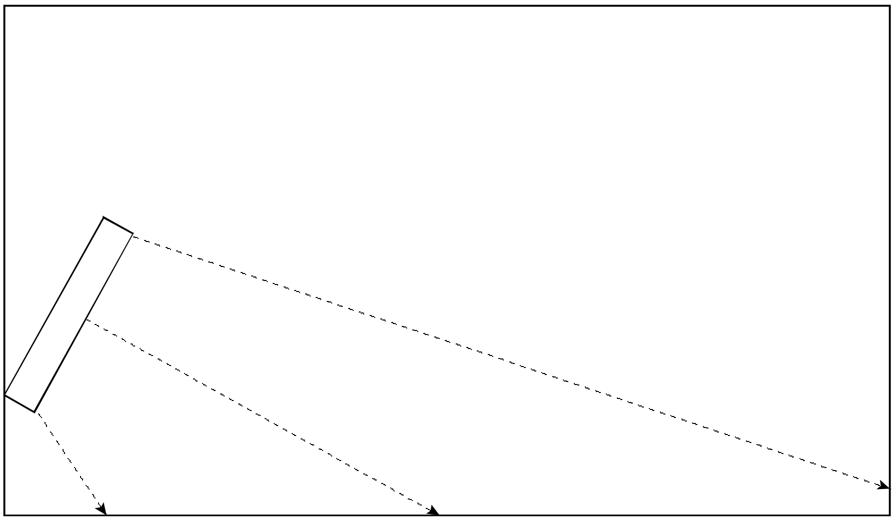
Q.1 (a)



Stage and loudspeaker Arrangements - Central Cluster + traditional Side-fills

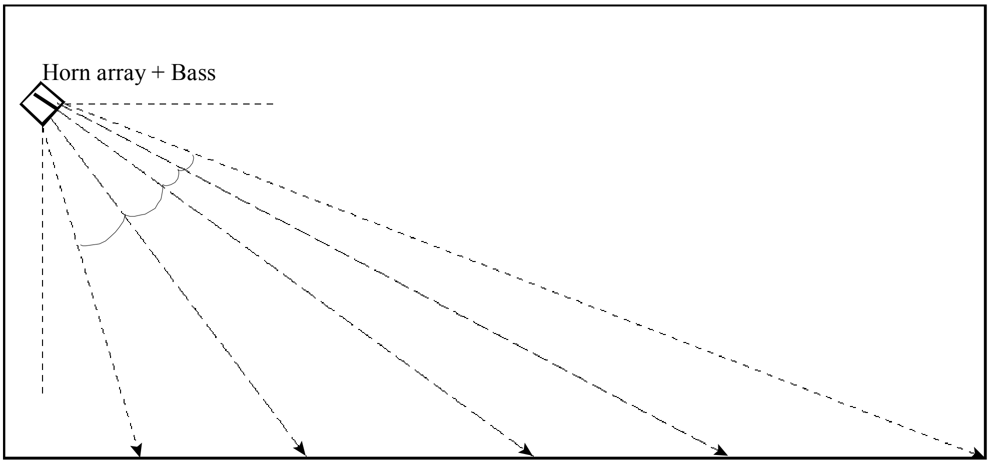
Side Columns:

Traditional column loudspeakers at the side are good for providing strong left or right effects, but in general the image placement will be off-stage for those members of audience not sitting close to the centreline of the auditorium.



Large SPL variation from front to back due to inverse square law. Basically audience at the front is just too close. column can be beamed down either physically or using progressive delays from top to bottom (beam steering).

Centre Cluster
A central cluster places the main image in the centre. Fortunately human hearing is not all that sensitive to image position in the vertical plane. In-fills may be needed in the front side areas.



Mid-frequency CD horns can be used to provide good coverage over the audience area. However CD horns are not good < 500Hz so low frequency bass units are needed to compensate.

(b) In a large auditorium with balconies, delayed balcony in-fill loudspeakers are needed to provide adequate coverage.

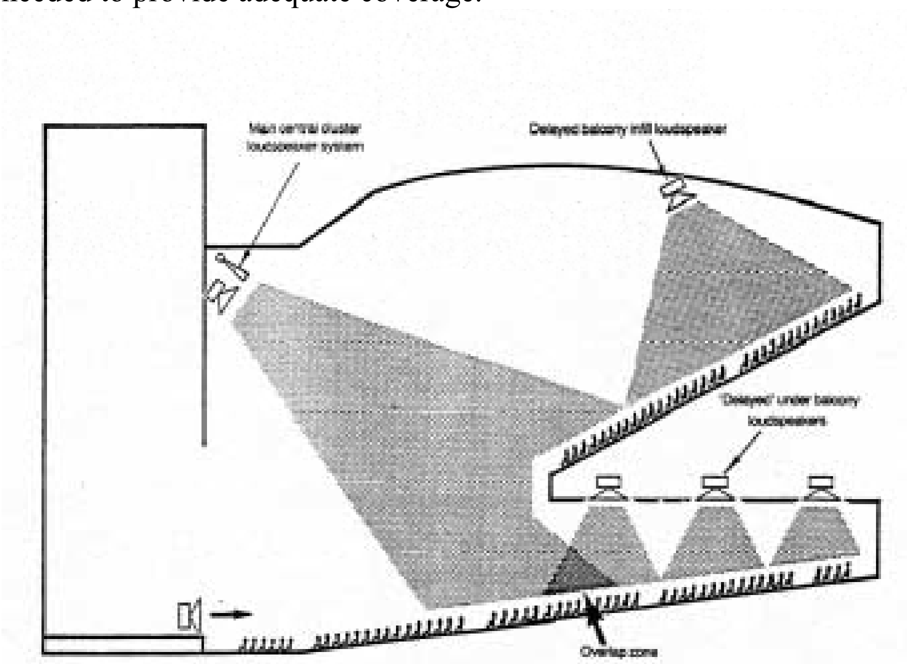


Fig. 18.10 *Under balcony loudspeaker delay system.*

Proper delay is essential to ensure that the first arrival is from the stage to use the precedence effect for proper source localisation.

(C) A CD horn is designed to give constant directivity over a frequency over a frequency range of typically 500-15kHz. It has a rectangular mouth and the beamwidths on the two axes are identified in degrees, e.g. 9040 has a beamwidth of 90˚ in one plane and 40˚ in the other. It is particularly useful because the sound can be directed to the audience and the footprint can be predicted with confidence over a wide frequency band.

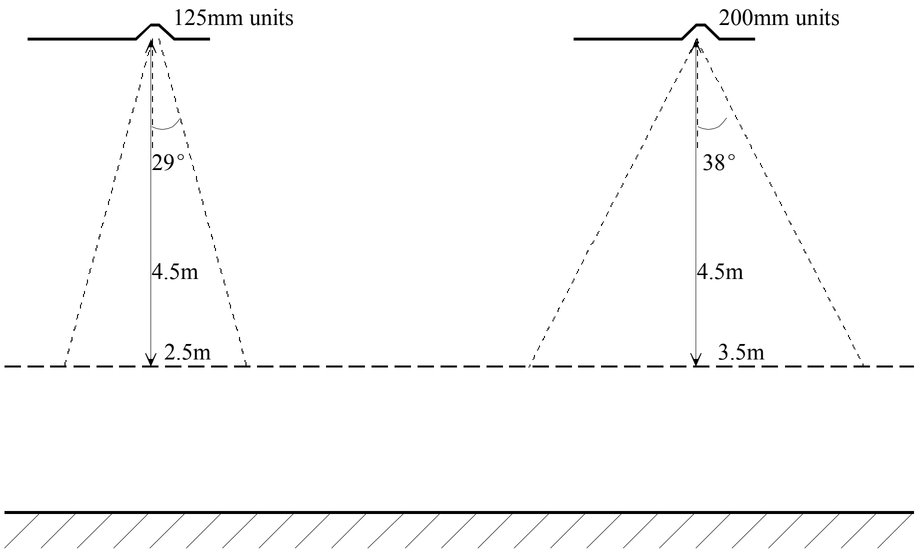
Its limitations are that it will not operate at low frequencies below 500Hz and low frequency bass units are required to compensate. Its power handling capacity is usually limited to a maximum of 40W and is also moderately bulky to use.

Q.2

(a)

To cover a large areas there are inevitably long cables running from the amplifiers to the loudspeakers. Consequently serious power losses can occur in the cabling. By using a high voltage, 100V, line the amount of current running through the cable is reduced, and the power losses are less. They also avoid the problem of matching impedance of a large number of loudspeakers to the amplifiers when they are connected directly. A step-up transformer is used to step up the amplifier voltage to the line voltage. At each loudspeaker a separate step-down transformer is used to get the voltage and the power required. Since each loudspeaker is independently connected, one failing would not affect the rest of the system, although it would create a dead zone.

(b)



200mm unit has a linear coverage of 7m per unit. To cover 120m x 15m using 200m unit requires:

120/7 => 17 units leaving 1m gap in the 120m direction
15/7 => 2 units leaving 1m gap in the 15m direction
Therefore a total of 17x2 = 34 units.

125mm unit has a linear coverage of 5m per unit. To cover 120m x 15m using 125m unit requires:

120/5 => 24 units in the 120m direction
15/5 => 3 units in the 15m direction
Therefore a total of 24x3 = 72 units.

For 85dB at head height, power required for the 125mm unit is

$$85 = 92 + 10\log W - 20\log(4.5)$$
$$= 92 + 10\log W - 13$$

⇨ $10\log W = 6$

⇨ $W = 4$ Watts

For the 200mm unit,

$85 = 96 + 10\log W - 20\log(4.5)$

$= 96 + 10\log W - 13$

⇨ $10\log W = 2$

⇨ $W = 1.6$ Watts

Number of amplifiers required:

For the 125mm unit is

$N = (4 \times 72)/200$

Rounded to 2

For the 200mm unit is

$N = (1.6 \times 34)/200$

Rounded to 1

Total cost for

125mm units:

72 x 30	= 2160
+ 2 x 475	= 950
Total	= £3110

200mm units:

34 x 60	= 2040
+ 1 x 475	= 475
Total	= £2515

Therefore the large units cost less and therefore appear to be more cost effective. However it is risky to use only 1 amplifier which will cause complete system failure should the amplifier fail. To safeguard 2 amplifiers will be needed for the 200mm unit system. This will add the cost up to £2990, close to the 125mm system. Since the 200mm unit system also has 1m gaps in each direction, the 125mm would be better after all.

Q.3

(a)

A column loudspeaker array has the directional characteristics of a line array in one plane and the directional characteristics of a single driver in the other. This allows sound to be concentrated in one plane but spread over in the other. When put up as a vertical column, this allows the sound to be focused to the audience in a hall in the vertical plane but spread widely in the horizontal plane.

(b)(i)

To calculate the beamwidth (-6dB) for the vertical plane, use the line array directivity (single driver directivity much smaller and can be ignored),

$$k(b/2)\sin\varphi \approx \frac{2}{N}$$

with b=0.18 and N=5

for the horizontal plane, use the single driver directivity,

$$ka\sin\theta \approx 2.2$$

with a=0.075

Hence

F	k	k(b/2)	ka	φ_V	θ_H
1000	18.265	1.64	1.37	14.1	>90°
2000	36.53	3.29	2.74	7.0	53.4

Therefore beamwidths (-6dB to –6dB) are

	1kHz	2kHz
Vertical Plane	28.2°	14°
Horizontal Plane	>180°	106.8°

(b)(ii)

First major side lobe occurs when the denominator of the directivity function goes to zero for the first time, which occurs at

$$k(b/2)\sin\varphi = \pi$$

At 1kHz, => $\sin\varphi = \pi/1.64 > 1$

⇨ $\varphi > 90°$

At 2kHz, => $\sin\varphi = \pi/3.29$

⇨ $\varphi = 72.7°$

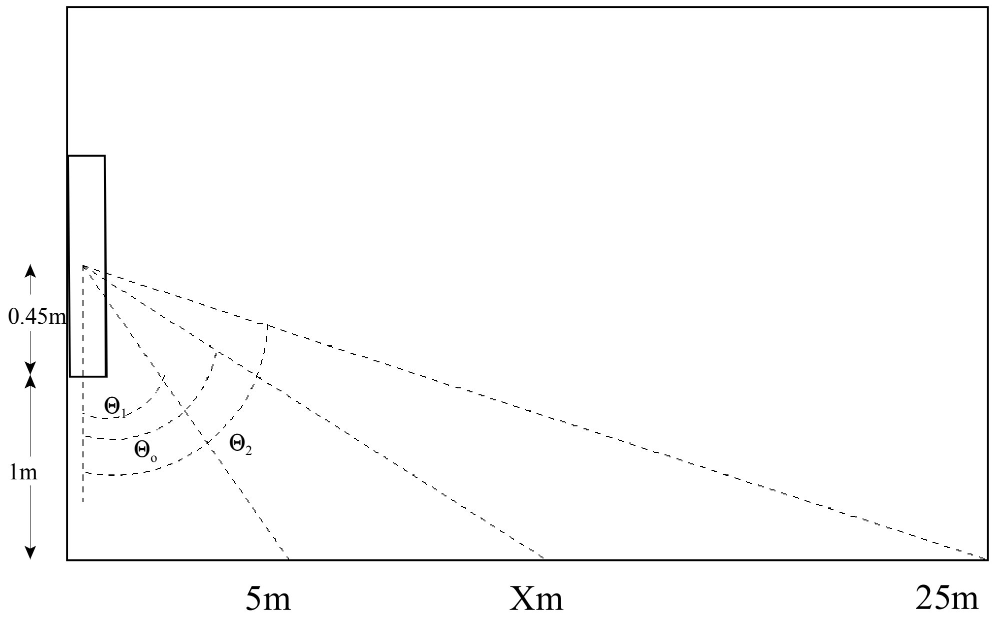
Therefore there is no major side lobe at 1kHz.

At 2kHz the first major side lobe occur at 72.7° above and below the main lobe axis in the vertical plane.

At these major side lobes the line array directivity is 1 and the magnitude is determined by the directivity of the single driver at the major side lobe angle (at 72.7° in this case).

Major side lobes are undesirable because they put energy into the reverberant field rather than to the audience. Besides lowering the direct energy and raising the reverberant level it can also cause echoes and reduce intelligibility.

(b)(iii)



$\tan \theta_1 = 5/1.45 \Rightarrow \theta_1 = 73.8^\circ$

$\tan \theta_2 = 25/1.45 \Rightarrow \theta_2 = 86.7^\circ$

$\theta_2 - \theta_1 = 12.9^\circ < 14^\circ$ the bemwidth at 2kHz.

However due to the much shorter distance to the front audience, the column should be aimed more to the rear than the front, say at

$\theta_o = \theta_1 + BW/2$

$= 73.8 + 7 = 80.8^\circ$

⇨ $x = 9$m

Even so the SPL at the front will be significantly higher than that at the rear due to inverse square law.

To achieve the proper angle, the column can be physically rotated or a progressive time delay be applied to “steer” the beam.

Q.4
(a)

$$D_c = \left[\frac{QA}{16\pi(1+n)}\right]^{\frac{1}{2}}$$

This is developed for a total of (1+n) sources of EQUAL power. Since only one source at a time is contributing to the direct sound while all sources contribute to the reverberant field, therefore the critical distance is shorter for more sources.

Q is the directivity factor of the source contributing to the direct sound. The stronger the directivity the stronger is the direct level and hence the longer $D_{c.}$

A is the total absorption area of the room and is inversely proportional to the reveberation time of the room. Hence the higher the A, the lower the reverberant level and hence the longer $D_{c.}$

If the sources are radiating into a highly absorptive area such as the audience the sound energy will be significantly absorbed by the audience before entering into the reverberant field. This does not affect the direct sound and hence D_c becomes:

$$D_c = \left[\frac{QA}{16\pi(1+n)(1-\alpha_A)}\right]^{\frac{1}{2}}$$

D_c increases. To be conservative this effect is usually ignored.

(b)
%ALcons is the Articulation Loss of Consonants. It is based on experimental work and states that intelligibility is closely related to the loss of consonants in speech. Consequently it is based on the frequency bands of 1kHz and 2kHz, which are most important for speech. The calculation procedure uses the formulae given which is dependent on the initial calculation of D_c.

It was found that when

%ALcons < 10 %	intelligibility is very good
%ALcons < 15%	intelligibility is very good
%ALcons > 15%	intelligibility will be a problem

Effect of background noise can be ignored if the signal to noise ratio (1 & 2kHz) > 25dB.

(c)
V = 1000m^3
T = 1.8s => A = 0.161V/T = 89.44

D_c = sqrt((2.5x 89.44)/16π) = 2.1m

At rear d = 16m > 3.5D_c (=7.4m)
Hence %ALcons = 9T + k = 18.2%
Therefore inadequate.

When a central cluster loudspeaker is used we can assume that the direct sound is mainly from the loudspeaker. To make the %ALcons acceptable, we have to increase D_c such that $d < 3.5D_c$ even at rear (since T has not been changed). We can also assume the the sound power of the human speaker can be ignored compared with the loudspeaker.

To be acceptable, %ALcons < 15%

⇨ $15 = [(200\ (16)^2\ (1.8)^2) / (1000Q)] + 2$
⇨ $Q = 12.8$

To check the D_c
$D_c = \text{sqrt}((12.8\text{x}\ 89.44)/16\pi) = 4.8\text{m} \Rightarrow 3.5\ D_c = 16.7\text{m}$

Hence $d < 3.5\ D_c$ even at rear with Q=12.8.

Therefore Q=12.8 is needed.

Q.5

(a)

The speech transmission index (STI) is based on the idea that speech consists of a wide range of frequencies which are modulated by low frequency envelops to produce the individual sounds. This modulation must be maintained in order for the speech to remain intelligible. The modulation can be degraded by reverberation and background noise. Note that the modulation is defined in terms of the intensity or p^2. By measuring the degradation of the modulation the intelligibility can be estimated.

To simulate human speech in the measurement, the simulation uses shaped noise split into 7 octave bands each of which is fully modulated by 14 different modulation frequencies F.

At source each of the 98 (7x14) frequency combination is fully modulated, A(1+cos(2πFt)) where F is the modulation frequency.

On reception each one is degraded by the reverberation and background noise to give B(1+ m cos(2πFt)) where m is a function of F and is ≤ 1.

The modulation is reduced by the room (reverberation + noise) to m(F) for each F in each octave band. In total 98 (=7x14) values of m will be obtained. The m(F) can be measured or calculated.

Analysis Procedure

1. Each value of m is converted into an apparent S/N ratio using

$$(S/N)_{app} = 10\log_{10}\left(\frac{m}{1-m}\right) \quad dB$$

2. Truncate $(S/N)_{app}$ to +/- 15dB. Nothing outside this range is significant.

3. A simple average is taken of the 14 $(S/N)_{app}$ values in each octave band to give 7 $(S/N)_j$ where j indexes the octave bands .

4. A weighted average is taken of the 7 $(S/N)_j$ values

$$\overline{S/N} = \sum_{j=1}^{7} w_j \, (S/N)_j$$

Where w_j = 0.13, 0.14, 0.11, 0.12, 0.19, 0.17, 0.14

5. STI is then calculated as:

$$STI = \frac{(\overline{S/N} + 15)}{30}$$

Assessment Criteria

STI	< 0.4	Poor

0.4 <	STI	< 0.6	Fair
0.6 <	STI	< 0.8	Good
0.8 <	STI	< 1.0	Excellent

(b)

1. If a human speaker is being simulated, then the directivity of the actual source should be the same as for a human source.

2. If background noise is a problem then source level and spectrum must be correct. If only RT is important then source level and spectral shape do not matter.

(c)

In a diffuse field, without the influence of background noise (S/N > 25dN),

$$m(F) = \left[1 + \left[\frac{2\pi FT}{13.8}\right]^2\right]^{\frac{-1}{2}}$$

and,

$$(S/N)_{app} = 10\log_{10}\left(\frac{m}{1-m}\right) \quad dB$$

Calculation gives

	T	F	m	$(S/N)_{app}$
500	2.0	1	0.739	4.5
	2.0	2	0.481	-0.3
	2.0	4	0.265	-4.4
	2.0	8	0.136	-8.0
2000	1.8	0.7	0.867	8.1
	1.8	1.4	0.657	2.8
	1.8	2.8	0.399	-1.8
	1.8	5.6	0.213	-5.7
	1.8	11.2	0.108	-9.2

			Average	-1.55

Therefore

STI = (-1.55+15)/30 = 0.45

Intelligibility is only fair.