Sound Reinforcement System

1.0 Introduction

The purpose of this chapter is to discuss some design concept and guidelines for the installation of sound system in halls. Quite often sound reinforcement system is required in multipurpose halls such as school hall with a stage, flat floor and with some intruding noise. Normally, such halls are long, wide, rectangular in shape and with high ceiling. If a hall is acoustically treated i.e. the walls are treated with sound absorbent material, ceiling is installed with perforated material and the floor is covered with parquet flooring of carpeted, reflection and reverberation of sound will not cause too much problem. However, halls with poor acoustical environment will produce sound that is not intelligible and quite reverberant. In such cases, special care must be taken in the positioning and selection of loudspeaker system and other audio equipment.

2.0 Design concept

The intended sound system shall pick up, amplify, distribute and reproduce voice and all other signal program satisfactorily. Most of the halls are probably designed for multipurpose, such as speech reinforcement, music, drama performance etc. Therefore system versatility is direct sound, early reflections from halls, ceiling, stage and reverberant sound. The rectangular hall with stage at the parallel walls which may give added full nest. The disadvantage is that the hall is usually long and thus the front and at the back will be great. As a rule of thumb for every doubling of distance from the sound source, this results in a decrease in sound pressure level of 6dB. The ambient noise for hall can be in a range of 40dB to 75dB, depends on the capacity of seated audience. Exhaust fans and wall bracket fans in the hall contribute noise when they are running. Generally, the speech from the loudspeakers must be louder than the ambient noise by at least 10dB in order to be intelligible.

It is generally reckoned that four factors contribute to articulation at any point in the hall. The first is the background noise level which ‘masks’ the required sound. The second factor is the level of speech above the threshold of hearing. This will depend on the distance from the speaker, the volume of hall and the nature of the surrounding (whether they are highly reflecting or not). The third factor is the reverberant time, if this is very short there will be insufficient reverberant energy to maintain the level of speech. On the other hand, if it becomes too long, sound of successive syllables overlaps, and the resulting ‘masks’ the speech. The last factor is the shape of the hall if the speech isn’t to sound. Unnatural, it is important that the amplified sound shall arrive at the listener just after the sound which has traveled direct from the speaker. Echo can be avoided if the direct and reflected path is kept less than 16 meter. However, if the difference between the reflected and direct sound is 10dB or more, the reinforcement system for halls must be loud enough with sufficient acoustic gain, articulation loss of consonants in speech and cover the listeners with uniformity.

The sound system shall be capable of supplying 90dB SPL program level plus 10db peaking factor. The system shall have even distribution of the reinforced sound throughout the seating area, typically plus or minus 3dB front to the back and site to site for one-octave band centre at 400Hz in addition to this, the system shall have uniform frequency response throughout the audience area. Typically plus or minus 3dB as measured with 1/3 octave band of pink noise at position across the main seating area. It shall have adequate dynamic range at an acoustic distortion level sufficiently low to ensure minimum listening fatigue. Signal to noise ratio of the entire system from inputs of main mixer to output of power amplifier shall not be less than 80dB.

Proper design and selection of equipment, especially loudspeaker system is important. If the acoustic of the hall is food i.e. the reverberation time is correct for speech and there are no specific acoustics defects such as echoes or 'dead' areas, a sound reinforcement system will only be needed to make the speech louder, particularly at the back of the hall. One of the methods that may employ is to have two column loudspeakers on either side of the centre – line, i.e. either side of the proscenium arch. Thus nobody will be very far off the axis of one other further from the microphone such that feed-back will be reduced. But, there are two disadvantages. The first is receiving two loudspeaker sounds. The real sound will arrive first, and if it is loud enough it will determine the apparent direction of the speech as coming from the man speaking, which is good. If however, it is not loud enough, the speech will appear to be coming from either one side or the other. This can be irritating. The other disadvantage is that for those who are nearer to either of the loudspeakers than they are to the man speaking, the speech will appear to be coming from the loudspeaker.

For large hall with balcony, second pair of subsidiary loudspeakers can be placed at mid-hall or about a third way from the stage. The subsidiary loudspeakers can be delayed. The time delay introduced shall of course correspond to the time taken for the sound from the stage to reach the rear of the hall, less the time taken for the sound from the mid-hall speaker to reach this area. The amplitude of this subsidiary loudspeaker shall be such that the sound from them reaching the listeners shall not be more than 10dB up on the sounds reaching them from the front of the hall. Therefore, for halls with bad acoustics, long reverberation time and distances to be covered by the loudspeakers are great, subsidiary smaller loudspeakers with proper time delay shall be introduced to solve the problems. In order to avoid the chance of indirect coupling, reflection directly from the rear wall can be reduced by facing the second pair loudspeakers towards to rear corners. We must make sure that maximum of the rated acoustic energy is directed towards the audience and not toward reflective walls and ceilings. The extant of reflections will depend on the nature of the seats. Some seats will act as absorbent. A well filled audience area has absorption very close to 100%. The important frequency bands which influence the intelligibility are the octave band of 2 and 4 KHz. It is important to match the angle of the loudspeaker with the height above floor level. Angling difficulties are increased with the height, 1.8 - 3.0 meter between floor and bottom of the loudspeaker will prove about right in most cases. Two separate amplifiers, each feeding a different pair of loudspeakers, fed from the same mixer can be employed.

The main reason column loudspeaker is used because the sound forward a column loudspeaker is projected from and with a flat beam. Most of the sound is concentrated and possess higher directivity. As one walks away from a column speaker, the drop in volume is barely perceptible until the limit of its range is approached.

Other method of loudspeaker system is to install cluster of loudspeakers in the centre of stage. This usually consists sets of horns and mid-base loudspeaker. The loudspeaker can be suspended from the ceiling using suspension bolts or steel wire ropes. Horn with long throw cover the back seating area and short throw horns cover the front. Stage monitor speakers are for people on stage or performers in concerts. Portable type is preferable and can be placed at both sides. Other areas such as backstage, restroom, dressing room, reception room, foyer etc can be installed with ceiling mounting speakers or box speakers.

In order to maintain the sound quality, low impedance connection is suggested to connect between the amplifier and the loudspeakers without using matching transformer. The loop resistance of the cable should at least be less than one tenth the load impedance of a loudspeaker. This is required to be so in order to reduce the power loss the cable resistance causes and also to make large enough a damping factor which has a close relationship to the sound quality of a loudspeaker. Usually a maximum length of 45m is allowed between a 8 ohm impedance loudspeaker and amplifier when 1.5mm2 diameter cable is used.

Care must be taken to have proper matching between power amplifier output and loudspeaker rating. To avoid damage of loudspeaker, the combination of the amplifier and the loudspeaker shall be determined as follows according to a degree of experience of a person who operates a system.

a) Inexperience person:-
   A continuous rating of the loudspeaker should be identical with a rating of power amplifier.

b) Well trained or experience person:-
Match an amplifier rating to a program rating of the loudspeaker.

c) Quality sounds are particular required.
Match an amplifier to a peak input rating of the loudspeaker.

Out of the three cases, case (b) permits the full use of performances of the loudspeaker and it is also economical.

3.0 Design formula

a) Reverberation Time (RT_{60}) Calculation:-

In order to determine articulation losses, reverberation time of the hall (RT_{60}) needs to be measured or calculated. Reverberation time is proportional to the volume of a hall and is inversely proportional to the area. The formula for halls where the expected RT_{60} > 2.0 seconds and absorption is relatively uniform and low in value is given as follows:

$$RT_{60} = \frac{0.163\ v}{Sa}$$

Where,
V is the volume of the hall in m3
S is the total surface area in m2
a is the average absorption coefficient.

b) Directivity Index (Q)

The directivity of a loudspeaker varies from 1 (omnidirectional) to values above 50 (highly directional). The directional properties of loudspeakers are generally frequency dependent and hence Q value shall always be qualified by a frequency.

c) Percent of Articulation Loss Consonants (AL_{cons}) Calculation:-

The percent of articulation loss of consonants (AL_{cons}) determined the articulation score in a hall. An AL_{cons} of 15% is considered to be a practical working limit. Formula for %

AL_{cons}:-

$$\%\ AL\ cons = \frac{200\ {D_2}^2\ {RT_{60}}^2\ (N+1)}{VQM}$$

Where,

D_2 is the distance from the loudspeaker to the farthest listener

RT_{60} is the reverberation time is seconds

V is the volume of the hall in cubic meter

Q is the directivity ratio

n is the number of loudspeaker groups identical to group 1 (like sources)

M is the Dc (Critical distance) modifier (usually 1 is chosen except special instances. Modifier due to audience absorption)

The above formula is used for D2 < DL and DL = 3.16Dc.

When D2 > DL, the formula becomes:-
$\% AL_{60} = 9RT_{60}$

It is necessary to assume a required signal to noise ratio of 25dB to make these calculations valid.

The basic formula can be further converted into the following useful formulas:-

Maximum D_2 that allowed an AL_{cons} of 15% $= \dfrac{15\ vom}{200\ RT_{60}^{2}(N+1)}$

Maximum RT60 that allowed an AL_{cons} of 15% $= \dfrac{15\ vom}{200\ D_2^{2}\ (N+1)}$

Minimum Q that allowed an AL_{cons} 15% $= \dfrac{200\ D_2^{2}\ RT_{60}^{2}\ (N+1)}{15\ VM}$

d) Critical Distance (Dc) Calculation:-

In an enclosed space like the hall, the sound field is made up of two components, the direct and reverberant. With increasing distance from the sound source, the component changes from direct to reverberant. Dc is defined as the distance in which the ratio between the direct sound and the reverberation sound comes exactly to 1:1. A knowledge of Dc provides us with distance at which the furthest distance is expressed as A x Dc i.e where A is the number of critical distances. The greater the reverberation time the less A can be in order to provide 15% $AL_{cons.}$ Critical distance (DC) can be obtained from the following equations:-

$$DC = 0.14 \frac{ORM}{N+1}$$

Where
Q = Directivity Ratio of loudspeaker

R = Room Constant

$$\frac{S\ a}{1 - a}$$

S = Total surface area in the hall

a = Mean absorption coefficient in the hall

e) Electrical Power Required (EPR) AND Sound Pressure Level (SPL) Calculations:-

When the definite acoustic sound pressure level (SPL) at a given distance (D2) from the loudspeaker is determined you need two important details in order to compute how much electrical power is required:-

i) The sensitivity rating of the loudspeaker, measured at 1 meter on axis when the loudspeaker is fed an input signal of one electrical watt.

ii) The acoustic level change and attenuation between the loudspeaker and the furthest listener position.

For example, we desire a 90 dB-SPL program level at 32m (D2). We have chosen a loudspeaker that has a sensitivity rating of 113 dB SPL at 1m from a 1 – watt electrical input. The wattage to be provided at the loudspeaker input allow a 90 dB-SPL program level to be reached is computed below:-

The acoustic level change (20 log 32) is 30dBAdding 10 dB to allow for the difference between program level and sine-wave levels plus 30dB acoustic level change gives 130dB-SPL at 1 meter from the loudspeaker. Now if 1 watt of electrical input can produce 113 dB at 1 meter, then 17 dB (130-113dB) above 1 watt for the required power will need:-

$$EPR = 10^{17/10} \quad = \quad 50 \text{ watts}$$

Hence, a loudspeaker which has a maximum power rating of 50 watt can be used. The above example is only applicable by assuming the installation is at outdoor. The formula for Maximum Program Level at a distance D2:

$$SPL_{D2} = 10 \text{ LOG (watts available)} - 20 \text{ LOG } d2 + \text{loudspeaker sensitivity (at 1 watt 1 meter).}$$

4.0 Sound equipment and other requirement

Audio equipment shall be of high quality, reliability, durability and good performance. Audio signal must be mixed, processed and amplified properly. The sound system configuration shall comprise standard 19” equipment rack, dynamic microphones, column loudspeakers, desktop mixer, monitor speakers, wireless microphones, intercom system etc. For large hall, control room is normally provided to place audio equipment and other stage lighting control equipment. The equipment rack in the control room shall be of stamped stainless metal plate and provided with side vents. It shall house all necessary audio equipment such as power amplifier, graphic equalizer, limiter-compressor, digital delay unit, wireless microphone receiver, cassette player recorder etc. The rack shall be so arranged that all equipment installed is withdraw able from the front for servicing and maintenance. There shall be enough space around the power amplifier to allow an escape of hot air form the power amplifier to allow an escape of hot air form the power amplifier to allow an perforated panel shall be mounted between the units mounted and mount a perforated panel large than one unit-size at the top of the rack.

4.1 Microphone

The type of microphone used for speech shall possess unidirectional characteristic. The recommended type is hand-held cardiod microphone mounted on telescopic floor stand with boom arm. It is especially recommended to use balanced type of microphone in order to reduce the external noise. In addition, high quality screened microphone cable high frequency largely varies according to the type of microphone cable. Wireless microphone such as tie-clip type is recommended for light drama on stage or other similar purpose. The microphone cable shall not be too long, the sensitivity decreases over high frequencies due to capacitance between conductors in case of high impedance microphone. The 3-pin XLR type connector is recommended. There shall be at least four microphone floor outlets on the stage and two outlets on both sides of the centre of seating areas and at the back of seating areas.

4.2 Mixer

Input level to the mixer must be properly set within the dynamic range. When the input is too large, a peak part of the program is clipped and the signal is distorted. On the other hand, when the input is too small, the inherent noise of a mixer masks the program signal. Besides level control, mixer panel also has tone control (bass and treble) function that equalizes the sound quality of the input signal. Other knobs and switches are grouping function, monitoring function and many other functions. It is recommended to specify XLR type connector with balanced of input, transformer-isolated and accepts low impedance sources.

4.3 Limiter-Compressor

Placing a limiter between mixer input and the power amplifier input suppress a peak component of the program, preventing the distortion from being caused by the peak clipping of a power amplifier. By using the compressor the overall program level may be higher for the same clipping level than in the case where it is not used. This effect proves advantage in high noise level areas. Therefore, limiters and compressors are used for suppressing a signal higher than the specific level.

4.4 Graphic Equalizer

Graphic Equalizer operates simultaneously at a number of preset frequencies, any of which may be boosted or cut Independently of the others. It is used of ensure that all frequencies can reach unity gain at the same time. After the sound system has been installed, the entire system can be equalized to its acoustic environment to ensure the specific tonal response and acoustic gain at each listener's ears. The recommended equalizer shall consists of 31 bands centred on frequencies at intervals of 1/3 octave and covering a frequency range of 20 Hz to 20 KHz.

4.5 Control Room

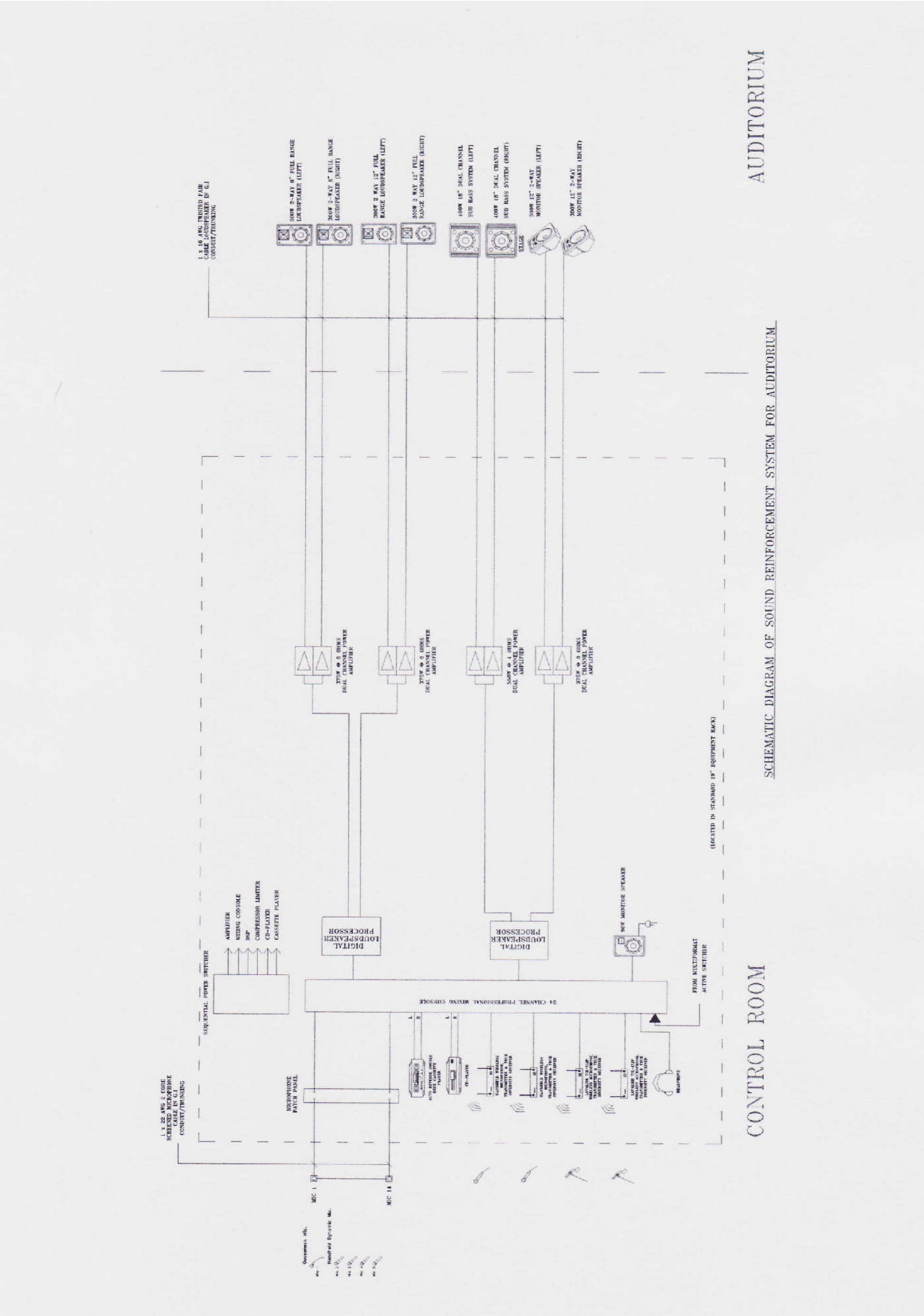
The control room is best located in the back of seating area. The size of window shall allow easy glance of the entire stage at his or her eye level. Suitable low level desk top shall be provided to place mixer console and other equipment. The input patch pave shall be installed near the console so that the person in charge will have easy access. The monitor speakers in the control room shall be installed above and behind the mixing console. The angle of the speaker shall be aiming at the controller's ears. Wired intercom system consists of one master station and a few slave units shall be installed for communication between the person on the stage and the controller in the control room.

4.6 Cable Selection

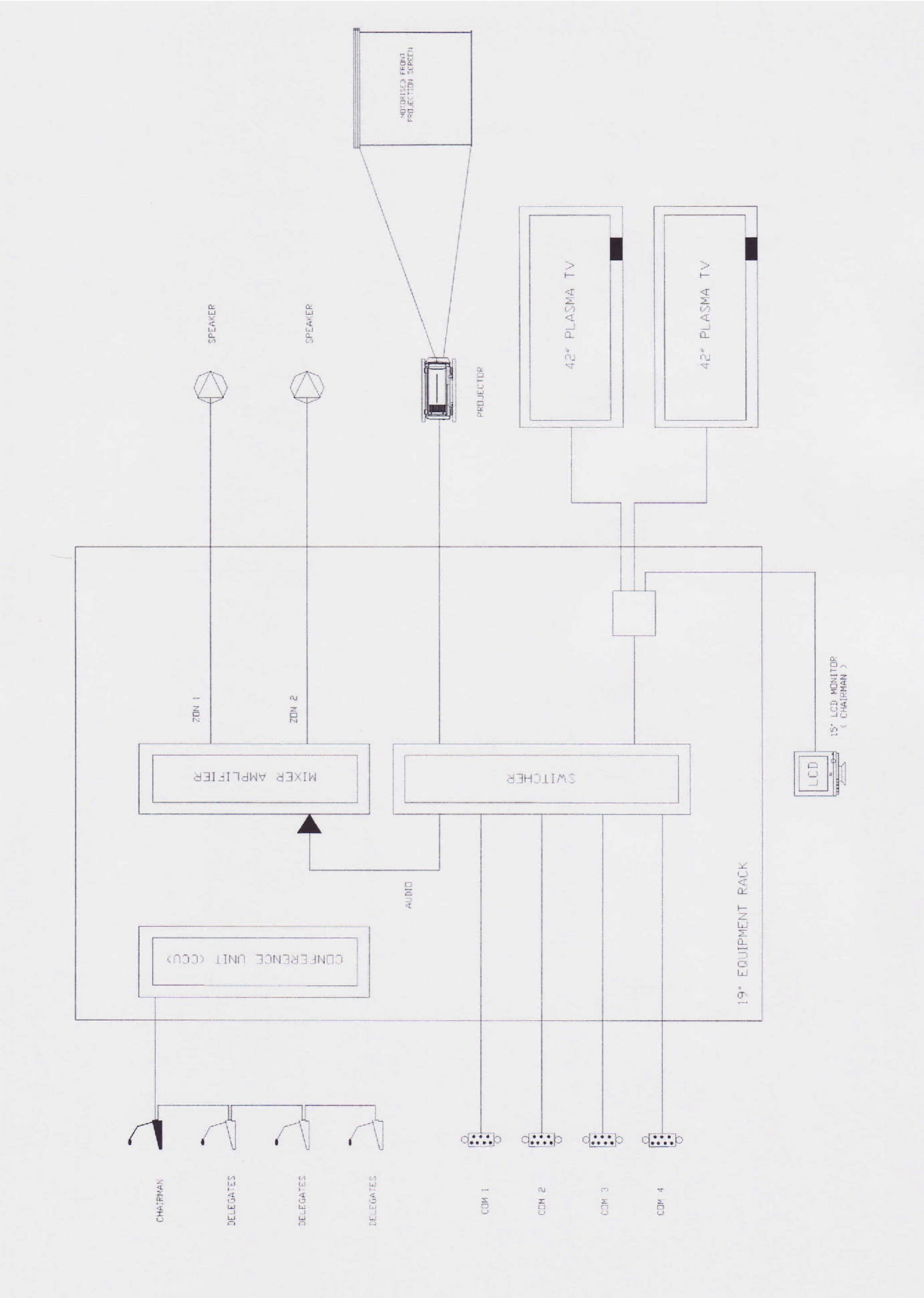
For wiring of -80 dBm to -20 dBm signal level, electromagnetic shielded wire shall be applied. For able with a -20dBm to + 30dBm signal level, cable materials must be equal to those of the microphone cable. For speaker cable to connect the power amplifier and loudspeakers, electromagnetic shielded type of speaker cable shall be used. Conductor of 1.5mm2 is suitable for high impedance connection of loudspeakers for law impedance connection of high output speaker, cables with a 5.5mm2 or large nominal sectional area shall be used. When using shielded cable between sound equipment, apply a one point grounding principle, the transmission-side shield must be electrically insulated from the chassis and ground it on the reception side. Also, to avoid oscillation, hum, induced noise, crosstalk and the like, the microphone line should be as separated as possible, from other high signal level lines, high frequency equipment, lighting control lines.

When lighting control system and sound control system are located on the same stage, it is advised to maximum the reduction of electromagnetic noise, by wiring the sound system at one of the wing of the stage and the lighting system at the other wing.

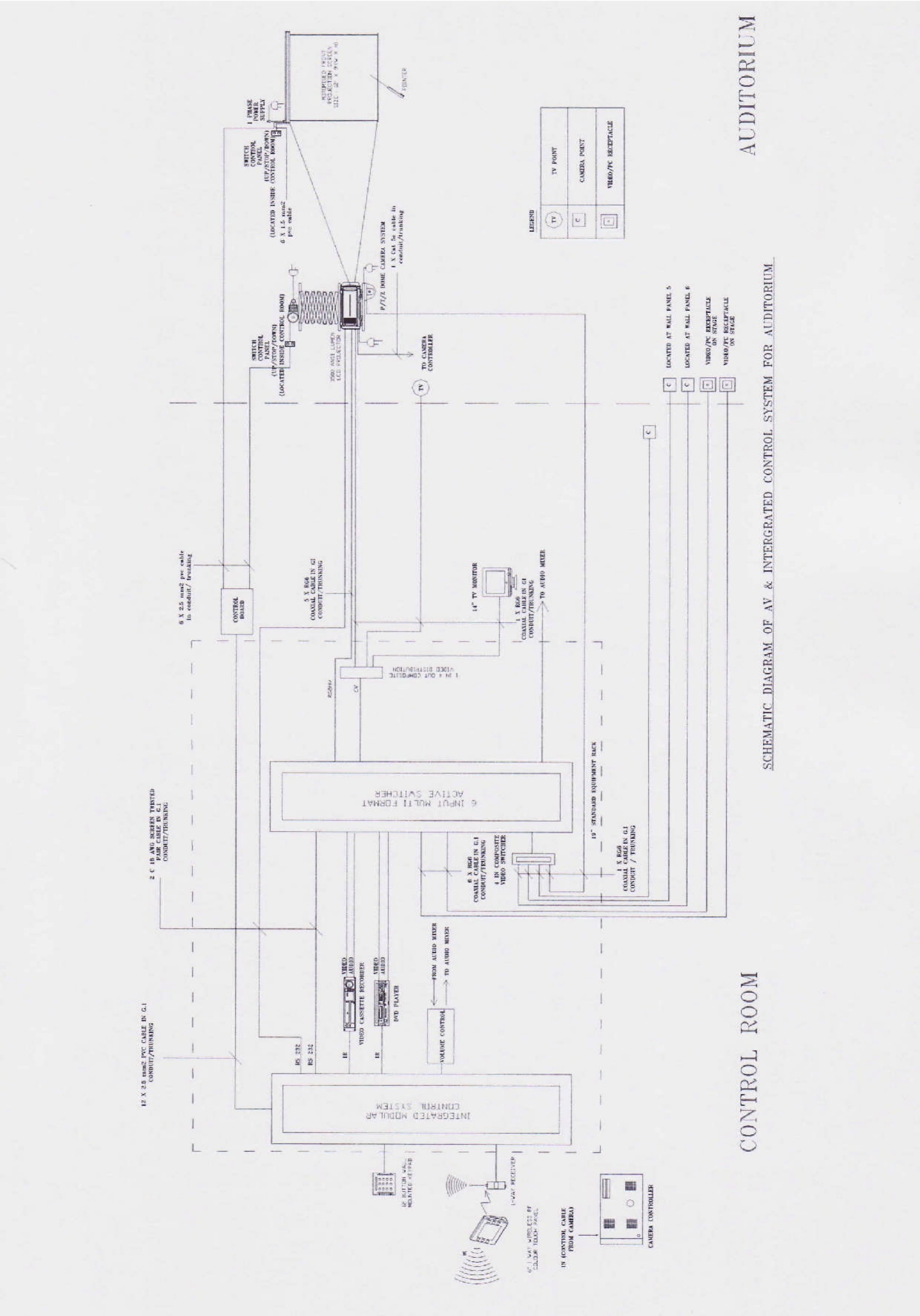
Appendix 1: Schematic Diagram of Sound Reinforcement System



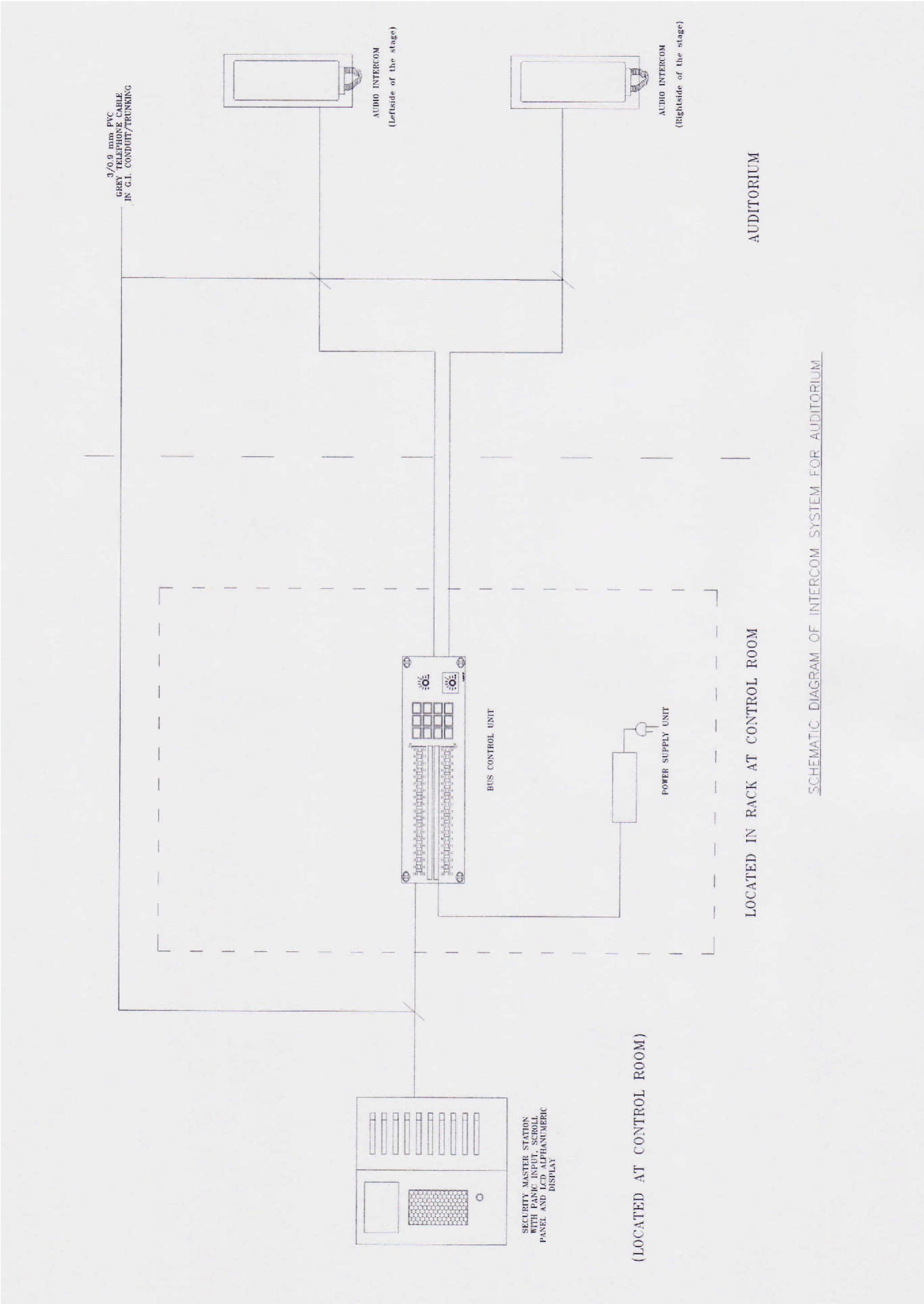
Appendix 2: Schematic Diagram of Conference System



Appendix 3: Schematic Diagram of AV and Integrated Control System



Appendix 4: Schematic Diagram of Intercom System



Appendix 5: Schematic Diagram of Stage Lighting System