Using Acoustic Enhancement To Improve Speech Intelligibility

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Abstract

The last decade, acoustic enhancement has been developed as an advanced tool of implementing variable acoustics in auditoria by electronic means. The implementations, however, have been limited to increasing the reverberation time of a hall, for instance to make a speech auditorium suitable for symphonic music. This paper describes the application of an acoustic enhancement system to improve the speech intelligibility of an auditorium only by adding early reflections, that is without increasing the reverberation time.

0. INTRODUCTION

Acoustic enhancement techniques [1-7] enable us to change the acoustic properties of a room by adding electronically, or by acoustic feedback, generated reverberation. This way variable acoustics can be realized by electronic means and the room properties can be changed at the touch of a button. This technique generally is applied to make a speech auditorium suitable for symphonic concerts and/or other music performances. A large number of theaters, however, do not require variable acoustics, but do lack sufficient natural speech intelligibility due to weak direct sound and/or insufficient early reflections. This can be caused by an inappropriate room size and/or shape. Particularly the continuous increase of seating capacity is related to this problem. Applying sound reinforcement techniques in such a situation is in most cases unsatisfactory, as both the naturalness of the sound and the localization of the actor are influenced in a negative way. In addition, these reinforcement systems require extensive operation during the performance and put extra constraints on the show sound design.

1. THE VIVIAN BEAUMONT THEATER

This theater (fig. 1) at Lincoln Center (NY) is an open or thrust stage theater, seating approximately 1100 people. The extreme flare of the side walls prevents the generation of any supporting early reflections. As the dispersion of the human voice (app 120°) is less than the expansion of the side walls, even the direct sound is inadequate for a large part of the audience. In addition, the contribution of ceiling reflections is minimal. Furthermore, large holes in the ceiling absorb much energy, reducing the overall sound level.
Ever since the opening in 1965, these problems resulted in serious complaints about the intelligibility for almost any production. Even extensive reinforcement with radio microphones on every actor could not solve these problems.

2. ACOUSTIC ENHANCEMENT

In 1995, Lincoln Center Theater contacted Systems for Improved Acoustic Performance (SIAP) on this matter. The wish was to have a solution for the intelligibility problems without the application of sound reinforcement. Freedom of movement for the actors without any restrictions to intelligibility or naturalness of sound was of the highest priority. Localization of the actors on stage should be fully correct. Facing away of an actor should result in a corresponding natural sounding change of tonal balance, but without a significant intelligibility loss. The solution should primarily be considered as an acoustic improvement of the auditorium and be of advantage to any production depending on good natural intelligibility. Implementations of our enhancement system in other theaters offered this capability as a standard facility, besides the more obvious variable acoustics [5,6,8]. The major difference was that speech enhancement would be the most important application in this theater.

2.1. System concept

The concept of our acoustic enhancement system is given in fig. 2. The sound from the stage is picked up by a number of microphones at strategic positions. The signal is processed in a multi-channel propriety processor and re-emitted in the auditorium. The major differences with traditional reinforcement systems are:

a) *Small number of microphones.*
A relatively small number of microphones are permanently installed to cover the entire stage. A large amount of coverage overlap is part of the design.

b) Multiple independent processing channels.

To simulate spatial diffusion, a high number of different output channels are generated. In addition, the number of un-correlated signal paths determines the maximum achievable acoustic gain. For instance, a system with 4 inputs and 25 outputs, is capable of 20 dB more gain before feedback than a single channel system. This is under the restriction that each input/output path is sufficiently de-correlated to all other input/output paths.

c) Large number of loudspeaker positions

Loudspeakers are distributed over the room with a large amount of coverage overlap. Each seat receives sound from multiple loudspeakers, each reproducing a differently processed signal.

2.1.1. Microphone configuration

The microphone configuration is such that each microphone evenly covers the entire stage, but from a different position. The super cardioid microphones are selected for a maximum similarity between on- and off-axis frequency responses. As a result the tonal balance and level of the sound remain constant with changing positions of the source. However, the signal being picked up changes with the orientation of the source. This enables one to maintain the natural change of tonal balance when an actor turns around. Also the delays from the source to the individual microphones vary with the position on stage. This enables one to maintain the natural localization of the original source on stage.

2.1.2. Processing

The acoustic processing in the processor generates different reflection patterns for each input/output combination. The envelope of the reflections can be programmed over a wide range. Frequency domain corrections can be implemented in each of the processing stages. Each output has its own delay settings to ensure proper localization of the original source. The level and delay settings are such that the individual loudspeakers are not noticeable for the audience and they experience a non-reinforced performance.
The system is pre-programmed with a fixed set of acoustic programs. The generated reflection patterns are programmed such that they fill in the reflections the auditorium's natural acoustic is lacking.
The processor consists of one or more frames. Each frame can be equipped with up to 20 DSP boards, each with either two inputs or two outputs. Depending on the installation and the application, the DSP boards can be loaded with 32kWord to 256kWord of data memory. Besides the DSP boards, each frame contains a master card that handles the communication between the DSP boards and on

which the data for the presets is stored during tuning of the system. A typical processor configuration is given in fig. 3. After tuning of the system, selection of the different settings is done by means of a small industrial terminal. Data can be entered by a numeric keypad and the activated setting is displayed in a 20 character display (fig. 4.). There are 6 function keys available which can be programmed to mute selected input and output groups. There are no operator controls to set levels, balance or other parameters to be modified during the performance. This implies that the sound engineer during the show can focus on show related sounds, like the reproduction of sound effects, instead of on maintaining intelligibility and avoiding feedback.

2.1.3. Loudspeaker Configuration

A large number of loudspeakers are distributed in the room. Each loudspeaker reproduces both early reflections and reverberation. The loudspeaker system can be divided in sub-systems related to the room geometry and acoustic requirements. Normally there will be a frontal system, an overhead system and a lateral system for the house's main volume. Changing the balance between these systems will influence the subjective spaciousness of the room. Balconies and under balcony areas, which generally are poorly coupled to the main volume, generally will be provided with their own sub-system, integrated in the main system design. Loudspeakers have to be of high sound quality. Any added distortion or coloration will directly be obvious due to the continuous A/B comparison to the natural source. Also there has to be a very smooth off-axis response to avoid any unwanted change of tonal balance over the auditorium.

2.1.4. Additional functions

Of course, the presence of loudspeakers and amplifiers with advanced digital processing invites the use of this equipment for other applications as well. In our opinion, however, this should never interfere with the basic function of the system, i.e., acoustic enhancement. Our processor generally is equipped with auxiliary inputs and outputs which, dependent on the installation, can offer a choice of the following functions:

a) *Microphone monitoring*
Unprocessed line level outputs make the system microphone signal available for other systems, like control room monitoring, lobby system, hearing impaired systems, dressing rooms, recording, etc.

b) *Effects reproduction*
Unprocessed line level inputs reproduce the signal distributed over the auditorium. These signals can be routed to all loudspeakers or pre-determined areas only. This can be of use to play background music before the show

or during the intermissions or to make announcements. These inputs also can be used to reproduce show effects.

c) Fill-in system

As the enhancement system provides loudspeakers in balcony and under balcony areas, it is only logical to use these as a fill-in system for reinforcement applications. The processor is programmed with the proper delays for the different areas but no other processing is performed.

d) Support of weak soloists

Sometimes a soloist lacks sufficient power compared to accompanying orchestra. In these situations the signal from a local microphone for the soloist, after pre-amplification, can be fed to the processed line level inputs to influence the balance. This signal will receive the same type of processing as the system microphones and therefore, will blend inconspicuously with the rest of the sound. Of course, an additional signal delay has to be introduced to compensate for the difference in microphone distance.

3. SYSTEM LAYOUT

For the Vivian Beaumont Theater the design had to be focused on speech intelligibility. For speech enhancement, the processing only introduces additional early energy, hereby increasing clarity without increasing reverberation time. The loudspeaker layout for the Vivian Beaumont Theater is given in fig. 5, the system Block Diagram in fig. 6.

3.1. Microphone configuration

To obtain an even coverage of the entire stage, 8 microphones are placed in a semi-circle around the stage. For actors acting upstage, 2 additional microphones are provided.
The system configuration is such that when an actor is facing away from the audience, the microphones across stage pick up the voice. In the processor, this signal is routed to the loudspeakers covering this audience area, compensating for the signal loss. The loudspeakers covering the area that the actor is facing, will receive less signal, both because less enhancement is required and to minimize feedback.

3.2. Processor configuration

The Vivian Beaumont processor has 10 microphone inputs and 48 outputs in two processor frames; each frame is loaded with 20 DSP boards. This configuration results in a matrix with 480 nodes in which each node has a different transfer function. Therefore, an extra gain of 26.8 dB can be achieved compared to a single microphone/loudspeaker configuration[2]. If we assume that, due to the geometry of the stage, only half of the microphones will pick up the proper sound and the on-stage loudspeakers do not contribute to the effective gain, still an extra gain of 23.2 dB can be achieved. The microphones can therefore be a large distance from the actors. As the average actor to microphone distance is about 10 m, the same gain could be obtained by using a single microphone - loudspeaker channel with an actor to microphone distance of about 0.7 m. However, with such a microphone distance it will not be possible to cover the entire stage and multiple microphones will be needed, reducing the gain before feedback.

3.3. Loudspeaker configuration

A total of 48 independent outputs are distributed to a total of 82 loudspeakers distributed over the auditorium. For monitoring on stage, 6 additional (moveable) loudspeakers are available. A total of 58 power amplifier channels with a total power capacity of about 15 kWatts is used.

3.3.1. Output level

Although the system for the Vivian Beaumont primarily is designed for speech enhancement, it has to be anticipated that loud sound effects will be reproduced during a performance. The enhancement system cannot exhibit non-linear behavior during such a situation. Therefore, the system must be capable of high peak output levels. In the Vivian Beaumont Theater, the system is designed for an output level of about 105 dB. Depending on the loudspeaker position, the lower frequency limit varies from 45 Hz to 90 Hz.

3.4. Equipment

Apart from the propriety processor, only standard commercial available equipment has been used. A full equipment list is given if appendix 1. A rack elevation drawing is given in fig. 7.

4. MEASUREMENTS

To illustrate the performance of the system for this particular situation, we measured the level distribution for a wide dispersion but directive source, roughly comparable to a human voice, facing one side of the auditorium. Measurements were taken at 5 comparable seats in the orchestra and 5 in the balcony as indicated in fig. 8.

The results show that without the enhancement system (fig. 9 & 10) the audience facing the back of the actor experiences a significant signal drop, particularly at the higher frequencies. With the enhancement system on, the level distribution is much more even (fig. 11 & 12). When the difference is expressed as gain, this results in a seat dependent gain as indicated in fig. 13 & 14.
The reverberation times of both the auditorium itself and with the enhancement system active are given in fig. 15. It can be seen from this that, despite the achieved acoustic gain, the reverberation time is not significantly increased. This also indicates that the system mainly provides early sound enhancement without also, undesirably, enhancing the late sound as well.

5. CONCLUSION

An acoustic enhancement system has been implemented to solve intelligibility problems by adding the missing early reflections. Natural localization and tonal balance are fully maintained for all actor positions and orientations. After installation of this system in the autumn of 1995, the complaints regarding speech intelligibility have disappeared and a number of plays have successfully be presented without any reinforcement of the actors. There is a high overall satisfaction from Lincoln Center and both audience and the companies playing the house.
Measurement results indicate a significant gain, particularly for actors facing away, without at any time giving the impression of reinforced sound.

6. ACKNOWLEDGMENT

In most situations sound (system) designers and lighting designers are in a continuous struggle for the best locations for their equipment in the auditorium and whether sound or light has the highest priority. We have to point out that this system never was implemented in the Vivian Beaumont Theater, if not for a lighting designer. We would like to aknowledge the indespensable assistance of Berverly Emmons, a lighting designer at Lincoln center, who initially heard the SIAP system during a large scale Aida production, introduced it to the management of Lincoln Center Theater and enthusiastically supported its use in the Vivian Beamont Theater.
We like to thank Lincoln Center Theater, in particular Bernard Gersten and Andre Bishop, for their confidence and the cooperation we received from them and their staff during installation. We would also like to thank ProMix Installations of New York, in particular Steve Shull and Peter Romandetti, who

handled the installation of the system in a very professional and cooperative way.

7. REFERENCES

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[4] D. Griesinger, "Improving room acoustics through time-variant synthetic reverberation", Audio Engineering Society Convention Paris 1990, Preprint 3014

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[8] B. Kok and W. Prinssen, "Electroacoustic correction of auditoria which have poorly coupled spaces using a SIAP system, a case study", Proceedings of the Institute of Acoustics, Vol. 17 "Opera and Concert Hall Acoustics": Part 1, 1995

APPENDIX

Equipment list Vivian Beaumont system

a)Microphones

10 Sennheiser MKH50

b)Processor

Siap MkIII, two frames, each frame configured with:

10 microphone inputs

24 processor outputs

4 auxilary inputs

2 auxilary outputs

c)Amplifiers

6 Crown CT-1600

8 Crown CT-400

15 Crown CT-200

d)Loudspeakers

8 Stage Accompany S26 (Side walls)

4 Stage Accompany S26 in custom cabinet (Side walls)

16 Stage Accompany F7 (front of thrust stage & Stage monitoring)

60 Kef Q10 (Ceiling)

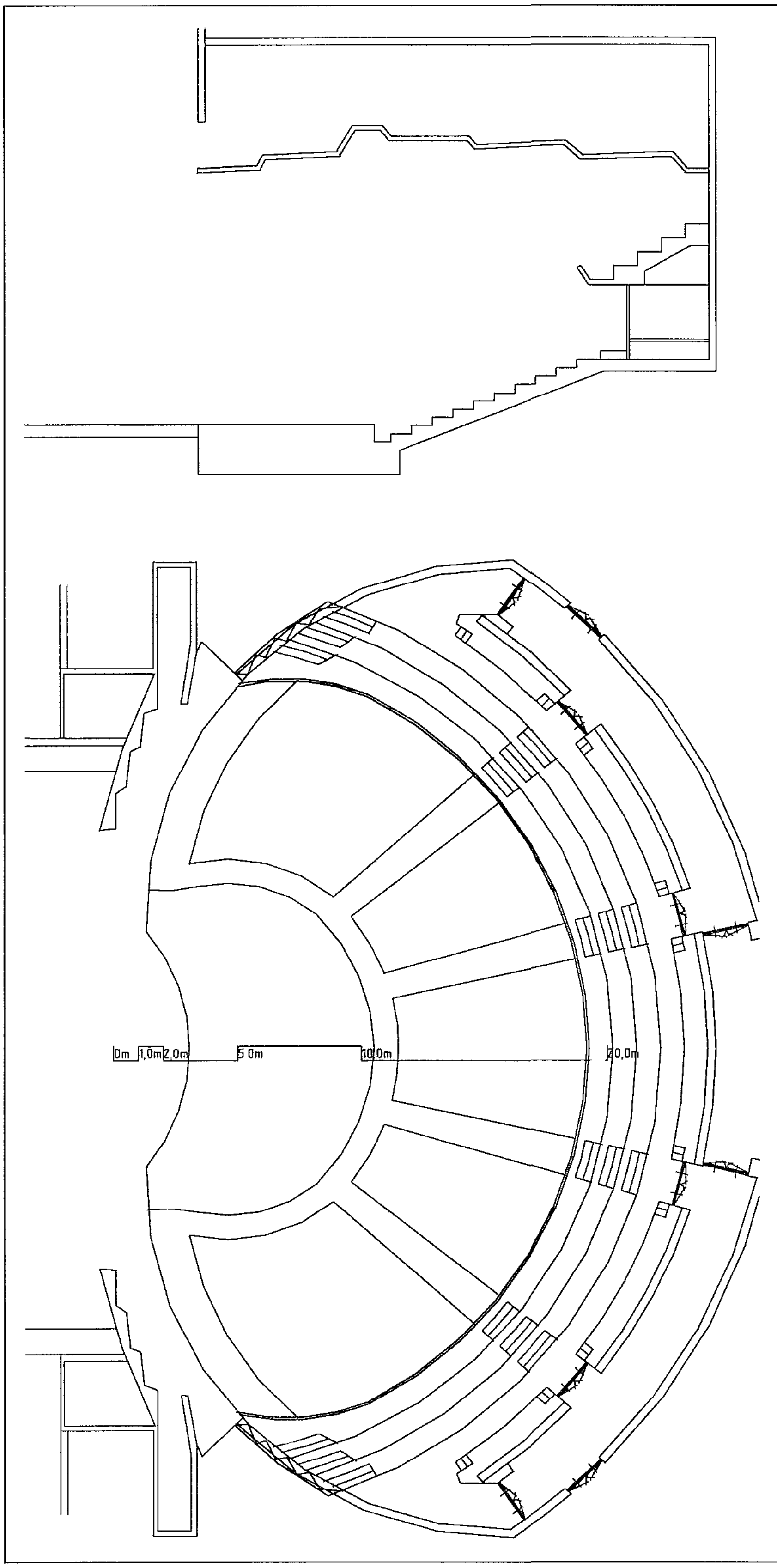

Fig. 1. The Vivian Beaumont Theater

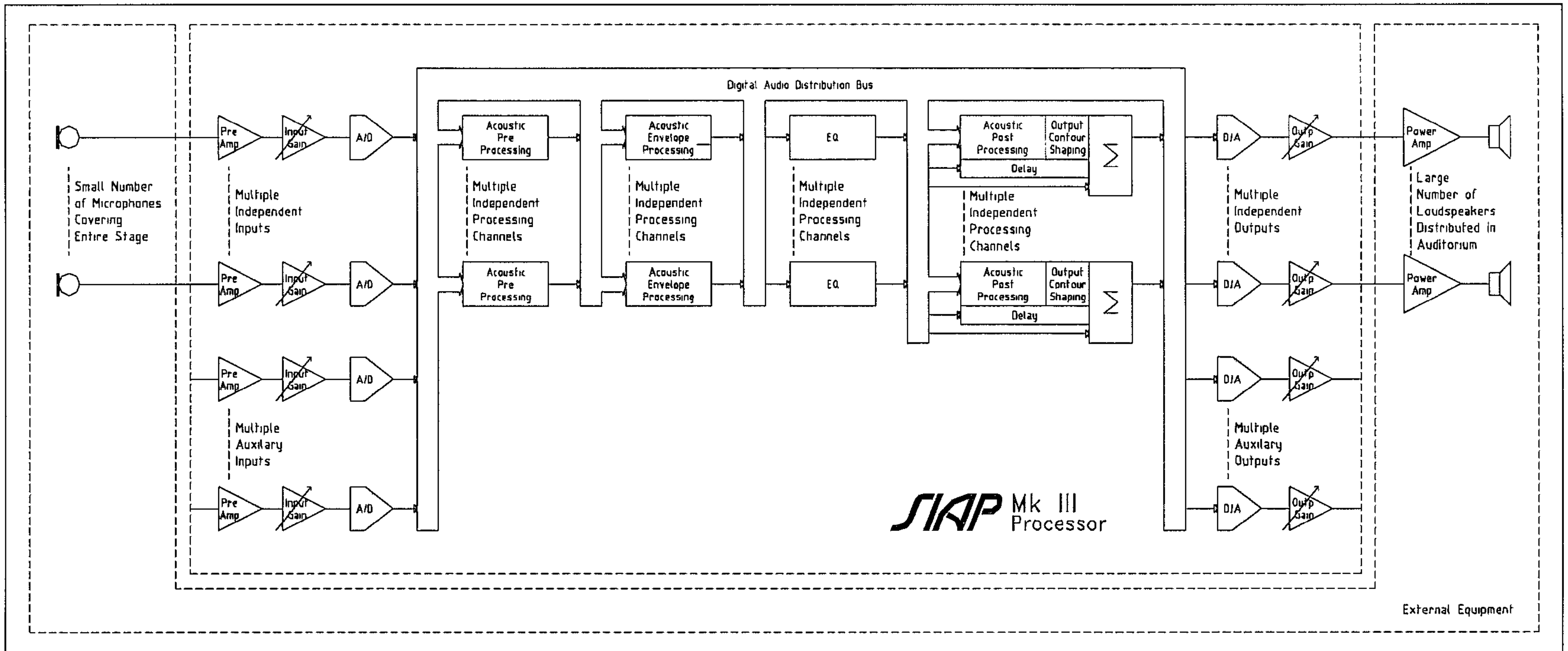

Fig. 2. Acoustic Enhancement Concept

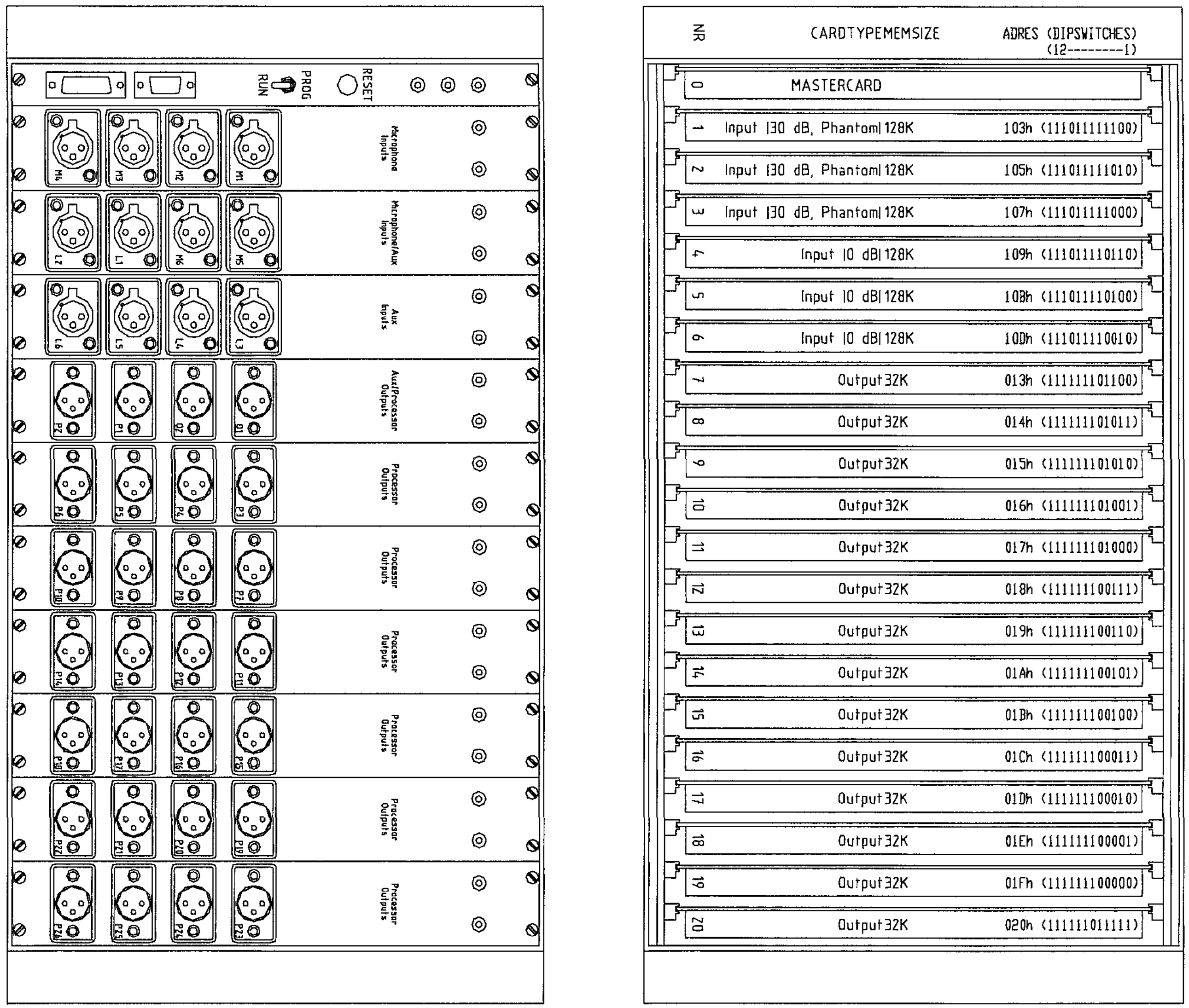

Fig. 3. Typical Processor Frame Configuration

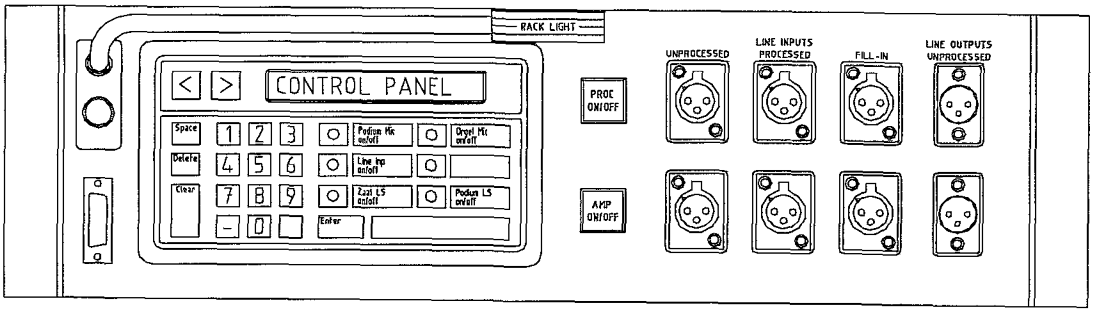

Fig. 4. Typical Control Panel Layout

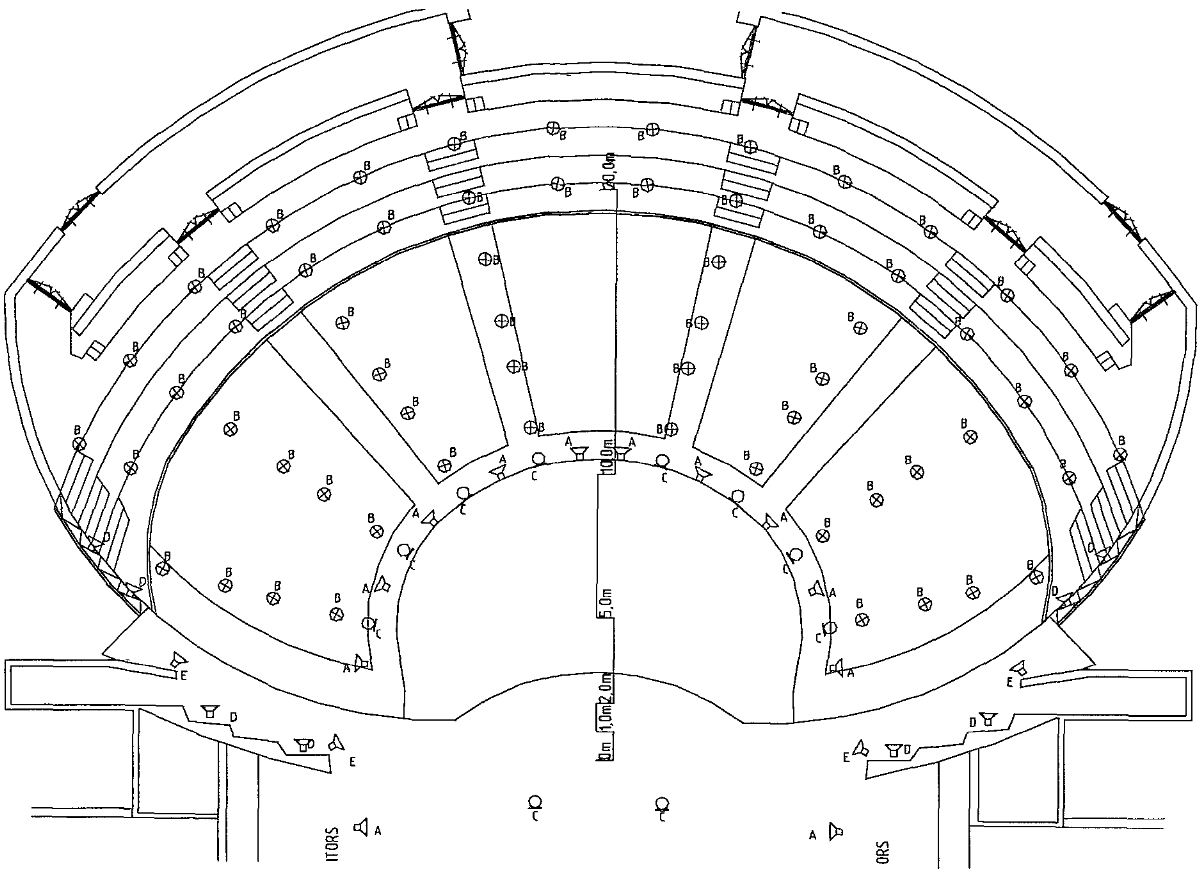

Fig. 5. Loudspeaker Layout

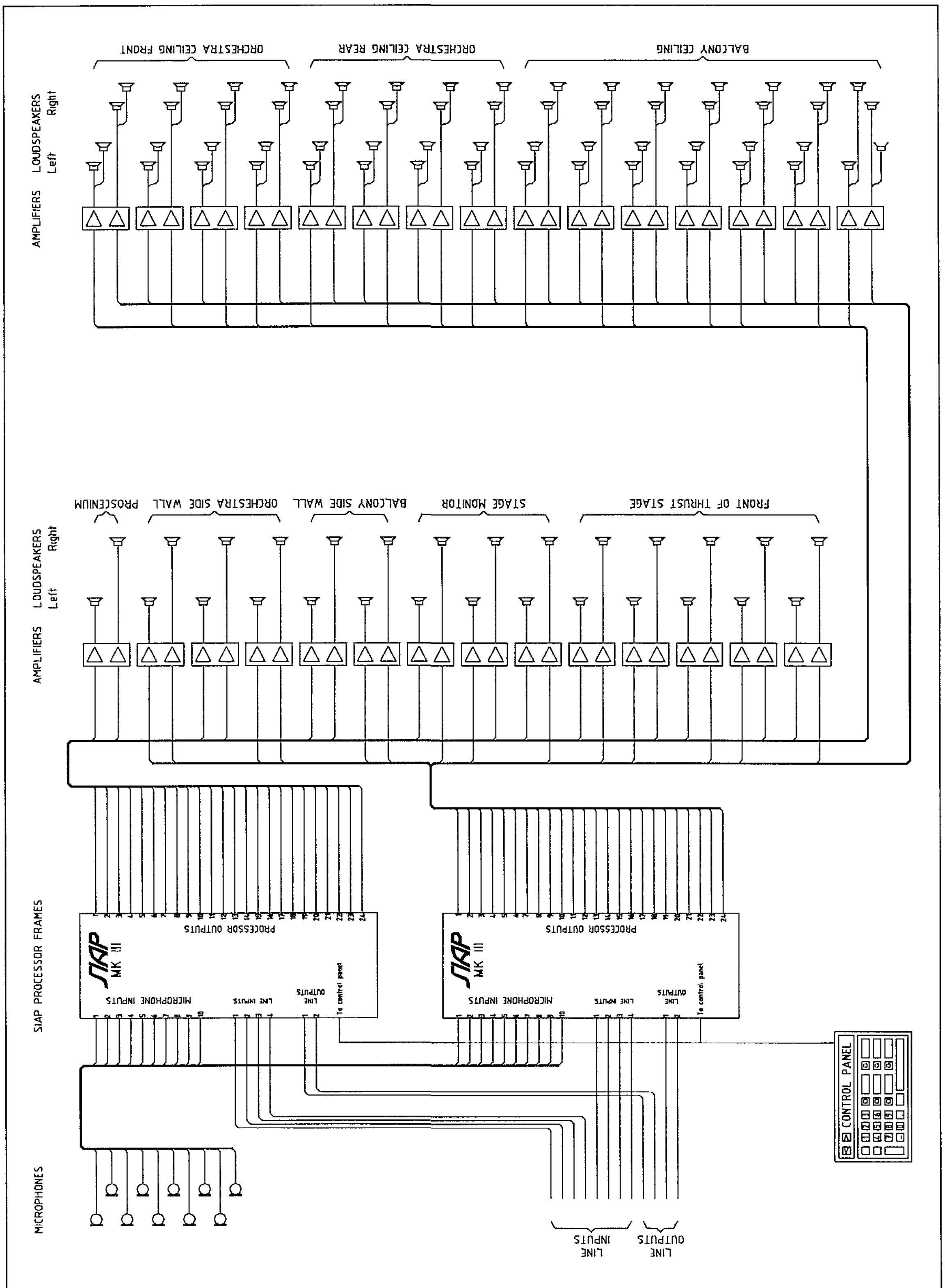

Fig. 6. System Block Diagram

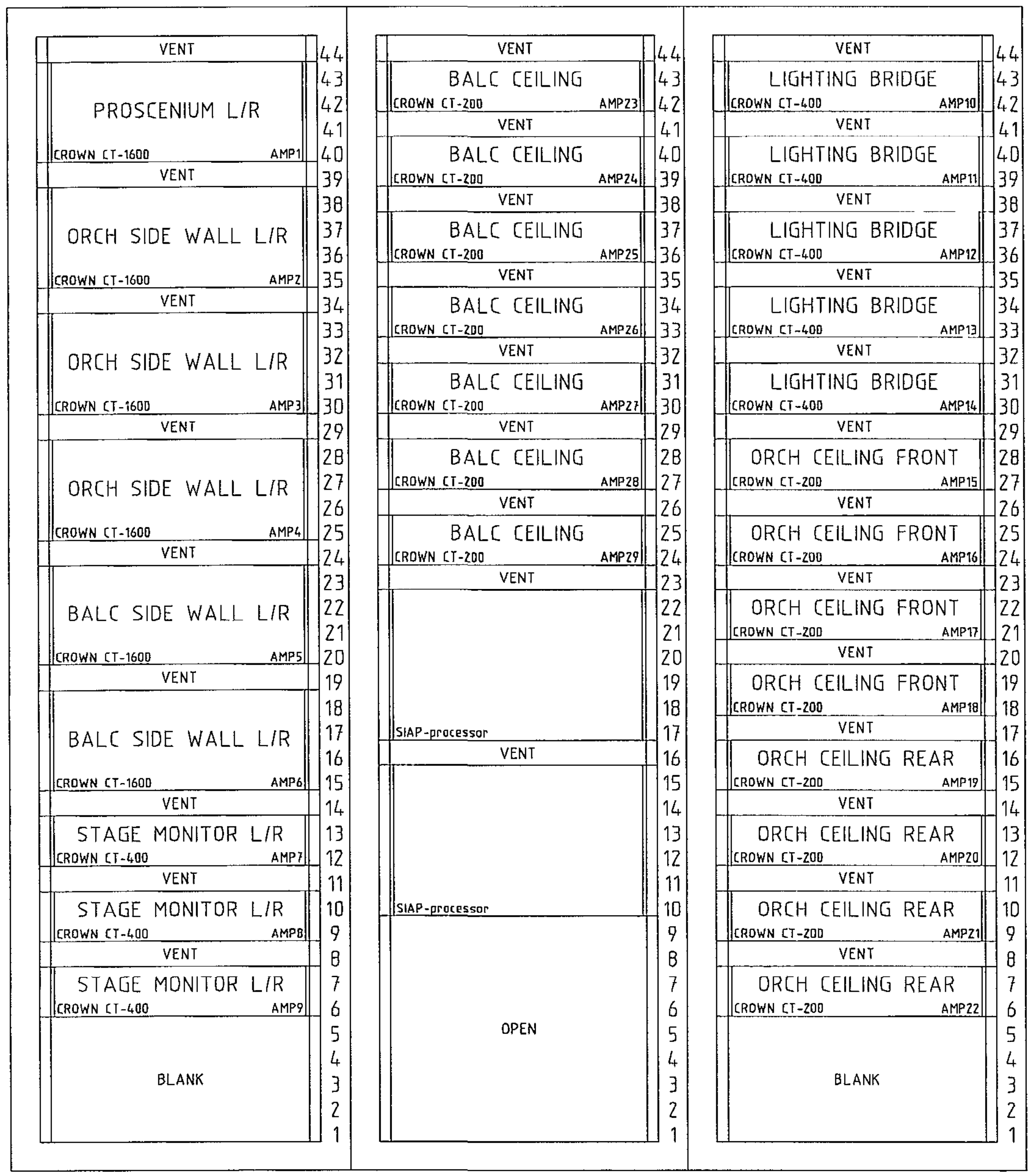

Fig. 7. Rack Elevation

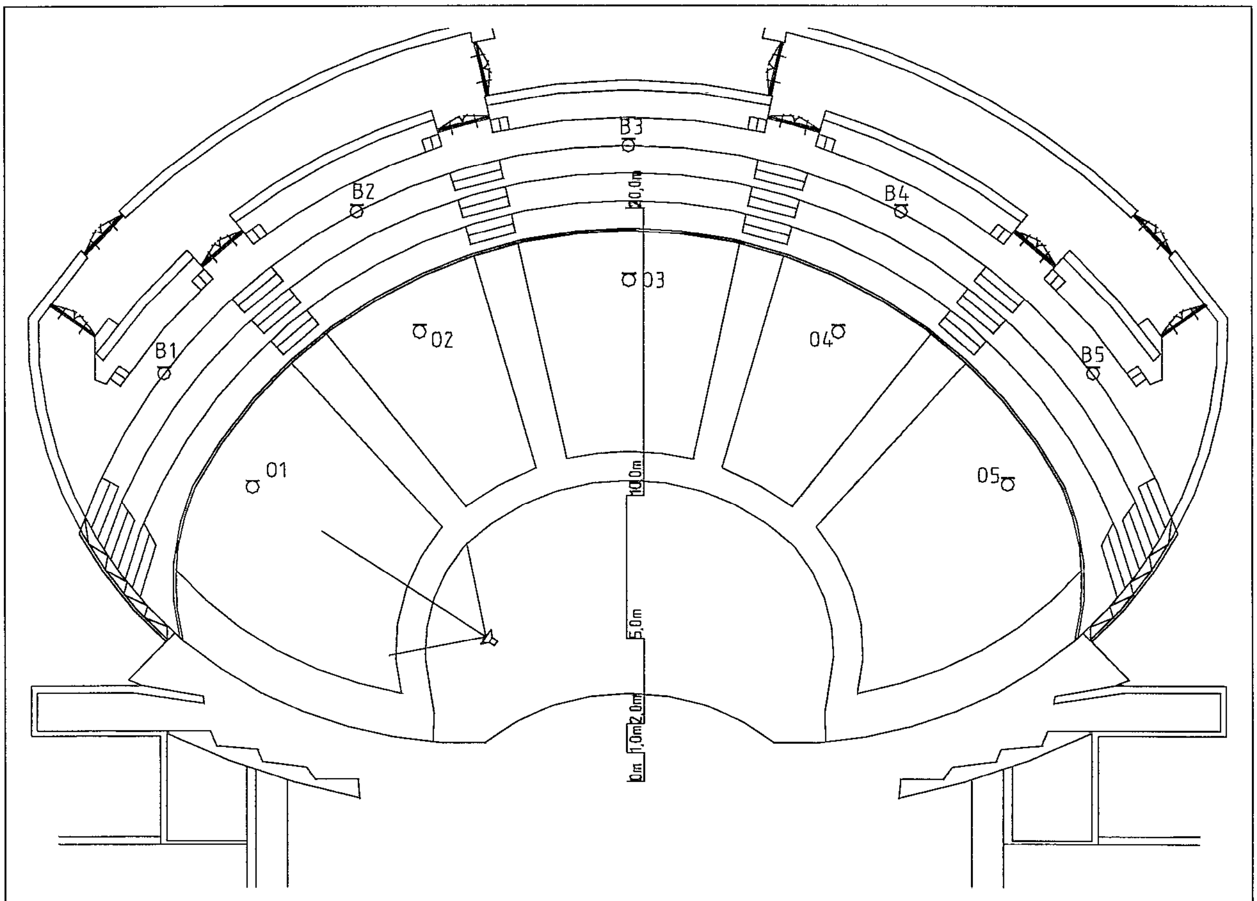

Fig. 8. Measurement Positions

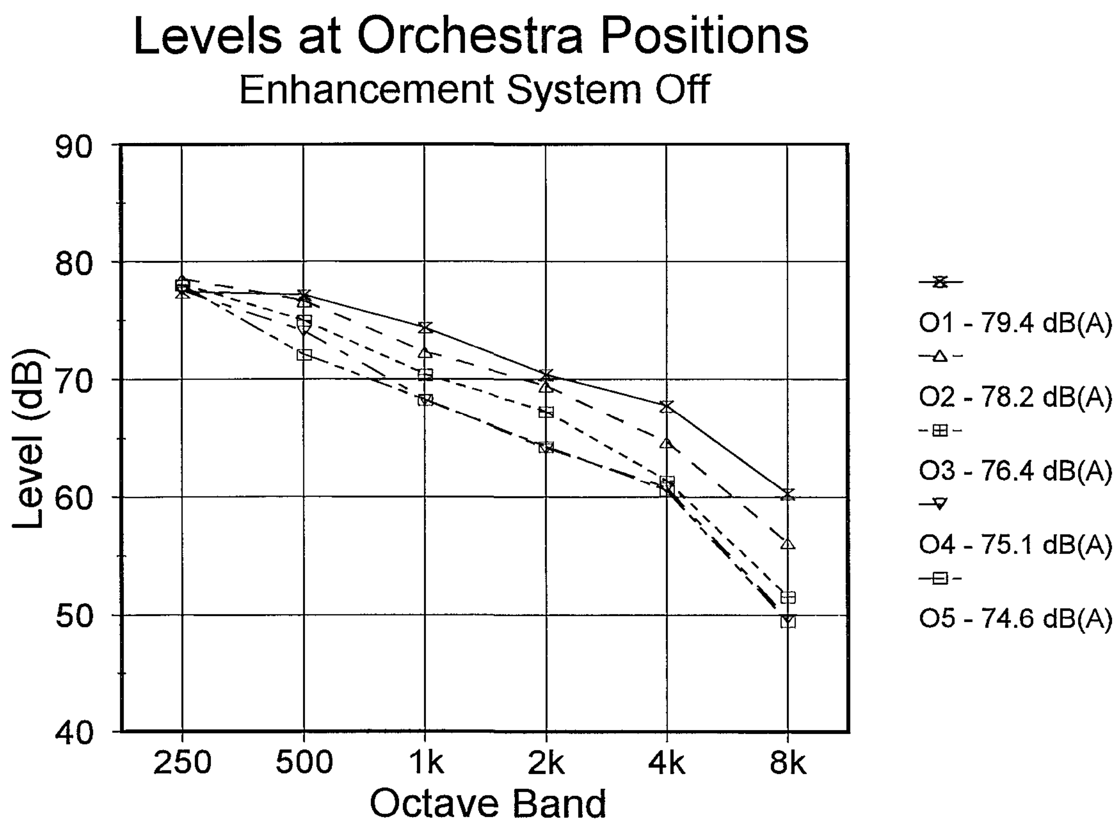

Fig. 9. Level Distribution,
Orchestra Positions, Enhancement System Off

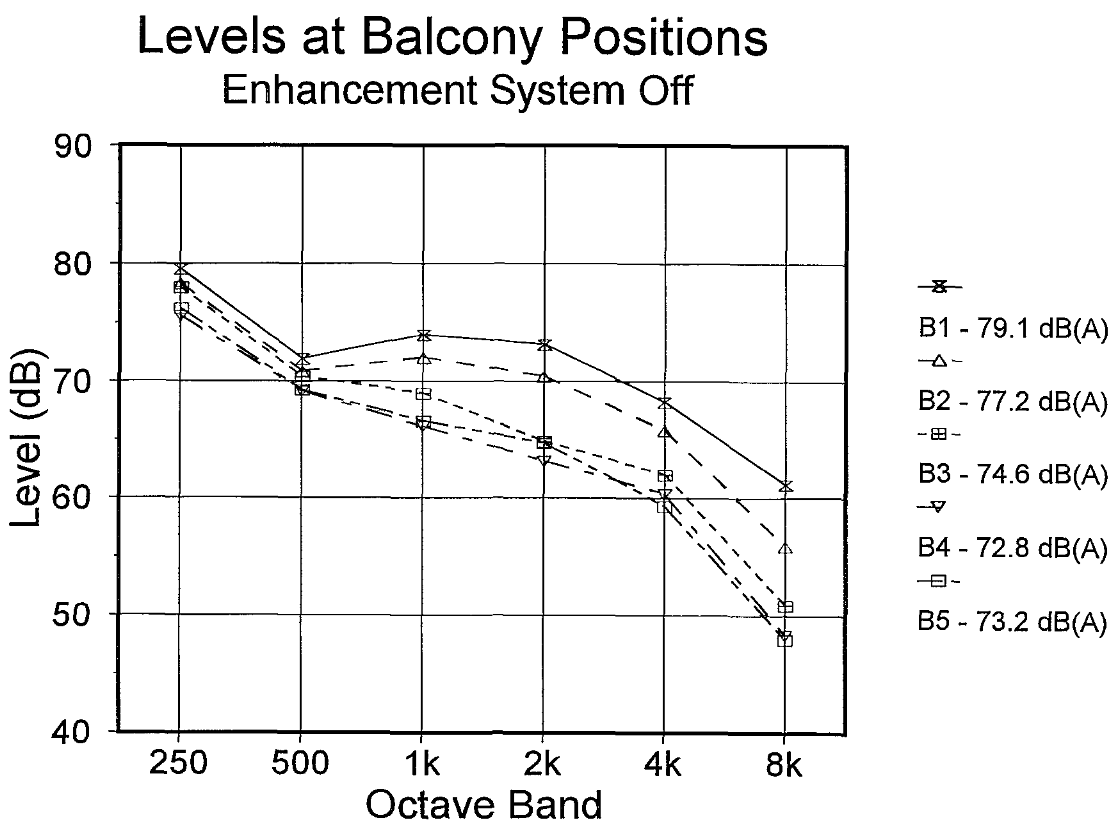

Fig. 10. Level Distribution,
Balcony Positions, Enhancement System Off

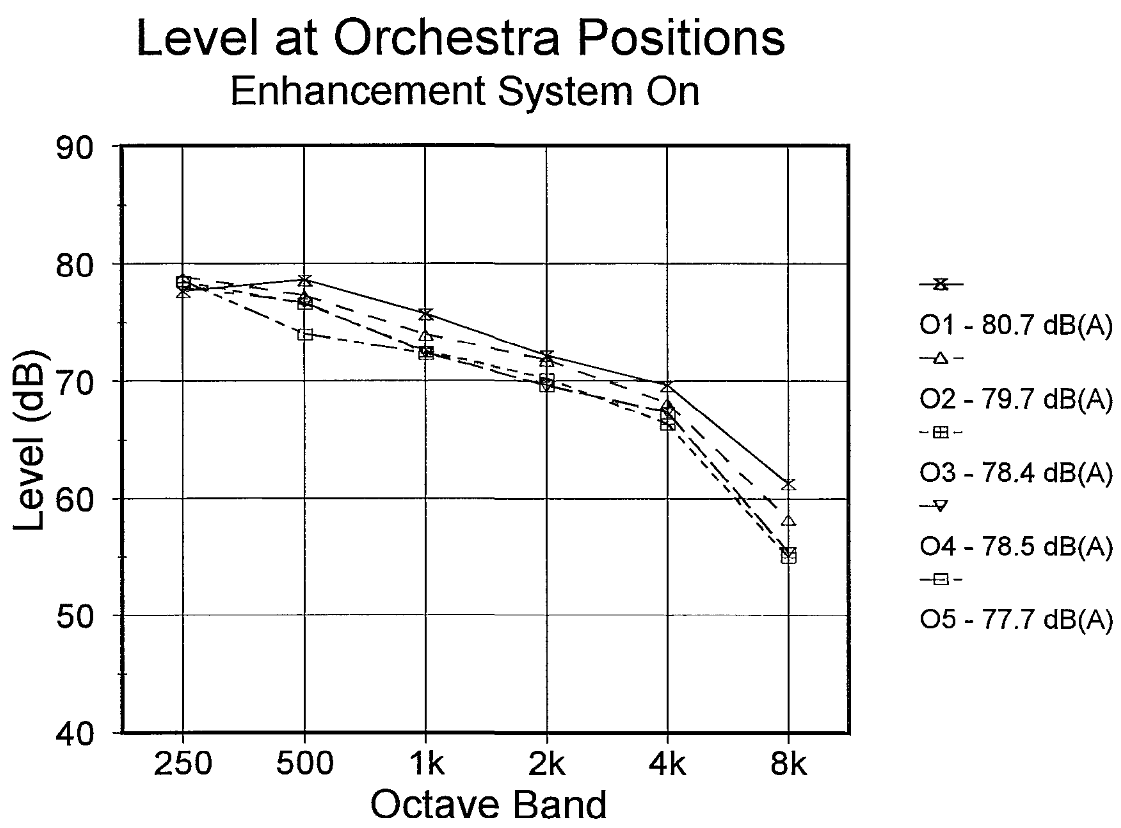

Fig. 11. Level Distribution,
Orchestra Positions, Enhancement System On

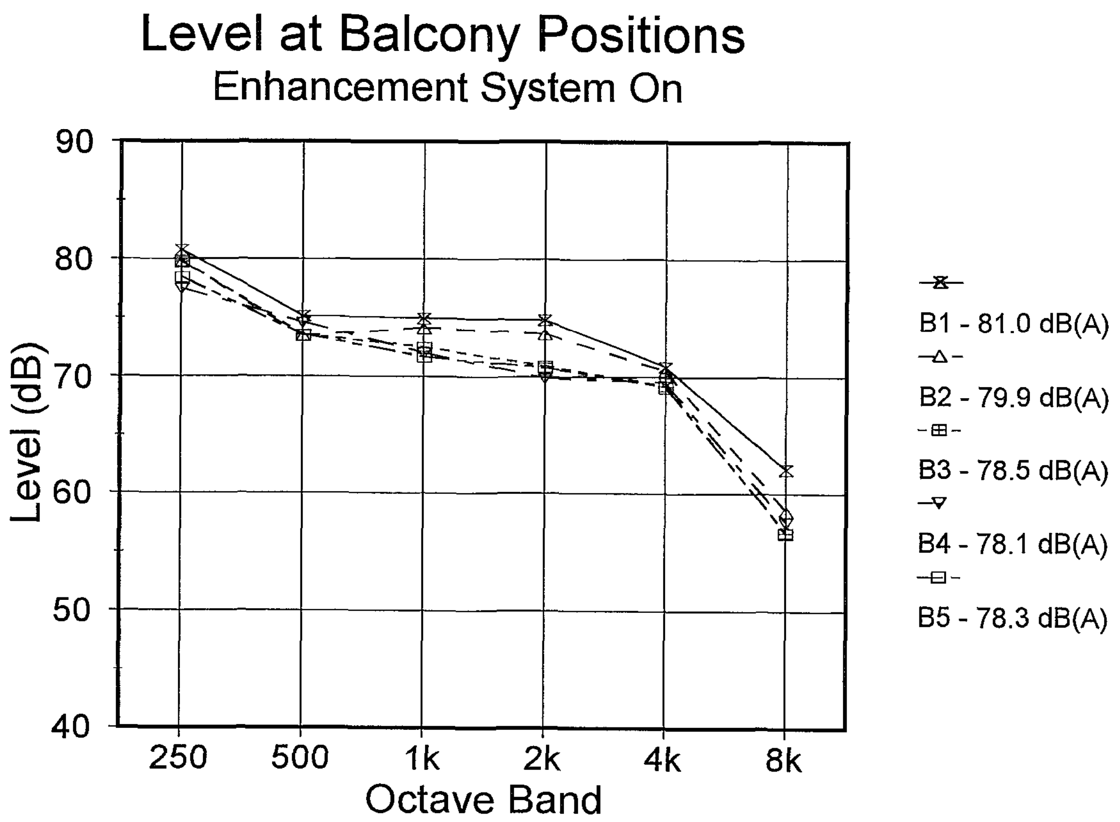

Fig. 12. Level Distribution,
Orchestra Positions, Enhancement System On

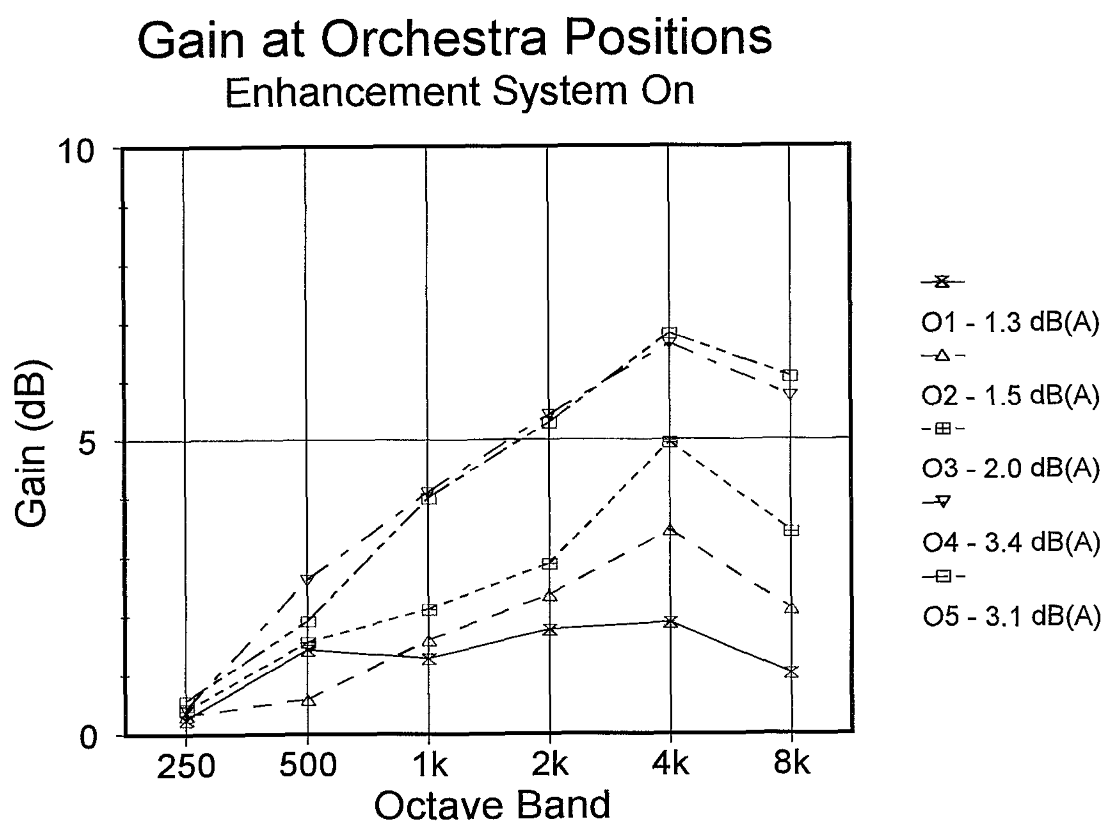

Fig. 13. Gain,
Orchestra Positions

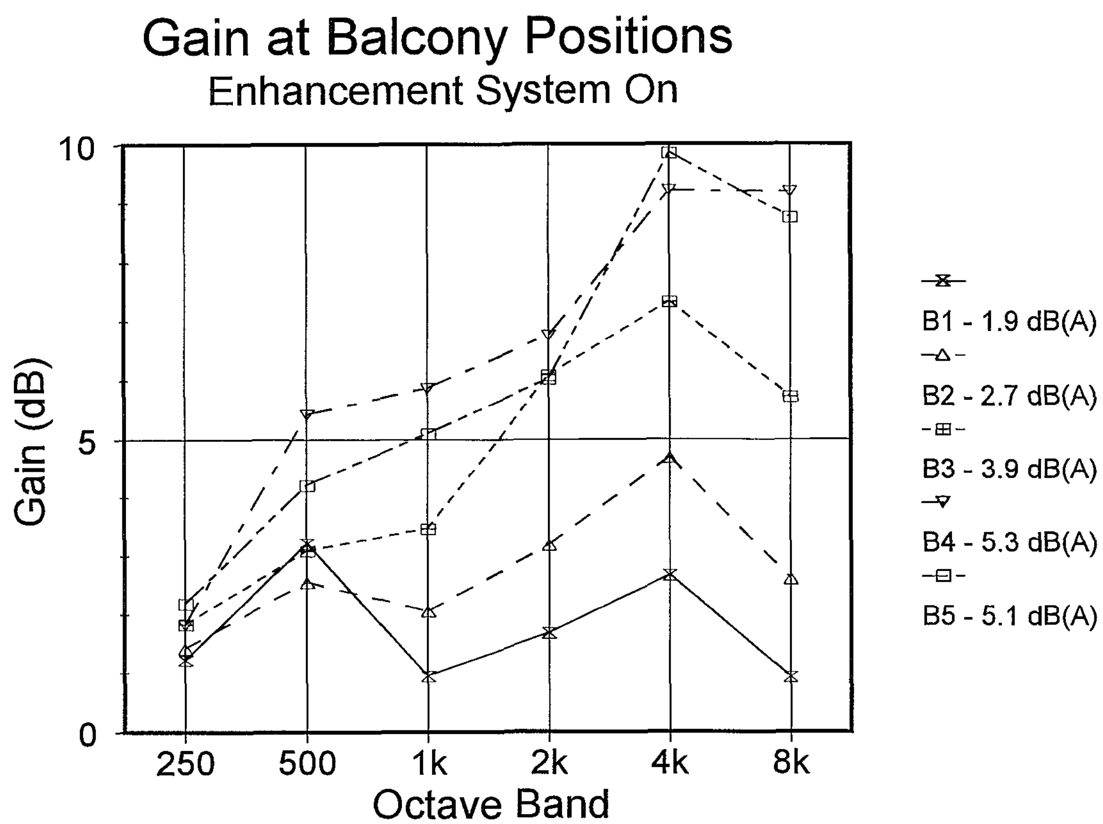

Fig. 14. Gain,
Balcony Positions

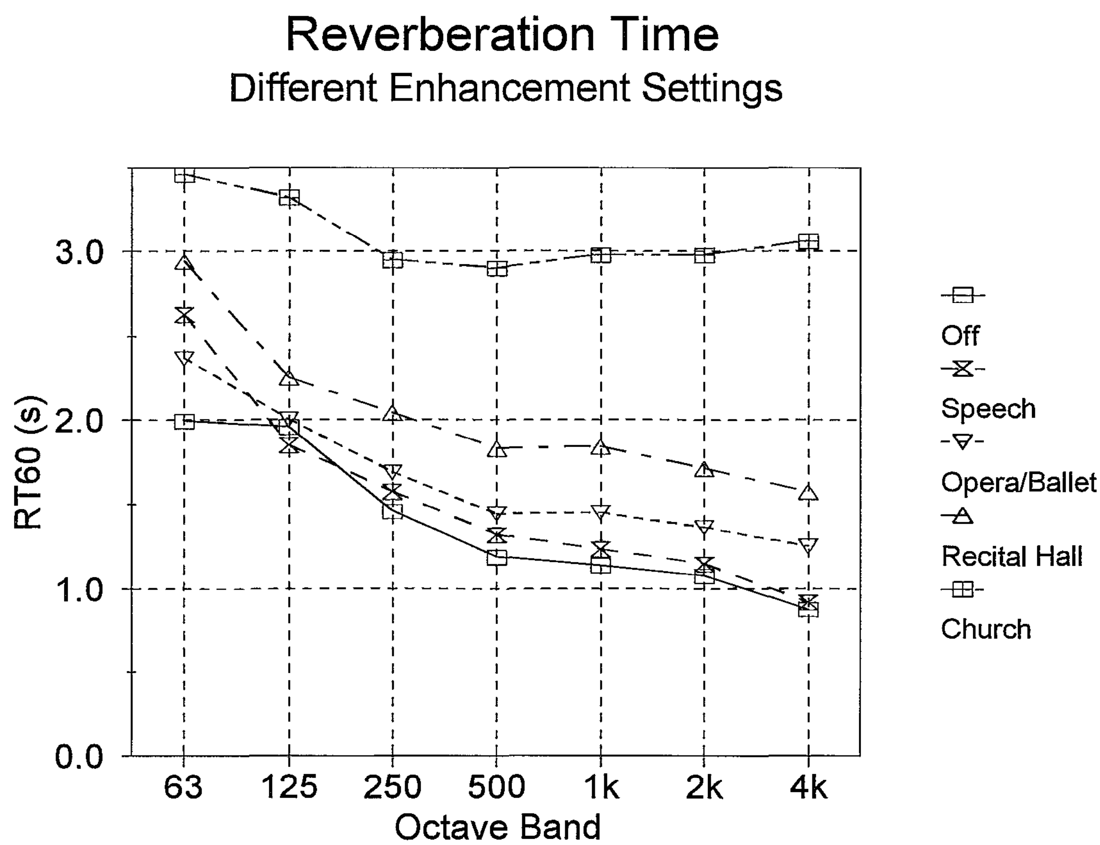

Fig. 15. Reverberation Time,
System Off and A Selection of Enhancement Settings