DESIGN CRITERIA FOR ACOUSTIC ENHANCEMENT SYSTEMS

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ABSTRACT
Acoustic Enhancement Systems have evolved significantly - both technically and by acceptance - over the past decade. Numerous publications are available covering the design philosophy and successful installations.
For people not directly involved in the implementation of these systems, it often is difficult to determine if a proposed design will perform to expectation in their situation. They are often confronted with large numbers of loudspeakers and microphones at - generally - undesirable places, but have no way of judging if the proposed configuration is the optimum, or that modifications are possible without negative influence to the end result.
This paper will describe basic design criteria for Acoustic Enhancement Systems like loudspeaker density, microphone positions, number of required processor in- and outputs, etc. These criteria can act as a tool to evaluate proposed designs, as well as a way to estimate the required provisions to be included in planning such an installation.
By presenting these criteria we hope to give our contribution to demystifying the "magic art" of designing Acoustic Enhancement Systems and thereby further increasing their acceptance as a viable tool in modern acoustic design practice.

INTRODUCTION

Acoustic enhancement systems are applied in situation where either the natural acoustics of an auditorium is inappropriate and/or when variable acoustics are desired. The basic principle is to pick up the sound from the stage with a number of microphones, process this signal in some way and to radiate the processed signal back into the room through a dedicated loudspeaker system. The intention is that the processing adds the desired properties to the natural acoustics of the room (or, in some philosophies, overrules the natural acoustics of the room).
As with any technical system, one has to distinguish the overall resulting performance specifications from the technical specification of the individual components. Both aspects will be covered in this paper.

PERFORMANCE SPECIFICATIONS

Reverberation time

As current enhancement systems add energy to the space, there always will be an increase in reverberation time [1], for reduction of reverberation time a kind of active absorption using anti-noise techniques would be required. In general, increased reverberation times are just what is required.

The earliest systems [1,2] were based on acoustic feedback to increase the reverberation time. In current systems [3,4,5,6], however, the reverberation is achieved using an electronic processor and the desired reverberation time can more or less freely be chosen within the limits of the processor. Nowadays reverberation time variations from 1 s to beyond 5 s easily can be achieved.

Level increase

Increase of reverberation time implies increase of level. In the feedback based systems the level increase will be directly proportional to the change of reverberation time. In the more advanced electronic processor based systems this relation is not this strict. With our SIAP system it is possible to obtain an increase in reverberation time from 1 s to 2 s, with an increase in overall level of less than 0.5 dB while maintaining a single slope decay curve. This is a significant advantage in small auditoriums where the otherwise required level increase of 3 dB would result in a too loud room, especially with a symphony orchestra. At the same time it is possible to obtain a significant level increase of 3 dB to 6 dB with only a minimum of increase of the reverberation time [7].

Clarity

By changing the shape of the reverberation envelope in the processor, one can manipulate the perceived clarity and subjective size and shape of the room. A small room can subjectively be made larger by applying a plateau in the reverberation curve, on the other side; the definition of a too wide room can be increased by the introduction of early energy.

Lateral efficiency

Not only the envelope of the perceived reverberation curve is of importance, but also the direction of the sound field. With enhancement systems this can be manipulated. Increasing the energy to sidewall loudspeakers will increase the lateral efficiency, while increasing the energy to frontal and/or medial loudspeakers will increase the definition.

Localisation

The enhancement system should have no negative influence on the localisation of the original source. At the same time, the loudspeakers should not be localised.

Coupled spaces

In natural acoustics, coupled spaces - like under balcony areas - often are troublesome. The limited coupling to the main volume results in reduced definition and reverberation coming “through a hole”. With enhancement systems it is quite easy to provide these areas with loudspeakers [8] and thereby offering the audience in these areas the same acoustic quality as in the main volume.

Stage

Lots of enhancement systems are used to provide (small to medium) drama theatres with variable acoustics. This enables municipal theatres to schedule chamber music and symphonic concerts. The fly tower with soft goods, however, is anything but a proper stage for these orchestras. Like with coupled spaces, it is relatively easy to provide the stage with increased reverberation and level. This alone, however, is not sufficient for good stage acoustics for orchestras. To improve the support very early reflections are required. Given the allowed microphone and loudspeaker positions the minimum delay of the enhancement system in a lot of situations will be too long to provide these very early reflections. The application of a - limited - orchestra shell can overcome this. As energy (level) considerations are covered by the enhancement system, this shell only has to provide early reflections at mid and high frequencies and can therefore be relatively lightweight and there is no need for it to be fully closed. In some situations even the ceiling can be omitted and replaced by suspended loudspeakers.

Output level
It will be without doubt that the enhancement system has to be capable of enhancing - without audible distortion - the full dynamic range of the performance. The output requirement of the enhancement system is directly related to the desired level increase. If the increase is 3 dB, the enhancement system has to be capable of the same level in the hall as the performance will deliver without enhancement. Is the required level increase lower, then the enhancement system will have to supply significantly less output; for a 1 dB level increase the contribution of the enhancement system will only be -6 dB.

Bandwidth
For acoustic enhancement purposes a frequency range of about 50 to 10,000 Hz is adequate. Due to absorption in the air and of the materials in the room, the reflected sound in good natural acoustics of a room contains only limited energy above 5 kHz, the extra octave, however, is required for proper transient response. Frequencies above 10 kHz appear to be of little significance for the perceived acoustic. Although some musical instruments go lower than 50 Hz, the presence of these frequencies in the natural room acoustic is generally sufficient. The energy and sound quality an enhancement system provides between 50 and 125 Hz is responsible for the perceived warmth of the sound. For this it is important the system does enhance the RT and the sound level to about the same extend (or maybe a little more if extra warmth is desired) as at mid frequencies.

Coloration
The enhancement system has to be free of coloration due to frequency selective decay or other causes. So both the applied processing and the feedback mechanism have to be such that this is avoided.

Self-induced noise
The self-induced noise of the enhancement system should not be perceivable by the audience. As the frequency content of the self-induced noise is different from the normal background noise in the room it will easily be recognised and disturbing, even if the level does not exceed the background level in the room.

Increase of background noise
As a result of the enhancement system, the overall background noise in the auditorium will be increased. This increase should be as low as possible and should never exceed the level increase provided by the system (system gain).

EQUIPMENT

In the previous chapter we described the performance specifications, which, to a certain degree, are system independent. The specifications of the equipment to use are more closely related to the design concept of a given system. As our experience is based on the application of the SIAP system as developed by our company, the criteria give hereafter are based on this system and may not be applicable in all situations for other concepts.

System concept

The concept of SIAP is given in fig. 1. 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.

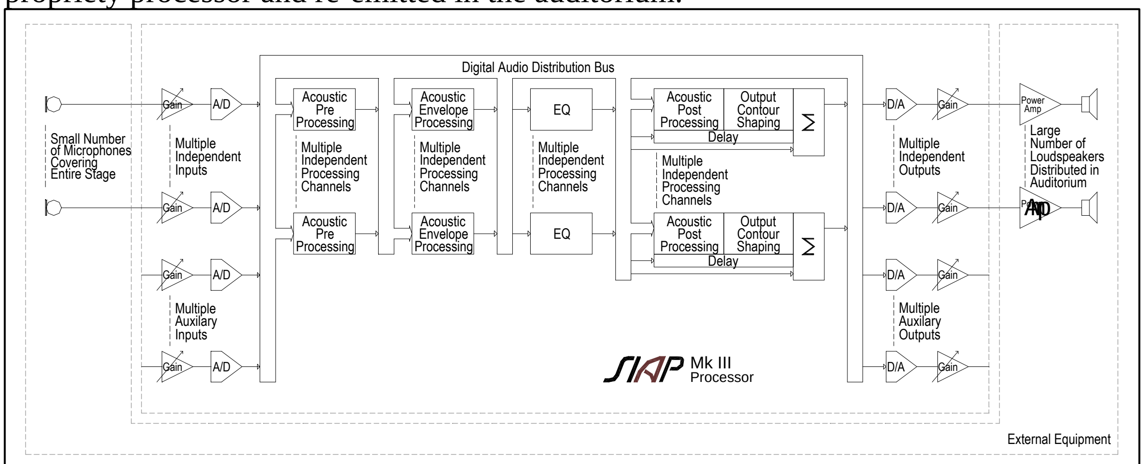

Fig. 1. Concept of SIAP system

Compared to traditional reinforcement systems the major differences with are:

- 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.
- Multiple independent processing channels.
  To simulate spatial diffusion, a high number of different output channels are generated. In addition, the number of de-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.
- Large number of loudspeaker positions
  overlap. Each seat receives sound from multiple loudspeakers, each reproducing a differently processed signal.

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 picked up changes with the orientation of the source. This enables one to maintain, for example, 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 localisation of the original source on stage.
The signal picked up by the microphones should be relatively free from reverberation and background noise; this can be optimised by both placement – close to the stage, away from noise sources – and directional properties. On the other hand, microphones have to be on a certain distance from the stage to enable proper blending of all the sources on stage and even coverage. Another consideration in positioning the microphones is the delay caused by the source-to-microphone distances. It will be clear that the minimum delay in the enhancement signal never can be less than this delay and particularly when increase of clarity is desired, the microphones should be relatively close to the source. A typical microphone position would be at 6 m height, just above the edge of the stage. In some situations, however, this will conflict with sight and/or light lines and the microphones have to move to a more distant position. Spacing between individual microphones should be sufficient to ensure a low degree of coherency between the signals being picked up, typically a distance of 3 m between the microphones is desirable. A typical microphone configuration is given in fig.2.
The microphones have to meet the following criteria:

- Smooth frequency range
- Frequency independent directional pattern
- High suppression of reverberation and background noise
- Small size, to minimise visibility

These criteria are met only by high quality, first order gradient, microphones. Microphones with a higher directivity – like shotgun microphones – are not suitable, as their directional patterns are highly frequency dependent.

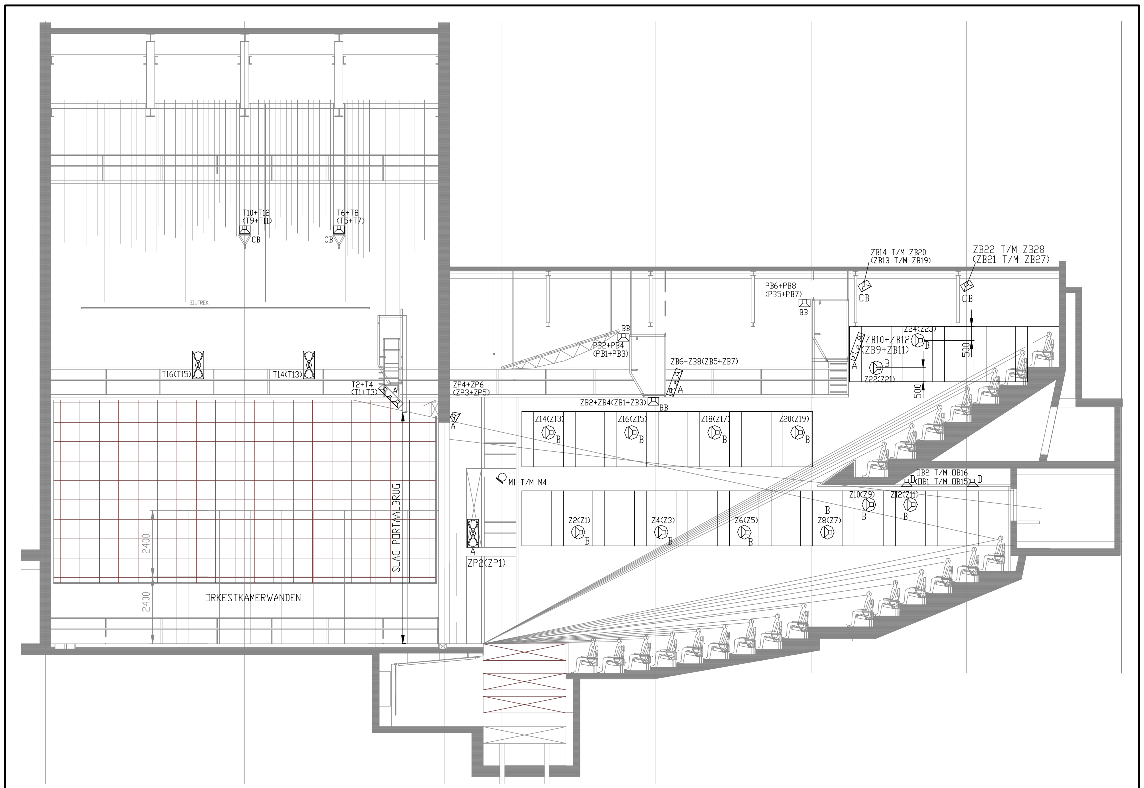

Fig. 2. Microphone and Loudspeaker placement in Medium sized Theatre

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 are often poorly coupled to the main volume, generally will be provided with their own sub-system, integrated in the main system design. An example is given in fig. 2.

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.

Generally a coverage overlap of 100 % or more is desired. This means that at any seat in the house the sound from the nearest plane is always handled by a minimum of two, but

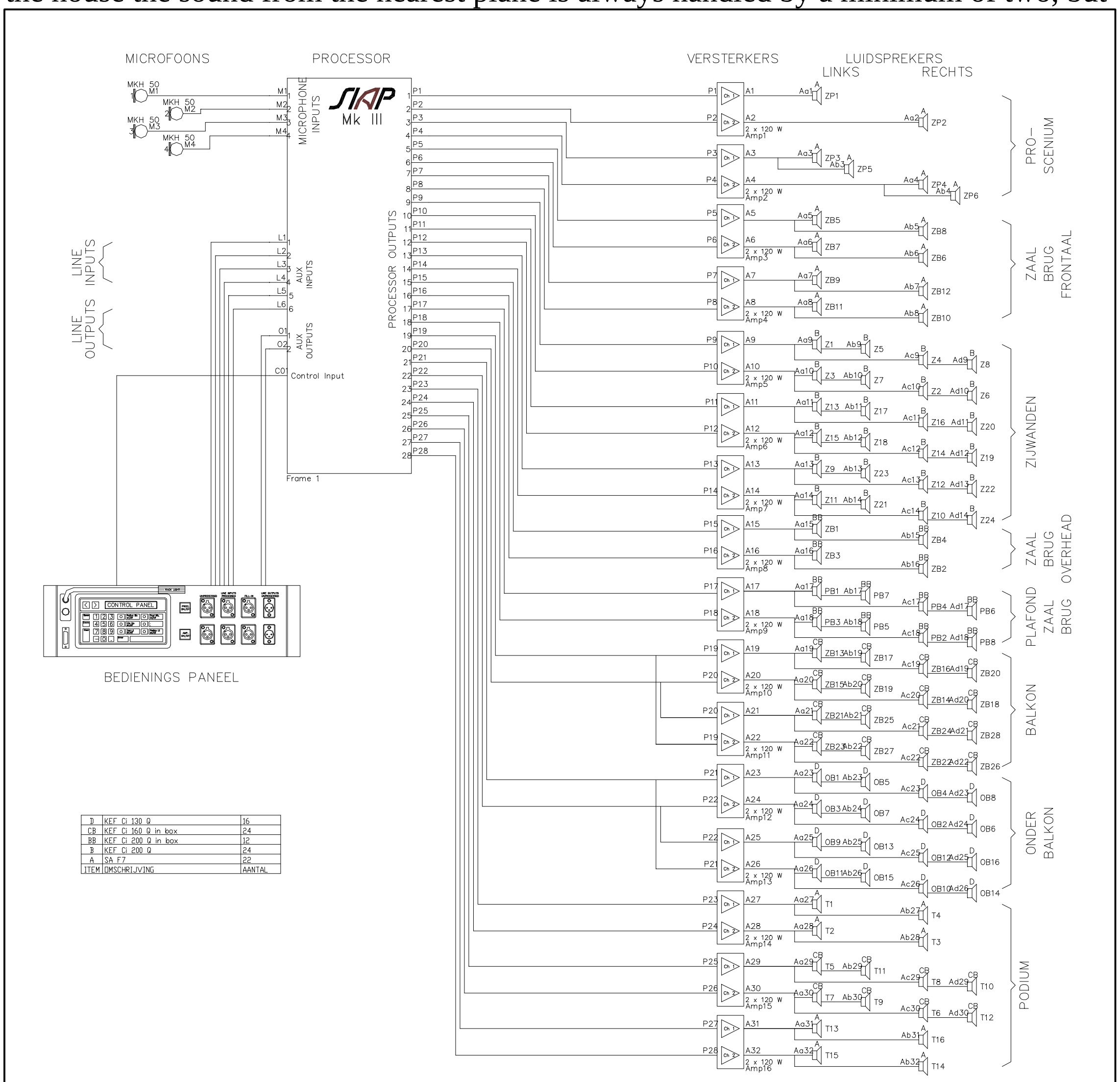

Fig. 3. Block Diagram of SIAP System

preferably four, loudspeakers. In addition, these loudspeakers each have to reproduce a different signal to avoid comb filtering and localisation of the loudspeakers.
By balancing the levels between frontal, lateral and overhead systems the spatial impression of the room can be influenced (in addition to the effects of the reverberation envelope).

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 localisation 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.
During the tuning, the system will be 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.
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. There are a number of 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.
In the design of the system a decision has to be made on the processor configuration, i.e. the number of in- and outputs. The number of inputs is typically 4 for a proscenium type theatre and can be more than 10 for more complex stages. The number of outputs will depend mainly on the number of loudspeaker groups required to obtain the proper coverage and balance between frontal, lateral and overhead energy. In addition care has to be taken that the system contains sufficient independent, de-correlated, signal paths. A single signal path allows for only –20 dB acoustic gain [1]. By creating multiple de-correlated signal paths, acoustic gain can be increased proportional to the number of signal paths. A 4 input, 25 output SIAP processor offers 100 de-correlated signal paths and therefore 20 dB more gain than a single channel system.
The dynamic range of the processor has to be sufficient to allow for both inaudible self-induced noise and undistorted maximum level reproduction. Dynamic range processing generally is undesirable, as this will alter the enhancement as a function of the level. The dynamic range of the SIAP processor is better than 95 dB, this means that at a level increase of 3 dB and a maximum level in the auditorium of 110 dB, the self-induced noise will be less than 12 dB.

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:

- 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.

- 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.
- 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.
- 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.

SUMMARY

Global design criteria for Acoustic Enhancement Systems are presented. In some aspects - output level, bandwidth, dynamic range - these are comparable to conventional sound systems; in other aspects - microphone and loudspeaker placement, parallel processing of signal, high degrees of overlap in both microphone and loudspeaker coverage - the design criteria are in contradiction with traditional sound system design. This is a result of the fact that not optimisation of the direct sound is the goal of an enhancement system, but just the opposite, the enhancement of the room response.

REFERENCES

[1] S.H. De Koning, “The MCR system – multiple channel amplification of reverberation”, Philips Technical Review 41 No. 1, 1983-1984

[2] P.H. Parkin and K.Morgan “Assisted Resonance in the Royal Festival Hall, London 1965-1969”, Journal of the Acoustical Society of America, Vol 48, November 1970

[3] W. Prinssen and B. Kok, “Technical innovations in the field of electronic modification of acoustic spaces”, Proceedings of the institute of Acoustics, Vol 16: part 4 (1994)

[4] W.C.J.M Prinssen and B.H.M. Kok, “Active and Passive Acoustics, Comparison of performance characteristics and practical application possibilities, presentation of a case study”, Proceedings of the institute of Acoustics, Vol 17: part 7 (1995)

[5] D. Griesinger, “Improving room acoustics through time-variant synthetic reverberation”, Audio Engineering Convention Paris, 1990, preprintnr 3014

[6] A.J. Berkhout, “A holographic approach to acoustic control”, Journal of the Audio Engineering Society, vol 36, No. 12 Dec. 1988

[7] B.H.M. Kok and W.C.J.M. Prinssen, “Using Acoustic Enhancement To Improve Speech Intelligibility”, presented at the 103rd AES Covention, New York, 1997, preprintnr. 4576

[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: part 1 (1995)