Acoustic Enhancement Systems

Bachelor-Thesis

by

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Abstract

This Bachelor thesis takes a closer look on acoustic enhancement systems, developed since the 50's of the last century. Synonym for such systems could be electro-acoustic enhancement systems or active acoustic systems. This thesis starts with a recapitulation of what's important in the matter of spatial hearing. It continuous by examining the most important aspects of room acoustics. The reader experienced in these topics can skip the first two chapters and go directly to Chapter 4. In this Chapter, acoustic enhancement systems are explored. Every part of a system is examined and some energy considerations are made. An emphasis is put on the issues of feedback and reverberation generation. Chapter 5 illustrates developed systems of the past and the present. This thesis constitutes an overview of acoustic enhancement systems and shows what these systems are about and what they are capable of.

Contents

1 Introduction

The needs for multi-purpose halls and venues has grown during the last half century. Seat capacities had to raise and the purpose of the halls should not be restricted to solely one, which implicates that the acoustics of a hall should not be restricted to one purpose, either.

It is possible to achieve acoustic variability in architectural means and many modern concert halls possess such options like hanging curtains, turnable panels and moveable surfaces. But in general architectural arrangements constitute a complex, laborious and costly method.

As technology grew and electronic resources became more usual, different trials of acoustic enhancement emerged. Back in the 50's they tried to prolong reverberation with a combination of microphones, resonators and loudspeakers. More and more research was put into this field and today a wide variety of acoustic enhancement systems are available.

Acoustic enhancement in electronic means is cheaper and more flexible. Systems of this kind are installed in venues around the world, whether in hotel event rooms or real concert halls, whether in houses of worship or opera houses, indoor or outdoor. For example, an installation in the Deutsche Staatsoper Berlin was made.

But what can these systems really do? Acoustic enhancement systems (AES), as the name implicates, were developed to enhance the present acoustics in a room. The acoustic problem, first, has to be understood and identified and a clear definition of the aim of an installation has to be made. This takes us back to traditional room acoustics. When these steps are taken, we face the problem of how can the set goals be achieved by installing microphones and loudspeakers, within the restrictions of the prevalent situation and without breaking the budget. If desired, the whole acoustics of a room can be provided by a system, but again a clear knowledge of the existing acoustics is necessary or, in newly built houses, the design of the acoustics has to be made in consideration of a later installation.

Something important to point out is that an AES is not to be confused with a sound reinforcement system or a public address system. In such cases the primary goal is to enhance speech intelligibility, to increase the acoustical arena, so that more people can take part in the event, and to increase the musical impact of the performance. This is achieved by raising the sound pressure to levels which won't occur in situations with non-amplified instruments or speakers. The goal of an AES is to provide a sound field as natural as possible. A critical listener should not be able to distinguish between real acoustics and enhanced acoustics.

To achieve a natural sound field with an enhancement system we face two major problems: the control of feedback, which occurs because microphones and loudspeakers are in the same acoustical environment, and the generation of reverberation. These two topics should be examined closer in this thesis.

2 Perception of Sound

Human perception of sound is a complicated matter and especially the neural perception is not yet described completely. Nevertheless it should build the basis of every sound engineers work.

Many scientist have done research in this field, even back to the middle of the 19th century [Helmholtz, 1863, Rayleigh 1877, Steinhauser 1877]. I want to point out especially on the work of Jens Blauert 'Räumliches Hören' ('Spatial Hearing') [Blauert, 1974].

This chapter should give a brief introduction into how the perception of sound and spatial hearing works.

Spatial Hearing

Intelligibility, *room impression*, *localization* and *envelopment* are the most important properties of interest concerning the perception of sound events in rooms.

When a sound enters the ear, first the mechanical processes of the ear, which are not drawn nearer here, are passed. After conversion from mechanical vibrations to electrical signals there are some processes happening at low neural levels (Griesinger, 1998):

- Analysis of sound in different frequency bands at the basilar membrane.
- The detection of rapid increases in level and the separation of sounds into individual units (which occurs later in the neural perception).
- Detection of interaural level differences (ILD) and interaural time differences (ITD).
- Detection of fluctuations in ILD and ITD.

The separation in different frequency bands states the first neural process. So the following processes must be seen frequency dependent.

Further the detection of sound endings and beginnings happen. This process determines intelligibility in the end. When the ability of this procedure is disturbed by noise or strong reflections, intelligibility drops.

As humans have two ears you can imagine this is an integral part of hearing. If a sound comes from anywhere beside the median plane there will be different signals at each ear, due to time delay and diffraction, shading and resonances of the head and the outer ear [Blauert, 1974]. There is general consensus that *inter aural level differences* and *inter aural time differences* are the main varieties detected by the hearing system.

Above 1500Hz-1600Hz the ILD becomes the prevalent influence on localization, because of shading of the head, and below the ITD is dominant, because of diffraction [Blauert, 1974], [Zölzer, 2002].

Fluctuations of interaural time and level differences are caused by interference of direct and reflected sound signals, and fewer by head movements (important for the front/back localization). These fluctuations are little variations in the differences and can be imagined as a low frequency modulation. Within a frequency modulation of 3Hz-20Hz, fluctuations lead to envelopment and below they cause a source movement [Griesinger, 1998].

Envelopment is also caused by the reverberation in a room. It is trivial to say that one senses more envelopment in a church than in a typical classroom. Important is not only how long the reverberation stays (reverberation time) but how strong the level is (reverberation level).

In higher neural processes the building of foreground and background streams occur. A foreground stream contains the individual units of speech or musical lines from instruments, and there can be more than one foreground stream (cocktail-party effect, distinction between different foreground streams). But there can only be one background stream which consist of noise, reverberation, etc. [Griesinger, 1997] .

The background stream can become the stream of focus when there is no foreground stream. Imagine a big crescendo of an orchestra during a concert and then an abrupt stop. Suddenly we concentrate on the reverberation which is information of the background stream.

David Griesinger gives in [Griesinger, 1997] a time scale of reflections and how they affect perception (time specifications mean arriving time after the direct sound):

0-10ms: affection of loudness, broadening of source image, change in timbre

10-50ms: cause room impression in terms of the size of the room

50-150ms: reduction of intelligibility and sensation of envelopment

150-end: reverberation

Because our daily hearing is generally reduced to hear sounds within the horizontal plane, it seems that lateral reflections constitute what mainly influences our hearing experience in spaces. This property is used in the reproduction of sound fields in means of wave-field synthesis.

After examining the physiological and neural properties of hearing we can go to observe what happens to sound in a room physically.

3 Room acoustics

A room for musical events or speech applications should provide a few general features to assure a good listening experience.

Intelligibility, for an easy understanding of what a speaker says or of musical lines.

Localization should not be disturbed (e.g. through supporting loudspeakers). This could cause an artificial listening experience.

An adequate *room impression* fitting to the size of the room and especially for music a good *envelopment* to provide a proper experience.

The first engineer dealing with room acoustics or concert hall acoustics was Wallace C. Sabine, a young Harvard assistant professor, at the beginning of the 20th century. He stated that the main attribute of a concert hall is the *reverberation time*, RT.

As technology and science grew, especially the system theory, a powerful tool for acoustically describing a room developed, the *room impulse response*, RIR (Figure 3.1). Mathematically the impulse response is the answer of a system on an ideal impulse, an infinite peak. Practically an RIR is measured with a gun shot or modern mathematical techniques like the maximum length sequence (MLS), the time delay spectrometry (TDS) or exponential sweep technique.

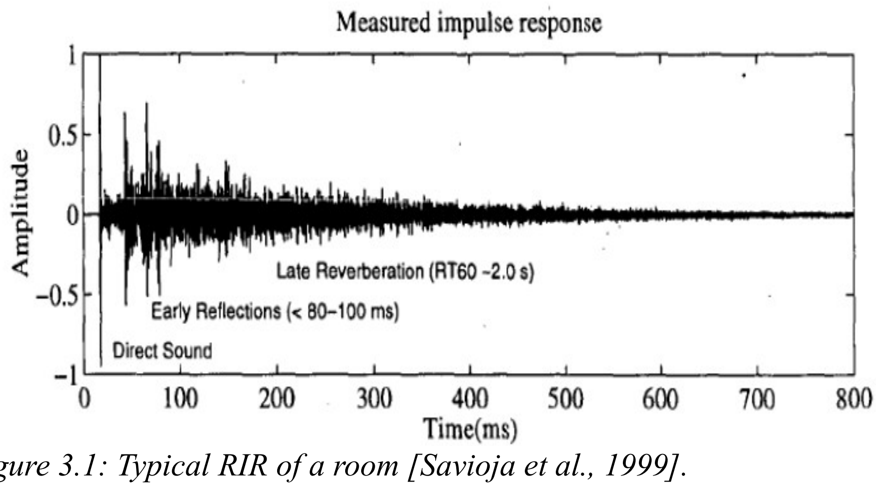

Figure 3.1: Typical RIR of a room [Savioja et al., 1999].

The RIR can be split into three sections of time. At first the direct sound arrives at a listener, the highest peak at the beginning of Figure 3.1. After the direct sound some discrete reflections happen, the early reflections. After the early reflections there is an onset of many, diffuse reflections, the late reverberation.

Out of the impulse response of a room one can derive many parameters which developed useful in acoustics and these are tried to connect with subjective impressions of listening.

A.C. Gade made an interesting experiment with musicians. He went on tour with an symphony orchestra and the musicians had to judge subjectively the concert halls acoustics after they played in it. The assessments were then correlated with objective measurements in these halls. Gade concluded that in halls the musicians didn't know the main property indicating quality was the reverberation time. But in halls they knew, other properties became of increasing importance [Gade, 1982].

So what are the main qualities of a good concert hall?

Leo Beranek gives in [Beranek, 1992] 7 acoustical properties which should be provided by a concert hall:

1) acceptable reverberation time
2) adequate loudness
3) a short initial time delay gap
4) a good number of early lateral reflections to the audience
5) a strong diffuse sound field
6) adequate ratio between the energy of the first 80ms and that in the next 2s
7) warmth of the sound

All these criteria came out of subjective statements and research of the last half decade was focused on valuing these criteria in objective formulas. The basis for the following acoustical parameters build energy considerations. Energy at different times are set in relation to each other. Energy is derived out of the RIR by integrating the squared sound pressure between any time.

$$E=\int_{t_1}^{t_2} p^2 \; dt \tag{3.1}$$

Where E is the energy and p denotes the measured sound pressure. Direct sound makes 5-15% of the entire energy, reverberation 10% and the early reflections 80%. So early energy seems to influence the overall sound impression the most, but that is of course seat location dependent.

The reverberation time is not directly an energy criteria but it could be derived out of the RIR (Schroeder-backward integration) [Weselak, 2007].

Reverberation Time

The reverberation time (RT) is that time a sound needs to decay 60dB of sound pressure level (SPL) independent of what SPL the sound was (how loud it was). After a few years of research in various halls at Harvard, Sabine developed an equation to predict the RT [Beranek, 1992].

$$T_{60}=0.163\frac{V}{A_{tot}}\ [s]\ \cdot \tag{3.2}$$

Where *V (m³)* is the cubic volume and A_{tot} *(m²)* the total sound absorbing surface at the measured frequency. Still today this equation builds a good basis in concert hall design. Eyring derived later the RT in a different way.

$$T_{60}=0.163\frac{V}{-A_{tot}\ln(1-\alpha_m)}\ [s] \tag{3.3}$$

So the volume of a room and the absorbing properties of the surfaces are the factors building the RT. Measuring the RT could be done by measuring the decay time of the RIR from -5dB to -35dB which is then doubled. The same could be done by measuring the decay time from -5dB to -15dB and multiply it by six, which is then referred to as the early decay time, EDT [Beranek, 1992].

During continuous music only 10 dB of sound decay is recognized and it gives music a singing sound and a feeling of envelopment.

Kuhl W. [Beranek, 1992] made a listening test on what is the best reverberation time for different music periods. It came out that for the classical period 1.5s seem adequate and for the romantic period 2.0s. These values are generally accepted since the best concert halls and opera houses have RT's in this range and they constitute the values architects and acoustical consultants head on when designing a new hall.

Loudness

Loudness means the loudness of the reverberation. The stronger the reverberation is the more envelopment occurs. A criteria developed for measuring the loudness is the strength factor, G:

$$G=\frac{\int_{0\text{ms}}^{\infty} p^2(x)\ dt}{\int_{0}^{\Delta t} p^2(r)4\pi\, r_s^2\ dt} \tag{3.4}$$

$\mathit{Optimal\ Values:}\ \ 0\text{dB}<G<10\text{dB}$

It relates the energy of a sound source to the energy of the sound source with radius r_s from the source, usually 10m.

A big *G* means that a source can fill the room with much energy, but intelligibility and clarity will be reduced [Weselak, 2007].

Initial time delay gap

Initial time delay gap t_i is the time it takes the first reflection to arrive at a receiver's position after the direct sound arrived. It was at first mentioned by Beranek [5]. He states that the acoustical quality of halls can be correlated with their width (except for wide halls with hanging panels from the ceiling). If the t_i is short, a big amount of early reflections can follow. A great width prevents the first reflections to come early.

Lateral reflections

Reflections from the side walls give a listener the feeling of envelopment or spatial impression, even more than reverberation, and it grows with the level of these reflections. A good number of lateral reflections seems to make the difference between feeling inside or outside concerning the sound event. [Baranek, 1992].

The best concert halls, especially with an rectangular shape (Vienna Musikvereinssaal, Boston Symphony Hall) provide numerous early lateral reflections. Beranek defines lateral reflections as these reflections that come from in between +-20° to +-90° from the median plane [Baranek, 1992].

An objective criteria forms the lateral energy fraction, LF. It is the ratio between the early lateral energy measured by a figure-8 microphone and the total energy measured by an omnidirectional microphone during the first 80ms.

$$LF = 10 \cdot \log \frac{(E_{5...80})_{side}}{E_{0...80}} \tag{3.5}$$

$$Optimal\,values: \; -6\text{dB} < LF(dB) < -4\text{dB}$$

Diffuse sound field

A diffuse sound field means that no localization of a sound source can be determined and that the overall energy in that area is everywhere the same. It occurs behind the critical distance, which means the distance where the direct energy of the sound source and the reflected energy are the same.

$$r_H = 0.057\sqrt{\left(\frac{V}{T}\right)}\;[m] \tag{3.6}$$

Imagine a steady sound source in a room. If you walk around, at some points the direct source energy will be the predominantly one you are exposed. If you go on, at some points far away from the source you will encounter primary the energy of reflections. And at some points the energies are equal, the critical distance.

So from a room-acoustical view it means that many reflections from every angle have to be provided, especially at high frequencies to avoid acoustical 'glare'. Irregularities and ornamentation, like seen in the great halls from the end of the 19th century, help to diffuse high frequencies.

Clarity

Clarity means the ability of distinguishing successive musical notes and different instrumental groups or the clear detection of words. It seems to be a degree for intimacy. As an objective criteria for music it means the ratio between the energy of the first 80ms and the energy after

80ms.

$$C80 = 10 \cdot \log\left(\frac{E_{0...80}}{E_{80...\inf}}\right) \tag{3.7}$$

$$\begin{aligned} &\textit{Optimal values:} \quad 0\text{dB} < C80 && \textit{for good Clarity,} \\ &\qquad\qquad -3\text{dB} < C80 < 0\ dB && \textit{for envelopment.} \end{aligned}$$

For speech applications the same criteria is used but with the limit of 50ms.

Warmth

The subjective value of warmth is scientifically translated into bass ratio. Because of the ears property of minor sensitivity at low frequencies a raise of reverberation time at low frequencies is desired.

$$BR = \frac{T_{125} + T_{250}}{T_{500} + T_{1000}} > 1 \tag{3.8}$$

$$\textit{Optimal values:} \quad 1.1 < BR < 1.3$$

All these criteria constitute tools for architects and acoustical consultants when designing a hall. Measurements of these criteria should build the starting point when considering an acoustic enhancement system.

4 Acoustic Enhancement Systems

An acoustic enhancement system (AES) is an electro-acoustic system consisting of microphones, a signal processing unit and loudspeakers (Fig. 4.1). These kind of systems take place in rooms for performances, where sound sources and receivers exist. The microphones pick up the sound of a source and pass it to the signal processing unit. It is the core of the system executing the demands and passing it on to the loudspeakers which playback the sound to the room. The exact design of the signal processing unit is dependent on the method used.

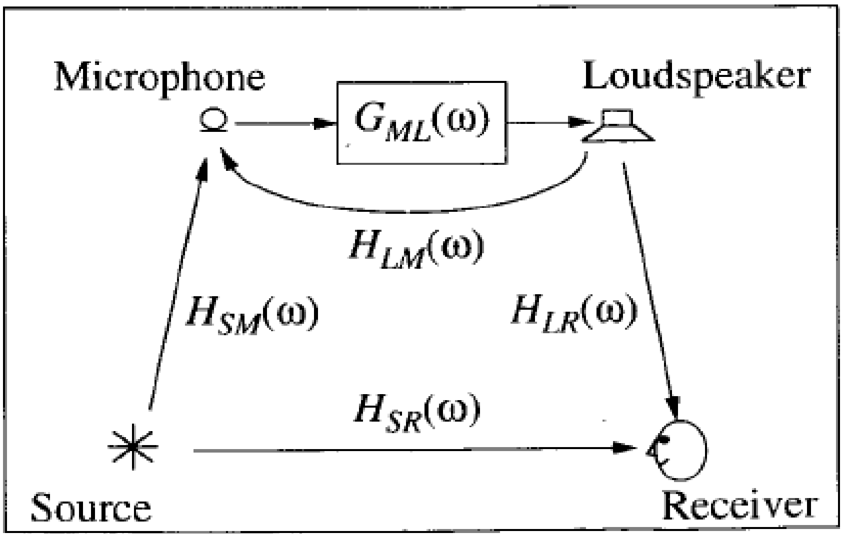

Figure 4.1: A basic model of an electro-acoustic system with the relevant transfer functions. Indices mean: S...source, R...receiver, M...microphones, L...loudspeakers. [Svensson, 1995]

These kind of systems have been emerged since the 50's and have been commercially available. Early systems used mechanical resonators or magnetic tape loops (see ch. 5.1). Today such systems can offer a wide variate, from enhancing reverberation time and level to providing the whole acoustics in an acoustical dry room.

A big advantage is the possibility of providing many different settings for multi-purpose rooms by using electronic signal processing units.

There are two different types of systems. In-line systems and feedback or non in-line systems. The main difference is the way of handling feedback or the way they handle the transfer function $H_{LM}(\omega)$. Some systems combine both, an in-line part and a feedback part. Generally, when installing such a system, it has to be made sure that the localization of the source is not disturbed. No signal from the system may arrive earlier than the original sound (Precedence effect). This is especially important for in-line systems where microphones are placed closer to the sound source.

It is possible to extract the impulse response of the acoustic system $H_{LR}(\omega)$ by measuring the impulse response of the room $H_{SR}(\omega)$ with the system on and with the system off. The subtraction of these two RIRs result in the IR of the system. Dependent on what the purpose is, this electro-acoustic IR has to be designed.

In-Line Systems

In In-line systems the direct sound of the stage is picked up by directional microphones placed inside the critical distance. The signals are processed and given back to the audience. Acoustical feedback between microphones and loudspeakers is not desired. So the signal processing unit should constitute the only process changing the impulse response at a listeners position with delay, phase and frequency response changes. Which means that reverberation generation in the signal processing unit is necessary. The focus then lies on reducing feedback and on the reverberation generation.

In-line systems therefore are highly flexible. The natural acoustics can be influenced strongly, especially by changing early energy behaviour.

An issue to consider is that if the microphones cover only the stage area the sources on stage cause more reverberation than sources in the audience area. That can cause an unnatural impression but on the other hand it increases the signal to noise ratio between stage and audience.

Feedback Systems

A feedback systems basis is the feedback between microphones and loudspeakers. Reverberation is generated by loops between microphones and loudspeakers. Omni directional microphones placed behind the critical distance pick up the sound and give it back via filters and amplifiers over the loudspeakers. No signal processing is necessary, though some systems use it.

The main problem is stability and the right calibration of the loop gain. If the loop gain in a channel is too high the system becomes unstable. So the loop gain must be kept low or a higher number of channels has to be introduced. Both constricts the influence of the system.

Such systems could be narrow or wideband. In narrow band use the loop gain could be higher but many channels are necessary. Wide band systems afford elaborate equalization to keep loop gain higher but are not that extensive.

Hybrid Systems

Hybrid Systems combine both methods. They use a feedback system and an in-line system simultaneously. The right combination of both is the main problem, which though leads to high flexibility.

Microphones

Which kind of microphones are used is dependent on the purpose of the system and on what kind of system or what method should produce the desired results. In-line systems use directional microphones, placed more closely to the source and need an adequate amount to cover a big area, it has to be in mind that directional microphones always lead to a coloured sound image. Non in-line system use omni-directional microphones, placed more far from the source. Less can be used because they cover a big area anyway.

In either way, microphones have to be placed at a distance sufficient far enough from the sources, so that no near-field sound effect of an instrument is picked up, which can lead to an unnatural result. This requirement displays the difference to public address systems. Additionally pick up from noise sources, such as light gear have to be avoided [Engel and Walter, 2006].

The quality of the microphones has to be extraordinary, especially the signal-to-noise ratio has to be high and they have to keep their properties over a long period of time otherwise a general calibration of the system has to be done over and over. The same counts for microphone pre-amplifiers which must have an excellent signal to noise ratio.

A problem is the view of the microphones. If they easily can be seen, the audience may have a wrong impression of what they are for or the view on the stage is disturbed. So an aesthetic point of view has to taken into consideration.

A good performance of the system is dependent on the placement of the microphones. It is the most sensitive part of the systems calibration.

Loudspeakers

Principally the same quality requirements are to be aimed for loudspeakers as they are for microphones. Additionally it is not acceptable to hear noise from the loudspeakers, which is important especially for listeners sitting near to loudspeakers. May a special solution for powering has to be found, like the speakers of the Meyer Sound Constellation system which uses active speakers with a DC input voltage between 12V-18V [Meyer Sound Brochure, 2009]. Noise gates can be used as well.

Directionality is a different problem because the majority of commercially available loudspeakers have directional radiation characteristics. Different types of loudspeakers and additional sub woofers for low frequency radiation can be used.

The direction of the speakers is dependent on the method used. In in-line systems the speakers should rather direct into the audience area so that no further undesired reflections occur. In feedback systems a compromise has to be found. The Assisted Acoustics System of the Yamaha Corp. uses loudspeakers for late reverberation directed towards the surfaces to decrease the direct sound components for the listener and the microphones.

To cover a big area and every seat in an auditoria, a big amount of loudspeakers has to be installed, mostly a few times more than microphones.

Again the same as with microphones, the view on the loudspeakers has to be made aesthetically acceptable. Architectural solutions have to be found.

Main challenges

Enhancement systems can cause unnatural behaviour of the reverberation, which causes an artificial sound experience. So the main problem is colouration of the sound field. It happens mainly due to undesired feedback, which will be examined closer, later. If an installed system in a hall is noticeable for a listener visiting the hall for the first time, it fails. May special focus should be put on the loudspeakers, because they are the weakest element in the system but have to be used extensively.

Another issue which can cause colouration of the sound field is the generation of reverberation. Reverberation has to be produced in the most sophisticated way possible. No artificiality should be hearable.

Before installing an active acoustic enhancement system it has to be considered that a system only adds energy. It cannot be used for reducing undesired energy or reducing noise. That means that the basic room-acoustic situation may has to be improved before installing an enhancement system.

An AES increases the sound pressure level (SPL) in a room. The ratio of reverberation time and reverberation level has to be considered carefully. A small room has loud reverberation and a low reverberation time. So a concert hall with too loud reverberation can seem smaller. Especially in small rooms additional absorption can be necessary for an AES to work appropriately.

In the next sub chapter it should be analysed what happens to sound energy in a room with an AES.

4.1 Energy considerations

The following considerations are based on diffuse field theory [Svensson, 1994] [Weselak, 2007]. An energy-based model considers only mean values of the energy density, or the pressure squared, over a frequency range.

Energy density means the energy stored in a given system or region of space per unit volume.

$$w=\frac{dE}{dV} \quad [\frac{J}{m^3}] \quad . \tag{4.1}$$

Herein w denotes the energy density, E the energy [J] and V the volume [m³].

The geometrics of the room are irrelevant, and no room resonances are taken into consideration. The assumption of an diffuse sound field is made, which has an equal energy density in the whole room. The sound source is seen as an energy source and the absorption of the surfaces as an energy appliance.

The energy balance is

$$\begin{gathered} V\cdot dE=P_0\cdot dt-P_\alpha\cdot dt \\ energy\,change\ =\ energy\,source-energy\,appliance \end{gathered} \tag{4.2}$$

where P_0 means the source or input power and

$$P_\alpha = A_{tot}\frac{E\cdot c}{4} \tag{4.3}$$

is the absorbed power, where A_{tot} is the total sound absorbing area and c the speed of sound.

Further the power balance is

$$V\cdot\frac{dw}{dt} = P_0 - A_{tot}\frac{w\cdot c}{4} \quad , \tag{4.4}$$

where $\frac{w\cdot c}{4}$ means the average incident intensity on the wall surfaces.

The steady-state answer then is

$$w_s = P_0\cdot\frac{4}{A_{tot}c} \quad , \tag{4.5}$$

and when the input power is set to zero the decay is

$$w(t) = w_0\cdot \mathrm{e}^{\frac{-A_{tot}c}{4V}t} \quad . \tag{4.6}$$

If there is an electro-acoustic system with n_L loudspeakers the power balance can be modified. In this case the second assumption is that the natural source and the artificial source can be seen as uncorrelated. This can be applied when the sources are at least one wave-length of the desired lowest frequency apart.

$$V\cdot\frac{dw}{dt} = P_0 + n_L\cdot P_L - A_{tot}\frac{w\cdot c}{4} \tag{4.7}$$

where $P_L = \mu^2 w$ is the power of the loudspeakers with μ^2, the gain factor.

The steady-state answer is then

$$w_{s,on} = P_0\cdot\frac{4}{A_{tot}c}\cdot\frac{1}{1-n_L\frac{4\mu^2}{A_{tot}c}} = w_{s,off}\cdot\frac{1}{1-n_L S^2} \quad , \tag{4.8}$$

where the power loop gain is

$$S^2 = \frac{4\mu^2}{A_{tot}c} \quad . \tag{4.9}$$

The decay is modified to

$$w(t) = w_0\cdot \mathrm{e}^{\frac{-A_{tot}c}{4V}\left(1-n_L\frac{4\mu^2}{A_{tot}c}\right)t} \quad . \tag{4.10}$$

If the expression nS^2 in eq. 4.8 approximates unity the system becomes unstable.

The issue gets more and more complicated if the desired effects are to change the early energy behaviour of the hall. For example if enhancement of intelligibility of speech or singers is desired, early energy has to added to the hall. Changing the early part of a hall's impulse response is a risky matter because it easily can give an inhomogeneous impression of the sound field. The issue of microphones and their positions gets even more important as they have to cover the entire source area and maybe to handle with moving sources [Svensson, 1994] . For the systems ability to provide early energy it is required that the systems latency is kept very low. If the signal path with signal processing lasts too long it is useless for giving early energy.

With these energy considerations the effect of an acoustic enhancement system can be simulated. But one has to be aware that the early response cannot be simulated with this method because of the diffuse field assumption. Further research in simulation was made by Svensson [Svensson, 1994].

4.2 Feedback

In every acoustical system consisting of microphones and loudspeakers in the same acoustical environment, energy picked up by the microphones and given back by the loudspeakers is again picked up by the microphones. Additionally reflections of the loudspeakers signals at walls are picked up by the microphones, as well.

When regeneration occurs, the direct and the reflected sound sum. At some frequencies they sum in phase and increase the amplitude at others out of phase and decrease the amplitude, which leads to a irregular frequency response of the transfer function between loudspeakers and microphones.

As a room has a very irregular transfer function (Fig. 4.2), the frequency response of the system should be as flat as possible, otherwise the irregularities sum. If the overall loop gain is increased, at some point the system will oscillate at the frequency with highest gain or at the path of least acoustical impedance. The system gets unstable [Barbar, 2005]. The transfer function of a room with an enhancement system is dependent on the acoustic environment, geometries (room resonances) and the properties of the devices used. As room resonances can't be avoided that easy, considerations about what can electronically be done have to be made. An analysis of the feedback issue should be made.

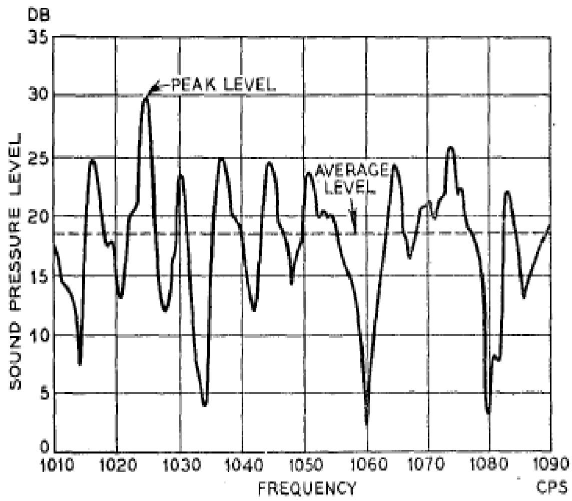

Figure 4.2: Section of a frequency response of a room. CPS means cycles per second. [Schroeder,1959]

To be concrete the average loop gain $\bar{g}$ can be introduced:

$$\bar{g}=\frac{E_{ML}}{E_{MS}} \quad , \tag{4.11}$$

or in logarithmic version

$$\bar{g}=20\log(\frac{E_{ML}}{E_{MS}}) \quad [dB] \quad . \tag{4.12}$$

Herein E_{ML} is the average energy the microphone picked up by the loudspeakers and E_{MS} the average energy the microphone picked up by the source. The loop gain is averaged over each frequency.

Everybody has experienced the effect of regeneration at live concerts with amplification. Instability embodies in a howling or ringing sound. The maximum loop gain in real applications is always less than unity, because the feedback transfer function $H_{LM}(\omega)$ has magnitude peaks of approximately 10 dB higher than the average magnitude [Svensson, 1994].

Schroeder shows in [Schroeder,1959] that the peak-to-mean ratio of the squared room transfer function in a diffuse sound field is

$$10\log(\frac{max(H^2(\omega))}{\bar{H}^2(\omega)})=10\log(\log BT)+3.6\,dB \quad , \tag{4.13}$$

where B is the bandwidth and T the reverberation time.

Further Schroeder introduces the gain before instability (GBI)

$$GBI[dB]=-10\log(\log(BT/22))-3.8\,dB \tag{4.14}$$

With a bandwidth of 20 kHz and a reverberation time of 2s one gets approximately -9 dB of maximal loop gain. Further Schroeder showed the relation of loop gain and probability of oscillation (Fig. 4.3).

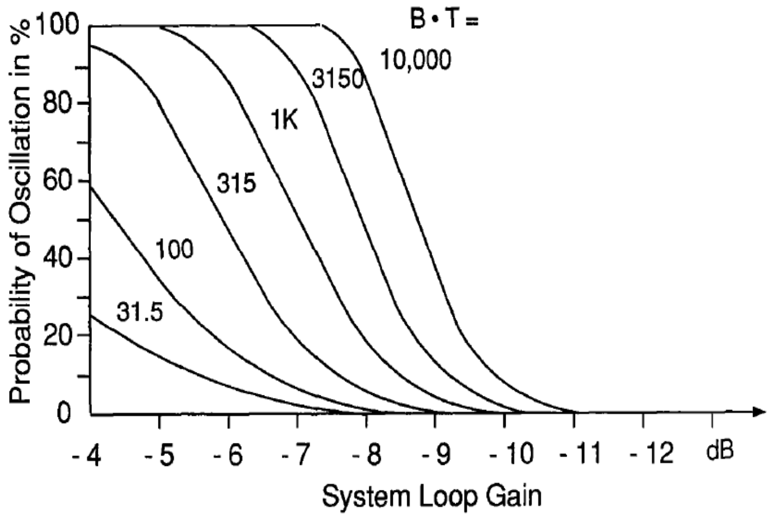

Figure 4.3: Probability of Oscillation with a distinct loop gain and bandwidth [Schroeder, 1954].

Steve Barbar and David Griesinger give in [Barbar, 2005] and [Griesinger, 1992] a maximal loop gain of -12 dB and with a headroom of 8 dB to avoid colouration, a single channel system should ensure a loop gain of -20 dB. Colouration here means a sound field which seems not natural. A cause can be, if a frequency band is more prominent than it normally is. Then a colouration of the sound can be observed.

The whole topic could be approached by using the 'concept of critical distance', as well [Griesinger, 1992]. The critical distance (CD) is the distance from the source where the direct sound energy and the energy of the reverberant field are the same. When an AES is used the CD is changed to the enhancement critical distance (ECD). If the ECD is smaller than the natural CD, all reverberant energy is potential feedback energy.

Now if a microphone in a simple one-channel system is distanced from the connected loudspeaker by exactly the ECD the loop gain is unity and the system gets unstable.

To avoid this the source distance (SD) of the microphone should be decreased.

Further the loop gain can be expressed as

$$\bar{g} = \frac{SD}{ECD} \quad . \tag{4.15}$$

In consideration of the above -20dB (= $\frac{1}{10}$) loop gain the ratio in (4.15) lies under the condition of

$$20\log\left(\frac{SD}{ECD}\right) \leq -20\text{dB} \quad , \tag{4.16}$$

or

$$\frac{SD}{ECD} \leq \frac{1}{10} \tag{4.17}$$

for a one channel system and an omnidirectional microphone.

If we add a second independent channel with same properties the energy level will raise 3dB. For a multichannel system each doubling of the number of channels will contribute 3dB of loop gain.

Equation 4.16 can then be written as

$$20\log\left(\frac{SD}{ECD}\right) \leq -20\text{dB} + 10\log(\textit{number of channels}) \quad . \tag{4.18}$$

By using cardioid microphones additional 4.8 dB can be achieved. The ratio for cardioid microphones is

$$\frac{SD}{ECD} \leq 0.173 \quad . \tag{4.19}$$

Let's consider a single channel system, where all the incoming microphone signals are mixed to mono. Further each microphone picks up only one sound source but every microphone picks up a little of the hall's reverberation. So if the microphones are properly spaced, the reverberant signals are uncorrelated and the addition of two microphones will give +3 dB of reverberation. The GBI will be 3 dB less [Griesinger, 1991].

So the equation 4.19 can be rewritten to

$$\frac{SD}{ECD} \leq \frac{0.173}{\sqrt{number\ of\ microphones}} \quad . \tag{4.20}$$

To maintain equation 4.16 the microphones have to be spaced pretty close to the sound source, which makes the placement of microphone in a single channel system very hard. The conclusion is, using many microphones can make the whole issue worse.

A single broadband channel with the required -20 dB of loop gain and without any artificial reverberation will raise the reverberation time by 1% [Griesinger, 1991]. For statistical independent channels the acoustical power will be the sum of the power of each channel.

Important for multichannel systems is that each channel should be delayed at least by the mean free path in the room. If not, the outcome will be that the reverberation level is too high in relation to the reverberation time. The room will seem smaller [Griesinger, 1991].

Solutions for Feedback Problems

The way of handling feedback is dependent on what kind of system is used. For non in-line systems, feedback is the basis and all the thoughts from above have to be considered carefully. For in-line systems feedback is to be avoided strongly because the generation of reverberation is handled different. In general some actions can be done to control feedback:

- reduce system's loop gain
- principal equalization to remove resonances
- direct loudspeakers to the audience
- introduce LTV (linear time variant) transfer functions
- carefully choosing of the positions of microphones, move closer to the source
- use of directional microphones (Cardioid, Hyper cardioid)
- increase number of independent channels

4.2.1 Time-Variance

An often used method for reducing feedback problems is introducing time-variance or frequency shifting. Using linear time-variant (LTV) systems helps to avoid superposition of frequencies, which have a response peak in a rooms transfer function. The main problem is to make it inaudible.

Schroeder introduced in [Schroeder, 1959] a frequency shifter to raise gain before instability. A room has, as mentioned above, a very irregular frequency response. The aim of frequency shifting is to shift a signal from a peak of the overall transfer function to a valley so that no superposition occurs. The distance of peaks and valleys determines the amount of frequency shift necessary. Schroeder tells a 5 Hz shift is inaudible for speech and nearly every music. The implementation is realized by modulators shown in Figure 4.4. The signal is modulated by a carrier of at least the frequency of the audio band. The lower side-band is rejected and then the signal is demodulated by a carrier of the same frequency minus the desired frequency shift.

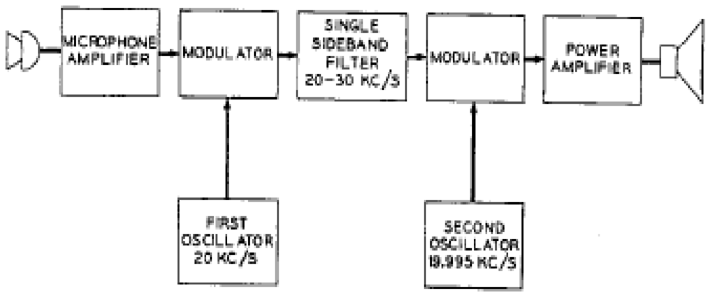

Figure 4.4: Block diagram of a frequency shifter. [Schroeder, 1959]

This method is claimed to give an extra GBI of +6dB. In short terms even more.

Time-variance can be introduced by changing filter coefficients or time-delays in reverberators over time. E.g.: the LARES system uses time-variant reverberators. Several reverberation units should be fully decorrelated and the autocorrelation of one reverberator has to be zero after one second. They use randomization. One single microphone can then be treated as if a number of microphones was used and they can be mixed after the reverberation units. Griesinger claims their implementation to give 6-8dB of extra GBI [Griesinger, 1991]. A problem arises through the digital handling of signals. The shortest delay possible to use is determined by the inverse sampling-rate. Interpolation has to be used to get continuous crossovers.

Developers of commercially available systems keep partly silent about the kind of time-variance used in their systems. Peter Svensson therefore presented in [Svensson, 1995] some general possibilities.

Amplitude modulation

Amplitude modulation, as used by Schroeder, can be applied. If we assume a LTI system with one single delay τ_0, the impulse response is

$$g(\tau)=A_0\delta(\tau-\tau_0) \quad , \tag{4.21}$$

with $\delta(\tau)$ the Dirac delta function and A_0 the amplitude value of the filter.
Amplitude modulation is then employed by adding a periodically varying function to equation 4.21

$$g(t,\tau)=(A_0+A_1\sin\omega_{mod}t)\delta(\tau-\tau_0) \quad , \tag{4.22}$$

where A_1 is the amplitude of the gain variation, and $\omega_{mod} = 2\pi/T_{mod}$ the modulation frequency with T_{mod} the period.

What results are two sidebands of equal magnitude and phase with a frequency shift of ω_{mod}.

Phase Modulation

Further a sinusoidal phase modulator can be used. The transfer function of the impulse response in 4.21 is extended to

$$G(t,\omega)=e^{j\Phi(t)}=e^{j\beta\sin\omega_{mod}t} \quad , \tag{4.23}$$

where $\Phi(t)$ is the phase modulation function and β the modulation index in radians. There is no dependence on ω. All frequencies are exploited to the same phase modulation. Bessel functions result as side bands.

Delay Modulation

Here only the delay time is modulated

$$g(\tau)=\delta[\tau-(\tau_0+\Delta\tau\sin\omega_{mod}t)] \quad , \tag{4.24}$$

where $\Delta\tau$ is the amplitude of delay variation.
Now there is a dependence on ω and the delay modulation is simply a phase modulation with modulation index

$$\beta=\omega\tau \quad . \tag{4.25}$$

The modulation function can be any function. A sawtooth, triangular or a pseudo-random function can be used as well.
Svensson concluded out of his simulations that the phase modulation gives an increase of GBI with the modulation index and that it seems robust against instability.
The delay modulation he found out is efficient in the mid to high frequency range, which means lower modulation can be used above a certain frequency. It's also easy to implement. For suppression of feedback in the lower frequency range it is not useful due to the extensive modulation which would have to be used. It would be audible.

Time-variance constitutes a powerful tool to increase GBI. The main exercise is to find a way to achieve high incremental of GBI and to keep it inaudible.

4.3 Reverberation Generation

The second, very sensible topic in the matter of acoustic enhancement constitutes the electronic generation of reverberation. It is only considerable for in-line systems and hybrid systems, as solely feedback systems generate the reverberation via regeneration.

Reverberation generation can be achieved mechanically, as well, by using plates or reverberation chambers, like broadcasting stations did in earlier times or like the KKL Luzern, acoustically designed by Russel Johnson, which uses big reverberation chambers accessible over rotatable doors to alter the reverberation time.

Electronic artificial reverberation, on the other hand, can be done in different ways, depending on the application. Basically there are two options:

1. using recirculating delay networks (FIR and/or IIR filters)
2. using convolution with an impulse response

Griesinger states in [Griesinger, 1989] that natural reverberation can't be simulated electronically for two main reasons:

- the high complexity
- the multitude of sources

High complexity means the complexity of impulse responses and multitude of sources means that in real we don't hear one impulse response, but the superposition of many.

But now as this was pointed out we can go on and see what we can do.

A focus is made on recirculating delay networks here, because this method is the most common for reverberation generation in acoustic enhancement systems due to their real time capability.

Convolution with impulse response is commonly used in auralization, because real time synthesis is not a necessity and it gives more realistic results. For integrity some methods for electronically generating an impulse response are mentioned. Of course an RIR can be measured, as well, but this topic is not described here.

This chapter should give a basic introduction to electronic reverberation algorithm.

4.3.1 Requirements

At first we should make a step back to room acoustics and examine what properties a natural reverberation unit, a three-dimensional room, has got.

Schroeder gives in [Schroeder, 1961] two main properties of large rooms which directly effect the design of artificial reverberation.

- frequency response
- transient behaviour

The frequency response of a room is characterized by its modes. Above a certain frequency, the critical frequency, the density of modes becomes so high that the modes overlap and can't be distinguished from each other.

$$f_c = 2000\sqrt{\frac{T}{V}} \quad [Hz] \ . \tag{4.26}$$

Because of many interfering modes, the amplitude-frequency response above the critical frequency is very irregular. These irregularities are so rapid on the frequency scale that they cannot be heard when listening to non-steady sounds. It seems that the room has a smooth response.

This apparent smoothness leads us to use artificial reverberators which offers a flat frequency response.

The transient behaviour of large rooms is the response of the room to a short impulse. When measuring an impulsive sound in a room, one can see that at first the direct sound arrives to the measure point, then a bunch of discrete reflections arrive and then a continuous distribution of echo's can be observed.

The time the continuous distribution starts, the critical time, is given by

$$t_c = 5 * 10^{-5} \sqrt{\frac{V}{\Delta t}} \quad , \tag{4.27}$$

with Δt being the width of the impulse.

Above the critical time the echo response is determined by the interference of many overlapping echo's.

Conditions to be met

Further Schroeder [Schroeder, 1961] gives some conditions for artificial reverberation which should be met to provide high quality.

(1) The frequency response must be flat

(2) The normal modes of the reverberator must cover the entire audio range

(3) The reverberation times of individual modes must be the same or nearly the same

(4) The echo density after a little time gap must be sufficient high (1000 echo's per second)

(5) The echo response must be free from periodicities (flutter echo's)

(6) The amplitude-frequency response must not show any periodicities (comb like)

4.3.2 Recirculating delay networks

Schroeder published in 1961 [Schroeder, 1961] and 1962 [Schroeder, 1962] the first methods for simulating reverberation.

He gave two simple recirculating delay networks, shown in Figure 4.5 and 4.6.

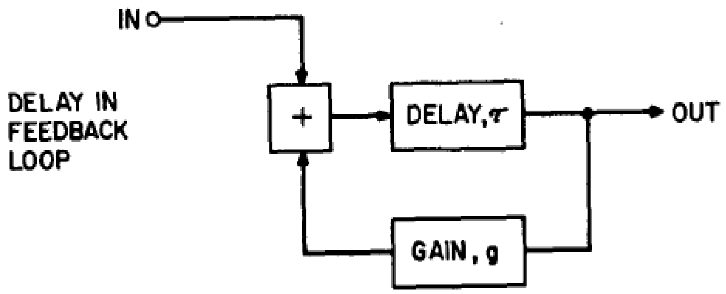

Figure 4.5: Comb filter [Schroeder, 1961]

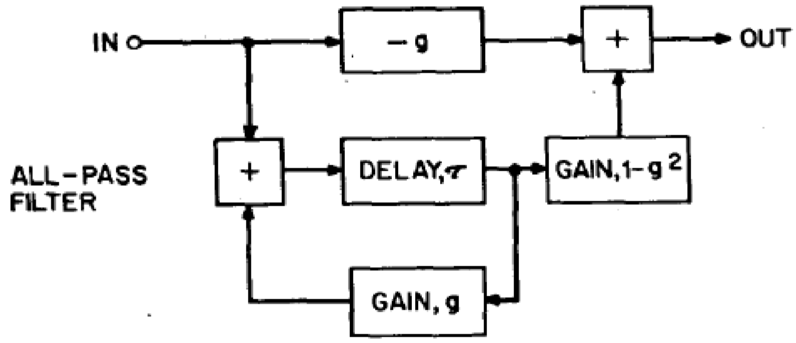

Figure 4.6: All-pass filter [Schroeder, 1961]

The first (Fig. 4.5) is a comb filter with a transfer function of

$$H(\omega)=\frac{e^{-i\omega\tau}}{1-g\,e^{-i\omega\tau}} \quad , \tag{4.28}$$

where g denotes the loop gain (between 0 and 1) and τ the delay.

It is called comb filter, because it has a comb-like frequency response with periodic maxima and minima, shown in Figure 4.7. Each maximum corresponds to one normal mode.

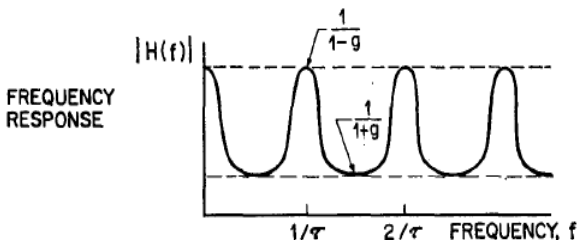

Figure 4.7: *Frequency response of a comb filter. [Schroeder, 1962]*

A comb filter has a 'hollow' or 'metallic' sound character.

The second (Fig. 4.6) is an all-pass filter, which has a linear frequency response. The transfer function is

$$H(\omega)=\frac{g+e^{-i\omega\tau}}{1+g\,e^{-i\omega\tau}} \quad , \tag{4.29}$$

with

$$|H(\omega)|=1 \quad . \tag{4.30}$$

The relation between loop gain, delay time and reverberation time is

$$T=[\frac{60}{-20*\log(|g|)}]*\tau \quad [s] \quad . \tag{4.31}$$

Schroeder claims it to have a 'colourless' sound quality [Schroeder, 1961] which states a remarkable improvement of the comb filter. Although it's flat frequency response, it has a very complex phase response, which can lead to a perceptually non transparent sound image when using non-steady-state sounds. It then creates a periodic flutter echo with a low echo density.

As the results were unsatisfying, Schroeder suggested to combine the filters. Figure 4.8 illustrates these suggestions.

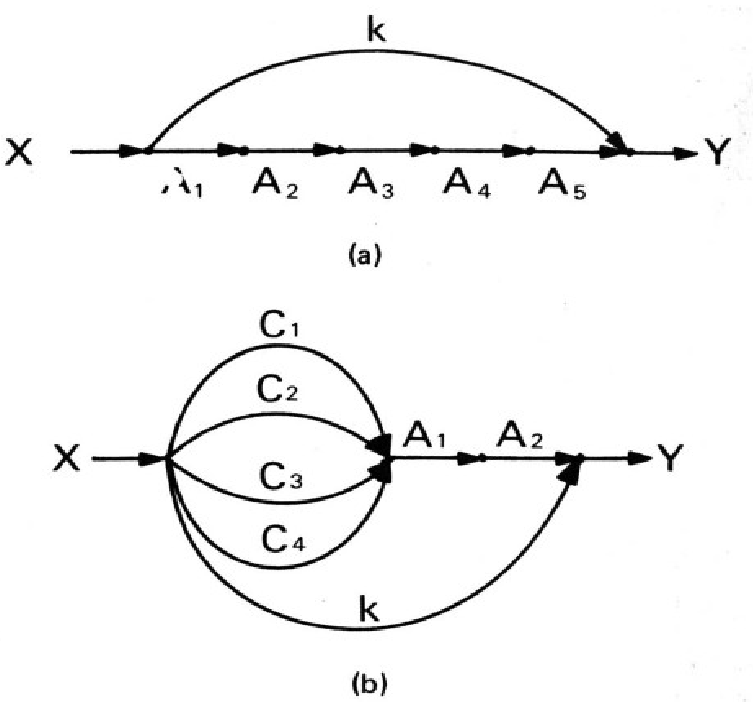

Figure 4.8: *a) Series combination of All-pass filters* A_i *, b) Comb filters* C_i *in parallel and two All-pass filters in series. Here k denotes the gain factor for direct sound [Moorer, 1979]*

Figure 4.8a) shows a series combination of all-pass filters, each with a different delay time. *X* denotes the input and *Y* the output. It is suggested to set the loop gains to 0.7. To avoid echo superposition and cancellation prime numbers should be used as delay times [Moorer, 1979] . (E.g.: 100, 68, 60 19.7, 5.85 ms [Schroeder, 1961])

It is also possible to adjust the direct to reverberant ratio by adjusting *k* and to set the initial time delay gap between direct and reverberant sound by setting τ . It is claimed that the series connection of all-pass filters eliminates flutter and colouration [Schroeder, 1962] . Anyhow, Moorer gives in [Moorer, 1979] some empirically learned problems, immanent to this reverberator:

- the decay was not dense enough and it did not die out exponentially
- smoothness of the decay was highly dependent on delay times and gain factors
- the tail of the decay showed an ringing effect, which lead to a metallic sound

Figure 4.8b) shows four comb filters in parallel and afterwards two all-pass filters in series. The idea is that the comb like frequency response is no problem, when the gaps between the maxima are closely spaced, which is the case in real rooms.

One comb-filter with a delay of 0.04ms would give 25 echo's per second. Four in parallel would have 100 echo's per second which are by far too less compared to the required 1000 echo's per second. To further improve echo density two all-pass reverberator units in series are set behind the comb filters. Each of them should multiply the echo density by a factor of about 3. Schroeder states that with time-delays of 5ms and 1.7ms and gain factors of 0.7, this is achieved. The delay times of the comb filters are again prime numbers and the gain factors are set according to equation (4.31).

This realization showed not exactly the same problems as the all-pass combination. Problems were:

- distinct patterns of echo when using impulsive sounds
- metallic sound especially when using longer reverberation times

To get more realistic responses, frequency dependence in form of low-pass filters are introduced for the loop gain factors. The low-pass filters are supposed to simulate the frequency dependent air absorption.

Figure 4.9 shows a modification of a comb filter. A simple one-pole low pass filter was used by Moorer. Stability condition gives

$$\frac{g_2}{1-g_1}<1 \quad . \tag{4.32}$$

Further g_2 can be found by setting it to

$$g_2=g(1-g_1) \tag{4.33}$$

where g_1 is determined by the air absorption consistent with the distance according to the delay time, and g determines the reverberation time.

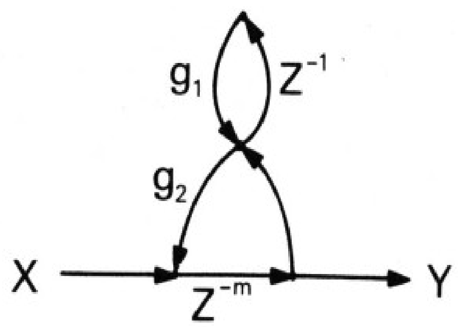

Figure 4.9: *Comb filter with a one-pole filter in the loop. [Moorer, 1979]*

In Figure 4.9 *Z* denotes the unit advance operator with a delay *m*.

Although this simple model for air absorption, Moorer claimed that the sound became more realistic. High frequencies decayed faster than lower frequencies and transient sounds were treated better. Even the high sensitivity of the delay time varying became less.

Schroeder gives in [Schroeder, 1970] further refinements. Some discrete delays are set in front of a reverberator, *R(z)* simulating the late reverberation (Fig 4.10). The delay times and gain factors are set following the room geometries.

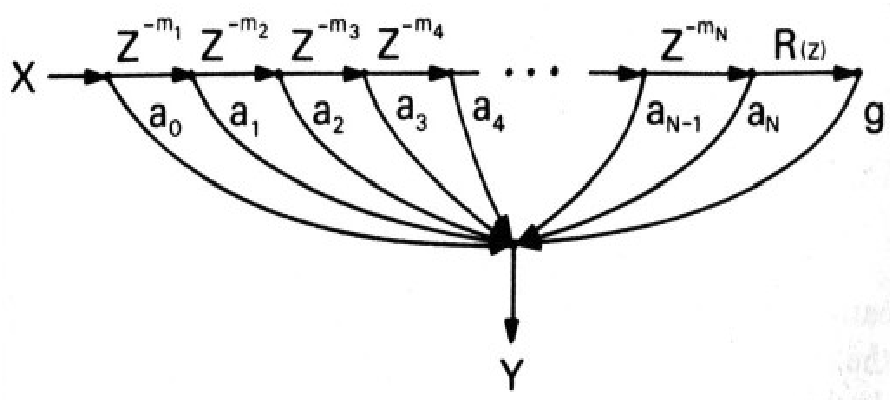

Figure 4.10: *Reverberator suggested by Schroeder in 1970. [Moorer, 1979]*

Moorer claims in [Moorer, 1979] that the most pleasing results for late reverberation *R(z)* where achieved when using six comb filters in parallel and one all-pass in series.

Figure 4.11 gives a possibly parameter constellation. The all-pass delay is set to 6ms with a gain of 0.7. The value *g* should be at constant level, but one have to observe if one comb filter's pitch is dominating and if necessary one have to adjust it. It is suggested to set the delay times linearly distributed over a ratio of 1:1.5. Then the nearest prime numbers should be picked .

	Delay (in ms.)	25 Khz g_1	50 Khz g_1
COMB 1	50	0.24	0.46
COMB 2	56	0.26	0.48
COMB 3	61	0.28	0.50
COMB 4	68	0.29	0.52
COMB 5	72	0.30	0.53
COMB 6	78	0.32	0.55

Figure 4.11: *Example of parameter constellation. [Moorer, 1979]*

Moorer points out that the problem of the recirculating delay systems is that they can never simulate the reflection patterns of a real room. Reflections do not come in regular sequences and there are no equally spaced gaps between reflections.

To improve the density of echo's, Moorer suggested an extension of the reverberator shown in Figure 4.12. Not only the direct sound but the whole FIR section is put into the reverberator *R(z)*. The delays *D1* and *D2* are set so that the last delay of the FIR section corresponds to the first delay of the IIR section.

With this reverberator it is claimed to approach the sound of a real concert hall.

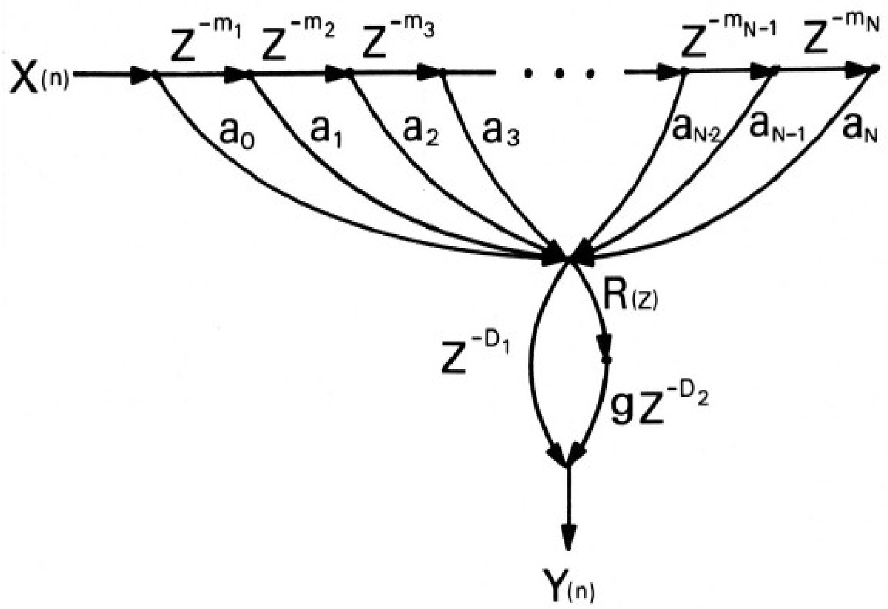

Figure 4.12: *Reverberator to improve echo density. [Moorer, 1979]*

An important property of real room reflection behaviour has been left out in all the considerations: The effect of diffusion. Walls in real rooms, especially in concert halls are always more irregular then flat. The diffusion is causing an impulse response to not having any gaps between reflections. Until the end of the response there are continuously some reflections occurring. This is a property hardly realizable with recirculating delays.

Moorer and his team have made an interesting discovery in [Moorer, 1979]. They found that the impulse responses of concert halls around the world sounded likewise white noise with an exponential decay. They tested this assumption by generating an impulse response with Gaussian unit-variance pseudo-random sequences with exponential envelope with the desired length. For including the direct sound an impulse was set at the beginning. Dry recorded music signals were then convolved with the impulse responses resulting in a natural, concert hall like reverberation. By filtering the impulse responses one can tailor the sound as desired. They stated this procedure to be the most natural sounding one of all they tried, except for using measured impulse responses.

Feedback Delay Network

Gerzon [Gerzon, 1976] generalized the all-pass filter of Schroeder to a N input, N output system with the same unitary network property. Stautner and Puckette [Stautner and Puckette, 1982] introduced then the feedback delay network (FDN). The FDN constitutes of an extension of the comb-filter. A set of delays are connected to a feedback matrix. The feedback matrix contains the gain factors and is an unitary matrix.

J. M. Jot [Jot and Chaigne, 1991] studied these networks and developed techniques for a good reverberator design [Gardner, 1997].

The FDN uses N delay lines with a delay τ_i and a feedback matrix of order N shown in Figure 4.13.

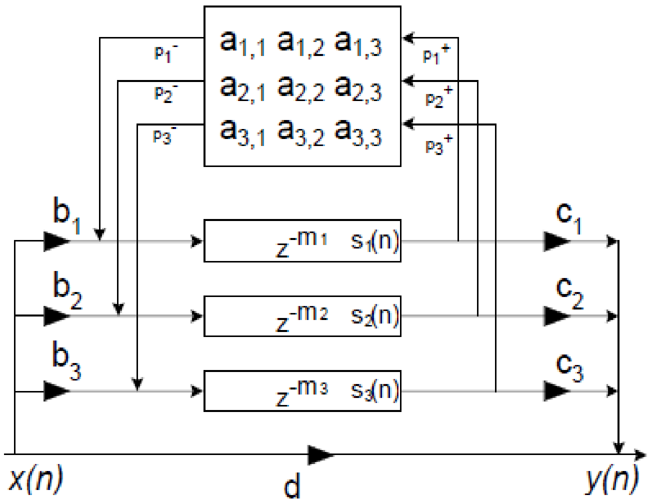

Figure 4.13: *Feedback Delay Network of 3rd order. [Gardner, 1997]*

Here, *x(n)* is the input series and *y(n)* the output series. The output relation is

$$y(n)=\sum_{i=1}^{N} c_i s_i(n)+dx(n) \quad , \tag{4.34}$$

where

$$s_i(n+m_i)=\sum_{j=1}^{N} a_{i,j} s_j(n)+b_{ix}(n) \tag{4.35}$$

are the delay line outputs and m_i is an integer denoting the delay according to the used sample rate.

Applying z-Transform gives in matrix form:

$$\boldsymbol{Y}(z)=\boldsymbol{c}^{\boldsymbol{T}}\boldsymbol{S}(z)+d\,\boldsymbol{X}(z) \tag{4.36}$$

where

$$\boldsymbol{S}(z)=\boldsymbol{D}(z)[\boldsymbol{AS}(z)+\boldsymbol{bX}(z)] \quad . \tag{4.37}$$

S*(z)*, ***b****(z)* and ***c****(z)* are vectors with length N, ***D***(z) is a diagonal matrix with the N different delays in it and ***A*** is the feedback matrix. For multiple input/output systems **b**(z) and **c**(z) become matrices.

The transfer function of the system is

$$\boldsymbol{H}(z) = \frac{\boldsymbol{Y}(z)}{\boldsymbol{X}(z)} = \boldsymbol{c}^{\boldsymbol{T}} [\boldsymbol{D}(z^{-1}) - \boldsymbol{A}]^{-1} \boldsymbol{b} + d \quad . \tag{4.38}$$

The feedback matrix **A**, again is a unitary matrix, which means all eigenvalues are on the unit circle and poles are of magnitude 1.

If **A** is a diagonal matrix, there would be no difference to Schroeder's parallel comb filters.

An example for an 4th order feedback matrix is shown in Figure 4.14. The *g* controls the reverberation time.

$$\mathbf{A} = \frac{g}{\sqrt{2}} \cdot \begin{bmatrix} 0 & 1 & 1 & 0 \\ -1 & 0 & 0 & -1 \\ 1 & 0 & 0 & -1 \\ 0 & 1 & -1 & 0 \end{bmatrix} \qquad (g < 1)$$

Figure 4.14: *A 4th order unitary feedback matrix. [Jot and Chaigne, 1991]*

To simulate absorption, filters can be set behind the delay lines.

Important to note is that the FDN reverberator is only supposed to provide late reverberation.

4.3.3 Ray based methods

Popular methods for creating an impulse response is the image source method and the ray tracing method [Savioja et al., 1999]. They both treat sound as rays with specular reflection behaviour. The result in most implementations is an energy-time curve (ETC). The goal is to simulate all possible sound reflection paths. The distinction between the two methods is the way they try to achieve this.

Ray Tracing

In the basic algorithm, a sound source sends out rays, which are reflected at walls with certain rules and a listener tracks which rays are incident rays to him. The easiest reflection rule is specular reflection. More advanced ones can take diffusion and diffraction effects into account. The listener can be modelled in several ways. The most common model is a simple sphere. The model has to be chosen so that statistical valid results are achieved.

Image Source Method

For the image source method a room has to be chosen in shape and size. The method then finds the reflection paths of the sound source by finding mirror sources of arbitrary order, dependent on the computational power available. After all mirror or image sources have been determined, a visibility check has to be made, to check if the reflection really arrives at a listeners position.

The locations of image sources is not dependent on the listener position, only the reflection path changes with listener position change.

Diffusion and diffraction is in the basic algorithm not included. The visibility check would become much more complicated if the effects are implemented.

Because of the high computational effort, the image source method is mainly used to simulate the early response of a room.

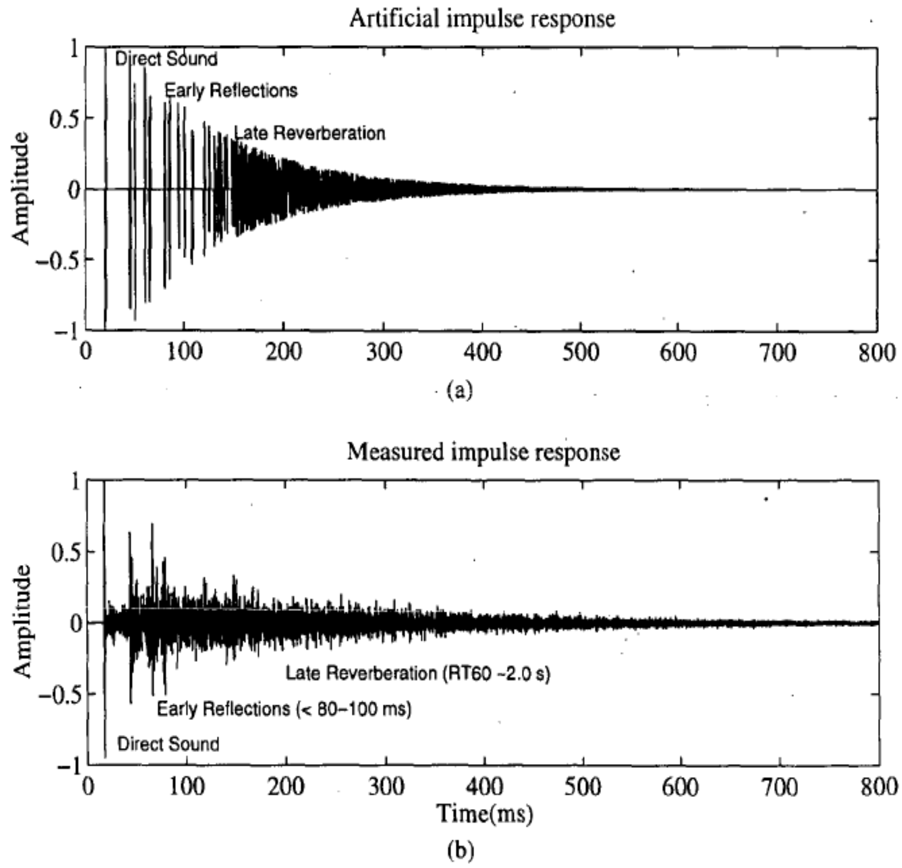

Figure 4.15: *a) Simulated Impulse response of a room b) measured IR. [Savioja et al., 1999]*

5 Developed Systems

Over the last half century, engineers and researchers have developed a variety of systems. The most important developments are shown in this chapter. Some systems exist only in a few venues and are not produced anymore and some are still commercially available.

5.1 Assisted Resonance

The 'Assisted Resonance' [Parkin, 1970] system belongs to the first active enhancement systems. It was first installed in the Royal Festival Hall (RFH), London, 1964. The installation in RFH tried to enhance within a frequency range of 70-340Hz, later 58-700Hz when a permanent installation was made. The principle of the system is resonance. It used a moving-coil microphone in a Helmholtz resonator with a high quality factor, an amplifier and loudspeaker either in a quarter-wavelength tube (up to 100Hz) or in tuned boxes (above 100Hz). Above 300 Hz quarter-wavelength tubes were used again. With channels spaced 3Hz, later a relative spacing of 2 - 4% was used, a high number of channels is necessary. The permanent installation used 172 channels. The positions of the microphones and loudspeakers were in the ceiling and were chosen at anti nodes of room resonances and a phase-adjusting is used to get positive feedback. By controlling the gain of the amplifier the reverberation time at this frequency can be increased.

In the installation process problems were found concerning the interaction of channels. It was not possible to separate some channels fully and some positive interaction was accepted. Another problem was that the anti nodes, or pressure maxima, changed over time and that no constant position was found. As a compromise a nearly constant position was chosen. The gain of the amplifier was set, after weeks of monitoring the RT, to a level, so that a particular RT was not exceeded.

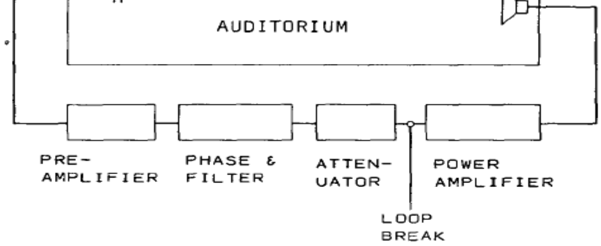

Figure 5.1: *Components of a Assisted Resonance channel [Parkin, 1970].*

Generally, after installation, big effort was put into finding out how the system behaves. In spite of the huge technical complexity and it's inflexibility the system was installed in several other halls.

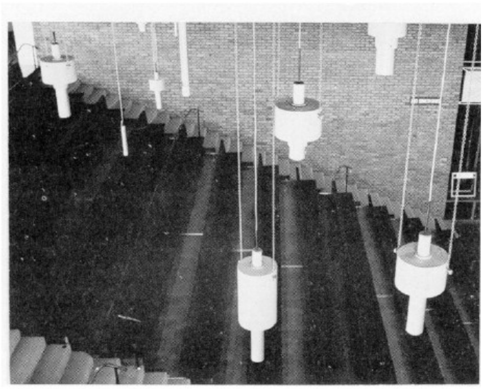

Figure 5.2: *Resonators and tubes from an installation in York [Parkin, 1970].*

5.2 Multiple-Channel amplification of Resonance (MCR)

In 1968, Franssen [Franssen, 1969] suggested using a large number of individual broad-band channels with low loop gain to enhance acoustics. Reverberation will be generated by feedback, but colouration would be no problem, because of the low loop gain of each channel. Of course equalization has to be applied to smooth the resonances.

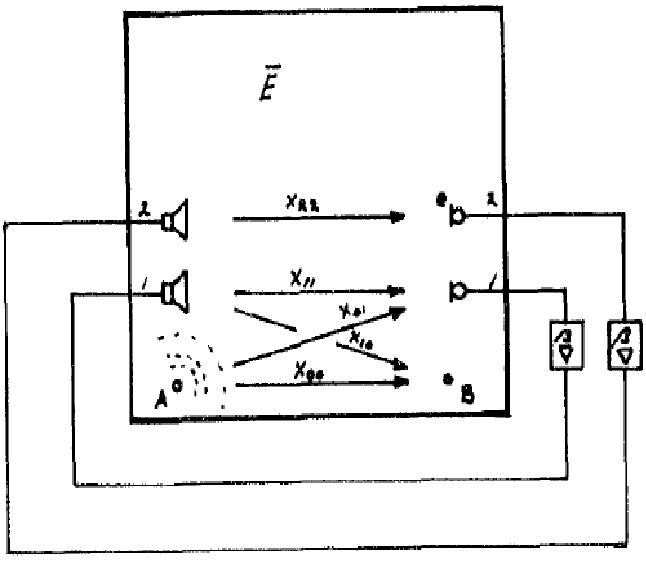

Figure 5.3: *Block diagram of an general MCR system [Franssen, 1969].*

Franssen calculated the transfer function for a MCR system (see Figure 5.3). For a sound source *A* and a receiver *B* the transmission at a certain frequency can be written as

$$e_0 = X_{00}\overline{E_{00}} + X_{10}\overline{E_l} \tag{5.1}$$

Where e_0 is the observed energy density, X_{00} and X_{10} the transmission factors and E_{00} and E_l the source and loudspeaker energy densities.

The average loudspeaker energy density is

$$\overline{E_l} = X_{01}\overline{E_0}\left(\frac{\beta}{1-\beta X_{11}}\right) \quad , \tag{5.2}$$

where $X_{01}E_0$ is the energy density picked up by the microphone.

When 5.2 is injected in 5.1 one gets

$$e_0 = X_{00}\overline{E_{00}} + X_{10}X_{01}\overline{E_0}\left(\frac{\beta}{1-\beta X_{11}}\right) \quad . \tag{5.3}$$

Further

$$e_0 = \left(\frac{X_{00} - \beta X_{00}X_{11} + \beta X_{10}X_{01}}{1-\beta X_{11}}\right)\overline{E_0} \quad . \tag{5.4}$$

If the transmission factors are averaged over frequency, they are all the same. 5.4 can then be simplified to

$$e_0 = \left(\frac{\overline{X}}{1-\beta\overline{X}}\right)\overline{E_0} \quad . \tag{5.5}$$

And for an n-channel system

$$e_0 = \left(\frac{\overline{X}}{1-n\beta\overline{X}}\right)\overline{E_0} \tag{5.6}$$

It can be seen in 5.6, the higher the number of channels, the higher the degree of amplification.

To avoid colouration Franssen states that the power loop gain should be 17dB less than 1 or 0.02. Thus an one channel system would amplify the energy density by 51 over 50 (or approximately 0.1 dB). For doubling the energy density, 50 channels are necessary with an average open-loop gain of 1 over 100 (or -20 dB).

5.3 Electronic Reflected Energy System (ERES) and Reverberation-on-Demand-System (RODS)

ERES was developed at Jaffe Acoustics and was patented in 1977 by Christopher Jaffe [Jaffe, 1992]. It uses microphones positioned in the stage area and connects them to multi-tap filters. The first tap is provided with full bandwidth to speakers in the proscenium, later taps are low-pass filtered and fed to speakers in the ceiling [Barbar, 2005]. A 72 tap delay (Figure 5.4) is used where each delay has an offset between 28ms-32ms. Originally analog realized, it was rebuilt in the digital domain in 1990. ERES generates no reverberant energy. It is claimed to model reflectors of specific size and mass.

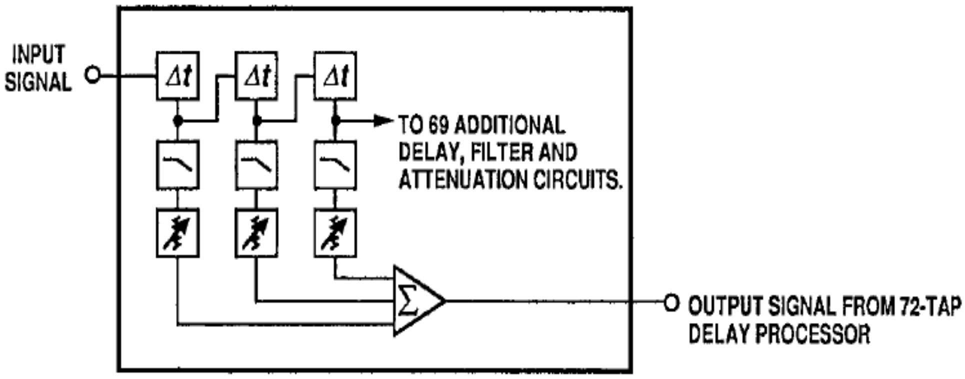

Figure 5.4: *Simplified 72-Tap Delay Processor Schematic* [Jaffe, 1992].

In 1982 a cooperation between Jaffe Acoustics and Acoustic Management Systems, Ltd. was made to put research in the development of reverberation enhancement systems. Peter Barnett developed 1985 the RODS which was added to the ERES. The attempt with RODS was to avoid the issue of feedback and colouration. This is achieved by using switches ahead of and after a reverberation generator. Dependent on whether the incoming signal is raising or falling, different switches open and close (Figure 5.5).

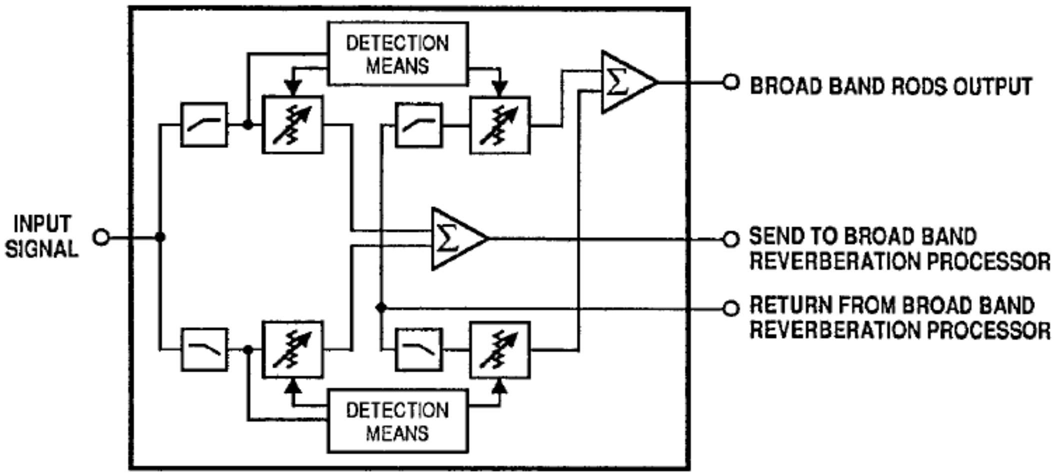

Figure 5.5: *Simplified RODS Schematic* [Jaffe, 1992].

5.4 Active Field Control (AFC)

Yamaha Corp. presented the AFC system, or AAS (Assisted Acoustics System), in 1985 [Miyazaki, 2003]. It uses an array of 4 or 8 omni directional microphones, placed at or beyond the critical distance (non in-line). The microphone spacing depends on the lowest frequency to be enhanced (at least half the wavelength). The microphone signals are fed through an electric microphone rotator (EMR) which incorporates a matrix to change the I/O combinations periodically (between 0.4-3.0Hz). An FIR filter is then used for wider control possibilities. It is claimed that by using the FIR filter either the energy of each tap can be raised which corresponds to an incremental of sound pressure level or the gaps between taps are increased which is equivalent to bigger volume (Figure 5.6). It is a special FIR filter (*fluc-FIR*) which smooths the frequency response by moving each FIR tap periodically on time axis. A graphic equalizer is used then to smooth the loop transfer function.

The reverberation generation relies fully on feedback and with the above mentioned methods, instability and colouration is avoided.

Two separate systems can be used to split the generation of early and late reflections. The system for early energy doesn't use *fluc-FIR* to preserve the reflection pattern.

Loudspeakers for late reverberation are directed to the walls to decrease direct sound to the audience and the microphones and to increase diffusion.

A Block diagram of the AFC system is given in Figure 5.7.

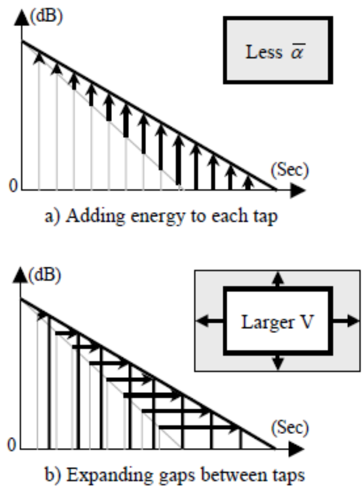

Figure 5.6:*FIR filter control possibilities* [Miyazaki, 2003].

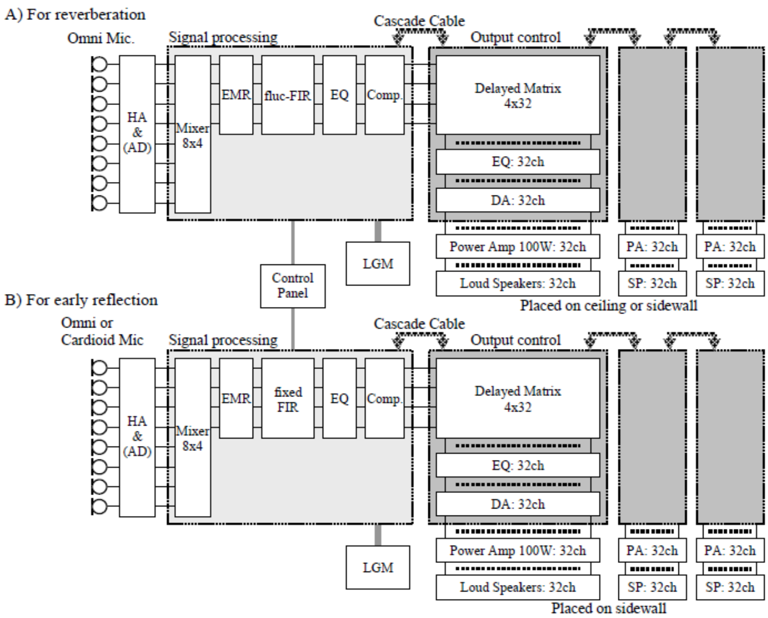

Figure 5.7:*Block diagram of AFC* [Miyazaki, 2003].

5.5 Acoustical Control System (ACS)

1987, Prof. Berkhout of the Technical University in Delft, presented the Acoustical Control System for reverberation enhancement [Berkhout, 1988].

The system is based on acoustic holography. The 'real' hall can be overlaid by a 'desired' hall by simulating the wave field appearing in a hall with different size, shape and absorption (Figure 5.8). Each loudspeaker has it's own impulse response depending on it's spatial position. The superposition of all used loudspeakers should form the coherent desired wave fields.

Practical installations are limited compared to the demands of wave field theory, because a huge amount of loudspeakers, closely spaced, would be necessary. Instead wider spaced loudspeakers are placed around the audience.

It is claimed that ACS gives the architect and the acoustician 'maximum degree of freedom'.

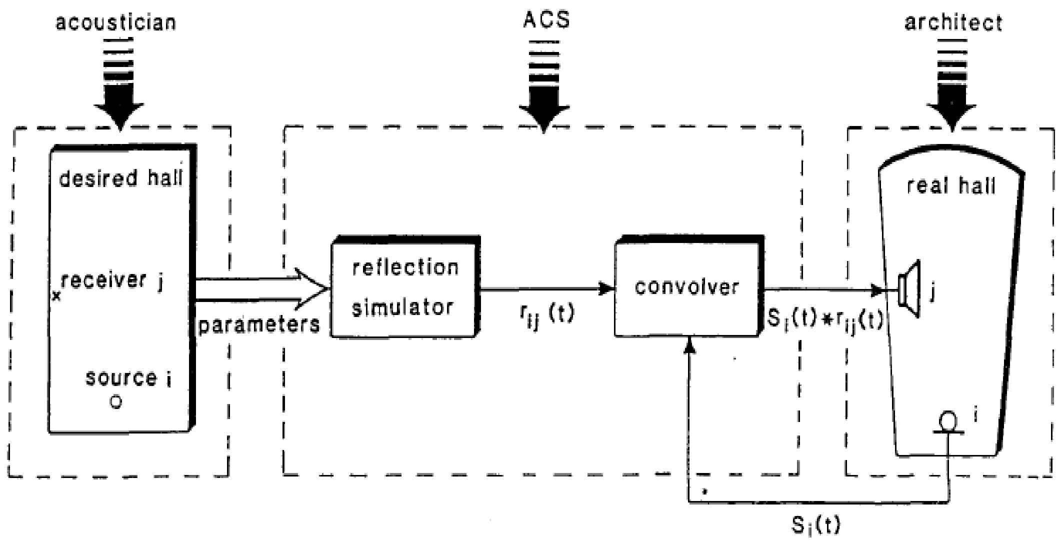

Figure 5.8: *Principle of ACS for one loudspeaker pair [Berkhout, 1988].*

ACS consists of three subsystems:

1. Wave field recording
2. Central processor
3. Wave field emission

The Wave field recording subsystem picks up the sound field in the stage area with a high amount of directional microphones. Thus ACS is an in-line system.

The microphone signals are fed to the central processor which consists of a number of simulation units (image-source method) to form the wave field. For higher order reflections many simulation units are necessary. The electronic equipment is analog.

If $\vec{s}$ is the input vector (microphones) and $\vec{p}$ the output vector (loudspeakers) then $\boldsymbol{T}$ forms the relation in between.

$$\vec{p}=\boldsymbol{T}\,\vec{s} \tag{5.7}$$

where

$$\boldsymbol{T}=\begin{bmatrix} t_{11} & t_{12} & \cdots & t_{1N} \\ t_{21} & t_{22} & & t_{2N} \\ \vdots & & & \vdots \\ t_{M1} & t_{M2} & \cdots & t_{MN} \end{bmatrix} \tag{5.8}$$

The matrix element t_{nm} is formed by the transfer function between microphone m and loudspeaker n.

$$t_{nm}=A_{nm}(\omega)e^{-j\omega\tau_{nm}} \tag{5.9}$$

The matrix that is necessary for the creation of higher order mirror sources would be huge and computationally intensive so that just first and second order reflections are generated.

Further the system is divided into modules for the creation of early reflections and late reverberation.

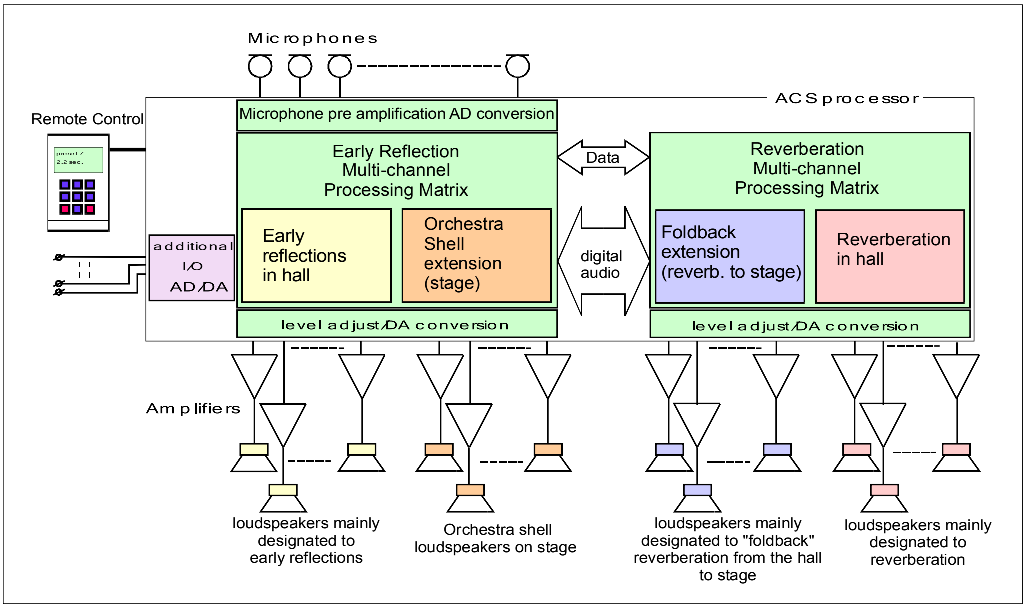

Figure 5.9: *ACS Block Diagram [ACS, 2009].*

As feedback is not the basis of the reverberation generation, feedback is to be avoided. Small loop gains of the ACS is achieved by:

- A high amount of microphones positioned near the stage. Gains of hall microphones are set to a safe level.
- Directional Microphones are used.
- Directive loudspeakers, turned to the audience.
- Matrix elements are time-variant.

An important fact and claimed advantage of ACS is that for the generation of the ACS responses, the 'real' hall does not play any role. Of course, and here the critics set on, the 'real' hall's acoustics do play a role on the impact of the ACS [Barbar, 2005].

The design of the ACS has to be made under consideration of the prevalent acoustics and if necessary some acoustic treatment has to be made in architectural means.

5.6 Lexicon Acoustic Reinforcement and Enhancement System (LARES)

David Griesinger made a first installation of the LARES in 1989 in Elgin Theater in Toronto [Griesinger, 1991]. The system is an in-line system. It consists of a small number of microphones (2 to 4, cardioid), positioned as close as possible to the sound source, at least for speaker banks and a high amount of independent time-varying reverberation units, which build the connection of the microphones and the speaker banks (Figure 5.10). The reverberation units are all of the same structure, what requires decorrelation between each unit. The method to achieve this and the main development of the LARES is the effective time-variance.

The time-variant reverberation units randomize the microphone signals in a way that is seems as if a higher number of independent microphones exist. The number of channels is determined by the number of reverberators (ideally the product of microphones and speaker banks). Griesinger claims the system to be very practical, because of the low number of expensive microphones and the high number of relatively inexpensive reverberators.

The time-varying directly adds 6 dB of gain before instability, which makes the system very effective. Figure 5.11 shows two impulse responses of one reverberation unit, where the second is made one minute later. The response has completely changed.

The installation in Elgin Theater consists of 2 microphones, 8 speaker banks and 16 reverberators. Griesinger writes that it achieves a source distance to enhancement critical distance ratio of approximately 1.75 (Bandwidth=15kHz, Reverberation Time= 1.5s...2.0s).

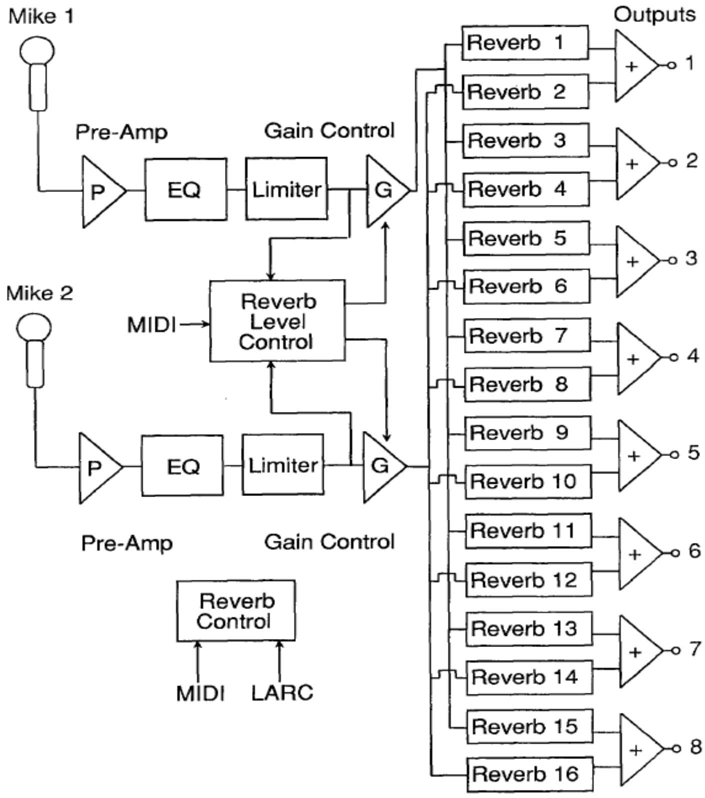

Figure 5.10: *LARES Block Diagram of the Elgin System [Griesinger, 1991].*

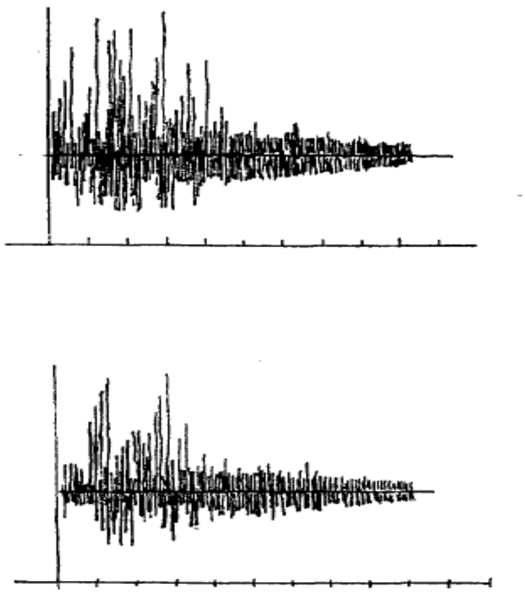

Figure 5.11: *Impulse response of one reverberator. One minute apart [Griesinger, 1991].*

5.7 Variable Room Acoustic System (VRAS)

The development of the VRAS was introduced by Mark Poletti in 1993 [Poletti, 1996]. It is a non in-line wide band system consisting of a N*N reverberation matrix and N loudspeakers and microphones.

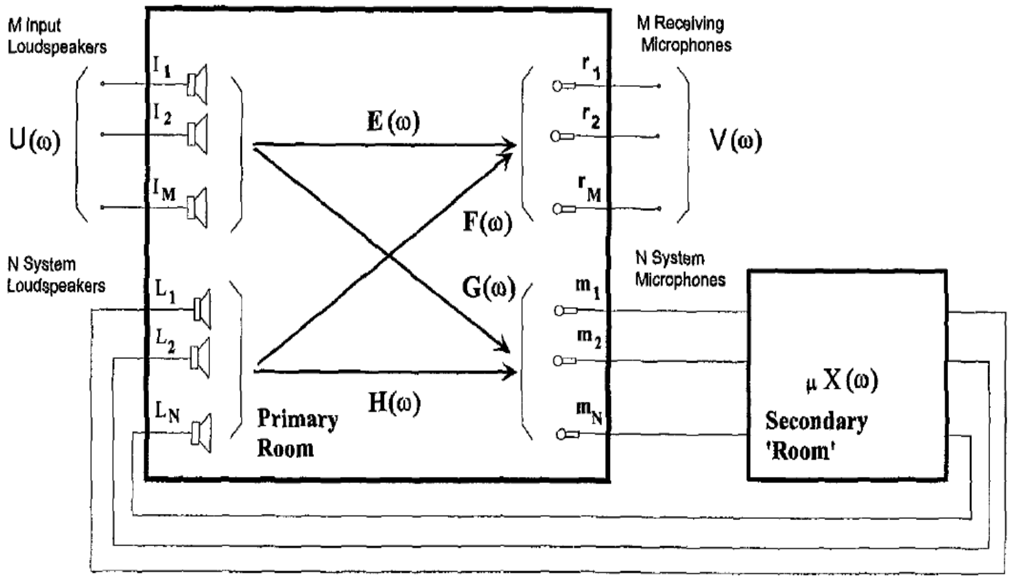

Figure 5.12: *Block diagram of the VRAS [Poletti, 1996].*

The reverberation matrix is claimed to place a 'secondary room' in a feedback loop around the 'primary room' . The secondary room (or reverberator) has dispersive properties, which raises the reverberation time of the primary room, but not the steady state level. As a result the apparent volume is increased. The steady state level can be raised by increasing the loop gain of the system which is equivalent to lowering the absorption.

Figure 5.12 gives a block diagram of the system. The *M* input loudspeakers refer to sound sources and the *M* receiving microphones represent listeners.

The reverberation units are designed in a way so that the power gain is unity (cp. FDN).

If the matrix **X** is unitary about all frequencies the power output is *N* and the power gain is unity. This is achieved by cross-connections in the reverberation matrix **X**.

$$\boldsymbol{X}^{\boldsymbol{H}}\boldsymbol{X}=1 \qquad (5.10)$$

The *H* subscript denotes the conjugate transpose of **X.**

The amplification by the system gets

$$\frac{p_{on}^2}{p_{off}^2}=\frac{1}{1-N^2S^2} \quad , \qquad (5.11)$$

compared to

$$\frac{p_{on}^2}{p_{off}^2} = \frac{1}{1 - N\,S^2} \tag{5.12}$$

of a normal system.

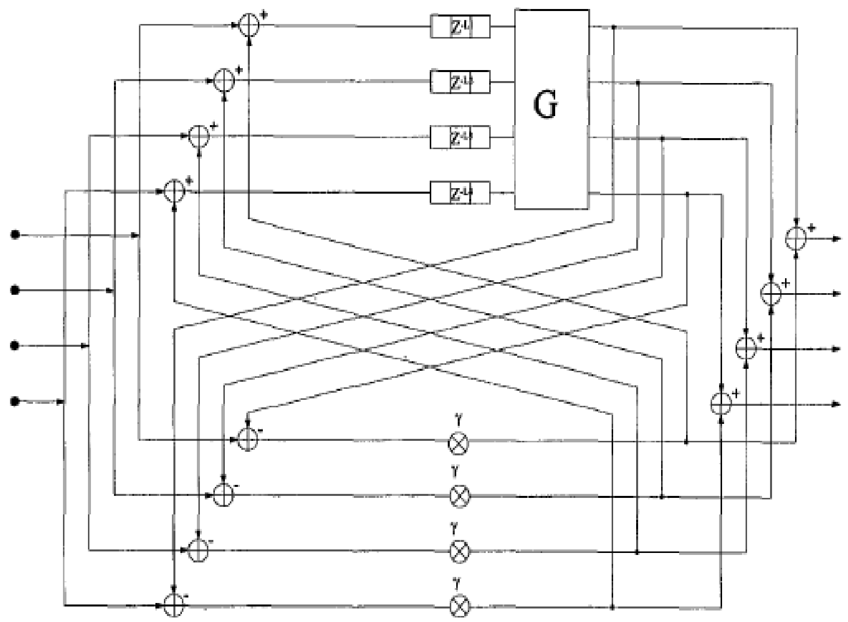

Figure 5.13: *Block diagram of unit reverberator [Poletti, 1996].*

The consequence therefore is that a unitary reverberator placed in an enhancement channel will not increase the risk of colouration and a rise of reverberation time is not achieved by increasing the loop gain. The stability of the system is therefore independent of delay times and amplitudes. Figure 5.13 shows the block diagram of an unitary reverberator (cp. FDN).

In 2007 Poletti introduced a patent for an in-line system, which can be combined with the non in-line system to create a hybrid. The properties of this in-line system are close miking to avoid feedback and discrete delayed reproductions of the input, in a way the precedence effect is never disturbed (to keep the right localization). Typical response lies around 80ms. Again, the delay matrix has unitary power gain, which means the stability of the system is independent of delay times and amplitude [Poletti, 2007] .

Meyer Sound Constellation

The Meyer Sound Constellation System is a commercially available acoustic enhancement system build on the basis of the VRAS technology. The system is a hybrid system, which means in-line and non in-line technologies are mixed. There are three underlying patents of Mark Poletti, [Poletti, 1998], [Poletti, 1999] and [Poletti, 2007].

5.8 System for Improved Acoustic Performance (SIAP)

SIAP was presented 1992 by Prinssen an Holden [Prinssen and Holden, 1992]. The aim of the SIAP is to provide missing directional reflections and reverberation. It uses a small number of microphones in the stage area (in-line), enough to cover the whole stage area. The signals are passed to the SIAP main frame where flexible signal processing happens with software routing possibilities. A layout containing four microphones, four acoustic processors and 28 output channels gives 448 of independent decorrelated reflection patterns. The decorrelation happens at the input and output stages. This would give a 26.5 dB higher output than for a single channel system. Time variance can also be used which would give another 6 dB of GBI. The processed signals are provided at multiple outputs and fed to a large number of loudspeakers.

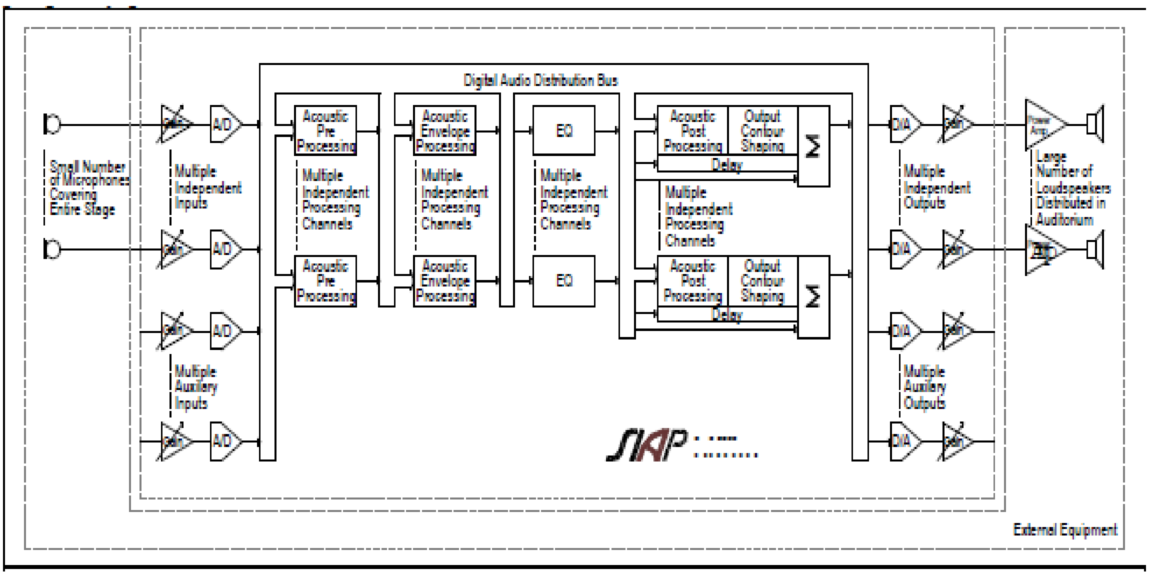

Figure 5.14: *Concept of SIAP system [Kok and Prinssen, 2009].*

In [Prinssen, 1997] they give a pricing diagram which says for small installations cost are between 100.000 – 200.000 U.S. Dollar, medium 150.000-300.000 U.S. Dollar and for large installations 200.000-500.000 U.S. Dollar. Just to get a picture of prices.

5.9 Contrôle Actif de la Réverbération par Mur virtuel à Effet Naturel (CARMEN)

Carmen is a system by the french company CSTB and was introduced 1996. It is a non-in-line system, which consists of a number of channels (active cells). One channel comprises a microphone, an electronic filtering unit, a power amplifier and a loudspeaker. The loudspeakers in this system are placed to create an acoustically virtual wall, which mimics the desired properties [Carmen-Homepage, 2009].

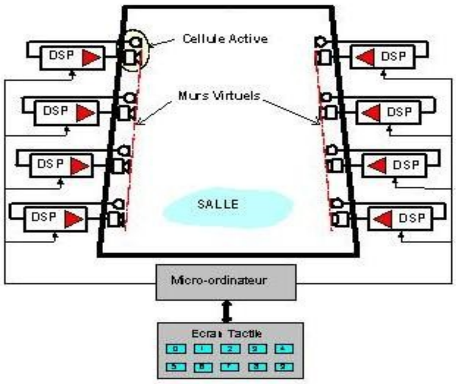

Figure 5.15: *Block diagram of the CARMEN system [Carmen-Homepage, 2009].*

5.10 Vivace

Vivace is a system developed by Müller-BBM around 2006 [Engel and Walter, 2006], [Mueller-BBM-Brochure, 2008]. Therefore it is the newest of the systems in consideration. It is an in-line system (Fig. 5.16), which uses 2-4 broadly directional microphones to cover the whole stage area. A DSP matrix is used to A/D convert and filter the signals and to send them to the main processor. The main processor puts out decorrelated signals with an early and late reverberation part. It is claimed that for a typical installation four independent signals are sufficient to provide adequate results. The signals get back to the DSP matrix, which distributes the signals to the speakers. Delay-lines and frequency response equalization of speakers take also place in the DSP matrix. As loudspeakers broadly directed speakers are used.

Special about Vivace is that it uses convolution for reverberation generation. Up to 16 impulse responses can be chosen with tools to edit these. A newly designed algorithm is used to provide fast results (a latency of 6ms is achieved). To avoid colouration, time variance is used in the convolution algorithm.

Additionally Vivace provides adaptive algorithm adjustment to adapt the reverberation to the input signal.

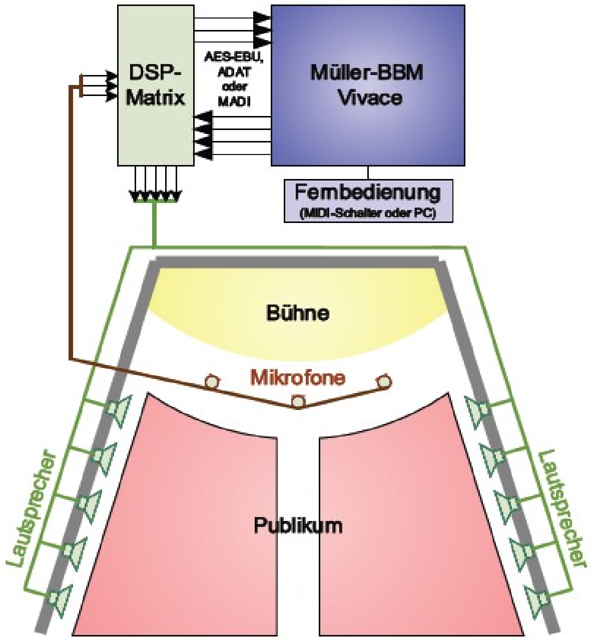

Figure 5.16: *Schematic of Vivace [Mueller-BBM-Brochure, 2008].*

In April 2009 the company Stagetec took over the system. They provide an link-up of Vivace with their mixing consoles.

6 Conclusions

Acoustic enhancement systems (AES) constitute powerful tools in the matter of acoustic variability. The application ranges from rehearsal rooms to big concert halls, wherever high flexibility is necessary.

The main challenge in the design and the installation of an AES is the demand for naturalness. No signs of artificiality of a sound field are to be perceived by a critical listener. The two major concerns in the design are the handling of feedback and the generation of reverberation.

Acoustic feedback in such kind of systems is unavoidable. There are two methods, which can be distinguished by the way of how they cope with the matter of feedback. Non in-line systems use the feedback between microphones and loudspeakers to generate reverberation, in-line systems generate the reverberation electronically, which means feedback is to be avoided. The outcome of undesired, extensive feedback would be howling or ringing tones in extreme cases and even earlier colouration of the sound field occurs, which is fatal. This leads to low loop gains, which means the impact of a system decreases. Solutions had to be found and were found by engineers. Basic equalization of channels is obligatory. Incremental of the number of channels raises the possible SPL output. Placing microphones more far away from loudspeakers and using directional microphones increases the gain before instability. And using time-variant filters has emerged as very useful. All these techniques created powerful systems in the use of acoustic enhancement.

The generation of reverberation is the second sensible topic. Engineers of the last half century worked on creating 'natural' artificial reverberation algorithms. With today's computational powers sophisticated algorithms can be designed and even real-time convolution can be applied.

Many systems have been developed during the last decades and the principal methods of system design seem to work. Future research should be put in how the control of feedback can be improved and what kind of loudspeakers should be used in which way, because they seem to be the weakest element in the system.

Every installation is unique because every room or hall is unique. So no receipt for installing a system can be given. The existing acoustic situation has to be determined before the installation. Questions arising are: Where do I have to put the microphones in order to achieve the best performance? Where can I put the microphones? Maybe location dependent compromises have to be made. The same problems hold for loudspeakers. What's the budget? All these are questions which could restrict the possible impact of an installation. And even when a proper installation was made, the right calibration of the system is not a trivial problem. Companies keep silent about their way of calibration.

An important point which was left out in this thesis is a psychological and social point of view. Not everybody will receive the installation of an AES euphoric. Performers and the audience have to be introduced to the system and it has to be made clear what the system is able to do and what it's not able to do. It has to be pointed out that the system exists to raise enjoyment of performances, not for any scientific experiment. And as we can't expect an audience visually not caring about a concert hall, it's understood that aesthetic solutions have to be found to integrate loudspeakers in the architecture. A system can fail, not because it's not working technically, but because performers and the audience didn't accept it. For what reasons ever.

Acoustic enhancement using electro-acoustic systems constitute a very interesting method for designing room acoustics, which is principally not that complicated, but gets tricky in realization.

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- *http://www.yamaha-afc.com/*
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