Variable acoustics
by means of reverberation amplification
using multiple channel amplification (MCR)

by ing. Cees Mulder

Introduction

Ever since the Roman amphitheatres man is adapting the acoustics to the performance held. Even in ancient Persia one created measures to improve acoustics. At the end of the 19th century Wallace C. Sabine derived a formula which enables us to calculate the reverberation time as function of room volume and absorption present in that room. The ability was created to design the acoustics on forehand, but also to design measures to make the acoustics variable. This was done by removable curtains, rotating panels etc. Making the acoustics variable meant one can use a room for both speech as (symphonic) music, so a venue could be used more economically. Although mechanical variable acoustics is not outdated and is still being improved, after WWII one has been searching for electro acoustic solutions to vary the acoustics. These solutions would also create possibilities for the acoustics when architectural limits had been reached.

History

The history of variable acoustics using electro acoustic means started at the Philips Laboratories during the 1950's by the development of the end-less tape loop machine (early versions used a disk in stead of a tape loop). This machine had one recording head and four or eight playback heads located on intervals along the tape path. Reflection patterns were enlarged by feeding back the signal of one of the playback heads to the input. A microphone was located above the stage and the signal fed to the input of the tape machine. The signals from the playback heads were fed to amplifiers that fed loudspeakers in the hall. In this way early reflections were created to increase spaciousness and reverberation by the recorded feedback. This technique was called “Ambiophonics”, nowadays called “room enhancement”, and is actually more artificial acoustics than variable acoustics. As the amplification was limited due to the acoustical feedback, the reverberation time of the room didn't increase noticeably.

Early 1960's Parking and Morgan developed Assisted Resonance (AR) for the Royal Festival Hall in London. This system consisted of a large number of amplification channels each consisting of a microphone, a narrow-band filter, an amplifier and a loudspeaker. Each channel was tuned such only a small frequency band (3-4Hz) was amplified. A large number of channels (>>100) was required in order to amplify a sufficient frequency range (60 to 700Hz). Both microphones and loudspeakers were located in the reverberant sound. In this way the reverberant sound was amplified causing the reverberation to increase. The system in the Royal Festival Hall was decommissioned in 1999 and removed end 2006. The last Assisted Resonance system was installed early 1990's in Tokio.

In 1967 dr. N.V. Franssen and ir. S.H. de Koning of the Philips Research Laboratories at Eindhoven, The Netherlands, developed Multiple Channel amplification of Reverberation (MCR). As with assisted resonance the reverberant sound is amplified by multiple channels, but each of these channels amplifies the sound over the entire audio frequency band. It took however till 1977 until a commercial system became available, due to the demand for a system from a German Broadcasting company. The system was fully analogue, but adjusting was done with the aid of a CPM operating system based computer and special tools. Even with these tools the alignment took three to six weeks, depending on the number of channels used in a system. Some of the analogue systems, the last one installed in 1994, still operate in 2007. In 1991 Philips started the development of a digital version. This system had automated procedures for adjusting the system. This resulted in a reduction of alingment time and more efficient tuning due to which less channels could be used for the same result as with an analogue system. After the introduction of the digital system and having installed three systems, Philips has quit the activity.

In 2004 Teamprojects started the development of a MCR system using the design criteria and experience from the Philips digital system as well as the experience with analogue systems. The system was created on an open digital DSP platform (Peavey Media Matrix, later NION). Special software has been developed for tuning the system. This software is under continuous development. This has led to an even more efficient and finer tuning.

Variable acoustics

Variable acoustics can best be explained by explaining the following formulas. The first one is W.C. Sabine's formula for the reverberation time:

$$T = 0.161 \frac{V}{A} \qquad [1]$$

where: T = reverberation time [s]
V = volume of the room [m^3]
A = total absorption in the room [m^2 open window]

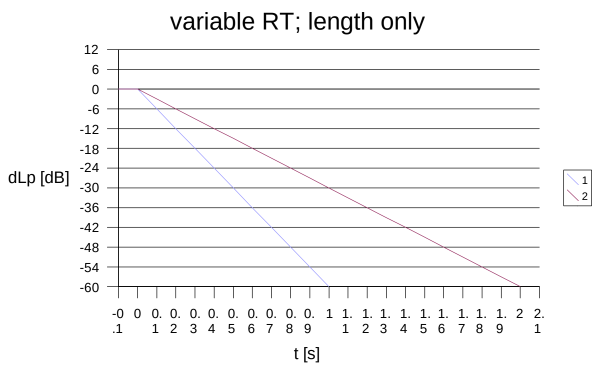
The reverberation time is the time sound decays by 60dB after the sound source has stopped. Another important formula is the one for the averaged diffuse sound pressure:

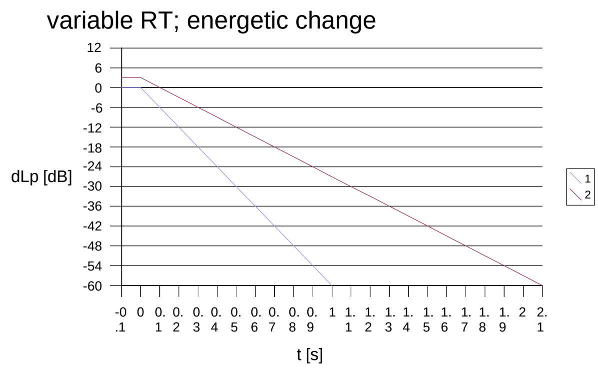
$$p^2 = 4\rho c W / A \qquad [2]$$

where: p = sound pressure [Pa = Nm^{-2}]
ρ = air density [kgm^{-3}]
c = speed of sound in air [ms^{-1}]
W = acoustic power of the source [W]

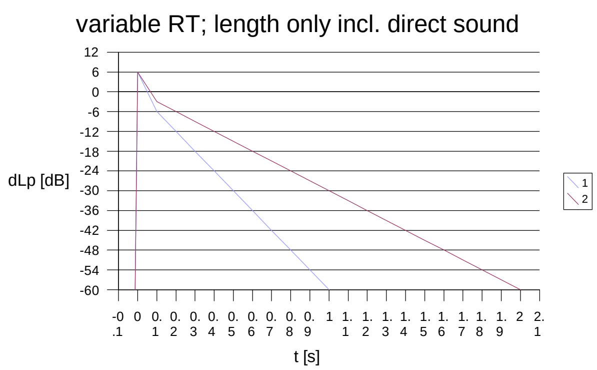
Sabine's formula shows one can alter the reverberation time in two ways:
- by changing the volume of the room;
- by altering the absorption in the room.

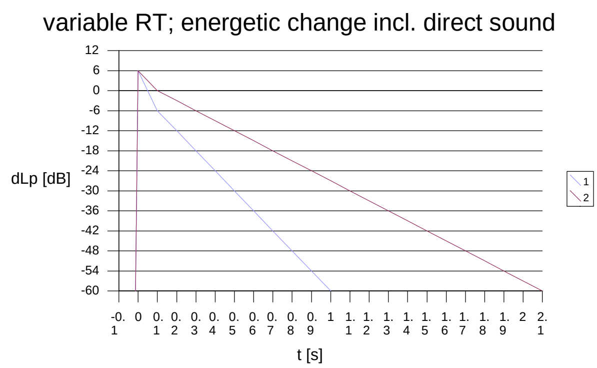
If one keeps the total absorption constant and changes the volume of the room one can alter the reverberation time without changing the average diffuse sound pressure in the room. This can be done by increasing the volume e.g. by increasing the ceiling height. Consequently some absorption has to be removed to keep the total amount the same as before increasing the ceiling height.
However, if one changes the reverberation by altering the absorption, e.g. by removing draperies or closing panels with behind them absorption, one also changes the diffuse sound pressure. So the way one changes the reverberation time makes the difference. In the figures below in a simplified manner the effect is graphically shown when a source is switched off in the diffuse sound field. In one situation the reverberation is varied only by increasing the length, in the other the variation of the length of the reverberation time is proportional to the increase of the diffuse sound level (energy density).



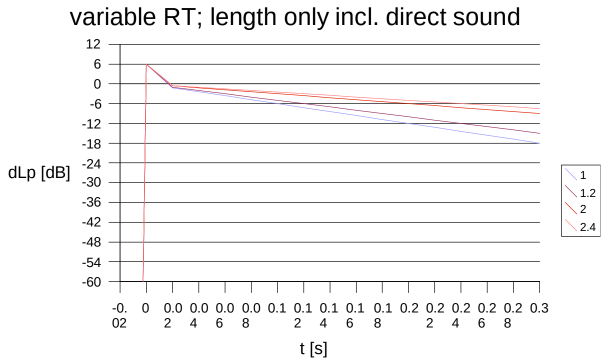


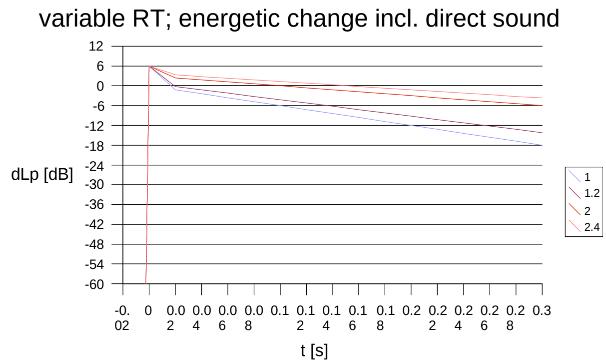
If the graph is recorded closer to the source such the direct sound is included the effect of the change of level of the diffuse sound is even more clear.





So, by changing the reverberation time, the early-late ratio is more affected by increasing the sound pressure than by just increasing the length of the reverberation time (one must say however that changing the volume has also an effect on the early sound due to increasing distances, but not on the direct sound). This effect is especially noticed with running music. For running music only the span of approximately the first 300ms matters. In the figures below the same figures are shown as before, but now only for the first 300ms.





One can see that just increasing the length of the reverberation time has a small effect within the first 300ms span. The effect decreases with increasing reverberation time. Research done at the university of Göttingen and Berlin has shown that the length of the reverberation time by it self is only of importance if the reverberation time is considerably shorter than 2s (Göttingen) or shorter than 1.7s (Berlin). Above 1.7s the increase of reverberation time is only noticeable in running music when this increase is due to the increase of the level of the reverberant sound (increase of energy density).

The reverberation time that is required in a room for a certain kind of performance depends mostly on shape and volume of the room. Reverberation time, shape and volume determine various acoustic parameters that influence our perception of the performance. A parameter that shows the influence of the reverberant sound on the direct and early sound for music is the Clarity C_{80} (for speech C_{50}, where an integration time of 50ms is used):

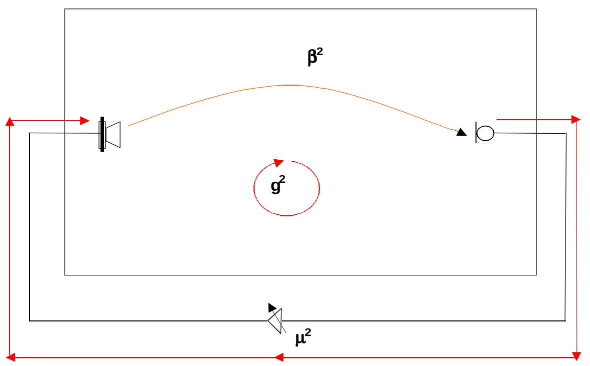
$$C_{80} = 10\log\frac{\int_0^{80} p^2\,dt}{\int_{80}^{\infty} p^2\,dt} \qquad [3]$$

where: C_{80} = Clarity [dB]

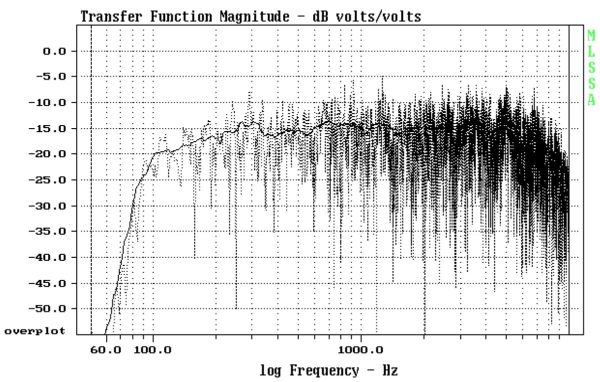
If the diffuse sound pressure alters, the denominator is affected most resulting in a change in Clarity. The Clarity is a very important parameter as it is a measure for instruments and instrument groups to blend, but also to blend the sound of a single instrument having very wild directivity patterns. A high Clarity gives a high definition of instruments, which can be positive. But instruments can sound as if they are single. Also the fullness of the sound of single instruments can suffer as blending of the different tones (frequencies) is not sufficient. If the Clarity is too low, an orchestra sounds “muddy”, there is low definition. The effect of both high and low Clarity is very well explained in the book by L.L. Beranek, “Concert and Opera halls, how they sound”, page 32 and 33. This means there is a preferable range for the Clarity that depends on the kind of performance. For symphonic music the desired Clarity C_{80} is approximately between 0 to -2dB.

Varying the acoustics by amplification of the diffuse sound

Formula 1 and 2 show the acoustics can be altered by changing the absorption. This can be done by removing absorptive materials from the room. But as formula 2 shows, increasing the diffuse sound while the acoustic source power is kept constant will also lead to a decrease in total absorption and thus an increase in reverberation time. To increase the diffuse sound, one can amplify the diffuse sound. Therefore a microphone and a loudspeaker should be placed in the diffuse sound field, such the microphone is placed outside the critical distance of the instrument(s) and of the loudspeaker.



The maximum increase that can be obtained (theoretically) is 2%. This can be explained as follows. The loop gain g^2 of an amplification channel should never exceed 1 (0dB) to prevent oscillation. Furthermore, the peak to average ratio of the frequency transfer function of an amplification loop through the diffuse sound field is 10dB. This means the average level of the loop gain may never exceed -10dB. To prevent colouration the highest peak in the frequency transfer function of the loop should be 7dB below unity gain for music, 5dB for speech. This results in a maximally allowed loop gain of -17dB for a single amplification channel. A practical value is -19dB (1.25%), as one can't obtain a fully flat frequency response. This is not enough to alter the acoustics noticeably. In the figure below an equalized frequency response of an amplification loop is presented.



Multiple Channel amplification of Reverberation (MCR)

By adding loudspeakers and/or microphones to the same amplification channel, the loop amplification will increase. Mathematically the loop amplification is written as follows:

$$g^2 = \mu^2 \beta^2 \qquad [4]$$

where: g^2 = loop amplification
μ^2 = electrical amplification
β^2 = acoustical transfer

This means the electrical amplification μ^2 of that channel has to be reduced to prevent colouration and oscillation. In order to obtain more amplification of the diffuse sound, each microphone loudspeaker combination should have its own amplifier. The only coupling between distinct channels should be via the diffuse sound field (β^2) in the room. This is called multiple channel amplification. The amplification obtained by each channel can now be added up (energetic addition). When using e.g. 48 amplification channels, an increase of the reverberation time can be obtained of (at least) 60% and an increase of the diffuse sound level of (at least) 2dB. A system with this size will then comprise 48 microphones located in the hall each feeding an amplification channel. On each amplification channel one or more (distributed) loudspeakers are connected.

Determination of system size

One amplification channel is capable of increasing the reverberation time and sound energy by 1.25%. If the architectural reverberation time T_{pas} (system passive) and the required reverberation time T_{act} (system active) are known one can determine the number of channels N as follows:

$$T_{act} = T_{pas}(1 + Ng^2) \qquad [5]$$

As guideline to calculate the required reverberation time for symphonic music as function of the room volume V one can use the following empirically obtained formula:

$$T = 2/3 \log V - 0.9 \qquad [6]$$

In order to prevent a very large system one should design the architectural reverberation time as long as possible for the application that requires the shortest reverberation time.

Other applications of Multiple Channel amplification to influence the acoustics

Affecting the lateral contribution
Spaciousness is determined for a large part by lateral reflections arriving 20 to 80 ms after the direct sound (Marshall and Barron). Wide halls lack sufficient lateral sound. One can make a room acoustically smaller by placing loudspeakers in a strategic way and so increasing the lateral energy. Making a room acoustically larger is not possible without changing wall properties.

Adding early reflections
Early reflections support the direct sound. An increase in early reflections means an increase in Clarity. If a theatre is designed well (for speech, high Clarity), no additional early reflections are required when this theatre is used for symphonic music. In many cases they are not wanted if one wants to decrease the Clarity. In large (multifunctional) halls adding early reflections can be desirable. This can be created with microphones above the stage area in order to limit the travel path. But one has to understand that adding early reflections electro-acoustically will also increase the reverberation time. Sometimes it is better to solve this architecturally.

Under balcony support
In theatres with a relatively low and deep balcony, the area under the balcony is screened from the room by the balcony. This creates a dry impression under the balcony. To bring the acoustics of the hall to under the balcony, loudspeakers can be placed under the balcony with corresponding microphones in the hall (e.g. balcony parapet).

Improving ensemble playing conditions
Reflections on stage can be created by placing microphones and loudspeakers above the orchestra. But also here one has to consider an architectural solution first before considering an electro-acoustical solution. The amplification will also increase the reverberation time (on the stage).

Variable acoustics vs. room enhancement

In the cause of time enhancement systems have been developed. In stead of varying the acoustics of a room these systems create a new “virtual” acoustics in a hall, comparable with the early Philips' Ambiophonics. These systems pick-up the sound from the stage area. This sound with a high direct component is processed and than reproduced in the hall. This processing comprises roughly two actions:

1) creating discrete reflections by a tapered delay;
2) adding artificial reverberation by means of a reverberator.

With microphones and loudspeakers in the same room, there will be reverberation amplification. This means to make an enhancement system effective, the effect of the reverberation amplification by this system should be minimized. This can only be obtained if:

- the architectural reverberation time is not substantial longer than one second;
- the hall is not too large, as the larger the hall the more electrical amplification is required and larger halls tend to have a longer reverberation time than small halls.

If this is not the case the electrical amplification should be limited in order to prevent the natural reverberation time becomes too long to create a virtual acoustics. The result will be that the level of the reflections created by the system and the artificial reverberation are too low in level. The effect on the Clarity will be minimal and so the system looses its effectiveness to enhance the acoustics.

Furthermore, if the increase in reverberation time is realised by increasing the length without increasing the energy density, in running music the effect can be that one can have 4s. reverberation time, but that it will sound like e.g. 2s, or even less.

references

- Leo L. Beranek, *How They Sound, Concert and Opera Halls*, Acoustic Society of America, Woodburry NY, 1996
- Heinrich Kuttruff, *Room Acoustics*, third edition, Elsevier Science Publishers Ltd., Barking, 1991

CM070411