ELECTRO-ACOUSTIC ROOM ENHANCEMENT: HISTORY, BASICS AND MISUNDERSTANDINGS

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1 INTRODUCTION

In the 1950s, while working on solutions for other industries, Philips developed the first electro-acoustic device for room acoustic enhancement: a tape machine with an endless loop, called Ambiophony. Later, around 1960, Parkin developed Assisted Resonance (AR) to improve the acoustics of the recently completed Royal Festival Hall. In 1967 Franssen from Philips created the Multiple Channel amplification of Reverberation (MCR) system.

Later, Griesinger, and others, developed further the Ambiophony method by using digital signal processing and sophisticated algorithms. Poletti used Franssen's idea, to add a secondary room in a digital signal processor to an MCR system, in order to achieve longer reverberation times and to decouple reverberation level and time.

At their core, most systems today are based on the two Philips developments. Contemporary systems add inline reverberation processors, processing to enhance stability against feedback, and more sophisticated signal distribution to what are essentially MCR networks. The fundamental acoustical and electrical principles of MCR systems as well as those of expanded electro-acoustic room enhancement systems are discussed, as well as some common misunderstandings

2 THE INITIAL PRINCIPLES

2.1 Ambiophony

In 1953 Roelof Vermeulen of Philips Research developed the idea of using a tape machine with an endless loop to create a device for creating early reflections and an approximation to reverberation. The machine, Philips LBC 7100, used for this was also used (and even sold) as a signal delay device for public address systems (version /00). To use this device for Ambiophony, the signal from a playback head was fed back to the recording head, creating a "train" of reflections that can be regarded as a simple kind of reverberation (version /01). In the figure below this principle is shown. This device was installed, amongst others, at Teatro alla Scala Milano.

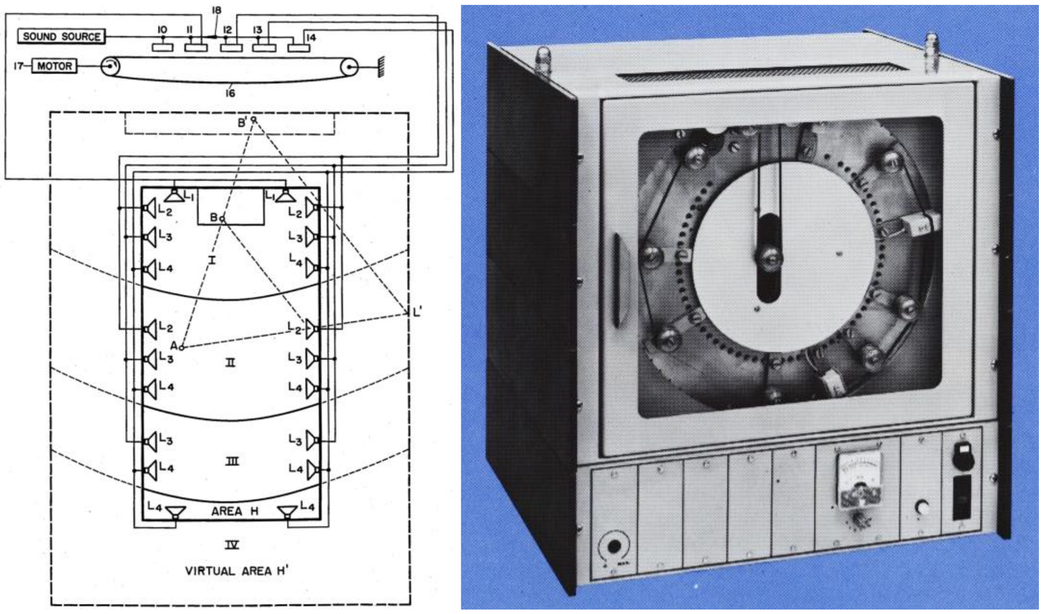


Figure 1: Philips LBC 7100

2.2 Assisted Resonance (AS)

A lack of sufficient reverberation in the Royal Festival hall led to the development of Assisted Resonance by Peter Parkin. For this, Parkin developed a system based on very narrow band tuned amplification channels. In each channel the microphone and the loudspeaker were placed in separate Helmholtz resonators, tuned to the same frequency. A correction circuit in each channel provided a constant phase difference of 0°. The amplification of each channel was such that it was below oscillation, but enough to maintain the energy for the duration of the required reverberation time. The schematic principle of a single channel is shown below.

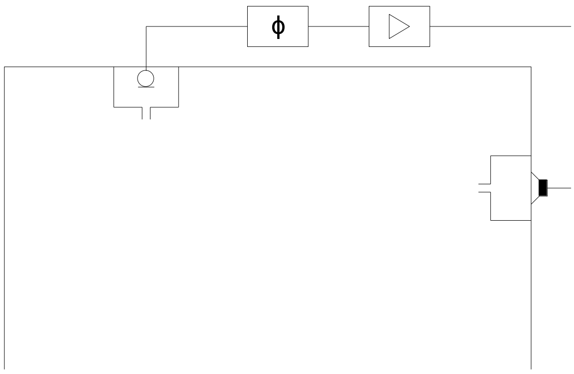


Figure 2: A single Assisted Resonance channel

The effective bandwidth of each channel was initially 3Hz, with later improvements this was increased to 7Hz. The frequency on which a particular channel can function is limited to below 800Hz due to the wavelength with respect to the physical dimensions of the components one channel works with. 800Hz corresponds to a wavelength of approximately 43cm. This meant a limited frequency range could be achieved, but required still a lot of channels to cover this frequency range. Due to the complexity and the limited useable bandwidth, only a few systems were installed.

2.3 Multiple Channel amplification of Reverberation (MCR)

In the fall of 1967 Nico Franssen of Philips Research came up with the idea of increasing the reverberation time by amplifying reverberation. Strangely, this development started because insufficient gain could be obtained with the Ambiophony system. According to Franssen a single channel had a limited available gain regardless of the number of loudspeakers connected to it. In order to increase the gain of a system, a second Ambiophony device should be installed in parallel without the coupling of the electrical signal paths, only the coupling over the reverberant field (via the microphones and the loudspeakers connected to each device) would enable an increased gain per channel. In this way the gain of each individual device could be added to the total system gain. As a gain increase leads to an increase in the energy density of the reverberant field, the reverberation time will increase proportionally (for a constant source sound power). Franssen concluded that the Ambiophony devices could be left out now, as the increase in reverberation time was established by use of multiple amplification channels.

3 METHODS FOR INCREASING GAIN

In order to amplify the reverberant field sufficiently to obtain a noticeable (and desirable) increase in reverberation, sufficient gain needs to be achieved. The gain of a single amplification channel is limited by colouration and feedback: The maximum channel gain is primarily determined by the difference in magnitude between the peaks and the average of a measured loop response of an amplification channel (measured from loudspeaker to microphone) and the necessary difference between the highest peak and unity gain in order to prevent colouration. The first aspect is approximately 10dB as predicted by Schroeder [6, page 72]. The second aspect is 5dB for speech systems and 7dB for music applications. This means, for music applications, the maximum gain to be obtained from a single channel is -17dB or a 2% increase in reverberation time and level.

Various methods have been developed to increase the channel gain. One of the first methods to obtain an increase in gain is the moving microphone technique described by Zwicker in 1928, as shown in the left picture below. By moving a microphone continuously, the fine structure, and thus the peaks, continuously changes preventing a channel to get into oscillation.

A subsequent method was described by Schroeder in 1959 by the introduction of a time variance circuit in an amplification channel, as shown in the right picture below, in order to continuously shift the “peaks” in the channel loop response to a “dip”.

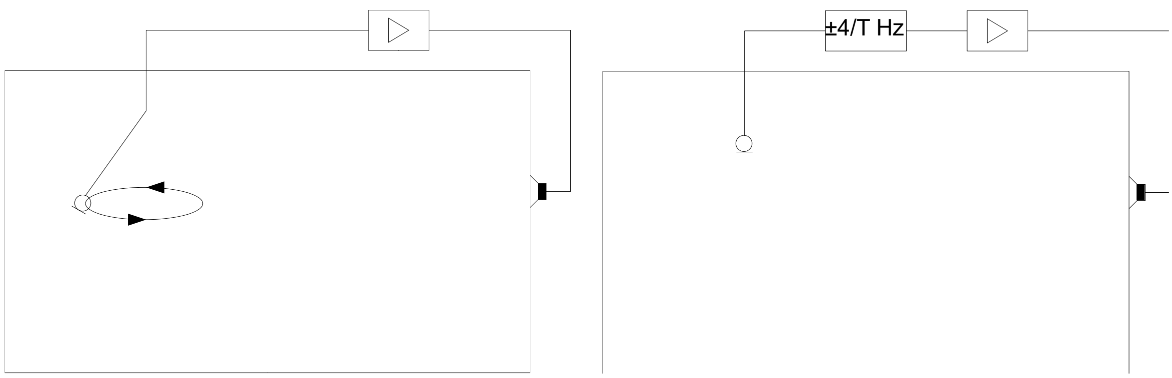


Figure 3: Methods to increase channel gain. Left: moving microphone technique. Right: Time variance in the electrical path.

The final method is the use of multiple independent amplification channels, as described by Franssen, however, it is remarkable that the increase in reverberation time and level was first discovered when the MCR approach was developed, almost 40 years after Zwicker's idea of the moving microphone.

4 BASICS REVERBERATION AMPLIFICATION

4.1 Loop gain vs. forward gain

When looking at an amplifier channel one has to regard the electrical behaviour (left picture), the stability to prevent colouration and oscillation, and the acoustical behaviour (right picture), the fact that it amplifies sound. For the amplification of the reverberant sound, the loop amplification is equal to the forward amplification of the reverberant sound; one puts back the signal in the same room as where it comes from. This makes it possible to calculate the corrections needed, as the loop gain g^2 [-] is easy to measure.

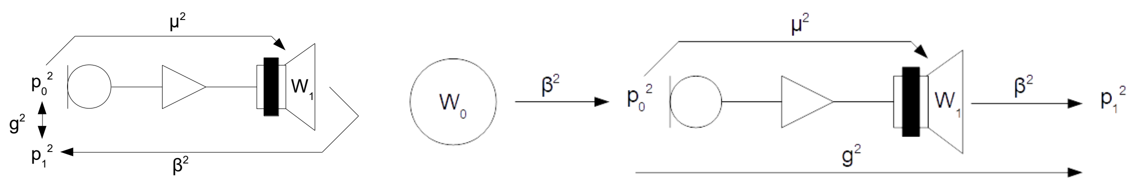


Figure 4: Left: electrical behaviour of an amplification channel. Right: an amplification channel from an acoustical perspective.

In the figure above, β^2 [Pa^2W^{-1}] represents the acoustical path from sound power to sound pressure and μ^2 [WPa^{-2}] the electrical path from sound pressure to sound power.

4.2 Negative absorption

A loudspeaker that is fed by a microphone in the same room can be regarded as a reflector, with the difference from a physical architectural reflector being that the point of incidence is not equal to the point of reflection. Furthermore, a loudspeaker is capable of "reflecting" more energy than is incident on it, so actually behaving as negative absorption. For this negative absorption introduced in a room with a system with N active amplification channels, the following applies:

$$A_{neg} = -A_{arch}\frac{Ng^2}{1+Ng^2}$$

The negative absorption A_{neg} [m^2] is thus determined by the architectural absorption A_{arch} [m^2] and the system gain Ng^2 [-].

4.3 Required gain correction for a target reverberation time

When using a system for reverberation time increase or correction, three situations are regarded:

- the reverberation time of the actual room: T_0 [s];
- the reverberation time of the room with an active system: T_{act} [s];
- a target reverberation time: T_{targ} [s].

When one wants to correct an active system for a target reverberation time one can calculate the gain correction $\Delta\mu^2$ [-], for which implies:

$$\Delta\mu^2 = \frac{1 - \frac{T_0}{T_{targ}}}{1 - \frac{T_0}{T_{act}}}$$

4.4 Reverberation amplification vs. direct amplification

The use of multiple channel amplification is not limited to the reverberant field, but the technique can also be used for direct amplification, e.g. to create an electro acoustic sound reflector by multiple loudspeakers in front and above a stage and microphones above an orchestra. One has to take now into account that there are two acoustical transmission paths, but only one electrical path. This means one can use the electrical path (the electrical gain and equalisation) to correct only one of the two which leaves an error for the other.

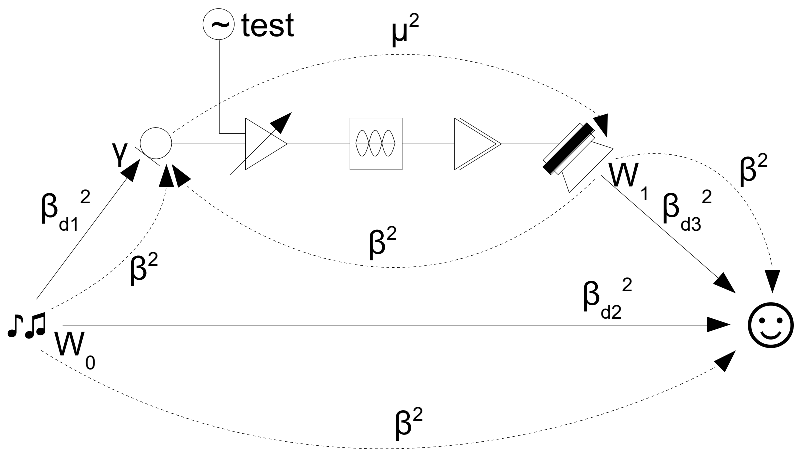


Figure 5: The transmission paths, direct, reverberant and electrical, for an amplification channel in a room

When a system is created where both the direct sound and the reverberant field are amplified, each part of that system should have its individual tuning. The channels for the direct field amplification (that uses d1 and d3) will introduce an error for the reverberant field. This error can be compensated for during the iterative tuning of the channels for the reverberant field amplification by taking the direct amplification channels into account.

5 AMPLIFICATION VS. REGENERATION

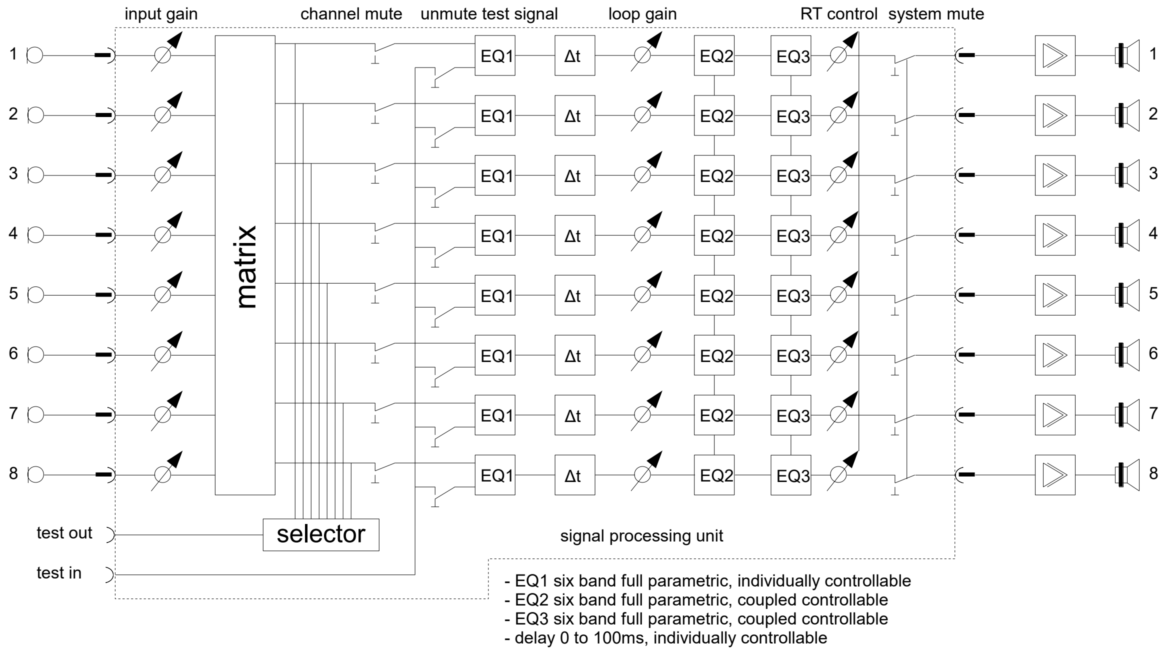
Although AS and MCR look similar, the working principle differs fundamentally - however the differences are often confused. AS uses the feedback of the channel to obtain a sustain effect, the principle design is an oscillation circuit that is kept below continuous oscillation. Therefore, AS should be considered as a (re)generative system. It is also of the utmost importance to keep each channel stable to prevent it from continuously oscillating (feedback).

With MCR, the loop amplification is low enough that the effect of the loop can be ignored and therefore it can be considered as an amplification system, using the forward amplification effect.

There is no temporal elongation or regeneration of the signal involved. Moreover, regeneration is even unwanted with reverberation amplification as it causes ringing and therefore undesirable colouration.

6 DIY SYSTEM

A simple acoustic enhancement system can be created from freely available standard components such as mixing consoles, audio signal processors, microphones, amplifiers and loudspeakers. Simply connect or route a single microphone to a single loudspeaker. An example of an eight channel system is given below, but when required a system with more or fewer channels can be created.



To tune a system one must be able to measure the loop frequency response of a channel. For this a measurement device should be temporarily inserted into the signal path of the channel under test. With most DSP devices and mixing consoles one should be able to temporarily insert this into the signal path, switching from one channel to the next. The measurement procedure is now as follows:

- switch all channel into “mute”;
- activate the first channel and equalise the loop to a sufficiently smooth and “flat” frequency response;
- bring the channel close to oscillation such that the tone just sustains (ringing);
- reduce the gain by 10dB;
- switch the channel into “mute”;
- repeat this for all channels;
- when done, unmute all channels.

In order to optimise the overall “gain before feedback” for the system, check each channel for its “distance to feedback” by bringing each channel individually to oscillation, as described above, but now with all other channels active. Note down this “distance” and bring it back to its original position. Repeat this for all channels, determine the average and correct each channel for that average. By changing the overall gain, e.g. by coupling the level controls, one can now change the reverberation increase in the room. If needed, one can use a second equaliser bank to correct the overall response for specific frequency bands.

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