# DESIGN AND TUNING OF ACTIVE ACOUSTIC ENHANCEMENT SYSTEMS

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## 1 INTRODUCTION

A significant portion of the design and tuning of an active acoustic enhancement system goes into avoiding and removing ringing artifacts in the system's response. However, most of the design and tuning is aimed to achieving a good and natural sounding system. This paper presents the use of modular design and tuning options of hybrid regenerative acoustic enhancement systems, providing an insight for those involved in the planning of such systems. Early reflection modules and diffuse reverberation modules are presented to allow for the control of the delicate balance between the early and late/diffuse energy in the system's response. Furthermore, hybrid regenerative systems are presented to operate in a theoretical range from full regenerative to full in-line, where in practice the optimal and most natural results can be achieved in a well-balanced hybrid of the two. The main challenges are in finding both the ideal balance between early and late energy and the right blend of regenerative and in-line techniques.

## 2 ACOUSTIC ENHANCEMENT TECHNOLOGIES

### 2.1 Regenerative systems.

In 1968, Philips NV patented a regenerative acoustic enhancement technology named Multiple Channel Reverberation (MCR[1]), applying multiple independent microphone/amplifier/loudspeaker loops in an acoustic space to enhance ('regenerate') acoustic reflections. As anyone involved in live sound is very aware of, wherever such a closed loop is applied, measures have to be taken to prevent ringing or, in the worst case, feedback-induced oscillation. If the purpose of the system is to amplify sound from a stage to an audience, then directional microphones can be placed close to the sound sources on stage, and speakers can be pointed away from the stage to the audience to reduce the loop gain. This way high sound pressure levels can be achieved without ringing or oscillation. However, if the purpose is to amplify diffuse reflections in an acoustic space, this method can no longer be used because omni-directional microphones and speakers are placed at locations beyond the room's critical distance in order to pick up and play back the acoustic reflections to be amplified, actively using the many reflection paths between the two. The solution in Philips' 1968 patent was to use many independent loops, with individual microphones and speakers placed at appropriate locations in the room (usually at ceiling level) so that valleys and peaks in the individual loop transfer functions spread out over the audible frequency range. As channel loop gain is needed to be less than -21 dB to avoid audible ringing, 20 to 100 multiple channels must be combined to provide enough gain to amplify the diffuse reflections, hence increasing the reverberation time of the room.

## 2.2 In-line systems.

In 1987, the Dutch company ACS introduced one of the first in-line acoustic enhancement systems named Acoustic Control System (ACS[2]). SIAP[3], Lexicon LARES[4] and Stagetec Vivace[5] all followed with similar approaches, each using the in-line technique,. Falling back to using directional microphones close to the sound sources on stage, very low loop gains are used in these systems to amplify sound, so less channels have to be used to achieve a stable high SPL sound field. By adding reverberation algorithms in each loop, a diffuse reverberation field is simulated. Modern versions of these systems use convolution, applying acoustic fingerprints of existing venues to theatre and concert halls.

## 2.3 Advantages and disadvantages

Pure regenerative systems work by amplifying existing reflections – which is good if the existing reflections are themselves beneficial (for music). However if there are anomalies in the existing acoustics, such as peak resonances or ‘slap’ echo’s, then the system also amplifies these. In more recent MCR systems, equalizers and delays are used to partially control such anomalies, but the scope of possibilities to influence the reverberation pattern is limited. A problem of similar importance is the so-called ’bath chamber’ effect of regenerative enhancement: the only method of increasing reverberation time is to amplify the reflections, so the reflections take longer to die out. It means that with increased reverberation times there also will be an increased volume. For limited enhancement of reverberant rooms this is not a problem, but for smaller rooms, or rooms with a short reverberation time, it can be a problem. Finally, since regenerative systems use microphones and speakers placed on or beyond the critical distance of the room, early reflections are often difficult to cover because of the acoustic delay from sound source to microphones, and from speaker to listener. In a medium sized concert hall these distances can be more than 15 meters, covering a time gap of 50 milliseconds or more before the first regenerated reflection can reach the listener’s ears. It is why regenerative systems as a rule are less suited to enhance early reflections.

Pure in-line systems have a full freedom of influencing the reverberation pattern, and because they do not rely on amplification, there is also full freedom in setting the volume of the diffuse reverberation field. Even very long reverberation patterns can be used at a low ‘convenient’ volume. Additionally, because the system’s microphones are placed close to the sound sources on stage, early reflections are much easier to generate. However, of course, all these advantages come at the expense of a disadvantage: the system works only one-way; from the stage to the audience. For theatre arts this is not a problem, but for the acoustic arts it is – for several reasons. Firstly, the performers on stage and the listeners in the audience are connected visually by the lines of sight, enhanced by the visual design of the concert hall. This connection also exists acoustically, through direct sound and through reflections. The interaction of the acoustics between stage and audience creates intimacy, connecting the performers to the audience. With an in-line system, this connection is lost: the stage area sounds different from the hall and vice versa. Secondly, the stage itself cannot be acoustically enhanced without regeneration, since that would require the microphones and speakers to be aimed into the same area, and spurious regeneration will occur – contradictory to the in-line concept. Since stage acoustics affect the quality of the performances happening on the stage, this is a serious problem – a performer will hear the reverberation coming from the audience area, but will not be surrounded by it.

## 2.3 Hybrid-Regenerative systems.

In 1987, The Japanese company Yamaha patented the AFC system[6], applying a regenerative system concept based on the Philips patent, while using reflection algorithms in the individual regenerative loops. The USA based company LCS applied a similar concept in 1993 with the VRAS system[7], later adopted and renamed to Constellation by Meyer Sound. Both the AFC3 and

Constellation systems aim to achieve a stable, natural sounding diffuse reverberation field by combining the two enhancement methods through a hybrid and modular approach, avoiding the limitations of the individual technologies.

First, a clear distinction between early reflections and diffuse reverberation is adopted, each served with the most appropriate method: an early reflections subsystem using directional microphones close to the performers on stage to achieve short acoustic delay paths, and a diffuse reverberation subsystem using microphones on or beyond the critical distance of the room to achieve a natural diffuse reverberation field.

Second, the diffuse reverberation system utilizes a regenerative method with reverberation algorithms (VRAS) or convolution patterns (AFC3) in the regenerative loops to allow the original acoustics of the room to be changed, with a high degree of freedom to influence the level and spectral characteristics of the diffuse reverberation field.

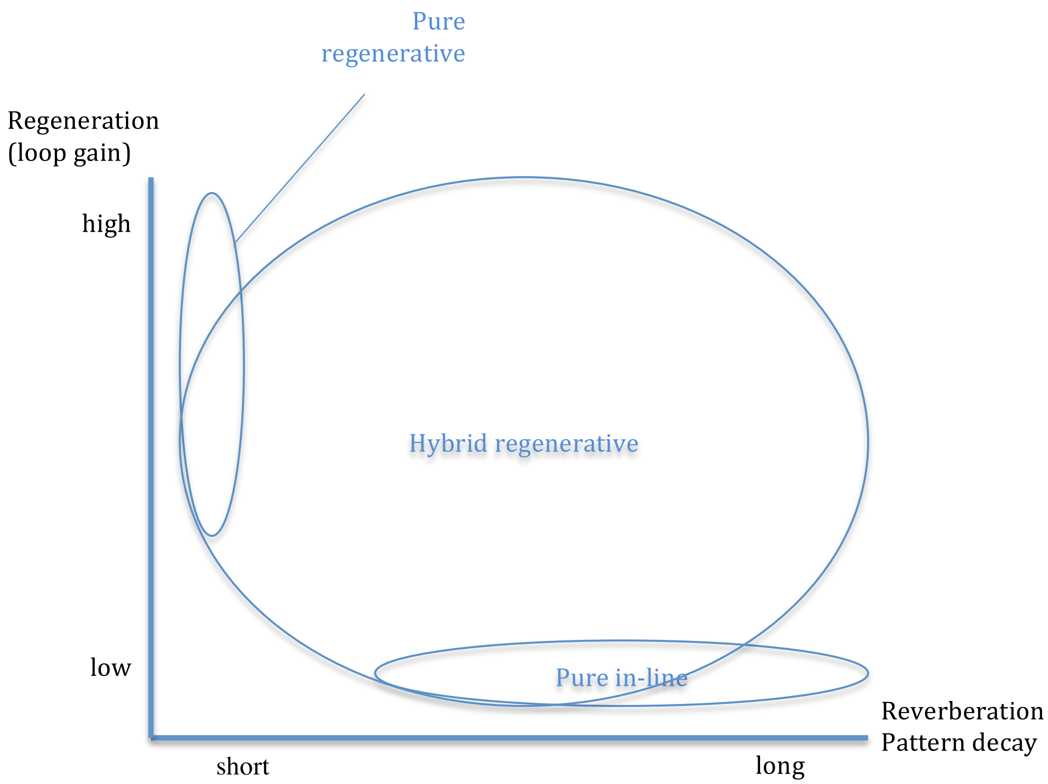


figure 1: system concepts

# 3 DESIGN AND TUNING CONSIDERATIONS

The modular concept of separate early reflection and diffuse reverberation subsystems supports a control of the delicate balance between early and late energy, in the room's acoustic response. It is in the first milliseconds after a direct sound where the auditory cortex of the listeners works at its hardest to determine the meaning of auditory events – integrating early reflections below a certain level with the direct sound – increasing the intensity of the sound but at the same time losing spatial resolution, widening the 'Auditory Source Width' (ASW). The later and/or higher level reflections are perceived as 'Listener EnVelopment' (LEV), representing the acoustic environment surrounding the listener. The perception of early reflections is influenced by the level and time location of each individual reflection as shown in figure 2: the precedence effect threshold curve[8].

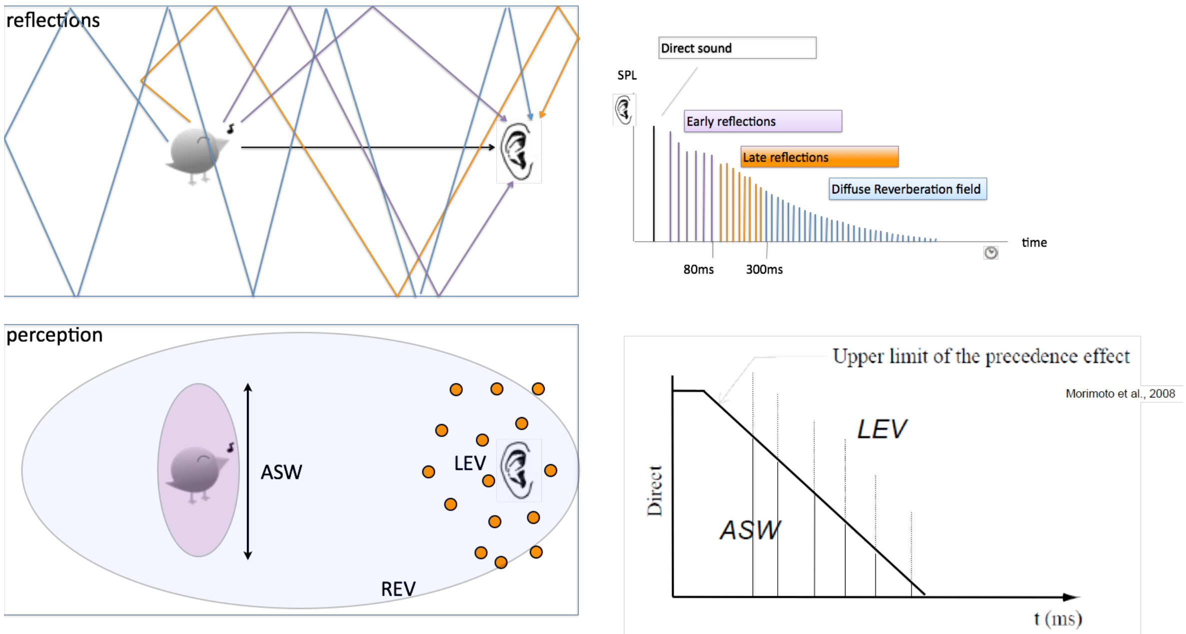


figure 2: reflection level and time attributes influencing ASW and LEV according to the precedence effect (Haas Effect).

## 3.1 Placing Early Reflections transducers and applying delay

By designing the locations of the transducers (microphones and speakers), and by applying appropriate delay times in the system's DSP engines, reflections can be 'tuned' to match the existing acoustic reflections, falling either inside or outside the precedence effect threshold. Of course, the lower time limit for first order reflections is the distance between sound source and microphone plus the distance between speaker and listener. The placement of the microphones is a particular consideration at the design stage, as they almost always are in conflict with sightline requirements - even if the microphones are very small. Early reflections designed to fall outside of the precedence threshold can be utilized to simulate an increased room size by increasing path delays, setting additional delay times in the system's DSP delay matrix.

## 3.2 Choice of Early Reflections transducers

Other than the locations of microphones and speakers, and the setting of delays, the choice of transducers offers further control of the early reflections patterns. By using directional microphones and speakers, direct feedback can be avoided, using the early reflection system in 'in-line' mode. The repetition of reflections by acoustic feedback is suppressed, allowing second and higher order reflections to be designed in great detail by using additional delay lines in the system's DSP. An alternative method is to use omnidirectional microphones and speakers, allowing a degree of regeneration to occur. The higher order reflections are then caused by the acoustic feedback between speakers and microphones, possibly creating a more natural response. This method can also be used as a 'primer' to enhance early energy in relatively 'dry' spaces prior to applying the diffuse reverberation modules.

## 3.3 The inverse square law challenge

Speakers for early reflections subsystems are preferably mounted on the sidewalls of a room to produce lateral reflections, affecting the sound perception most effectively. If the audience would be sitting in a limited area in the center of the room then wave front synthesis can be used to exactly control the time, level and direction of early reflections. But for audiences in concert halls, normally starting already at a few meters from the sidewalls, the possibilities are very limited. Unlike acoustic reflections – where an inverse-squared drop-off per doubling of distance property starts from the sound source, keeping the relationship also when they are reflected - systems using speakers start the inverse-squared relationship at each speaker. See figure 3. This means that, where acoustic reflections have a relatively constant level for listeners in seats close to the walls as well as in the center of the audience, reflections played out by loudspeakers drop off much faster, making it impossible to exactly simulate an acoustic reflection pattern. Even for the first order reflections, the squared drop-off causes a significant level difference between the side seats and the middle seats in an audience – causing either the listeners close to the wall to hear too loud reflections, or the listeners in the middle of the room to hear too weak reflections. Using long column speakers or large flat surface speakers helps solving this problem, but often is not acceptable for visual reasons - especially in historic venues. The most commonly used solution is to mount the speakers on a high position on the wall so the influence of the squared decay is less, but the reflections still count as lateral. As a compromise, a longer acoustic delay in the early reflection system has to be accepted because of the larger distance of the speakers to the listeners.

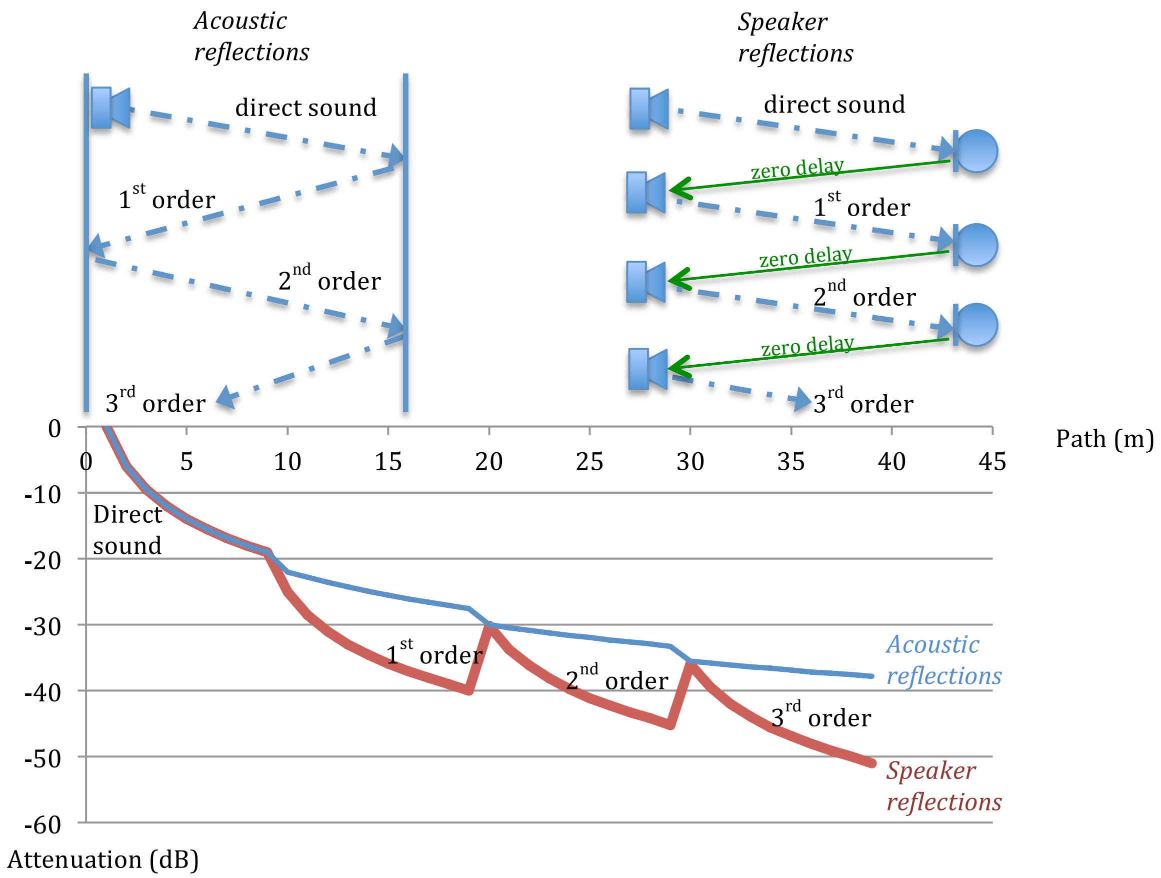


figure 3: inverse square law: simulation of acoustic and active single reflection paths with a 10m wall distance and -2dB wall absorption / matching DSP attenuation, and no diffusion and air absorption.

### 3.4 The hybrid 'sweet spot'

The hybrid-regenerative concept allows for the diffuse reverberation field to adapt to the existing acoustics, as well as, if applied, to the enhanced early reflections. By varying the length of the reverberation algorithm in the regenerative loops, acoustic energy can be extended in time to fit the goal: short patterns for short reverberation times, long patterns for longer reverberation times. At the same time, reverberation time is also affected by the amount of regeneration ('liveness') of the system. Of course, the liveness directly affects the level of the diffuse reverberation. These two parameters – 'decay time' (of the reverb pattern) and 'liveness' (loop gain) - interact with the physical positioning of the system's transducers, and theoretically can provide any result between pure regenerative and pure in-line. A fourth parameter is the selection of the DSP reverb pattern affecting the sound characteristics. For pure in-line mode systems, the DSP reverb pattern determines the system's response completely; often convolution patterns of good sounding concert halls are used. It's no problem for hybrid systems to offer some presets in this mode using directional microphones, although it is not the primary target. Instead, the primary target is to provide a natural response, based on the existing acoustics, fitting the venue best. The reverb algorithm is then used as 'make-up' template to correct issues in the existing acoustic response - for example to reduce or increase low frequency reverberation time to affect the Bass Ratio of the room's response.

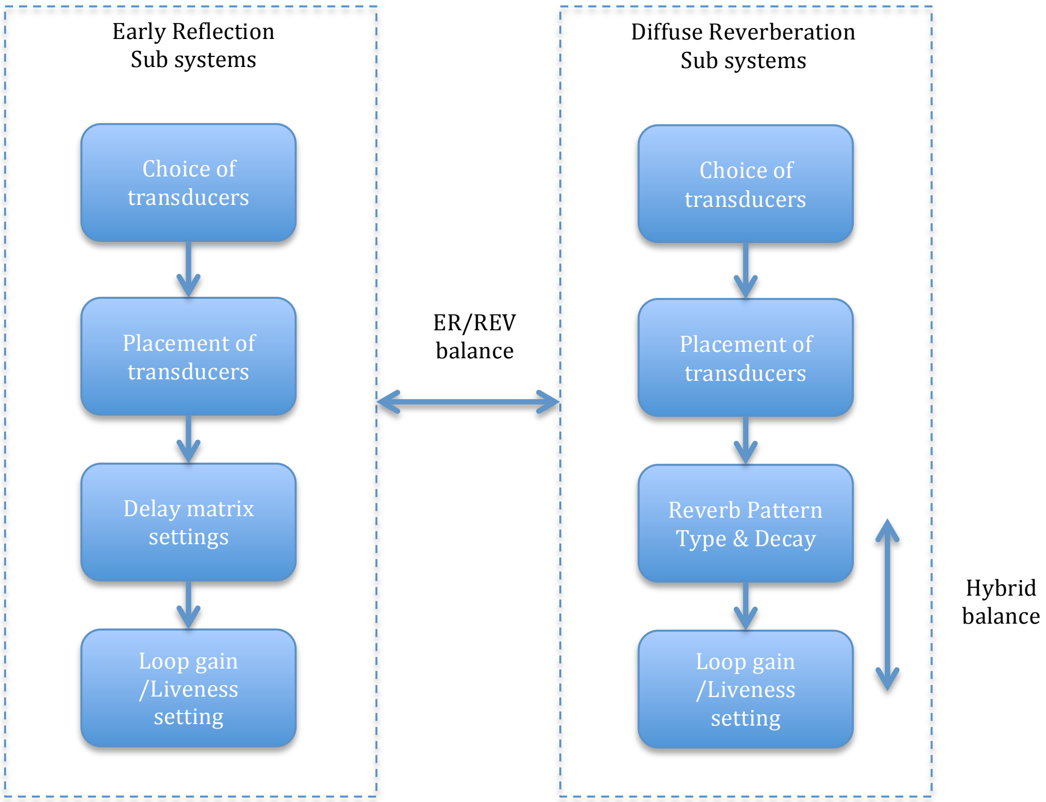


figure 3: design & tuning tools

### 3.5 Listening vs ISO3382

The described set of design and tuning tools allow for virtually any result to be achieved – whether the goal is to fulfill a set of ISO3382 parameter requirements, or to sound fantastic and natural. Although the two goals are correlated to some degree, the correlation is not 100%. Often formal requirements such as EDT to RT ratio, strength (G) or clarity (C80) will not necessarily produce a good sounding result for a particular room. Vice versa, a result tuned 'by ear' can sound natural and appropriate, but not comply to parameter requirements set for the project. We propose to assume that a good sounding result, to be assessed by listening, is the ultimate goal of any acoustic project. The practice of designing and tuning of acoustic enhancement systems therefore mainly involves listening, and using previous design and tuning experience to achieve the optimal result. The modular approach with hybrid regenerative systems offers many tools to achieve this result, but our ears remain the main tool.

## 4 CONCLUSION

With modern active technology, using microphones, DSP engines, power amplifiers and speakers, it is possible to enhance a room's acoustics at a very high quality level. However, the design and tuning of active systems differs greatly from the design and tuning of mechanical systems. This paper provides a first insight in the design and tuning challenges of active acoustic enhancement systems so consultants, system integrators, musicians and investors can understand the basic technical concept, and also understand the exciting potential of active systems to provide high quality results with a much broader application range and variability compared to mechanical measures. However, the key element common to the design and tuning of acoustic enhancement systems, both mechanical and active, is the listening. It's what we all hear that makes rooms sound great.

## 5 REFERENCES

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