Yamaha Commercial Audio Systems Europe

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**Title:** Active Acoustic Enhancement Systems - introducing Yamaha AFC3

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**Abstract:** With the introduction of advanced Digital Signal Processing technologies, a new generation of regenerative active acoustic enhancement systems is brought to the market. This development allows for systems to produce excellent results with less microphone/speaker loops than before, lowering costs and increasing flexibility so systems can also be applied in medium and small size venues. This paper presents a brief history of the field of active acoustic enhancement systems, describing the concepts and DSP architectures used in today's regenerative systems, and introduces the latest generation of Yamaha's Active Field Control system based on a 'hybrid modular' approach that combines multiple active acoustic enhancement concepts and technologies.

**Keywords:** AFC, AFC3, EMR, FIR, regenerative, in-line, hybrid regenerative, early reflections, modular.

## 1. Introduction

The field of 'Room Acoustics' describes the acoustic behaviour of a room (or rather a hall), analysing the way sound waves are reflected and absorbed by it's walls, floors, ceilings and objects. For concert halls, theatres and opera houses, the acoustic behaviour is a key factor in the success of the venue - with the result measured by the public's appreciation of the musical acts performed in them. It has to be noted that different acoustics are required to support different musical performances best - eg. an organ concert needs a longer reverberation time than a symphony orchestra concert. In many cases, a hall is built specifically to suit a particular performance type.

The acoustic behaviour of a hall can be described by analysing the acoustic energy at the listener location after an omni-directional sound source radiates an impulse at the stage position. From the resulting impulse response graph the direct sound, early reflections and the reverberation field energy levels can be observed, and several acoustic parameters can be extracted to describe the acoustic behaviour of the hall.

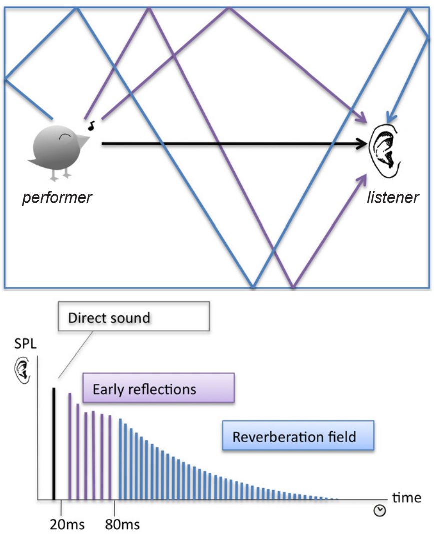


*figure 1: room acoustics*

Amongst the most important parameters are the Reverberation Time $RT_{60}$, defined as the time for the average sound energy density to decrease to -60dB after the source stopped. Also Clarity $C_{80}$ is an important parameter, calculating the ratio of the energy in the first 80ms to the energy after 80ms to represent how well the sound source signal's details over time can be heard. For a complete overview of parameters we refer to Beranek [1].

Sometimes the acoustics of a hall are not entirely optimal to support the musical performances held in it - mostly because trade-off's have been made between acoustical and visual quality of the hall. In that case the acoustic behaviour can be improved by changing wall materials to be more or less absorbent, or by placing reflectors or absorbers to increase or decrease the reverberation field.

In a growing number of cases, a hall is designed to be able to support multiple types of musical performances. Such a 'multi-purpose' hall can be used more efficiently - allowing higher return on investments, a very important factor in today's economic environment.

Because mechanical measures to improve a hall's acoustic behaviour - or to introduce variability - are very expensive, using electro-acoustical tools to achieve the same result have become increasingly popular. Not only because of the lower cost, but also because of the ease of use: a mechanical solution for acoustic variability often requires the assembly and placement of heavy acoustic panels, while an electro-acoustic solution requires only pressing one preset button. Last, but not least, with the latest DSP technologies, the use of electro-acoustic systems allows the acoustic behaviour of a hall to be changed far more than would have been possible with mechanical measures.

Basically, using an electro-acoustic system to enhance the acoustic behaviour of a hall constitutes placing one or more microphone(s) and speaker(s) in the hall, connected together through one or more amplifier(s). In practical cases, multiple microphone-speaker loops are required to achieve a stable system. Often more speakers are used per loop to achieve an equally distributed sound field.

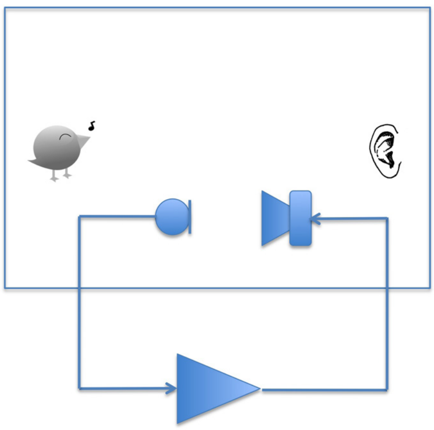

*figure 2: active acoustic enhancement concept*

### 2. System concepts: *in-line* and *regenerative*.

Basically, the enhancement of the acoustic behaviour of a room can be achieved in two ways: either by synthesizing reflections based on the direct sound, or by adding reflections based on the room's original reflections.

The first method is sometimes referred to as 'Synthesis of Sound Field' S-SF [2], or more commonly as 'in-line'. In-line systems work by synthesizing the required reflections in a room based on the direct sound, playing them back to the audience through a speaker system. If the room is highly absorbent then the result can be controlled completely by the synthesized reflections. If the room already has reflections, then the result is the sum of the original reflections and the synthesized reflections. The system offers a one-way response, generating acoustic energy only from the performer area to the listener area. If the performer steps out of the performer area (eg. stage), then the system no longer works. Also, acoustic energy from the listener is not included in the system's response.

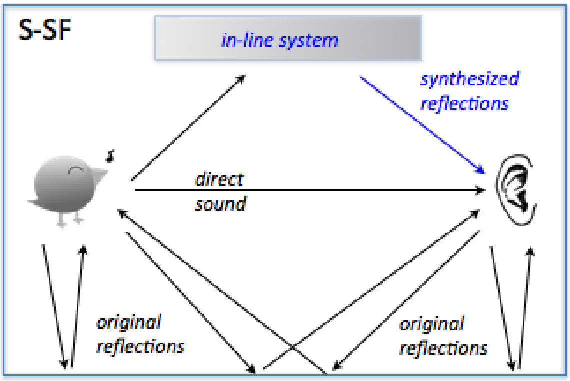


*figure 3: S-SF 'in line' active acoustic enhancement concept*

The second method is sometimes referred to as 'Assistance of Sound Field' A-SF, or more commonly as 'regenerative'. Regenerative systems work by amplifying a room's already existing reflections, so the result is completely based on the given acoustic condition. It is an overall response, enveloping both the performer and the listener.

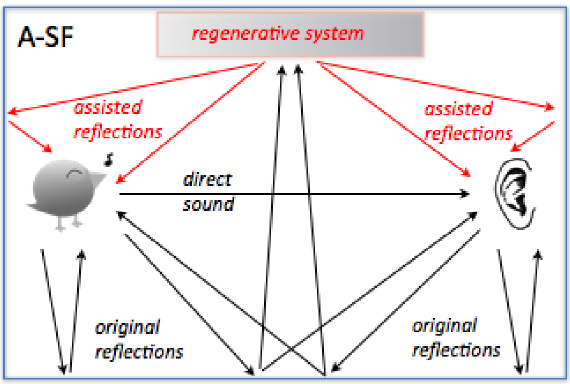


*figure 4: A-SF 'regenerative' active acoustic enhancement concept*

Figure 5 presents a historical overview of the most relevant systems brought to the market since 1955, with the numbers at the left denoting the number of installations found published by the manufacturers on the internet. Years with '...' attached indicate that the system is commercially available on the market in 2012.

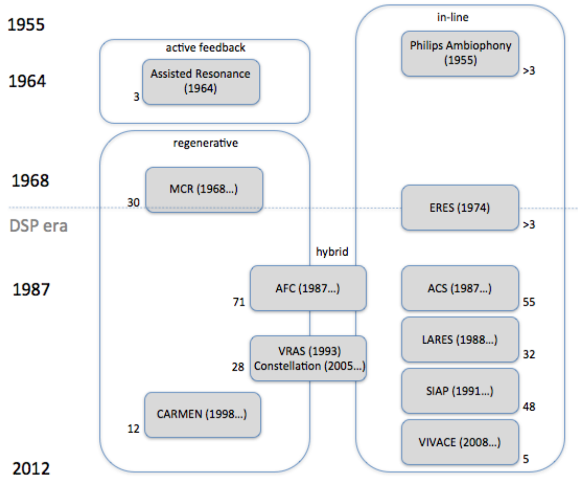


*figure 5: historical overview of active acoustic enhancement systems.*

### 3. Challenges

The main challenge for the acoustic enhancement system designers is a familiar one in the field of sound reinforcement: if a microphone - amplifier - loudspeaker combination with a high enough gain is placed in a sound field, the sound field will be amplified, but certain frequencies will stand out, colouring the sound. If the gain is set to an even higher level, the system will start to oscillate at a certain frequency. The reason for this is that the open loop gain $G_O$ of the created loop - including the electro-acoustical transfer function μ (from microphone to amplifier to loudspeaker), and the acoustical transfer function β (from the speaker to the microphone) - becomes close to or greater than 1.

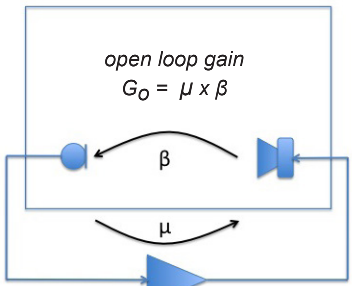


*figure 6: open loop gain.*

The acoustical transfer function β consists of the sum of all reflections that occur in the hall between the speaker and the microphone. Depending on the size and shape of the hall and the position of the microphone and speaker, reflections wil cancel each other out for some frequencies, and add up for others. The difference between cancellation valleys and addition peaks can be tens of dB's; figure 7 shows an example acoustic transfer function β. Because β is part of the open loop gain, it becomes obvious that if the electrical gain of the amplifier is increased (increasing μ), the open loop gain becomes greater than 1 first for the frequency with the highest peak - this is the oscillation frequency. But for open loop gains slightly lower than 1, the peaks will generate long reverberation times for the frequencies involved, acting as a filter, causing colouration.

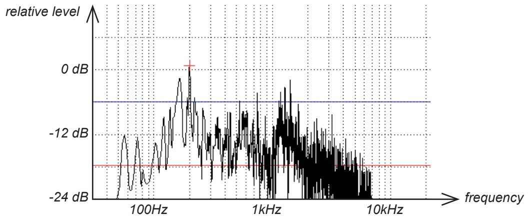

figure 7: example of an acoustic transfer function β

To prevent colouration or oscillation at the peak frequencies, several countermeasures can be taken. Table 1 shows the available options as used by today's commercially available active acoustic enhancement systems. Countermeasures and system types will be explained in the following chapters.

| | regenerative | | | | in-line | | | |
|---|---|---|---|---|---|---|---|---|
| | MCR | Constellation | Carmen | AFC3 | ACS | SIAP | LARES | VIVACE |
| use many independent channels | ✔ | ✔ | ✔ | | ✔ | ✔ | | |
| use time variance | | | | | | | ✔ | ✔ |
| use spatial averaging | | | | ✔ | | | | |
| use DSP/FIR | | ✔ | | ✔ | | ✔ | | |

table 1: countermeasures

## 4. The first attempts: 'Ambiophony'

Around 1959, R. Vermeulen of Philips N.V. patented one of the first active acoustic enhancement systems on the market using a tape wheel or loop with a recording head and multiple reading heads to generate multiple instances of a sound field [3].

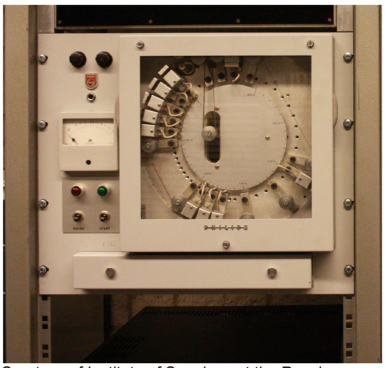
Courtesy of Institute of Sonology at the Royal Conservatoire, The Hague, Netherlands

figure 8: Philips Ambiophony tape unit (1959)

In 1975, J.C. Jaffe of Jaffe Acoustics presented a similar system - ERES - based on digital multi-tap delay lines to generate early reflctions [4]. Both systems picked up the stage sound - including the direct sound and the early reflections on stage - and repeated the reflections in appropriate patterns to construct a realistic reverberation field in the audience part of the hall. The resulting signals are played back by loudspeakers pointed to the audience, away from the stage microphones, creating enough gain before feedback to provide a stable system. Today, the results would not have satisfied the expectations, but in 1959 the results were perceived as excellent - reason why the ambiophony system was built into many halls in Europe, including the Scala in Milan.

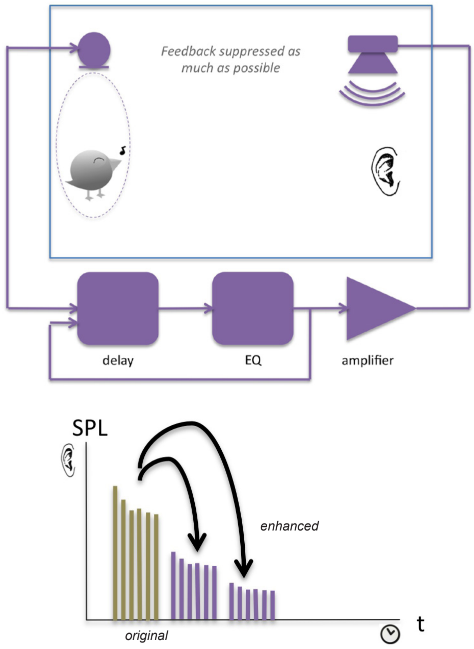

figure 9: multi-tap delay concept

## 5. Assisted Resonance

In 1964, P.H. Parkin and K. Morgan of the UK department of Scientific and industrial Research presented an experimental system installed in the Royal Festival Hall in London [5]. Although the system is not commercially available on the market, the scientific concept is so fundamental to the field of active acoustic enhancement that it is included in most publications on the subject. Basically, Parkin and Morgan acknowledged that playing back an amplified signal from a microphone in the same space results in severe colouration, or oscillation with higher amplification. Their solution was to construct multiple microphone - amplifier - speaker loops each tuned to a very narrow frequency band using microphones placed in tuned Helmholtz resonators, installed at places where the loop transfer function at that frequency was at its maximum. By adjusting gain and phase for each loop individually, the energy increase for each individual frequency range could be controlled to achieve a stable, uncoloured result with a higher energy level, and with it a longer reverberation time. Note that this system did not need to avoid feedback at all, simply because it utilized feedback as the basic principle. Although the method was very elegant and straight-forward, the minimum frequency range to be controlled was found to be just a few Hertz, so large amounts of loops were required to cover the target frequency spectrum. In the initial stage of the project, 89 loops were used to cover a frequency range from 70Hz to 340Hz. To target a full frequency range up to 8kHz, more than a thousand loops would be required, which is both physically and financially not possible for most halls. Nevertheless, the acoustic challenge in the Royal Festival Hall was the lack of 'warmth' - or energy in the low frequencies - so the Assisted Resonance system was a perfect solution. The system stayed in service for many years, it was even enhanced to include double the amount of speakers to cover up to 700Hz in a second stage of the experiment.

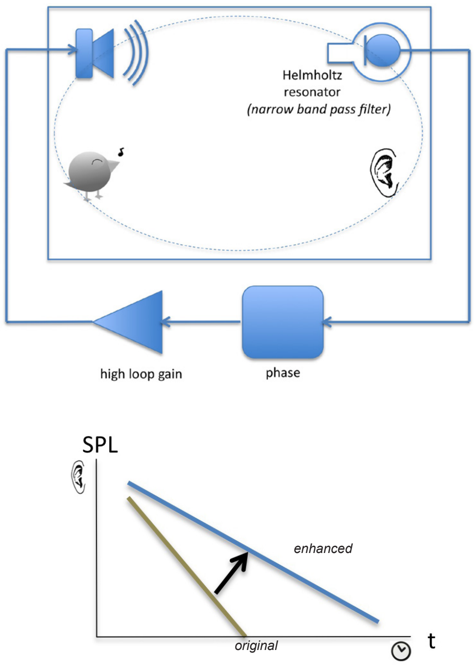


*figure 10: assisted resonance concept*

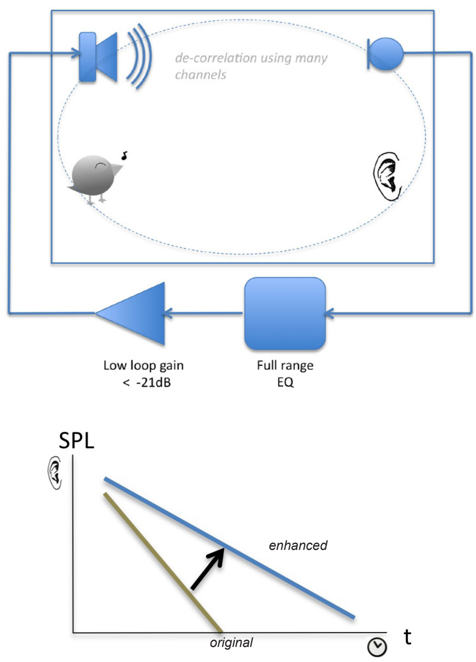


*figure 11: MCR concept*

### 6. Regenerative systems: MCR

In 1969, N.V. Franssen of Philips NV, patented the concept of 'Multi Channel Reverberation' or MCR, later developed further by S.H. de Koning [6]. The scientific concept of MCR is as fundamental and elegant as the Assisted Resonance concept, presenting a different approach to basically the same challenge: how to prevent colouration and oscillation when fitting a room with microphone - amplifier - speaker loops. Where the Assisted Resonance method uses channels with a narrow bandwidth and high gain, the MCR concept basically shows that full bandwidth channels can be used as long as the loop gain per channel stays below -21dB. Channels can be added without the risk of colouration and oscillation provided that the channels are not correlated - that is: they have independent open loop transfer functions. This can be achieved by carefully distributing the microphone/speaker loops across the hall. This means that to double the acoustic energy in a room (and with it, an increase of the reverberation time), about 100 channels are required - a lot, but still way below the amount of channels that Parkin needed for a full range solution. The MCR system has been built into many concert halls in Europe, and is now still offered by the Dutch company Event Acoustics as XLNT-MCR. The French public research organisation 'Centre Scientifique et Technique du Bâtiment' (CSTB) developed the Carmen system, an alternative way of using the MCR concept by offering integrated microphone/speaker modules to form a 'virtual wall' [7].

The advantage of regenerative systems is that they re-use ('re-generate') the existing acoustic response of the hall, sounding very natural because the system does not add artificial content to the enhanced response. This of course goes with a disadvantage: the enhancement of the response is limited to amplifying what is already there. Also, making the reverberation time longer always means that the amount of acoustic energy has to be amplified: longer means louder, and louder means longer. This constraint corresponds with the slope of the reverberation tail in figure 11 changing with increasing loop gain.

### 7. In-line systems

From 1987 to 1991, three systems were brought to the market taking a completely different approach that would break away form the regenerative 'longer is louder' constraint: ACS (1987, ACS bv, van Berkhout) [8], LARES (1988, Lexicon, D. Griesinger) [9] and SIAP (1991, SIAP bv, van Munster & Prinssen) [10]. In 2008, Stagetec brought the Vivace system to the market (Stagetec, Muller-BBM) [11]. Each system uses specially developed reverberation algorithms running on DSP hardware that became available in these years, avoiding acoustic feedback as much as possible by placing directional (cardioid, supercardioid) microphones as close as possible to the stage. Additionally, time variance is sometimes applied to modulate the reverberation algorithm delay times a little (LARES, Vivace). Although it is reported to be slightly audible in some circumstances, it suppresses feedback, avoiding colouration and instability for systems using a limited amount of independent channels. If an in-line system is installed with many independent channels, de-correlation occurs automatically, and time variance is not needed anymore (ACS, SIAP).

Assuming that in-line systems are feedback-free, any reverberation pattern can be added to the existing acoustics. If the existing acoustics are 'dry' (low energy / low reverberation time), the result is almost fully dependent on the active system, which is ideal to achieve multi-purpose usage of venues. Also, because the reverberation and early reflection patterns can be designed in detail, and directional microphones are used, powerful Early Reflections and localization features can be supported.

A disadvantage of in-line systems is that only the area covered by the directional microphones - eg. the stage - is enhanced. Sound coming from other areas - eg. from the audience - are not included unless they are equipped with their own system. It is very difficult for in-line systems to support a natural acoustic behaviour covering a complete hall.

Because feedback can never be avoided completely, in-line systems still include a slight regenerative effect. Also, when a hall already has a significant reverberation field, applying an in-line system adds the original and the in-line field together; the listener hears two fields. Both effects have to be managed carefully by the system designer to achieve a natural sound.

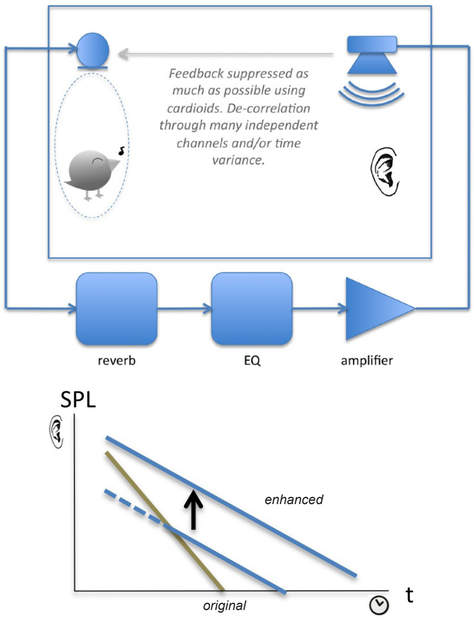


*figure 12: in-line concept*

## 8. Hybrid regenerative systems.

Two companies researched the possibility of combining regenerative and in-line concepts to achieve a system that uses a hall's existing acoustics to get a natural sound, but at the same time add artificially designed responses to have more control and to get out of the fixed energy / reverberation time constraint. The concept of enhancing a hall's acoustic response using an external acoustic space was already known and applied as an architectural 'mechanical' solution, placing a second hall adjacent or around an existing hall, opening the doors between them if a longer reverberation time was needed. A good example is the concert hall in Luzern [12].

Yamaha presented the Active Field Control system (AFC) in 1987 (Yamaha Corporation, Kawakami, Shimizu, Watanabe) [13], and LCS presented the Variable Room Acoustic System (VRAS) in 1991 (LCS, M. Poletti) [14].

Both use reverberation modules inserted into each of the system's microphone loops. AFC uses loop flattening algorithms to achieve a stable and colour-free response with just a few physical channels. VRAS (later renamed to Constellation by Meyer Sound) uses multiple digital reverberators per channel to reduce the amount of physical independent channels. Parametric equalizers can be used to further flatten the open loop gain, allowing these systems to be stable and colouration-free with less channels compared to pure regenerative systems.

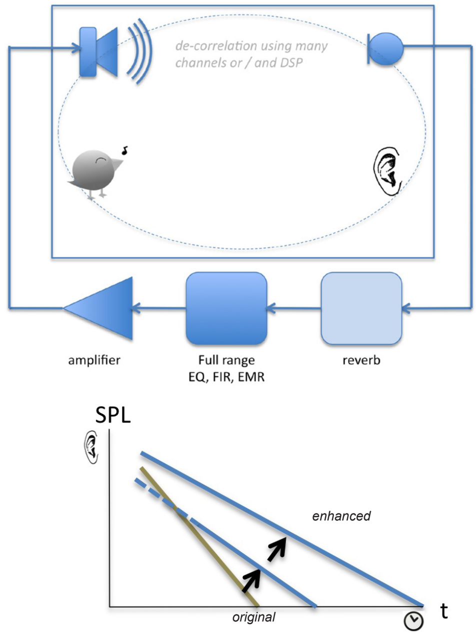


*figure 13: hybrid regenerative concept*

## 9. Modular systems.

If a regenerative approach is used then the system's microphones should be placed at or beyond the 'critical distance' of the system - the distance from the stage where direct sound energy and reverberant energy are equal. Placing microphones further away makes it easier to generate a flawless reverberation field, but makes it impossible to generate early reflections simply because of the distance between stage and microphones. Placing them closer to the stage allows for Early Reflections to be included, but it disturbs the regenerative part because the direct sound starts to play a role. In practice, the designer can achieve an appropriate and (financially) acceptable balance with one system, or decide to use two systems: one for early reflections and one for reverb. Further more, individual modules of these systems can be optimized to enhance different issues in the hall: the main reverberation field, under-balcony reverb, early reflections, side reflections, reflections on stage ('electronic stage shell') and foldback of the reverberant field to the stage.

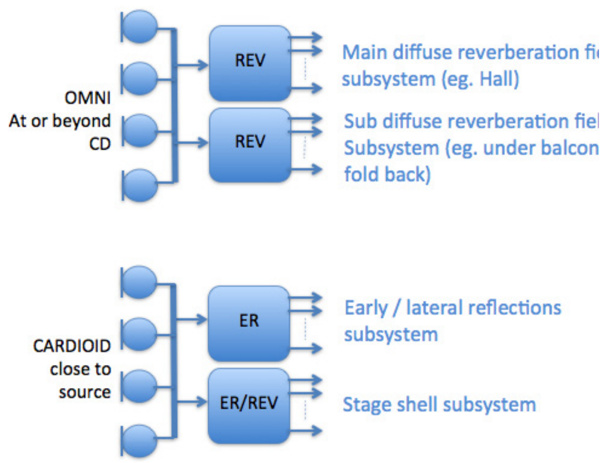


*figure 14: modular system example*

## 10. Regenerative system architectures

Figure 15 shows a pure regenerative system based on the MCR regenerative method, using many independent channels to achieve a stable and colour free system. Each channel comprises of a microphone, an equaliser, a power amplifier and a loudspeaker. For a small system with moderate enhancement about 50 channels are required. For larger enhancements (more energy, a longer reverberation time), more channels are required. An example of a pure regenerative system is the XLNT MCR system.

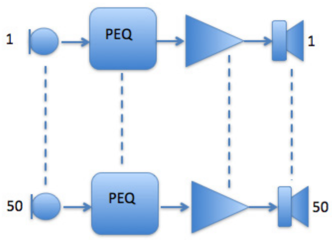


*figure 15: MCR concept using many independent channels*

Figure 16 shows a hybrid regenerative system using 16 microphones and 16 loudspeakers, constituting 16 physical channels. By applying multiple digital reverberators per channel, the amount of effective channels is increased, achieving a stable and colour free result with less physical channels than MCR. The use of reverberators in the channels allow the system to achieve more freedom in changing the acoustic response compared to using only many independent channels. An example of a hybrid regenerative system using multiple digital reverberators per channel is the LCS VRAS system.

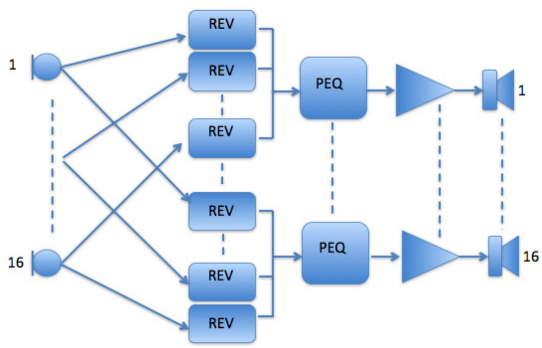


*figure 16: hybrid regenerative system using multiple reverberators per channel*

Figure 17 shows a hybrid regenerative system with 4 microphones and 16 loudspeakers. By using loop flattening algorithms [15], the open loop gain of the system's channels can be flattened to allow the use of less independent physical channels - in this case only 4. The available loop flattening alrogithms include a spatial averaging module that cross fades each system bus through the available microphones, preventing feedback energy from accumulating at peak frequencies. Also, Finite Impulse Response (FIR) filters [16] can be used to solve the highest peaks in open loop gain. Compared with the system presented in figure 16, less physical channels and less reverberation modules are used, achieving a similar result. An example of a hybrid regenerative system using loop flattening algorithms is the Yamaha AFC3 system.

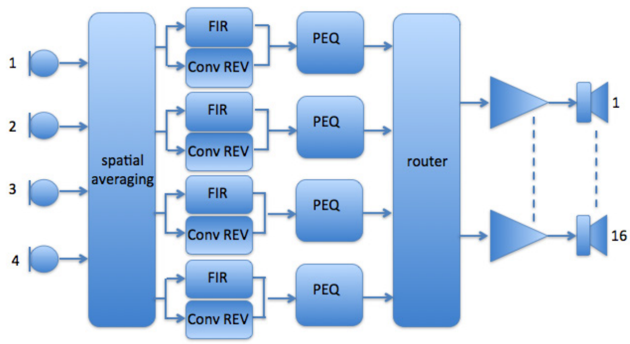


*figure 17: hybrid regenerative system using loop flattening*

## 11. Introducing Acoustic Field Control: AFC1, AFC2

The first and second generations of AFC, installed in over 70 venues world wide since 1987, use a FIR filter (AFC1 with time variance) to flatten the loop gain enough to allow the use of only 4 or 8 microphones. The reflection patterns are generated by dedicated FIR filter banks [16], while all output channels have parametric equalizers to tune the system further. The FIR algorithms had to be designed and set manually using an iterative process, with intensive communication between the users of the system (conductors, musicians) and the tuning team to achieve a good result. The Japanese tuning team spoke only Japanese and English, and not any other European language; this language constraint is the reason why the first generations of AFC were only marketed in Japan and the US.

Since 2004, Yamaha AFC systems are built using the Digital Mixing Engine (DME) series hardware platform with dedicated firmware to support AFC processes. For the first generation, AFC1, the DME32 DSP hardware architecture was used as a DSP building block. A system includes two units as a minimum, larger installations used 5 to 10 units. An example of a large AFC1 installation is the 5000 seat Tokyo International Forum hall A, initially including also the Jaffe ERES system, but later enhanced exclusively using 14 AFC1 units, 20 microphones and 197 speakers to achieve functionality for 7 subsystems [17].

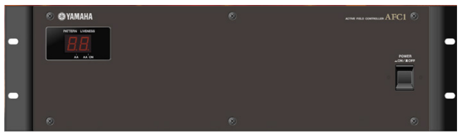


*figure 18: AFC1 unit (2004)*

In 2008, the second generation, AFC2, was presented, based on the hardware architecture of the DME64N Digital Mixing Engine launched some years earlier. As the DME64N was 4 times more powerful than the DME32, and had twice the i/o capacity, systems could be built with less AFC units. Also, the AFC2 DSP hardware could support larger FIR filters, and introduced a spatial averaging 'EMR' module to be sufficient for stabilization so time varying the FIR filter was no longer needed. An example of a compact AFC2 installation is the 490 seat auditorium of the Whitney Point Central School District, NY using two AFC2 units, 4 microphones and 28 speakers to achieve reverberation enhancement. [18].

*figure 19: AFC2 unit (2008)*

## 12. Introducing AFC3

Although the third generation AFC3 systems are based on the same DME64N hardware architecture that was the basis for AFC2, the DSP power has been significantly increased by applying an additional FIR DSP card in the unit. The MY4-AFC FIR DSP card adds four convolution FIR filters that can be inserted in the four buses to add extremely dense and natural reverberation. The algorithms are based on the database of reverberation convolution samples in Yamaha's library for the SREV1 sampling reverb. [19]. The convolution patterns are not designed for use as in-line reverbs, instead a choice of four convolution patterns is available to adjust the original acoustic response of the hall to suit the performance target. The patterns basically offer an increase or decrease of the reverberation times for the low and high frequency ranges.

*figure 20: AFC3 unit (2012)*

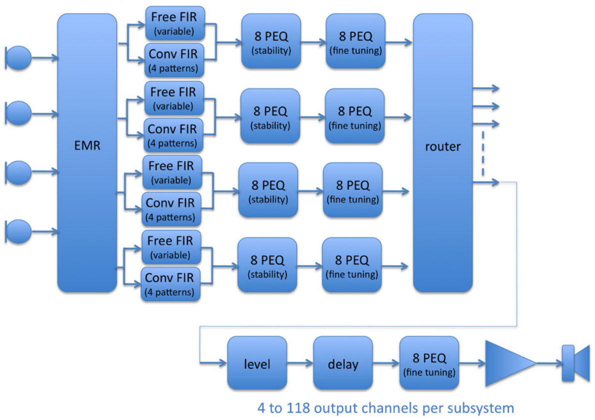


*figure 21: AFC3 REV module DSP block diagram*

Figure 21 shows the layout of an AFC3 core unit used for a reverberation module of a system. First, the EMR block takes in the four microphones (omni-directional, placed at or beyond the critical distance from the stage), producing four independent buses with spatial-averaged signals for further processing. Each bus is then split to two parallel FIR filters. The freely configurable FIR filter solves the most serious peaks left in the open loop gain, while the convolution FIR filter adjusts the original acoustic behaviour using a selection of one of the four available convolution patterns. A decay rate, initial time gap and overall level can be applied to the convolution FIR filter to match the reverberation time and level targets.

The decay is applied to the FIR filter's tap levels, so the reverberation time is still connected to the reverberation energy causing the decay setting to behave as if it is regenerative, with the overall level setting behaving as in-line effect. Each bus has an 8 band Parametric EQualiser (PEQ) dedicated to solve remaining colouration issues, with a second 8-band PEQ available for manual tuning. The four busses are routed to up to 118 outputs, with a delay and 8-band PEQ available for each output to further control energy and timing per speaker.

AFC3 systems can optionally support auxiliary inputs to provide routes to the loudspeakers for other systems, for example a mixing console using the AFC3 system's speakers for surround effects.

Since the AFC3 DSP unit has four 16-channel 'MY16' type interface slots, AFC3 system modules and i/o devices can be integrated as a networked audio system, including the support for Dante, CobraNet and EtherSound through optional interface cards. This allows for example to install remote controlled networked power amplifiers close to their speakers at distributed locations in the hall for maximum flexibility and power efficiency. Using the same network, AFC3 systems can be controlled through Yamaha control panels and third party media control systems such as AMX and Crestron, including wireless control options.

One of the main innovations of the AFC3 system is the automation of the tuning process. A software program supports automated tuning procedures for the free configurable FIR filter and the 8-band bus PEQ to achieve maximum stability, leaving the second bus PEQ and the speaker PEQ for further manual fine tuning. This allows an AFC3 system to be tuned by local Yamaha tuning engineers, speaking the same language and sharing the same musical background as the users of the system. This feature allows Yamaha to offer the AFC3 system in all areas of the world, including the European countries.

## 13. Demonstration.

To illustrate the regenerative character of a hybrid regenerative system, an AFC3 REV module was temporarily installed in the Rheinsaal of the Kolner Messe Nord during the 2012 TonMeisterTagung exhibition with one AFC3 DSP core, four DPA4060 omnidirectional microphones, one IPA8200 8x200Wrms amplifier and eight IF2108 loudspeakers. The system design, build-up and tuning was done shortly before the exhibition start in 3 hours. Using a 'voice lift' preset (basically adding appr. 2 dB Early Reflections energy to the room) during the presentation, two reverberation field presets were recalled at the end of the presentation, with a trumpet player walking around the audience whilst playing. This short demonstration illustrated that the small system achieved a good result, and that the room behaved completely naturally, with a diffuse reverberation response to all positions in the room.

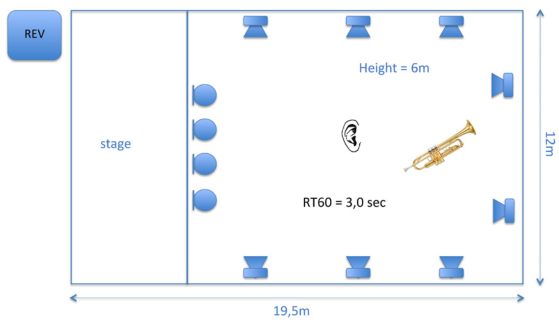


*figure 22: AFC3 REV module at the Rheinsaal R4*

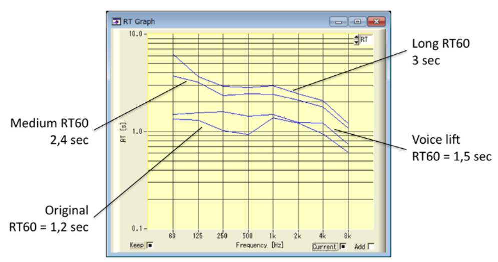


*figure 23: tuning result ($RT_{60}$, empty room)*

## 14. Conclusion.

The increasing DSP power available in today's audio industry (2012) allows the use of powerful algorithms that make it possible to combine in-line and regenerative reverberation enhancement concepts into a hybrid regenerative system with less independent channels compared to pure regenerative systems, offering increased functionality, a higher audio quality and lower implementation cost than before. Where active acoustic enhancement systems in the past were expensive and often could only be designed and implemented by the manufacturer, this development will drive the cost level downwards, and support the design and implementation by contractors, system integrators and acousticians - broadening the market scope to include also medium and small scale venues looking for variable regenerative acoustics at an affordable cost level.

## 15. Disclaimer.

Detailed technical information about the market's commercially available Active Acoustic Enhancement Systems is sometimes not easy to find. Many references can be found, but they are not always consistent. If any erroneous statement is found in this paper, please let the author of this paper know so it can be corrected in a future revision.

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