Acoustic Control Systems (ACS) and the science and art of acoustics.

Based on fundamental research and knowledge of how acoustics and the human ear intertwine, ACS together with the Delft University of Technology develops systems that can chance the acoustics of a hall. Natural sound is our starting point. An ACS-system can truly enhance acoustics and make it variable.

To be able to generate real acoustics ACS-systems uses the following principles:
- Multiple microphones to record the sound sources
- Multi-channel matrices for sound processing
- Separate control over earlier and later reflections (reverberation)
- Every loudspeaker getting its own correct signal, together building the required sound field
- Sound processing with essential knowledge about how sound waves are travelling within a specific environment, are is being influenced by different matter (wood/stone/curtains/air) and are interacting with other sound waves

An ACS system is - certainly not - comparable with a collection of modified studio reverbs. The system generates true and natural sounding acoustics, using specially developed and state of the art digital techniques. All built together in a modular structure that is very easy to control and service.
The system architecture as well as a balanced procedure for tuning of the system makes sure this only takes a very limited amount of time.
Natural sound, high dynamics, high signal density, high system stability and very low noise that's ACS.

Why we use the multi channel architecture described above:
Fact is that in nature sound within a venue is transferred in many, many directions. One can say acoustics is truly "multi-channel". This can of course only be simulated with a larger number of channels.

Multiple microphones therefore provide the input of our system. Typically 12-36 are used.
In most theatres they will be arranged in one or more arrays near the stage to provide a high density pick up of sound. In some situations, for example in churches microphones will (also) be placed within a defined pattern in the hall.
Depending on where sound originates microphones will pick up signals with different timing and sound levels. Use of directional microphones (e.g. super cardioid) will further improve the directional information and allow the system to generate sound reflection patterns, as in nature, depending on where the sound originates.

Using a limited amount of microphones (say 4) will give insufficient directional information to be able to generate a natural propagating sound field. Also large phantom sources will emerge. Similar to when you listen to a stereo set and it feels that the sound source is somewhere in between your loudspeakers. The negative effect of this in is that a sound source (for example a violin player) will not be contained to its original position but be everywhere in the hall (become larger than life). If you close your eyes then it will also become very difficult to pinpoint where the sound is coming from.
This problem also occurs when microphones are spaced further apart throughout a hall (distributed microphones). Another problem of distributing microphones is that background noise, sound of air handling systems etc. become easily picked up and transferred throughout the whole space.
The distance between microphones and sources should not be too large. This way the travel time of sound waves from the source to the microphones is be kept relatively short and the system can generate early reflections (see below). In some venues (like churches) this may be of lesser importance and microphones can be placed a bit further away.

When microphones are placed further away form the stage, in a hall with a typical theatre set-up (a stage in front of people seated in the hall) it is impossible to generate and render correctly timed early reflections at the required locations. (Sidewalls near the stage, forestage reflector.) Simply because the sound will arrive too late at the microphone.
There are systems in the market that are based on the old fashioned principle of acoustic feedback. Microphones are placed deeper in the hall, recording the reverberation, loudspeakers will render the reverberation, which is again recorded. Like making a picture of a picture and again, the result deteriorates Systems like these are known to be very hard to tune and, also because easily become unstable, the increase reverberation time is often very limited

To further simulate nature, **multi-channel matrices** process the input signals. Using many channels and microphones will result in a system that has a very high gain before feedback. This is further improved by the multi channel architecture that allows the signals to be de-correlated. All resulting in a very stable system.

Each of the connected loudspeakers gets its own and correctly timed signal, and the rendered signals together will build the sound field necessary for the required change of acoustics.
Depending on the place and function of a loudspeaker the type (e.g. dispersion pattern) and amplifier power necessary will be determined.

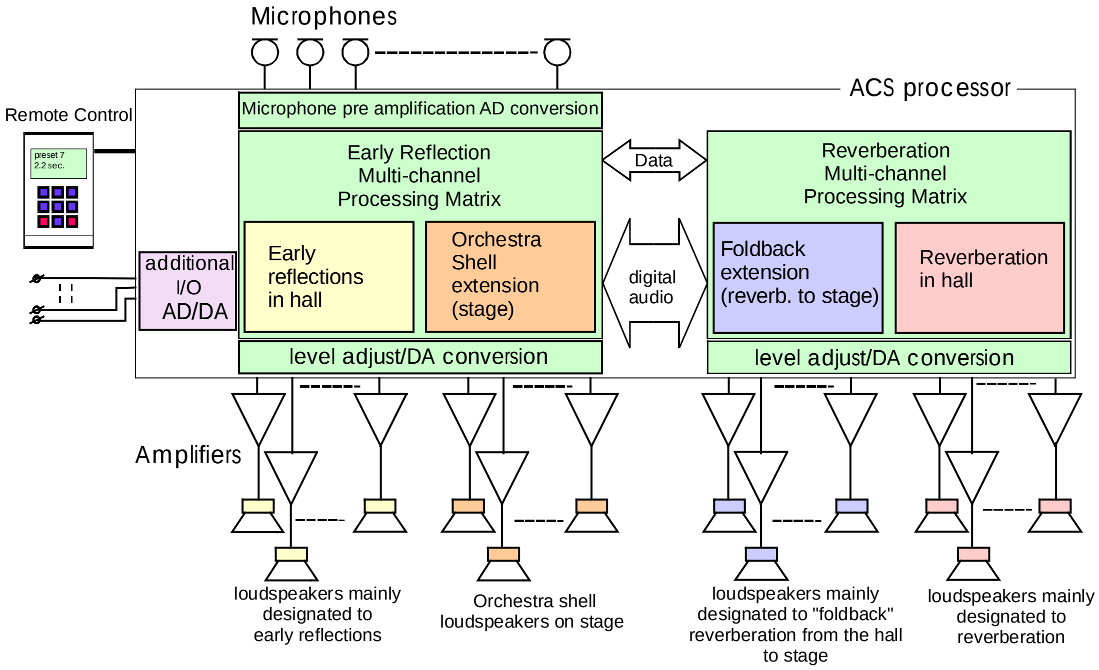
Using only a limited number of channels and microphones means that output signals have to be derived from these few inputs and these similar output signals are brought out to many position throughout a venue, this generally results in a woolly blurred and unnatural sound.

The system distinguishes between earlier and later energy (**early reflections and reverberation**)
Early reflections are the first reflections from ceilings / walls etc. they will give a listener information about the size of a space and improve clarity of sound.
Later re-reflected sound, or reverberation, will of course add the reverberant sound that is so very important for many types of musical performances, but it will not improve clarity.
(Think about speech in the long reverberation of a cathedral. One can only speak slowly to be intelligible.)
This is why earlier and later reflections can be controlled separately in amounts, levels, timing, frequency spectrums etc..
Also the place in the hall where earlier and later sound should be rendered is often different.
In many halls you see a forestage reflector, reflecting sound from the stage (early reflections) towards the listeners in the hall (improving clarity) alternatively this might be done with a number of loudspeakers forming a virtual forestage reflector.
Also ACS supplies the so-called orchestra extension, an array of loudspeakers surrounding the performers on stage radiating early energy. They form a virtual orchestra shell, as if there where walls surrounding them. Musicians can better hear each other, thus the ensemble playing conditions are improved. Generally it is not ideal to also render reverberation from these loudspeakers. ACS uses the foldback extension to render reverberation in the stage area, the loudspeakers for that are placed somewhat further away from the performers.
The exact control of the timing and levels of earlier and later en later energy on each of the loudspeakers has another advantage and this is that it can be adjusted such that it becomes impossible to localise that sound generated is coming from loudspeakers.

The ACS system generates acoustics based on **knowledge about how sound waves are travelling within a hall.** During its “lifetime” a sound wave is being (partly) reflected / absorbed by different matter like wood, stone, chairs, audience and curtains it will travel through air and interact with other sound waves. This all is be taken into account while processing.

Very often virtual acoustics are (for a large part) judged by the fact that the reverberation tail looks good on the display of the acoustics analyser. Although this may look good, it is not uncommon the sound is unrealistic and/or static. Making a reverberation tail that sounds natural and looks good on the display of the acoustics analyser that is the art.

ACS, Acoustics with sound natural, without digital artefacts, with good definition, experienced as real.



Typical system set-up