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BACKGROUNDS AND PRINCIPLES OF THE DELFT ACOUSTICAL CONTROL SYSTEM (ACS)

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1. INTRODUCTION

Unfortunately, many auditoria have been built and are still being built with unsatisfactory acoustics for music performances. In particular, many auditoria turn out to be "too dry", meaning that not enough reverberation is added to the music. Pronounced examples are modern churches and outdoor music pavillions, but also many music halls suffer from a severe lack of reverberation in the hall as well as on the stage.
The reasons that nowadays auditoria are still being built with unsatisfactory acoustics are two-fold:

1. Multi-purpose function
   The hall is used for significantly different purposes such as lectures, pop concerts, musicals, symphony concerts; these activities are acoustically difficult to combine in one hall.
2. Creative architecture
   The hall is built by an architect with the ambition to give a personal touch to the architectural design; unfortunately practical experience has taught that interesting designs do not always support good acoustics.

In conclusion, acoustical requirements may set a severe limit to the use of the hall and the creativity of the architect. Needless to say that any solution to circumvent both limitations would be wholeheartedly welcomed.

Traditional architectural acoustics recommand physical solutions which are often very costly or even prohibited for aesthetic or historic preservation reasons. An attractive alternative is the application of electronic solutions, which show the following basic advantages:

1. No physical changes

   The physical architecture is not touched except of the provision of microphones and loudspeakers; corrections of the sound field are made fully electronically ("electronic sound control").

2. Flexibility

   The setting of the electronic parameters can be optimally adjusted for each performance ("variable acoustics").

3. Low cost

   Looking at the present day prices of modern electronic components, electronic solutions for reverberant field enhancement are generally considerably less costly than physical changes in the architecture.

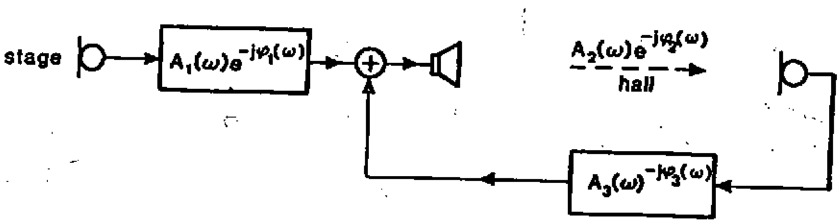


Figure 1: Principle of electronic reverberation enhancement systems based on acoustical feedback (single channel version); the stage microphone is optional. Note that for those systems the acoustic communication between hall microphones and hall loudspeakers is essential.

Since the sixties, a number of electronic solutions have been proposed aiming at optimization of reverberation times and impulse responses in enclosures. On one hand, there are the systems designed especially for the enhancement of reverberation time and level in auditoria (AIRO, Philips). Basically, those systems are built up from a (usually large) number of broadband and/or narrow band channels, each consisting of a microphone, amplifier/delay system and a loudspeaker, see figure 1. Reverberation enhancement is based on acoustical feedback, so that the loop gain is determined not only by the electronic transfer function $A_3(\omega)$ but also by the acoustic transfer function $A_2(\omega)$ between loudspeaker and hall microphone. Unfortunately, $A_2(\omega)$ is influenced by geometric variations in the hall (e.g., due to the entry of an audience) as well as temperature variations, so that the danger of colouration or even hawlback becomes apparent for such systems. These undesired effects can be avoided only by reducing $A_3(\omega)$ to a 'safe' level, which limits the effectiveness and flexibility of the system considerably.

On the other hand, there are the systems aiming at optimizing the impulse response at a listener position by adding synthetic reflection patterns radiated from loudspeakers placed at various positions. Nowadays small-scale (i.e. 4 to 6 channel) systems based on this principle are available for domestic use in order to overrule the natural acoustics of the listening room by acoustical conditions better fitting the music to be reproduced (Roland, Yamaha, etc.). Here, the contribution of each loudspeaker channel is determined by an electronic delay line, see figure 2.

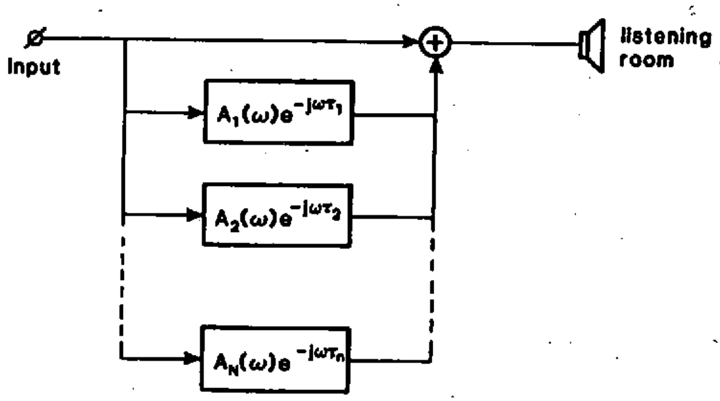


Figure 2: Principle of electronic impulse response simulation systems (single channel version). The transfer function $A_n(\omega)\exp(-j\omega\tau_n)$ defines the property of one simulated reflection path.

As long as the input of the system is electrical (e.g., CD or record signal), colouration and hawlback effects cannot occur. If the system is used to process the signal of a sound source within the room using a directive microphone as the input device, acoustical feedback can be reduced to a safe level without limiting its performance, since feedback is not the basis of the system.

The Acoustical Control System (ACS), as developed in the Netherlands by Electronica Griffioen B.V. in close cooperation with the Delft University of Technology, aims at optimization of the acoustical conditions of multi-functional auditoria and studios in a most flexible way. Its philosophy differs fundamentally from the above mentioned concepts. With ACS first and higher order reflections are simulated for a desired hall (size, shape and absorption coefficients should be specified), exactly according to the laws of physical acoustics. The result is transmitted into the real hall by a distribution of loudspeakers such that, temporally and spatially, the reflection pattern generated in the real hall is the same as the reflection pattern in the desired hall (figure 3).

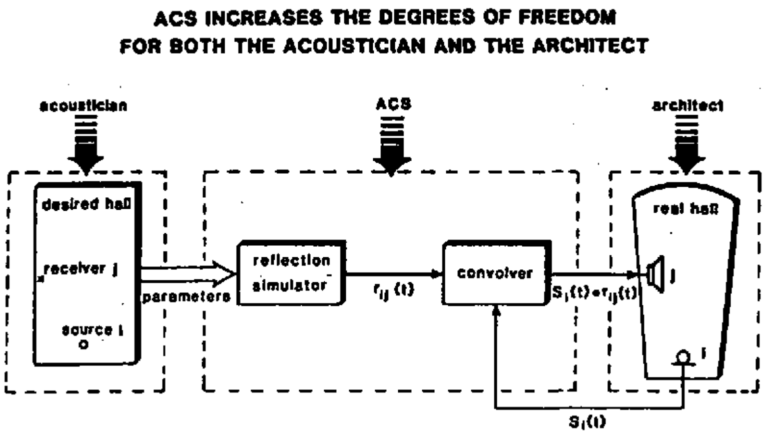


Figure 3: Principle of ACS for one microphone-loudspeaker pair. The central processor simulates reflections in a desired hall, the resulting reflection pattern is convolved with the microphone signal and the convolution result is transmitted into the real hall via the loudspeaker. Note that in this simplified picture the feedback from loudspeaker j to microphone i has been neglected.

It is important to realize that the impulse response of one loudspeaker depends on its mounting position. Hence all loudspeakers transmit different impulse responses. Therefore, ACS is in principle multi-channel. The resulting impulse responses in the real hall determine the reverberant field at each listener position. Note that in the ACS concept both architect and acoustician have maximum degrees of freedom in doing their job!

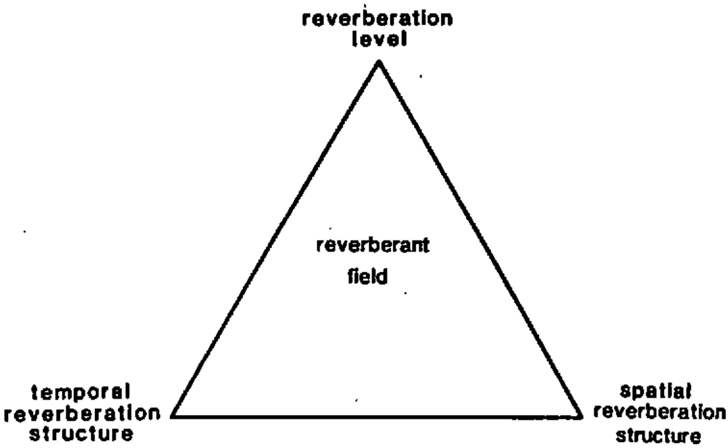


Figure 4: Three important properties of the reverberant field.

Due to the wave theory based impulse response approach, ACS enables independent control of the three properties which together describe the quality of the reverberant field (see figure 4):

1) <u>temporal reverberation structure</u>,
   determining early-to-late energy ratio criteria (definition D, clarity C_{80}, etc.) and decay parameters (early decay time T_{10}, reverberation time T_{60});

2) <u>reverberation level</u>,
   i.e., the 'energetic scaling' of the impulse response pattern with respect to the direct field;

3) spatial reverberation structure,
determining criteria related to spatial perception (early lateral energy fraction ELEF, interaural cross correlation IACC, etc.).

Note that with the ACS principle the above three properties can be optimized for the entire hall and not for a few positions.

2. DESCRIPTION OF ACS

2.1. The ACS architecture

The ACS may be subdivided in three subsystems (figure 5):

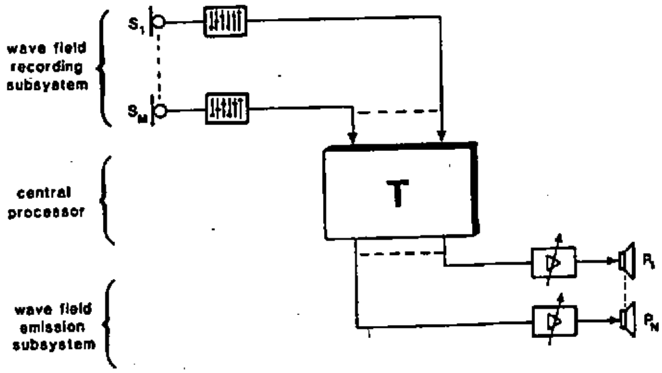


Figure 5: With ACS the source wave field is picked up by a matrix of microphones, wave theory based reflections are simulated by the central processor **T** and a spatially oriented reflection pattern is generated in the hall by a network of loudspeakers.

1. Wave field recording subsystem
2. Central processor
3. Wave field emission subsystem.

The wave field recording subsystem measures the sound field at the stage and in the hall with high quality broad-band microphones. The microphone signals are optionally equalized and sent to the central processor. The central processor consists of a number of reflection simulation units (see section 2.2). The wave field emission subsystem transmits the simulated reflections back into the hall by high quality broad-band loudspeakers. In ACS broad-band loudspeakers will be distributed over the entire hall. It is important to realize that at a given position in the hall the reflection tail is not made by one loudspeaker but it is synthesized by contributions of all loudspeakers: ACS is fundamentally multi-channel.

2.2. The central processor

If the M input signals of the central processor are represented by input vector $\bar{S}$ and the N output signals are represented by output vector $\bar{P}$, then input and output are related by transfer matrix $\mathbf{T}$:

$$\bar{P} = \mathbf{T}\bar{S}, \qquad (1)$$

where

$$\mathbf{T} = \begin{pmatrix} t_{11} & t_{12} & \cdots\cdots\cdots & t_{1N} \\ t_{21} & t_{22} & \cdots\cdots\cdots & t_{2N} \\ & & & \\ & & & \\ t_{M1} & t_{M2} & \cdots\cdots\cdots & t_{MN} \end{pmatrix}. \qquad (2)$$

Matrix element t_{mn} is specified by the transfer function between microphone m and loudspeaker n, to be written as

$$t_{mn}(\omega) = A_{mn}(\omega)e^{-j\phi_{mn}}. \qquad (3)$$

In ACS, the matrix elements are usually determined in octave bands, using the concept illustrated in figure 3.

The central processor consists of a number of reflection simulation units, each generating one delayed signal between each microphone-loudspeaker pair (figure 6), so that for one r.s.u. we write

$$\phi_{mn} = \omega\tau_{mn}. \qquad (4)$$

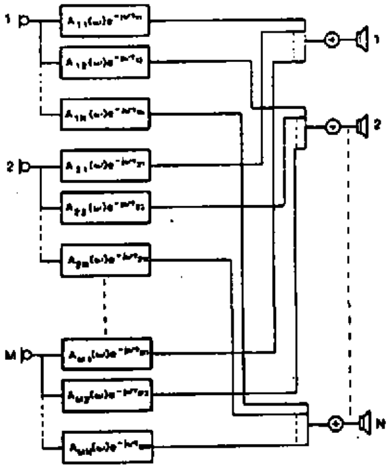


Figure 6: Schematic configuration of one ACS reflection simulation unit.

The subsequent units are coupled through an internal bus connection (figure 7). Different units can be used for, e.g.:

- early reflections on stage;
- early lateral reflections in the hall;
- n-th order reflections in the hall, the maximum value of n depending on the maximum T_{60} required.

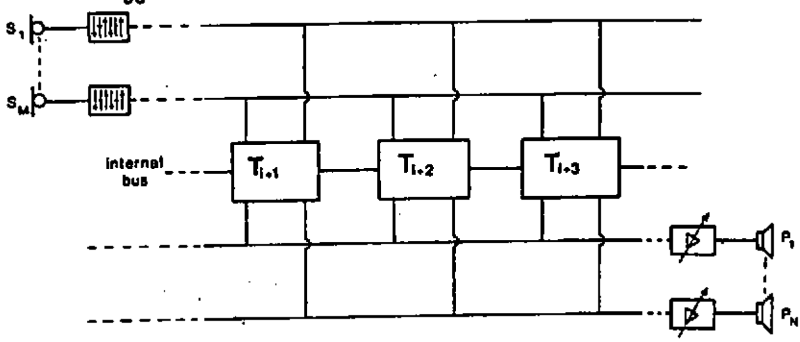


Figure 7: Internal coupling of subsequent reflection simulation units.

3. ACS AND THE FEEDBACK PHENOMENON

In reverberation enhancement systems based on feedback (acoustic communication between hall microphones and hall loudspeakers, figure 1) the loop gain $A_2(\omega)A_3(\omega)$ is the key function. If $A_2(\omega)A_3(\omega)$ is too low, then the system has no noticeable effect. If $A_2(\omega)A_3(\omega)$ is too high then colouration or even hawlback may occur. As the electronic impulse simulation systems (figure 2), ACS is not based on feedback and, therefore, the smaller the loop gain the better. This is achieved in four ways:

a. A significant amount of microphones are positioned near the stage to measure the direct field; a relatively small part of the reverberant field is picked up. The gain of the hall microphones is set to a safe level.

b. Directive microphones are used.

c. Directive loudspeakers are used; the main lobe of each loudspeaker is turned towards the audience.

d. The matrix elements of the central processor are time variant.

Experience with ACS shows that very large reverberation enhancements can be achieved without any sign of colouration.

4. THE ACS DESIGN AND TUNING PROCEDURE

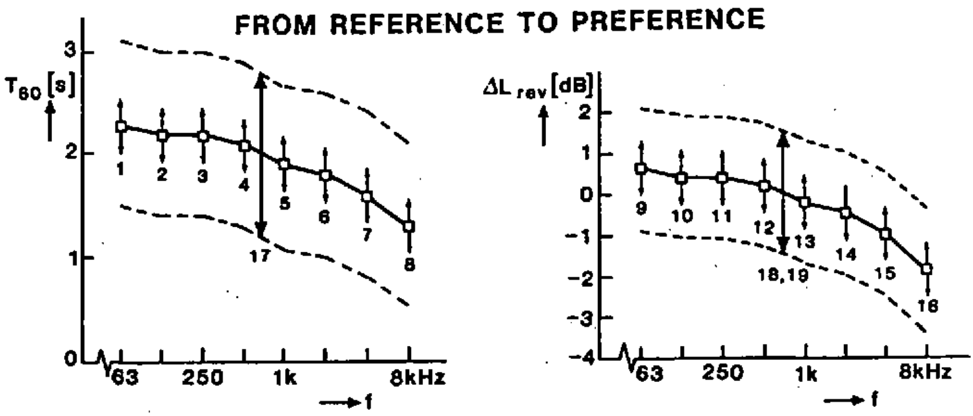
Design, manufacturing, installation and tuning of ACS are done in the following steps:

1. Analysis of the acoustical conditions of the existing or designed hall.
2. Definition of the acoustical conditions of the desired hall. (For a multi-functional hall: desired variations around a reference condition).
3. Determination of microphone/loudspeaker positions and the central processor transfer matrix $\mathbf{T}$; selection of microphones and loudspeakers.
4. Manufacturing and preprogramming of the system (in factory).
5. Installation of the system in the hall.
6. Fine tuning of the system in situ, so that the desired (reference) conditions are really satisfied ("calibration").
7. Adjustment of several preference settings for different hall functions by varying 19 system parameters (see below), starting from the reference setting ("from reference to preference").
8. Storage of preference settings in a memory, to be recalled by a push-button device.

Starting from the reference setting, that is stored in the ACS memory, the preference settings can be adjusted by varying 19 ACS fine-tuning parameters, labeled 1 - 19:

* 1 - 8: the individual reverberation time values in the eight octave bands from 63 Hz up to 8 kHz;
* 9 - 16: the individual pressure levels in the same octave bands;
* 17 : the scaling factor for all reverberation times;
* 18 : the input amplification of all microphones;
* 19 : the output amplification of all loudspeakers;

The principle is illustrated in figure 8, where the characteristics of the Amsterdam Concertgebouw are chosen as the reference.



(a) (b)

Figure 8: The effect of changing the ACS fine-tuning parameters 1 - 19.

Each fine-tuning parameter can be varied by means of a 16-step selection switch.

5. ACS MODULES AND MODELS

Several ACS models have been developed, consisting of different ACS modules. An ACS module is an independent system as described in the previous sections, dedicated to a specific function. Modules can be coupled acoustically and/or electrically if desired. The different modules available are:

1. Hall reverberation module:
   The hall reverberation module generates reverberation in the hall by at least 12 I/O ports. The decay rates and pressure levels can be set per octave band.
2. Stage reverberation module:
   The stage reverberation module generates short-delay reflections on the stage by at least 6 I/O ports. The decay rates and pressure levels can be set per octave band.
3. Speech module:
   The speech module generates first-order reflections into the hall by at least twelve output ports, using one or more PA microphones.
4. Theatre module:
   The theatre module generates first-order reflections into the hall by at least twelve output ports, using a matrix of directive microphones above the stage area.

The ACS configuration as installed in the auditorium of the Delft University contains all four modules. It is coupled with an IBM computer workstation. Besides for system parameter control, this workstation is used as an important tool in the research program around ACS. To emphasize the research function of the Delft system, it has been baptized DARS, standing for Delft Acoustical Research System.

6. REVERBERATION CHARACTERISTICS OF THE DELFT SYSTEM (DARS)

The lower curve in figure 9 shows the natural reverberation characteristic of the auditorium of the Delft University, having a T_{60}-value for the mid-frequencies of about 1.2 seconds. In this conditions, the hall can well be used for lectures and meetings, provided that the speech level is enhanced by a PA system. Using the ACS speech module for this purpose, speech intelligibility is satisfactory at all seats: Speech Transmission Index (STI) values are 0.60 or higher.

Reverberation times and reverberant sound level values of the Amsterdam Concertgebouw were adopted as criteria for the reference setting. From this setting, several preliminary preference settings have been derived, mainly based on listening tests by experienced listeners (musicians, acousticians). At first stage, reverberation characteristics for a 'normal' and a 'long' music setting were chosen as specified by the full lines in figure 9. The sound was judged as 'very brilliant' by some listeners, as 'too sharp' by others. Therefore, it was decided to drop T_{60}-values and levels for higher frequencies as shown by dotted lines in figure 9. These modified settings were preferred by almost all listeners. Since then, the modified 'normal' setting is used during most orchestral concerts, the modified 'long' setting for large choir performances.

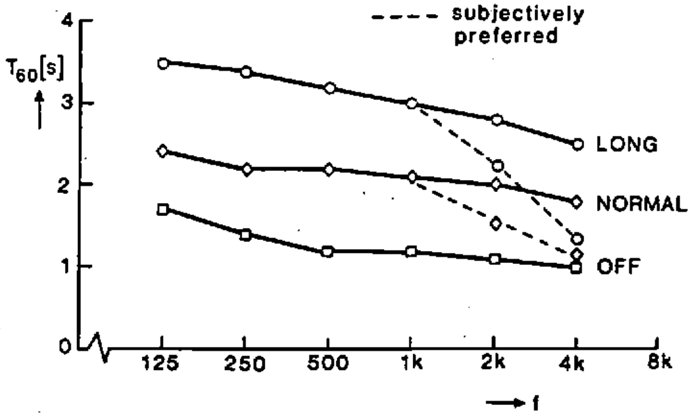


Figure 9: Reverberation characteristics in use at the Delft Auditorium.

Figure 10 shows the decay curves for the 500 Hz octave bands at the three settings given above measured well beyond the critical radius. It is seen that double decay shapes do not occur, which means that the reverberant levels of ACS are high enough to fully overrule the natural decays. Level measurements show that the levels are a few dB lower than the levels which would correspond with the T_{60}-values introduced by ACS. This appears to be in favour of speech intelligibility as well as musical definition.

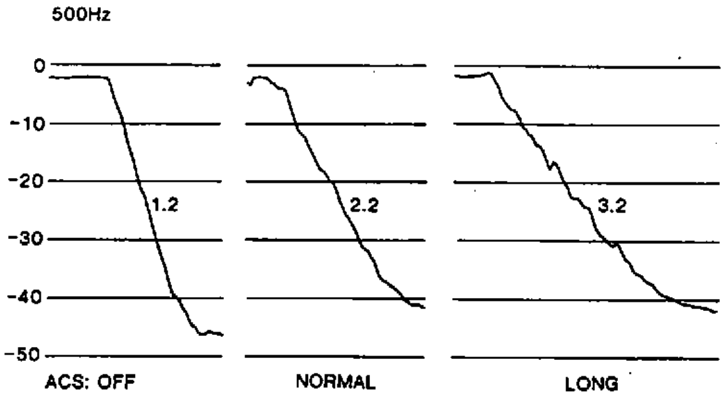


Figure 10: Three decay curves for 500 Hz at the Delft auditorium.

7. RESEARCH PROGRAM AROUND DARS

The following research projects related to ACS have been started in Delft, making extensive use of the DARS facility:

1. Development of algorithms for determination of the transfer matrix elements of the ACS central processor, taking the acoustical conditions of the real and the desired halls into account.
2. Exploration of new applications of the existing modules and reflection simulation units; development of improved versions.
3. Experiments on acoustical perception, offering test persons a fully controlled acoustic field under non-laboratory conditions.
4. Development of a self-adjustment procedure for ACS, given the desired acoustical parameters as input data.

8. CONCLUSIONS

The Acoustical Control System ACS aims at increasing the degrees of freedom for both the architect and the acoustician. With ACS first and higher order reflections are simulated for a desired hall according to the laws of physical acoustics. The result is transmitted into the real hall by a distribution of loudspeakers such that, temporally and spatially, the reflection pattern generated in the real hall is the same as the reflection pattern in the desired hall. ACS is fundamentally multi-channel: at any

position in the real hall the reflection tail is not made by one loudspeaker but it is synthesized by contributions of all loudspeakers.
For the design of a specific system the acoustical properties the desired hall play an essential role.

After installation a reference setting is realized by carrying out interactive measurements such that in each octave band T_{60} values and sound pressure levels are according to user specification. Starting from this reference setting, preference settings can be adjusted according to different hall functions. The preference settings are stored in ACS and can be recalled by a simple push button selector.

Experience with ACS shows that the variability of the system guarantees the realization of preferred reverberation characteristics as well as the realization of instantaneous variable acoustics.

The ACS configuration in the auditorium of the Delft University not only has made this hall multi-functional, but is also used as a tool for acoustical research activities which were not realizable until now.