ELECTRO-ACOUSTIC SYSTEMS FOR THE NEW OPERA HOUSE IN OSLO

Alf Berntson

Artifon AB
Östra Hamngatan 52, 411 08 Göteborg, Sweden
alf@artifon.se

ABSTRACT

In this paper the requirements and design of the sound reinforcement system for the large hall in the new opera house in Oslo is presented. Much effort has been laid on variable acoustics in adjusting the room for amplified performances and minimizing problems with disturbing echoes. The reinforcement system has been designed for highest level of fidelity. This includes the possibility of preserving correct localisation of sound sources on stage for most of the audience. Furthermore, almost all of the loudspeakers will be hidden which helps the audience in their belief in the "illusion" on stage.

1. INTRODUCTION

The new Opera House in Oslo is presently being constructed, and will be inaugurated in 2008. The house will have a large hall with 1350 seats and a flexible small hall with 400 seats, both with variable acoustics and electro-acoustic equipment for the highest level of fidelity. There will be many sound systems in the house such as sound reinforcement systems in the large hall, small hall, large rehearsal hall and foyer, intercom systems, fold back systems, sound effect play-back systems, sound distribution systems for calling and voice alarm, AV-systems in lecture halls, play-back systems in small rooms, recording and sound effect production systems capable of 7.1 surround which can be adjusted to the loudspeaker system in the main hall. The sound system for reinforcement and production will be fully digital with distribution of sound and control mainly on fibre. In this paper we will concentrate on the fixed loudspeaker system for the audience in the large hall.

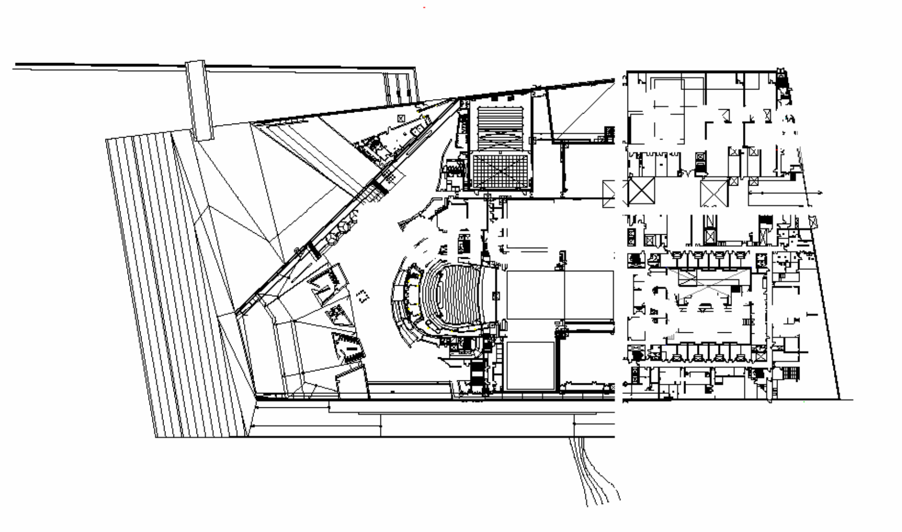

Figure 1. *Plan of the first floor of the opera house. The foyer is to the left, the large and small hall in the middle and BOH to the right (rehearsal rooms, workshops, administration etc).*

2. FUNCTIONS

The sound reinforcement systems for the audience and the variable acoustics in the large hall are designed for high quality amplification and play-back in operas, operettas and musicals. The usage in the standard repertoire operas will be very limited while musicals will demand all the more advanced functions.

The sound reinforcement system for the audience in the large hall is designed for the following functions mainly:

- Source oriented reinforcement (SOR) of artists on the stage i.e. amplifying sound while localising it back to its source.
- Natural sounding reinforcement ("inaudible PA") for speech (and sometimes singing) in operettas etc.
- Reinforcement of orchestra and instruments.
- Play-back of recorded music.
- Live reproduction of choir from other rooms.
- Sound effects from mobile loudspeakers on stage.
- Advanced surround effects in the hall.

3. REQUIREMENTS

3.1. Variable acoustics

Variable acoustics is very important for adjusting a hall for different type of performances. For loudspeakers, when often higher levels are used, avoiding disturbing echoes is at least as important as low reverberation time. Reflections back to stage good for singer support can cause disturbing echoes from high level loudspeakers. Furthermore, the surfaces near the loudspeakers shall not cause comb filtering.

The performances in this house will be acoustic opera without sound reinforcement, operettas and musicals. Operas by Wagner, Richard Strauss etc will use the longest reverberation time and modern rock operas and rock musicals will require the shortest. In the final design the mid-frequency reverberation time will be passively adjustable between approximately 1.2 s to 1.6-1.7 s.

3.2. Loudspeaker systems

3.2.1. Localisation

In theatres today it is common that the basic requirements for "good sound" such as low noise, low distortion, feedback prevention, adequate sound pressure level and coverage are fulfilled. However, the localisation of the sound sources (usually the singers) on stage is often wrong. This leads to

- reduced intelligibility and transparency
- difficulty to differentiate between different actors dialogue
- confusion and increased listener stress due to loss of correlation between visual and auditory perspective
- undermining of the audience's suspension of disbelief

Wrong auditory localisation means that the hearing gives wrong impression of direction and/or distance compared with the reality (sight). Since the actors are moving on stage (mostly) the sound system must be able to create multiple virtual sources which can change positions independently of each other. Moreover, the localisation should work for most of the spectators not only in a "hot spot" in the middle.

The auditory localisation is mainly tied to our binaural abilities and the precedence effect. In the free field the localisation in the horizontal plane is dependent on the inter-aural level, phase and time differences. In the median plane the pinnae (outer ear) give spectral cues. Head movements are important for resolving ambiguities between front, back and up. In real rooms with reflections the precedence effect (also called the “law of the first wave front”) gives us the possibility to perceive the direction of the source. We hear a sound as coming from the direction from which it first reaches us. The distance to the source is mainly determined by the relation between direct and reverberant sound level. That’s why highly directive loudspeakers often give an impression of an unnatural close sound. High directivity loudspeakers, such as line-arrays, are often used to get an even coverage and high intelligibility. However, a too even coverage is not the optimum for a natural sound image. Therefore, these types of loudspeakers can be difficult to use in a SOR system. This leads to a trade off between the requirements of intelligibility, coverage and localisation.

In performances of rock musicals etc, when high levels are used, it is no point in trying to achieve source localisation since the sound image anyway is unnatural. Therefore, a “high level system mode” was defined for which the system shall be adjusted for even coverage.

3.2.2. Intelligibility, maximum sound pressure level and coverage

Of course, requirements for intelligibility (STI) [1], maximum sound pressure level and sound pressure level coverage were set. There are some standards on max SPL but there is no one widely used internationally. Therefore, a clear definition was developed for this project. One attempt in standardising is the so called “Common Loudspeaker Format” [2] which may be the future standard for presenting loudspeaker data. In this context the NORDTEST method NT ACOU 108 [3] should be mentioned. This method is one of very few standards for validating system requirements in situ.

3.2.3. Maximum noise level

Requirement for electronic noise from the loudspeakers were set to be maximum 14 dBA / PNC-9 by the acoustic consultant. It is very important to define the prerequisites for this type of requirement. We therefore clearly defined the:

- frequency range
- percentage of the area of coverage
- number of channels routed
- source impedance
- groups of loudspeakers used
- gain
- acoustical conditions

4. CALCULATIONS

4.1. Variable acoustics

During the planning the acoustic consultant used ODEON [1] and the electro-acoustic consultant used CATT-Acoustics [1] for computer calculations. The ODEON model was converted to CATT format and verified by calculating room acoustical parameters in 36 positions spatially averaged. The differences between the two programs were very small (within subjective difference limen of appr. 5% for RT, EDT and D_{50}) except in the low frequency region (octave bands 125 Hz and 250 Hz) where the reverberation time etc predicted by ODEON was much longer. Of course, geometrical acoustics has a limited validity in the low frequency region but this holds for both the programs.

In the first phase of the planning four cases were analyzed in order to show the influence of different surface diffusion modelling.

- Estimated frequency dependent edge and surface diffusion
- Estimated frequency dependent surface diffusion only
- Frequency independent surface diffusion as used in the ODEON model
- No diffusion

An example of the results for the reverberation time T_{30} without variable absorbents is shown in figure 2.

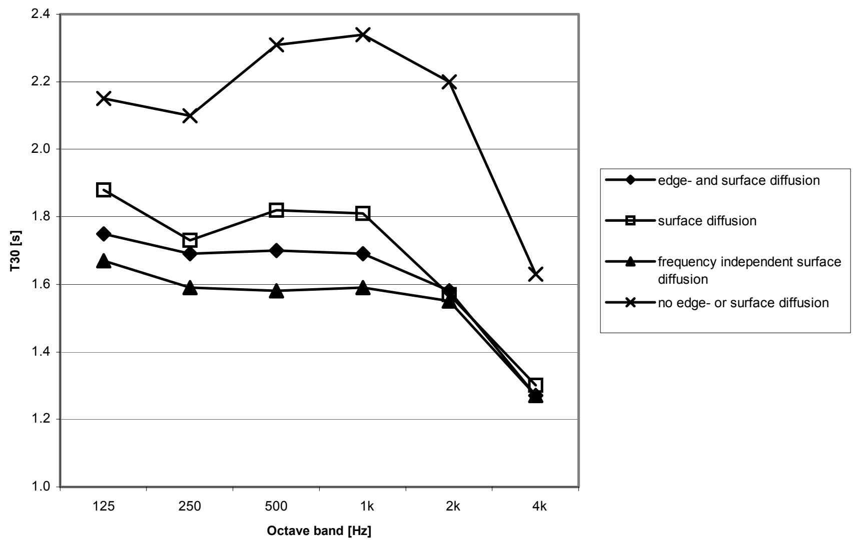

Figure 2. *CATT-predicted reverberation time T_{30} with different surface diffusion modelling.*

The results show clearly the high influence of the surface diffusion, especially for the mid frequency region. The influence is surprisingly high considering the complex model with 1259 planes. Estimating realistic values is therefore crucial for the calculated results.

With the variable absorbents extended the remaining reflecting surfaces may cause disturbing echoes. Besides echograms both auralisation and calculation of the echo criterion proposed by Dietsch and Kraak [6] was used in the design process for detecting disturbing echoes from loudspeakers and transient sources in the pit. It was clear that the Dietsch and Kraak echo criterion was not enough and that auralisation was a necessary tool.

4.2. Loudspeaker systems

Several CATT-calculations were performed in the design process. Three different types of main proscenium loudspeakers were calculated: normal point sources (clusters or individual boxes), curved line-arrays and straight line-arrays. The simulation of line-arrays in CATT consists of distance dependent directivity. The source position is still a point. Therefore, long arrays should be divided and added in phase. This is still an approximation but the influence of reflections near the loudspeakers will be more accurate.

In order to get maximum amplification with correct localisation the loudspeaker sound shall be delayed appr. 10-30 ms after the direct sound (Haas-zone). With this delay the loudspeaker sound can be at least 4-5 dB louder than the direct sound. With weak voices or high amplification it will be necessary to amplify the direct sound. Therefore, four loudspeakers will be hidden in the stage front edge for creating virtual sound sources on stage. When SOR is not used they can be used for normal front fill and image shift. CATT-calculations showed that SOR will not work on the sides furthest towards the stage on the first and second balcony. In these positions the level difference between the loudspeakers in the SOR-system will be too high. It was also clear that it was necessary to have four delay zones for the under balcony fill loudspeakers on the sides (two zones on each side).

5. DESIGN SOLUTIONS

5.1. Variable acoustics

Already in the program it was determined that electro-acoustic enhancement system should not be used for prolonging the reverberation time. It was later decided that the variable acoustics should be passive and motorized. The variation consists mainly of movable auditorium towers, on each side of the pit, adjusting the width of the hall and of absorptive textiles in front of the walls around the auditorium. Almost all of the vertical walls can be covered by textiles. The textiles consists of two layers of velour, each layer surface mass ≥450 g/m^2. Across the proscenium header and around 3rd balcony and technical gallery it will be hung in folds (50-100% overdrape). All variable absorption will be hung in motorized tracks and stored in closed boxes.

5.2. Loudspeaker systems

In the final solution all loudspeakers except the central proscenium overhead cluster are hidden. Figure 3 show the loudspeakers included in the SOR system and orchestra system in the CATT-model. Only the lowest level of the under balcony side fill loudspeakers are shown. E0-E3 are in the stage front edge. C3, C4, C6 and C7 are the orchestra loudspeakers in the loudspeaker towers, C1 and C5 are the voice loudspeakers in the loudspeaker towers, C2 is the central hanging voice loudspeaker. D0-D7 are the side fill loudspeakers under the 1st balcony. A6-A9 are the 3rd balcony fill loudspeakers (A6 and A9 are hidden in the back of the follow spot room).

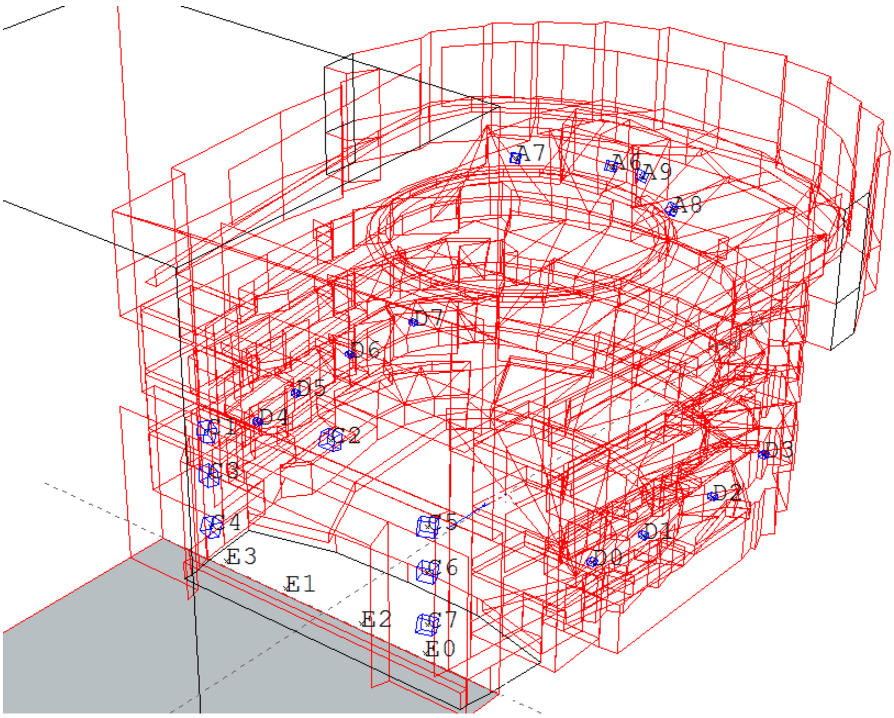

Figure 3. *CATT-model with SOR loudspeakers indicated (for clearness only side fills under the 1st balcony are shown)*

The loudspeakers for the audience can be divided in the following functions:

- SOR system
  Consist of four stage front edge loudspeakers, three main proscenium loudspeakers (left and right in mobile loudspeaker towers, centre hanging in a hoist) and 28 balcony fill loudspeakers grouped in nine zones.
- Orchestra system
  The orchestra loudspeakers consist of main systems (6 m over the stage floor) and lower fill systems (2.5 m) in the loudspeaker towers (left and right). Additionally, there are two (left and right) sub woofers placed in the auditorium towers next to the loudspeaker towers. The balcony fill loudspeakers shall be used as well.
- Surround system
  There will be totally 50 fixed speakers which can be used for surround. All will be hidden except for the 3rd balcony. Under the first balcony ten surround loudspeakers will be aimed towards the stalls. The rest will be used for under balcony seats (of which 24 will be used for fill as well).
- Sound effect loudspeakers
  Six active mobile loudspeakers and two sub woofers (woofers normally placed on the technical gallery).

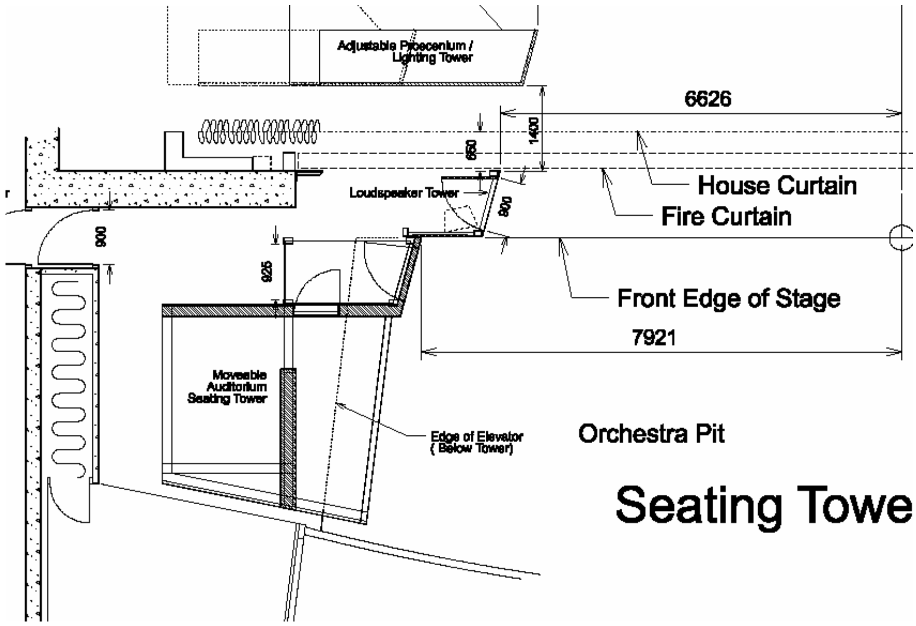

Figure 4. *Plan showing the movable loudspeaker and auditorium towers.*

Long line-arrays were rejected. The main draw back with long line-arrays with high directivity in the vertical plane and low directivity in the horizontal plane is the risk for disturbing echoes. Another draw back which is not often considered is the variation in frequency response with distance and angle. In this hall the risk for disturbing echoes and the fact that line-arrays are more difficult to use in SOR were the main reasons for not choosing line-arrays.

For controlling the SOR function a matrix mixer with adjustable gain and delay in the cross-points is needed. Every source (or group of sources in the same position) that shall be localisable simultaneously require one separate input in the matrix mixer. Every loudspeaker group which needs adjustable delay or level needs a separate output. The stage will be divided in nine localisation zones (each appr. 5x5 m). Totally, the matrix mixer will have 18 "sound images" for stage localising (9 with stage front edge loudspeakers and 9 without).

Eight inputs and 16 outputs turned out to be sufficient. In the final solution the SOR-mixer function will be embedded in the digital main mixer system.

6. SINGER SUPPORT

During the planning process the possibility to use a more advanced electro-acoustic enhancement system for improving the opera singer support on stage has been discussed. At present it is decided not to include such a system. A full fold-back system for "normal" monitor sound is of course included.

7. REFERENCES

[1] IEC 60268-16. "Sound system equipment – Objective rating of speech intelligibility by speech transmission index".

[2] Common Loudspeaker Format Group. *www.clfgroup.org.*

[3] NORDTEST method NT ACOU 108. "In Situ Measurements of Permanently Installed Public Address Systems". *www.nordtest.org.*

[4] ODEON room acoustic software. *www.odeon.dk.*

[5] CATT-Acoustic. *www.catt.se*.

[6] Dietsch, L., and Kraak, W. "Ein objektives Kriterium zur Erfassung von Echostörungen bei Musik- und Sprachdarbeitungen". *Acustica Vol. 60, 1986*.