Sound System Design in Public Environments—an Integrated Design Approach

Jakob Kraft, Rahe-Kraft GbR
Rahe-Kraft GbR, D-10963 Berlin, Germany; email: mail@rahe-kraft.de

1. Introduction

Public spaces nowadays require optical attractions like video-screens, but also an acoustic „environment“. Typical issues in public spaces are legal immission restrictions, difficult room acoustics, high background noise as well as specific expectations from clients and architects.

The Sony Center Forum combines all these aspects, requiring an integrated design approach. Some modern modelling tools and the used hardware made a high-quality solution possible which was unthinkable only a few years ago.

2. Project Description

The Sony Center is a public square with about 3500m^2 of audience area. To make the area more attractive, a large LED screen including a sound system was planned by the owner. Mostly background and atmospheric music should be played but the system also needed to function for speech transmission.

The project faced the following difficulties:

- an appartment directly facing the main speaker only 26m away, immissions restricted to 60 dB(A) by German law while 8m below, sufficient level was required
- a constant background noise of 60-62 dB(A) caused by the center's climate system
- an RT of 6 seconds at 500 Hz
- architectural limitations

Besides, the system was to be operated by unexperienced personnel and needed to be weather-proof.

The author was questioned if any system could meet these requirements or not, and if a sound system was judged to meet all criteria, to design this system.

3. Design Approach

Immission control was the limiting factor in this design. The system with the best ratio of useful signal in audience area to immissions at 10m height needed to be found, then the possible levels in audience area must be calculated as exactly as possible (see Figure 1).

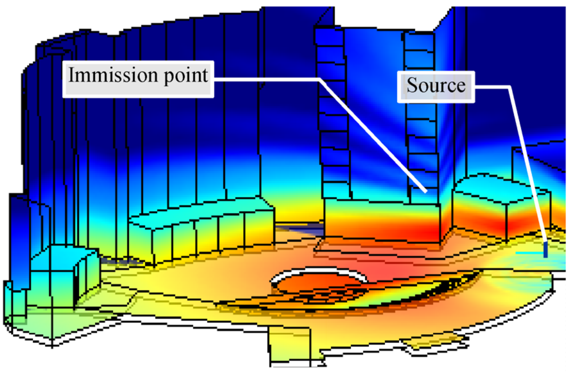

Figure 1: 1 kHz Direct sound mapping with an Intellivox 2c demonstrating the source to immission point distance (26m).

It was chosen to use a room/electroacoustic software instead of immission prognostic software due to the special directivity of the used speakers; CATT-Acoustic 8.0 (figure 2) was used for its capacity to handle special distance-dependant directivities with existing DLLs which seemed reasonable to best suit the requirements; some calculations were made with Duran Audio's DDA software.

Initial calculations were done with L-Acoustics dV-DOSC, Nexo Geo S, Duran Intellivox 1b/2b/2c/4c, Bohlender & Graebener Z-Line (different lengths).

Due to architectural requirements, a curved or mechanically aimed system could not be used. Thus, further calculations were limited to the Duran models due to their unique ability of electronic aiming and beam-forming.

Three basic appraoches were chosen:

- a distributed system with 12-15 smaller models
- a larger main system with only two delays
- a main system only.

All systems were optimized to evenly cover the audience area.

A number of receivers were modelled in the audience area and at the critical immission location. For all three systems,

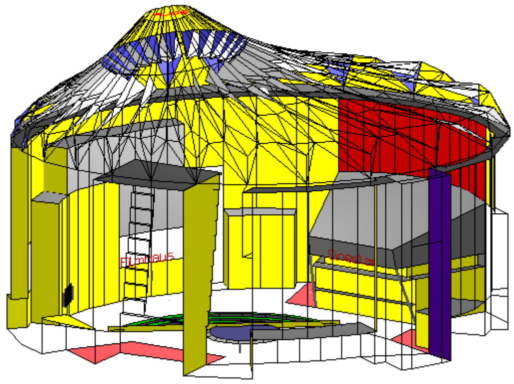

Figure 2: Detailed CATT-Acoustic model.

level ratio from the immission point to audience area receivers were calculated. It must be noted that due to the nature of speaker systems with distance-dependant directivity, it was necessary to equalize the direct sound for every system to result in the same spectral balance at audience receivers rather than 1m from system origin to make calculation results comparable.

Surprisingly, the performance of the distributed system turned out to be worse than those of the other two systems.

Both the main/delay system as well as the „main only“ system could have worked but the 4m-long Intellivox 4c needed for the „main only“ system did not fit beside the screen, and this solution could also not meet the client's request of splitting the audience area into several zones.

4. Turning simulation results into virtual immission measurements

Simulation results showed that possible levels in audience area could be only a few dB higher than at immission location and thus only little higher than the background noise. Noiseless STI was more than sufficient, but STI with noise varied around 0.50. The given information was simply not enough to make a decision for a system:

- measurement methods for immission control could not be directly compared to numerical simulation results; the „Taktmaximal-Mittelungspegel“ yields a long-term average of peaks in 5-second slots, measured with „FAST“ (comparable to “Impulse“ measurements)
- Spectral balance and dynamics of program played on the system had strong effects on the level differences as well as intelligibility, D-50, C-80 etc.

The above uncertainties suggested that a decision for a system could not be made based on pure numerical results. Auralizations were made for all three systems, every receiver location with speech as well as music. The wav-file at immission location was calibrated to the allowed level; all files were then mixed with calibrated background-noise recorded on site.

The results corresponded with numerical results: speech intelligibility was only sufficient with increased levels; for pure background music it was sufficient. It was concluded that the client needed to decide by himself based on subjective listening.

5. Subjective client decision based on auralization

The mixed binaural wav-files for 14 positions [1], both music and speech, were played to the client. Some initial disappointment over possible signal-to-noise ratio changed soon into a clear judgment that, with adequate content produced, the system could well do the job to create a new atmosphere in the Forum.

In this special case, auralization turned out to be the key tool for client presentation; thus it was not only „added value“ but pure necessity. Furthermore, the clients expectations were adjusted to be realistic; no complaints ever turned up after final system installation.

After a test installation, the client decided for the approach with a main system and two delays, consisting of two Duran Intellivox 2c and two Intellivox 2b. Two smaller systems for the nearfield and a 12“-subwoofer were added to round the system off.

6. Signal distribution and processing

Immission control officials asked that the system automatically be turned off at specific times, that levels be safely limited and the whole system be tamper-proof.

Cable lengths did not allow analog signal distribution; Cobranet turned out to be the appropriate standard which met all needs, support by many manufacturers enables future enhancements.

Very few DSP mainframes on the market offer hardware-based clock and calender for automatic switch-off and programmable level changes. Symetrix Symnet was chosen as the most cost-effective system.

The system is completely password-protected; level changes or other events can be programmed via an ISDN connection. A listening microphone was installed at the screen to verify the system function from the control room in the basement.

7. Architectural Solutions

Together with the project architect, the main system was integrated into the screen (Figure 4).

A remarkable design was found for one delay speaker by the project architect, giving an example for what audio can look like. A pillar was extended to 4m height and the speaker was integrated (Fig. 3).

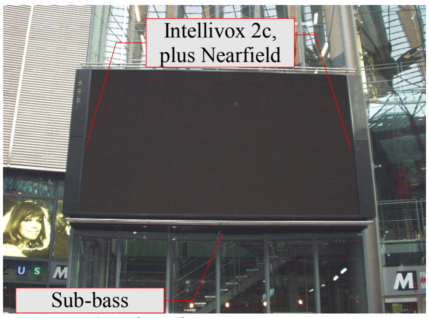

Figure 4: Speaker placement.

Figure 3: Design Speaker (Delay).

8. DSP-programming and system tuning

After the final installation, a considerable amount of time was spent on programming and system tuning using Monkey Forest measurement system. Some monaural recordings were made with the same wav-files used for convolution [1].

The used Intellivox speakers proved to be difficult to tune due to some resonances and the lack of very high frequencies of the 4“-fullrange speakers.

An automatic program-leveller not only assured a constant level independant of program dynamics, but also increased subjective levels while maintaining the measured levels at the immission point.

It became clear that for a system with critical parameters as this one, all modelling work done before was necessary but not sufficient; due to the special requirements in this case, it had to be followed up by careful filter, delay and aiming tuning on site by the designer rather than the installer.

9. Experiences after a 6-months period in use

Daily screen content is played from 10am to 9pm, can be heard anywhere in the Forum but is never disturbing which is what the client wanted (see Figure 5 for SPL distribution).

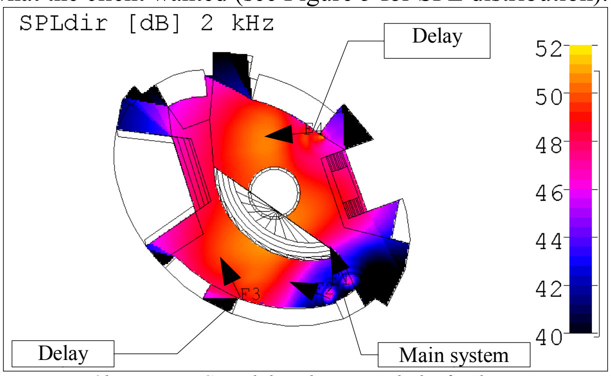

Figure 5: 2kHz Direct Sound distribution with the final system, nearfield speakers not yet included.

The system also proved its capabilities for 3000 people at a soccer game with sufficient intelligibility anywhere.

The client had to get used to the fact that he can, due to the necessary immission restrictions, only have limited control over the system. Problems with the DSP software were quickly fixed by Symetrix.

All in all, the system did meet the clients expectations.

10. Conclusions

After all, it seems that modern hard- and software are powerful tools for the consultant to create high-quality solutions which were not thinkable a few years ago.

At the same time, utilizing these technologies requires some in-depth knowledge about acoustic simulation and auralization, sound system design experience, signal processing and distribution, immission regulations, acoustic measurements and computer network design.

The many parameters from these different fields strongly influence each other.

Thus, the concept of a closely-woven design approach, integrating all different aspects from project kick-off all the way to the final system tuning, yields the best final results for the client.

Reference:

[1] Reference for downloading listening examples. URL: www.rahe-kraft.de/downloads/daga04wav.zip