Sound Localization or Speech Intelligibility?

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0 ABSTRACT

Sound systems exist since the invention of loudspeakers and microphones, but at that time mainly for the enhancement of the loudness of speech and musical signals. After WWII first delay lines in form of tape loops have been used to allow the sound coverage for larger areas or to create first delay lines for distributed ceiling speakers. In 50ties and 60ties huge congress halls have been built and the sound systems did use tape loops to assure certain sound localization and enhanced reverberation. So-called “intensity and runtime stereophony” has been developed. The paper will describe this way to create a high comfort in sound reproduction in halls or open air theaters in the 70 and 80ties. Afterwards, in the 90ties and in the first decade of our century this trend to design systems also for localization was stopped and such issues became more important like smooth coverage and wide frequency ranges, like maximum sound pressure and high intelligibility. But today the complex manipulation of sound fields by means of sophisticated sound systems allows both, best speech clarity and source localization. The paper describes the whole development over time and the state of technology for sound localization combined with needed speech intelligibility.

1 INTRODUCTION

Sound systems are in use since the 20ties of the last century, in the first years mainly for the movie industry. Especially after the sound film was introduced the need for good and clear sound transmission was obvious. But the rooms for the use of loudspeakers like in cinemas or for meetings were still small enough to install simple systems in front of the spaces. Larger existing spaces like churches or huge halls have been used for music or organ play but an understandable speech was difficult to achieve. In the thirties in Germany and Russia ruled gigantism and huge buildings with halls for 20.000 and more people have been planned but not realized, see figure 1. This hall was planned for 50.000 people. We don’t know which sound system should come into operation, but for the technicians this was certainly a challenge. In Russia was a similar development at that time. Already in 1931 a huge building was planned with a hall for 22.000 people, see figure 2.

Fig. 1: Planned hall in Nuremberg 1937

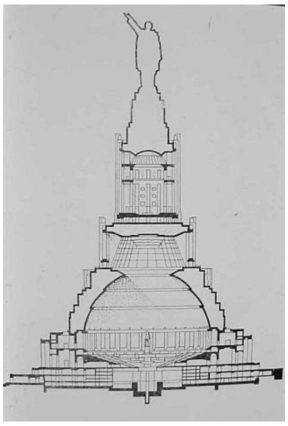

Fig. 2: Planned House of Soviets in Moscow 1931

In the last we know more about the background of this building. At the Soviet Academy of science especially an Acoustic Institute has been founded to solve the acoustic problems of this building. But the war did prevent the realization, only the basement had been built on the place of a famous Russian church. After WWII the basement was converted into a huge swimming pool and after 1990 the church was rebuilt at this place.
We know from studied documents that at this time a distributed Sound system was planned by using tubes for sound delay purposes. This way first localization effects had been prepared. Also customized loudspeaker had been designed.

The real research in using sound systems in halls and here especially in large ones did start in the late 40ties and the 50ties of the last century. In the following chapter we will touch the preconditions for advanced sound systems, the development of the technical tools and the different focus for the quality of sound in different times until now.

2 PRECONDITIONS FOR ADVANCED SOUND SYSTEMS

About sound impact and delayed reflections already Joseph Henry/1/ wrote in 1858: *“Reflected sound may tend to strengthen the impression or confuse it, according as the difference of time between the two impressions is less or greater than the limit of perceptibility.”* This and other experience for echo occurrence was known but a scientific research did only start almost 100 years later. In this context the precedence effect was discovered by Haas/2/. It was found that for small delays this delayed signal could not localized separately even if the delayed signal was until 10dB louder, see figure 3. The effect only did work with delay times until 30…40msec; with higher delays echoes became audible. This effect has been disregarded a couple of years and only at the end of the 50ties some sound designs may be found by using delay units to avoid echoes but also to realize localization to the original source, see Harry F. Olson, Acoustoelectronic Auditorium/3/ in figure 4.

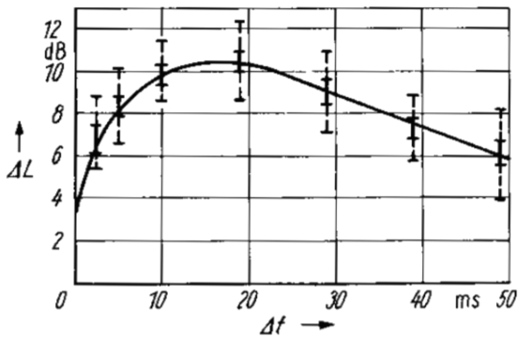

Fig. 3: Precedence Effect by Haas

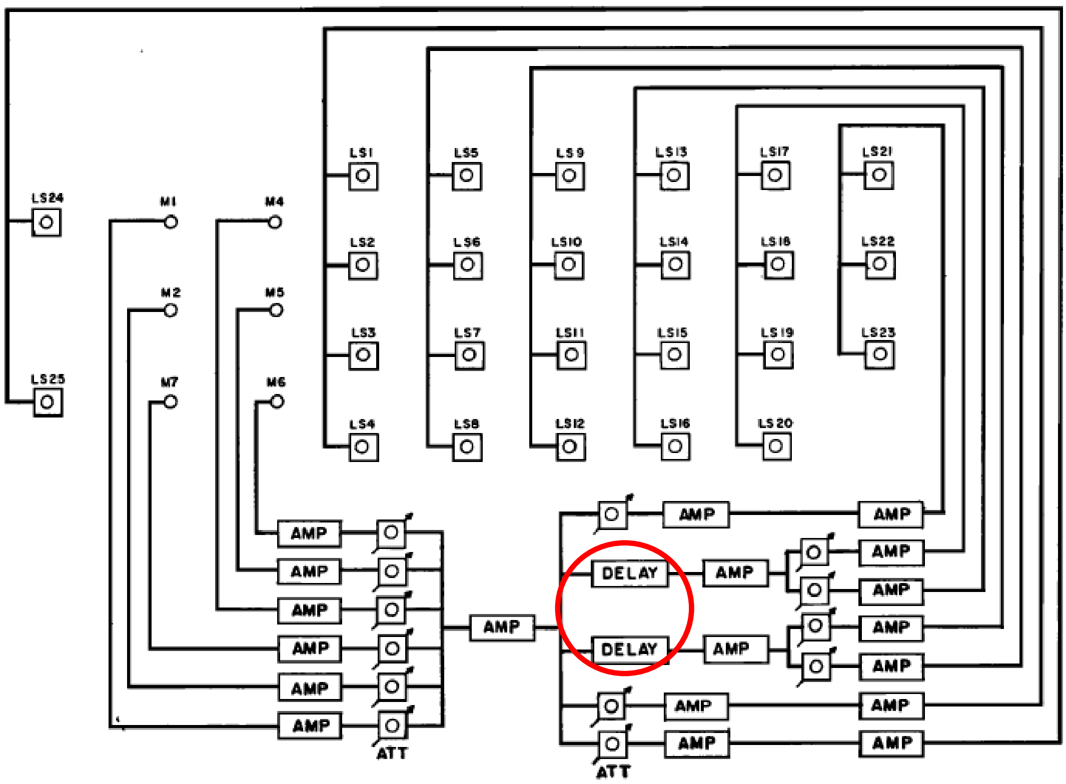

Fig. 4: Schematic diagram of the sound system of the Acoustoelectronic Auditorium

But the question was at that time what kind of delay devices could be used. Digital ones as used today did not yet exist. So in the Article of Olson two solutions are proposed, see figures 5 and 6.

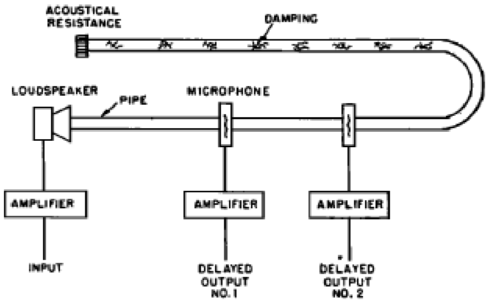

Fig. 5: Acoustic Tube delay

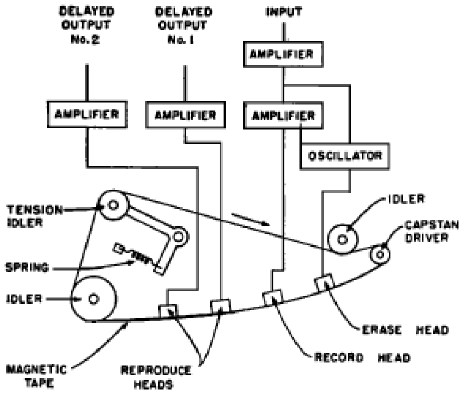

Fig. 6: Tape recorder delay

With such kind of delays the first installations have been made. Also the enhancement of reverberation has been done by pick up the signal close to the source and reproduces it with a multitude of delays along walls and ceiling of a room/4/, see figure 7.

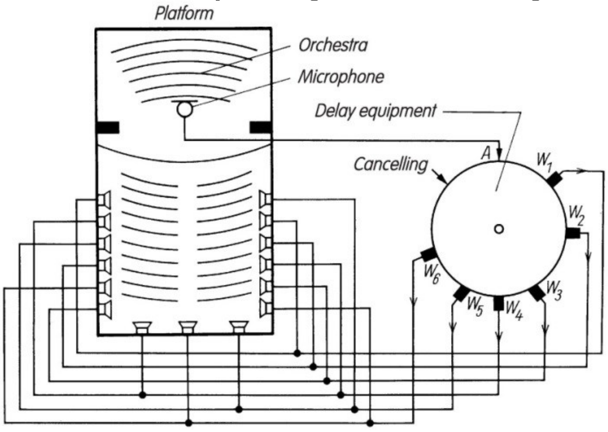

Fig. 7: Ambiophonic reverberation system 1958

At that time the Russian did construct now a huge hall for 6.000 people directly in the Kremlin territory. Such tape delay lines have been used to enhance if needed the reverberation time for the acoustically dry hall constructed mainly for speech. The schematic diagram is shown in Figure 8.

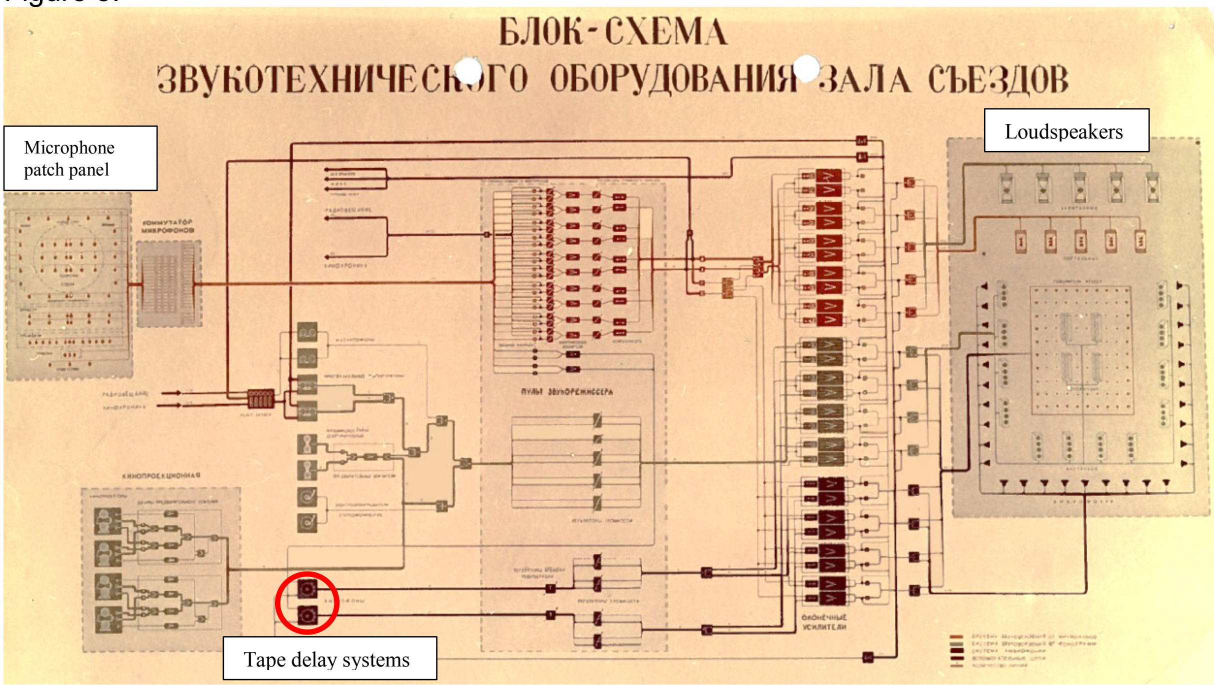

Fig. 8: Schematic sound system of the Soviet Congress Palace, opened 1961

3 USE OF DELAY UNITS IN SOUND SYSTEM DESIGN

The next step forward to use delay systems in the daily sound system design work was done by David L. Klepper/5/. In the late 60ties he published a couple of articles with the topic "Time-delay Units for Sound reinforcement". He used tape delay systems especially for under-balcony areas to keep the focus on the central cluster.

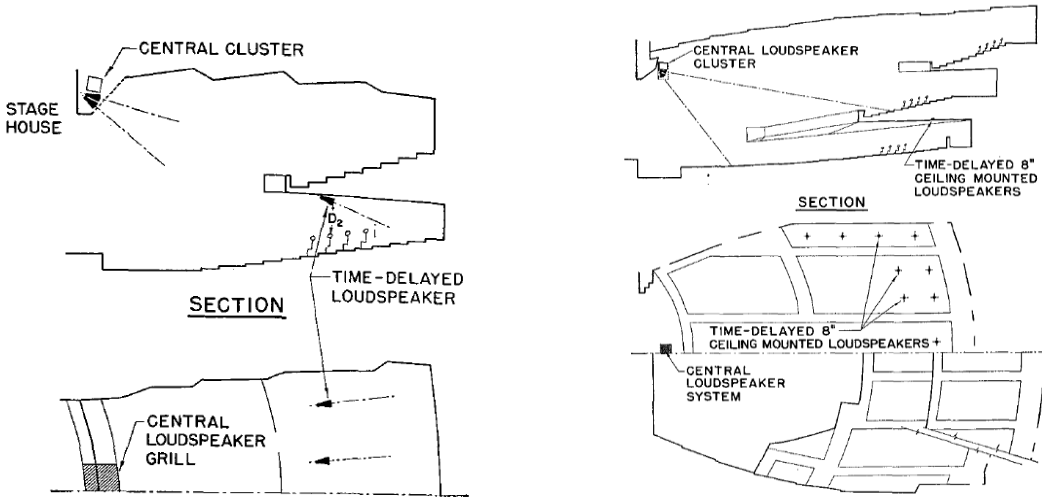

Fig. 9: Sound systems in Theatre proposed by D.L. Klepper

With the development of first analog (shift register based) but soon later digital delay lines in the 70ties a general use of delay line in sound reinforcement designs could be stated.
So at that time the sound system design was focused on the following features:

- The reproduction loudness should be adapted to the environs (the noise level).
- There should no unusual non-linear distortions be perceptible.
- The system should not produce any additional disturbing noises (noise, cracking, etc.).
- There must no feedback phenomena occur.
- The linear distortions occurring should correspond to the usual room-acoustical conditions
- The optical and acoustical localization's of the true or imaginary source (primary sources) should coincide.

Interesting, that at that time the speech intelligibility was not mentioned with one word.

4 HIGH END DELAY SYSTEMS TO ENSURE SOURCE LOCALIZATION

At that time the focus was on sound systems in large halls and this was also a topic of research.

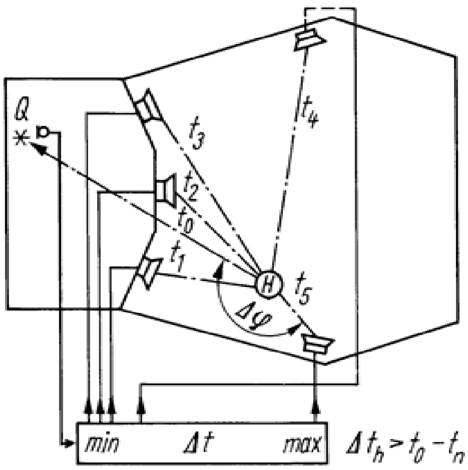

Fig. 10: Principle Deltastereophony DSS

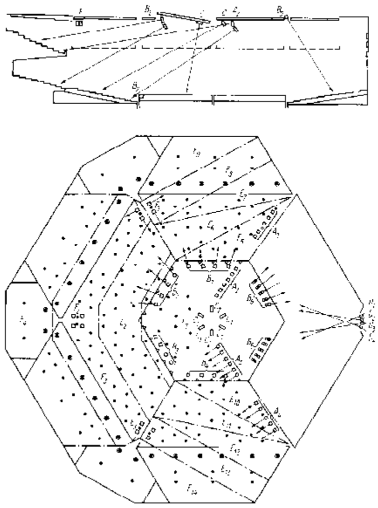

Fig. 11: First application in the Berlin Palace of Republic

So together with other colleagues the Deltastereophony concept has been developed at that time/6/, see figure 10.
With the DSS system the sound of the original source is heard first and then it is followed by signals coming from different loudspeakers with different delays and adapted levels. If the level of the original source is to low (singer or natural speaker in a large hall) a small loudspeaker (often hidden) must amplify the low level (so-called simulation speaker). All this also must work for movable sources, delays and levels are changed according to the actual source areas on stage.
The first application of this system was realized in the East-Berlin Palace of Republic and this for altogether more than 5000 loudspeakers (including the chair loudspeakers) and max. 15 movable sources (wireless microphones), see figure 11.
Another principle for sound coverage including sound source Localization in large halls was developed in the late 70ties by Plenge et al/7/. In 1979 in the International Congress Centre in Berlin the “Sound Reinforcement System with correct localization Image” was installed, see figure 12.

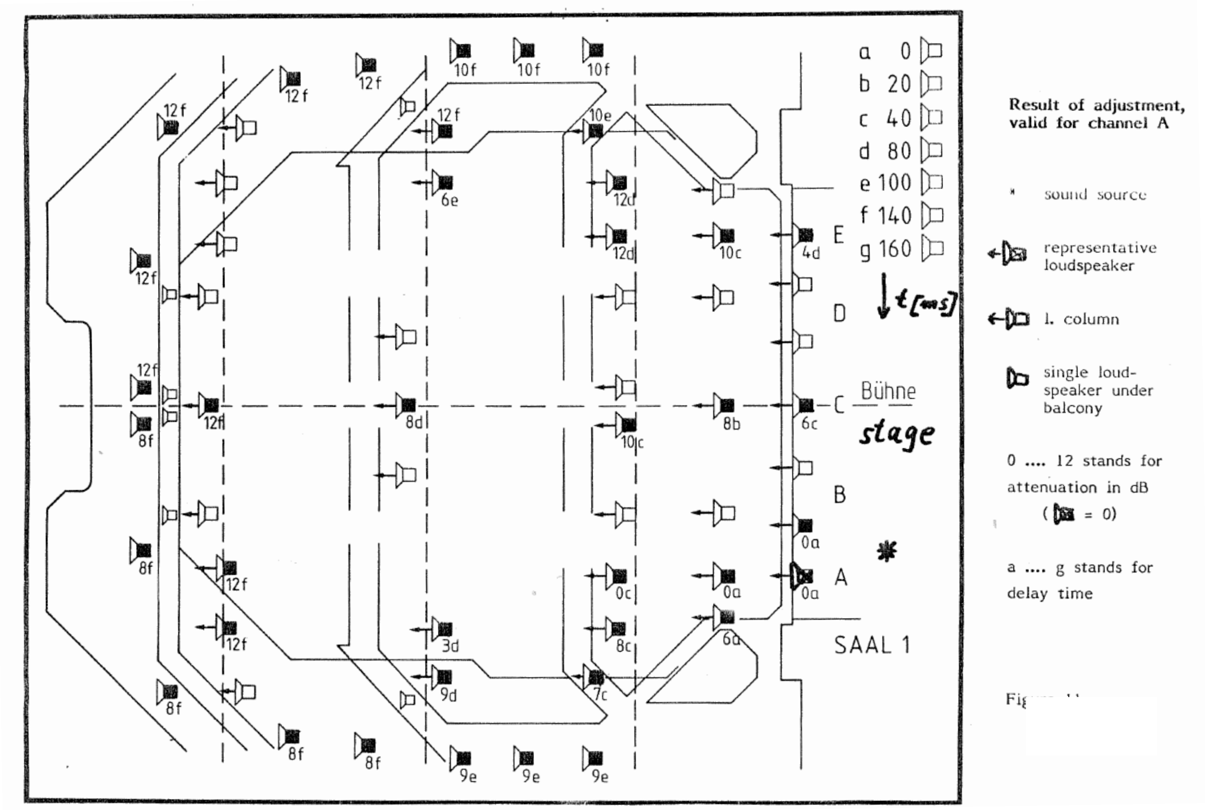

Fig. 12: Run-time delay system in the Hall 1 of the ICC Berlin

The figure explains the delay and level settings for the original Source * in stage area A. Only the black marked speakers are in operation and the numbers show the level attenuation in dB and the letters the delay in msec. A disadvantage was that by moving of the source over the stage some loudspeakers are switched off and other switched on. So the listener had different spatial impressions on his seat.
After more installations of the above mentioned DSS system in halls a first installation of the system was done on the sea stage in Bregenz / Austria in 1984. At that time the PC was just introduced and not yet available for us and so the calculations have been done with home computers. Fig. 13 shows the layout of the sea stage and some reflectograms calculated with the mentioned home computer. This kind of design is now since 30 years on the sea stage in operation, today of course now by using modern computer technology and extended equipment.

After the positive response to use such systems in large facilities the company AKG Acoustics GmbH decided to develop a DSS Compact device for 6 inputs and 10 outputs, see figure 14.

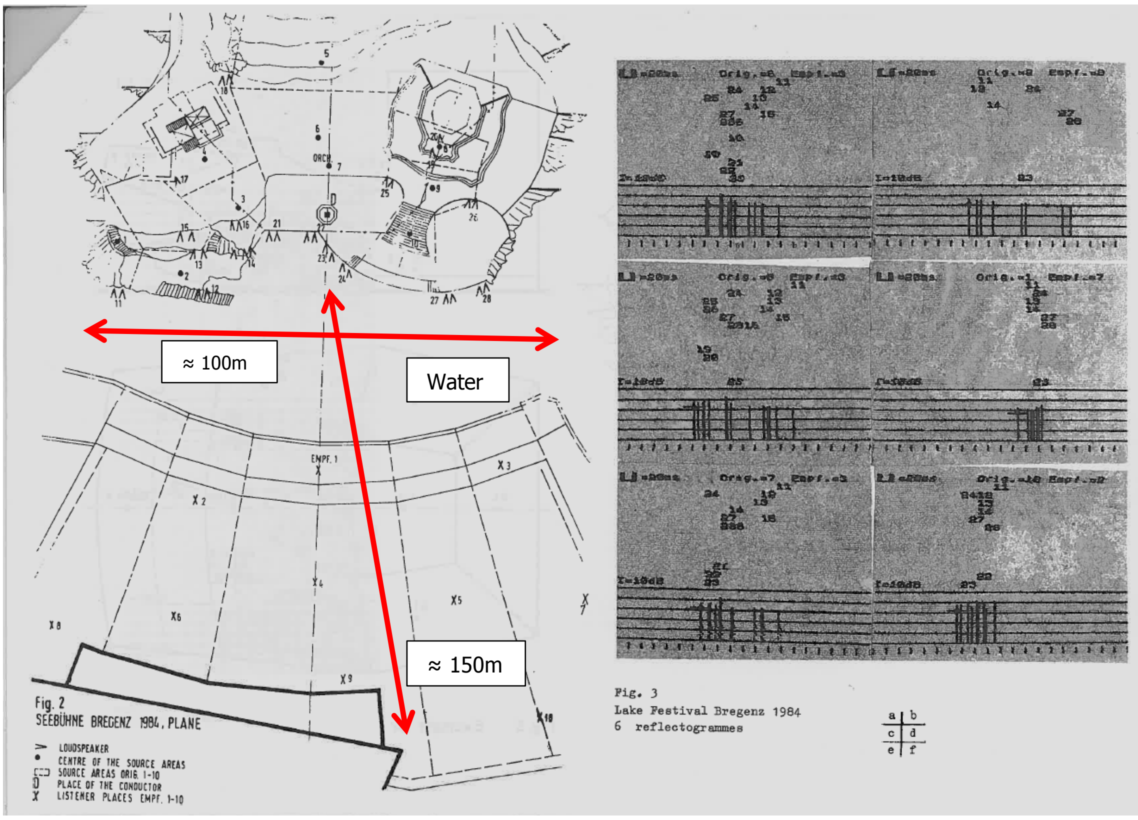

Fig. 13: Layout of the sea stage and the calculated reflectograms on selected seats

This Compact Processor DSP610 was also usable for mobile sources and the change of delay times has been done in live mode without any audible clicks. 1986 the first presentation has been realized in the Variety Art Centre in Los Angeles.

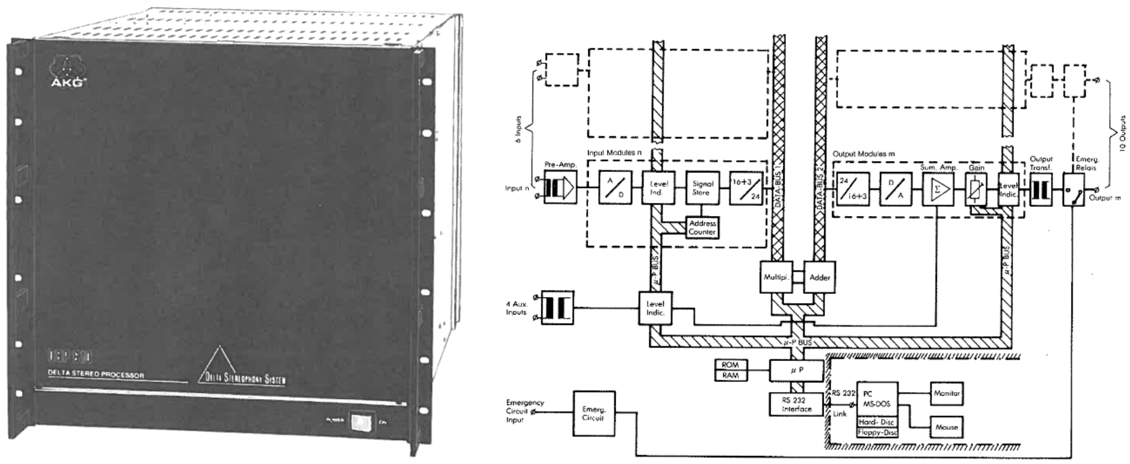

Fig. 14: DSS Compact device and block diagram

The first presentation of DSS procedure in Variety Art Centre was based on settings calculated with the home computer ZX Spectrum. The Data transfer happened with printouts on Electro paper and by entering of data settings into the AKG processor by hand.
The Data Transfer from ZX Spectrum has been done with 6 coefficients for each source:

$$t_V \,/\, \text{msec} = A1 + A2 * x + A3 * y + A4 * x^2 + A5 * x * y + A6 * y^2$$

x and y are the coordinates on stage at a certain height and the coefficients have been calculated with a developed software tool ADAS as an antecedent of the software EASE. ADAS versions have been used since the beginning of 1988, see figure 15.

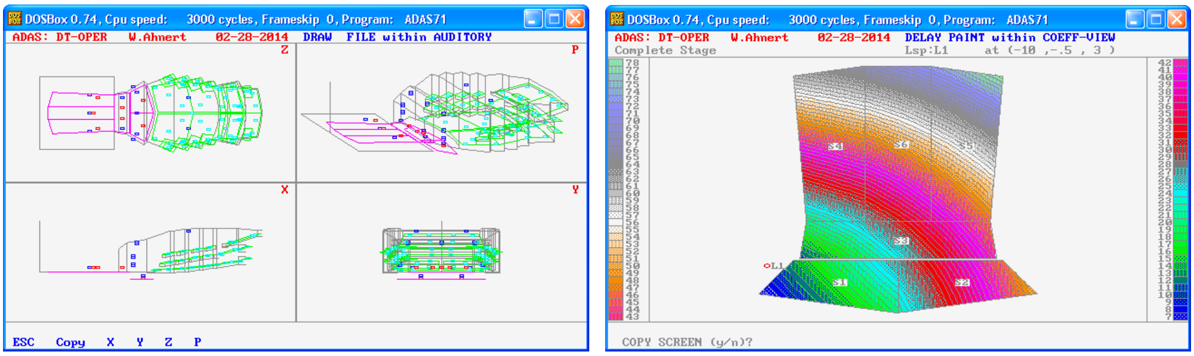

Fig. 15: ADAS printouts for simple room shapes with divided stage areas and delay mappings on stage

The mentioned coefficients calculated in ADAS could be read by the firmware of the DSP610 and the device was now ready to be used in the specific project, see figure 16.

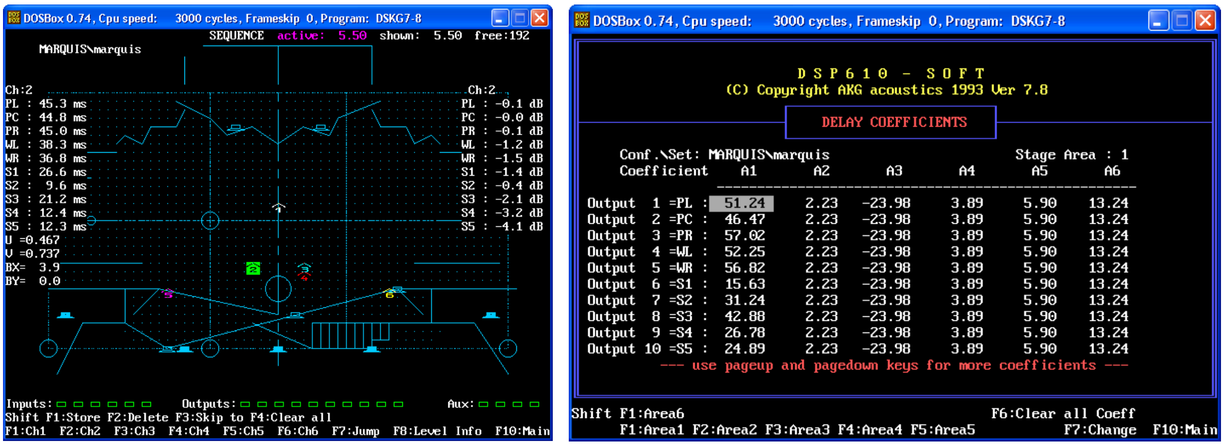

Fig. 16: Control panels of the DSP610 with 6 sources on stage for 10 loudspeaker lines

Because the movements of sources have had to be done by hand in 1989 started the elaboration of an algorithm for the DSS Processing of mobile sources on stage. Investigations of automatic Localization of sound sources were done at that time, compare figure 17. But this approach was given up because of missing hardware.

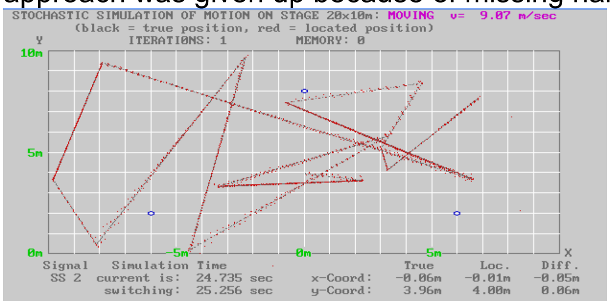

Fig. 17: First attempts with head tracking

5 SITUATION TODAY

Since the mid 80ties with the availability of the PCs started a Development of professional simulation software:

- Bose Modeler since 1985
- CATT Acoustic since 1986/87
- Odeon since 1988
- EASE since 1990

Parallel a Standardisation of intelligibility measures took place:

- RASTI in 1988 in IEC 60268-16
- 4 revisions until 2014 of the standard IEC 60268-16
- RASTI since 2011 not anymore recommended, now full STI
- 2002 Introduction of STIPa

Since the beginning of the 90ties was the Focus more on modern sound systems with new speakers like Line Arrays (L-Acoustics) and steered arrays (Duran Audio). It started the use of software to improve the prediction of reverberation, sound coverage and level and achievable measures like clarity and intelligibility. Observed could a decline of large projects with sound localisation demands and delay lines have been used also for small installations.
After September 11 worldwide more focus was given on emergency calls. It followed the development of STIPa and 2004 the update of IEC 60268-16. In prediction software packages the pre-calculation of full Speech Transmission Index was demanded by consideration of signal/noise ratio and Masking.
But parallel the interest in good sound source localization together with high intelligibility could be observed. A system was introduced, called source-oriented reinforcement SOR), which allows to localize a sound source on stage and to have high intelligibility of spoken words. Here like in the DSS system the stage is subdivided in sub-stages only known for the operator, see figure 18.

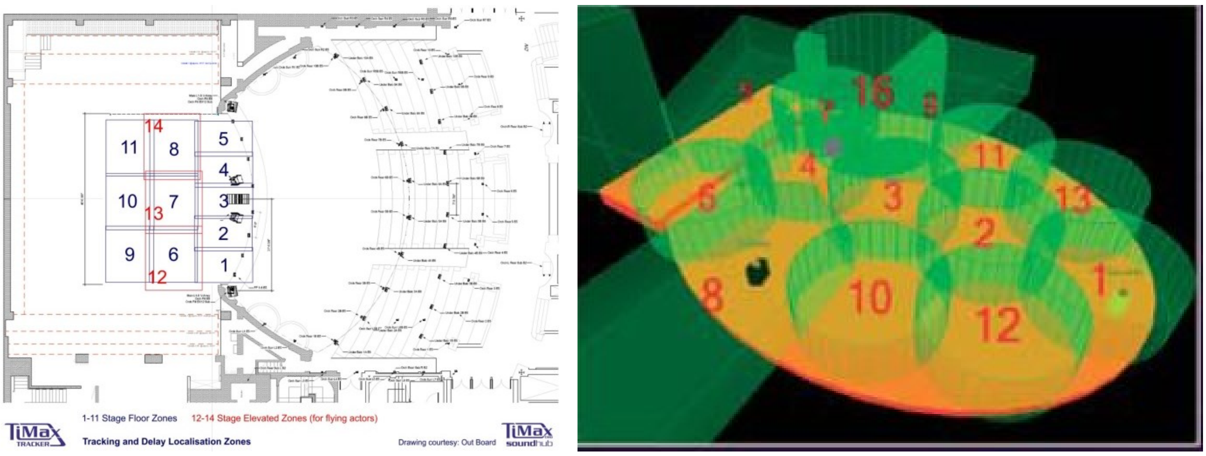

Fig. 18: Stage layout in the SOR system and visible limits of sub-stages in different shape

Quite often in the sub-stage areas a smaller loudspeaker are hidden to support the level of the original source in this area, see figure 19.

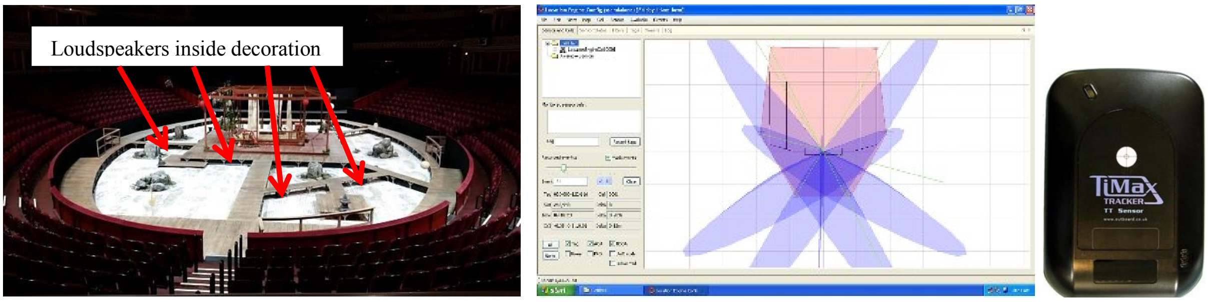

Fig. 19: Loudspeakers on stage for source support and Tracking device with corresponding software

Additional the now available head-trackers are used to localize the position of the source on stage and to change levels and delays of the system correspondingly.

But this approach with the SOR system is only a single one. Generally, by checking tenders for new sound systems in halls, theatres or large facilities you find the following demands:

- Needed sound level pressure
- Needed frequency response, flat or adapted to the design
- high speech intelligibility
- Sufficient signal to noise ratio
- Freedom of echoes

But still missing: Localization of the source on stage for talker or music groups.
Therefore the question "Sound Localization or Speech Intelligibility?" should be answered as follows: Both issues must be realized in the future to reach the needed sound comfort known from sound "coverage" by natural sources. This demand must be a part of any tender document for sound systems in new facilities.

6 LITERATURE

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