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Various Applications of Active Field Control

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ABSTRACT

The Active Field Control system is an acoustic enhancement system that was developed to improve the acoustic conditions of a space so as to match the acoustic conditions required for a variety of different types of performance programs. This system is unique in that it uses FIR filtering to ensure freedom of control and the concept of spatial averaging to achieve stability with a lower number of channels than comparative systems. This system has been used in over 70 projects in the US, Europe and Japan. This paper will provide an overview of the characteristics of the system and examples of how the system has been applied.

1. INTRODUCTION

When a person is enjoying music at a concert, the acoustical conditions of the venue have a significant impact on his or her subjective impression and evaluation of the music. It has also been reported that a musician’s performance is affected by the acoustical conditions [1]. However, performance spaces are chosen not only for their acoustical conditions, but also in consideration of other factors, such as the capacity and convenience of the venue. If a system could change the acoustics of a venue, it would help both musicians and audiences better appreciate the musical performances. To achieve this ability to change the acoustics at a reasonable cost, we have developed an Active Field Control (AFC) system that makes it possible to change the acoustical conditions of a space. To date, we have used this system for various projects.

Figure 1 shows an overview of the AFC system. Sound picked up by microphones is amplified and digitized by an HA and AD unit and then processed by a signal processing unit. The resulting signals are assigned to multiple output channels through the use of a level matrix. The signals are then amplified by amp units and reproduced through multiple speakers. Because the sounds reproduced by the speakers vary with the conditions under which the speakers are installed, compensations must be made for these variations in each individual speaker during the tuning period. Also, to adjust the spatial impression of the reproduced sound field, it is necessary to specify different delay and gain values for each speaker. That is why each speaker is normally driven by an independent amp channel.

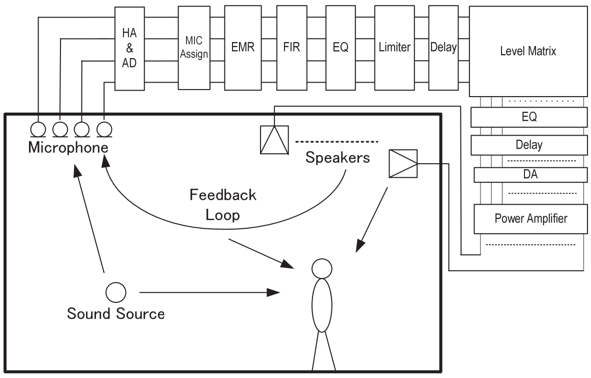

Figure 1: Overview of the AFC system.

Because the microphones and speakers are set up within the same space, the sound that is reproduced by the speakers is picked up again by the microphones and then reproduced again by the speakers. This creates a feedback loop between the microphones and speakers. When the gain between the microphones and speakers exceeds a given amount, this feedback loop results in self-exciting oscillation, which can cause feedback noise. The difference between the system operating state and the gain value that leads to feedback is called the feedback margin. When the overall gain of a system is constant, the feedback margin between a specific microphone and speaker increases as the number of independent channels increases. Conversely, when the feedback margin for each channel is constant, as the number of independent channels increases, a larger overall system gain can be achieved, and acoustic conditions can be changed with greater freedom [2]. That is why it is common to create a system that comprises at least four independent channels by setting up at least four microphones in a venue and sending the signal picked up by each microphone to a speaker without mixing the signals. In AFC, an EMR (electronic microphone rotator) [3] is used to switch combinations of microphones and speakers over time. This results in the spatial averaging of loop gain and the flattening of the loop frequency characteristics for the overall system. This in turn makes it possible to achieve stable control with a lower number of microphones.

Also, in some cases, we configured systems that comprised more channels and we adjusted the placement of microphones and speakers and adjusted signal processing parameters in each system. These adjustments sometimes resulted in the configuration of energy exchange systems and systems in which early reflection and reverberation could be controlled independently [4][5].

Because the adjustable range for each channel is determined by the combination of the electrical transfer characteristics from the head amp to the speaker and the physical transfer characteristics from the speaker to the microphone, increasing the size of FIR filters in the signal processing unit increases the controllability of the system and makes it possible to implement a variety of different changes. Normally, when the size of FIR filters ranges from the tens to hundreds, the modifications made by these FIR filters are used to flatten the frequency characteristics of the loop gain, which is a transfer characteristic of each channel. When thousands of FIR filters or more can be used, they can be used to adjust acoustic conditions, including time axis variation characteristics. Also, if a high-performance processor that can convolve tens of thousands of FIR filters is used, it becomes possible to convolve impulse responses (IRs) that were measured in a specific space and reproduce that space's acoustic characteristics in a different room [6]. Also, by using IRs created through processing or simulations, it is possible to produce acoustic spaces with extreme reverberation times and unnatural spatial impressions that would normally be impossible to experience. These acoustic spaces can be used to produce a variety of different acoustic experiences.

As indicated above, AFC with different structures and compositions can be used for a wide range of applications, from simply enhancing reverberation to making significant changes to sound fields for dramatic effect. In the following chapters, we will provide three examples of how systems with different characteristics have been implemented.

2. SYSTEM FOR IMPROVING STAGE ACOUSTICS

This example is of how AFC was implemented to improve the stage acoustics in two school auditoriums that do not have stage shells. In both auditoriums, the performers receive little support from reflections, and the deviation between the acoustics of the audience area and the stage area is significant. The system was used to make performances easier for performers and to reduce the acoustic deviation between the stage and the audience area. Four boundary microphones were installed in the proscenium and 12 speakers were installed on fixed batons above the stage. Figure 2 shows the system configuration and figure 3 shows a block diagram of the system.

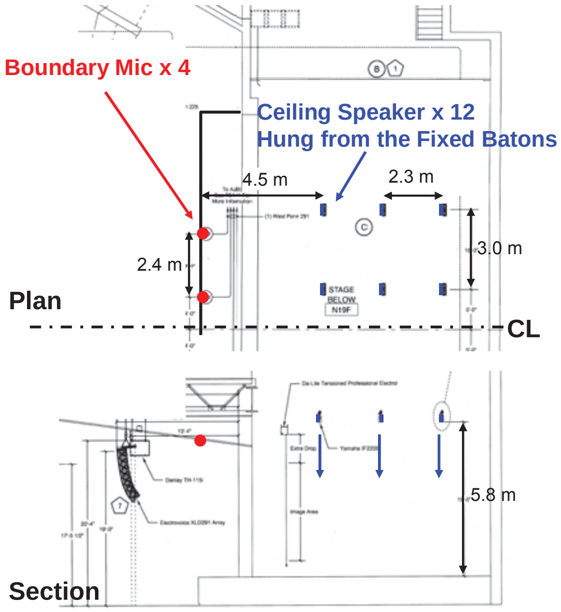

Figure 2: System configuration of the auditorium.

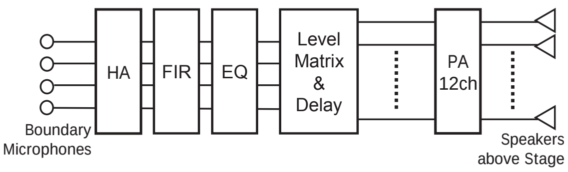

Figure 3: Block diagram of the system.

	System Off	System On
Auditorium A	-15.0	-14.0
Auditorium B	-14.7	-14.2

Table 1 Measurement results of ST_{Early} (dB).

After the system was tuned, acoustic measurements were made to evaluate the effect of implementing the system. First, as a measure of ensemble conditions, we measured ST_{Early} [7], formerly described as *ST1*, on the stage. Figure 4 shows the measuring points of ST_{Early}.

As shown in table 1, the result was that the implementation of the system resulted in an average improvement of 0.5 to 1.0 dB on the stage.

Figure 4: Measuring points of *ST* at each auditorium.

Also, we used *ts* [8], which represents the IR center time, to evaluate how much the acoustic deviation had improved. We placed a sound source at the boundary of the stage and the seats and performed measurements at various points on both the stage and seat sides that were equal distances from the sound source. Figure 5 shows the measuring points of *ts*.

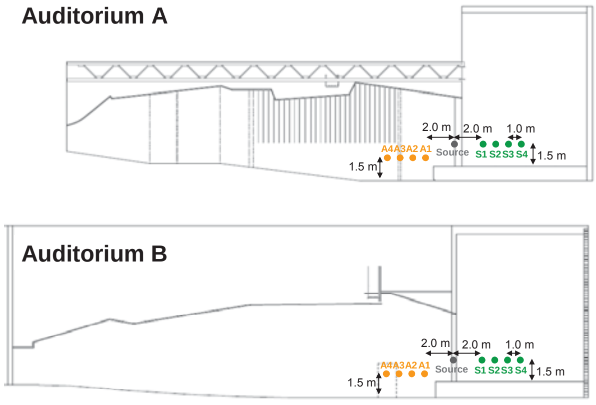

Figure 5: Measuring points of *ts* at each auditorium.

When the change in *ts* at each of the points located at equal distances from the sound source follows the same trend, we can assume that the acoustic deviation is small. The results are shown in figure 6. Using this method, we were able to confirm that the implementation of the system resulted in decreased acoustic deviation.

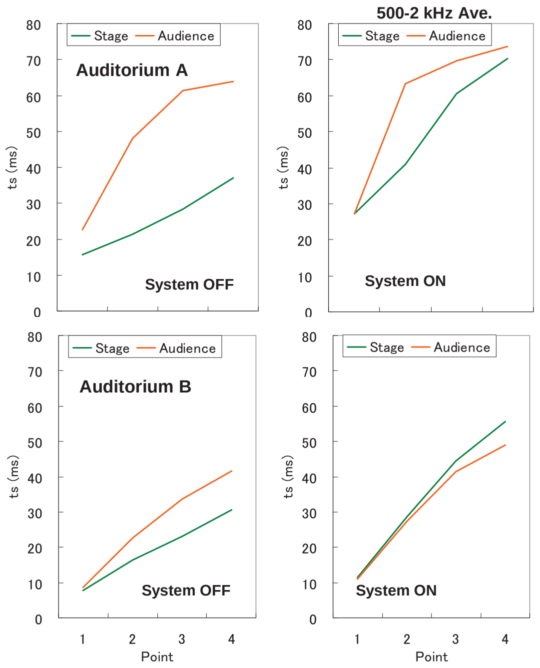

Figure 6: Measurement results of *ts*.

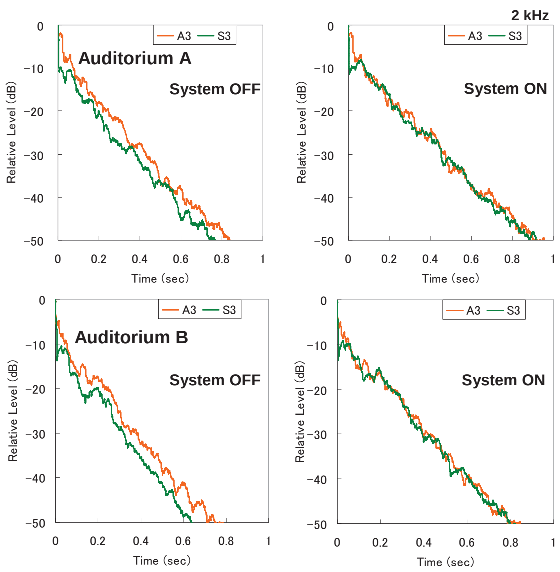

Figure 7: Measurement results of rms curves.

In addition, at each auditorium, we calculated rms curves using the results of measured IRs at measurement points on the stage and in the audience area. As shown in figure 7, the results of these calculations showed that the rms curves of the stage and audience area differed when the system was not running but were almost the same when the system was running. These results also showed that the implementation of the system resulted in decreased deviation between the acoustics of the stage and the audience area.

3. SYSTEM COMPRISING AN EARLY REFLECTION UNIT AND A REVERBERATION UNIT

The second example is of a multi-purpose hall with 400 seats in which two systems—an early reflection enhancement (ER) system and a reverberation enhancement (REV) system—were used. In addition to configuring an REV system mainly for reverberation enhancement, we configured an ER system to compensate for the lack of early reflection in the audience area, which was caused by the shape of the room. Four boundary microphones were installed near the stage, and sounds were processed separately by an REV processor and an ER processor. Then the sounds from the REV processor were reproduced from 22 ceiling speakers that were installed in the ceiling of the audience area, and the sounds from the ER processor were reproduced from 8 speakers that were installed in the walls of the audience area. In addition, 8 speakers were installed on the stage, and these speakers reproduced mixed sound from the REV and ER processors. Figure 8 shows the system configuration of the auditorium and figure 9 shows a block diagram of the system.

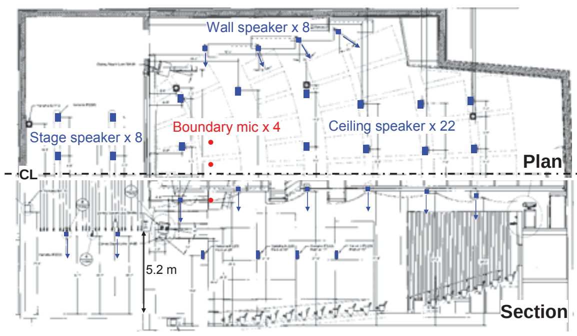

Figure 8: System configuration of the hall.

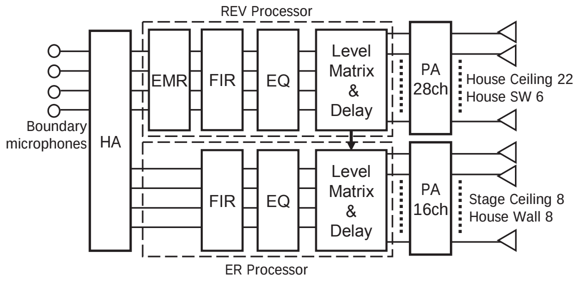

Figure 9: Block diagram of the system.

Figure 10 shows the measuring points and figure 11 shows the reverberation time (RT) of each preset. Thanks to the REV processor, the system was able to provide different RTs for different situations.

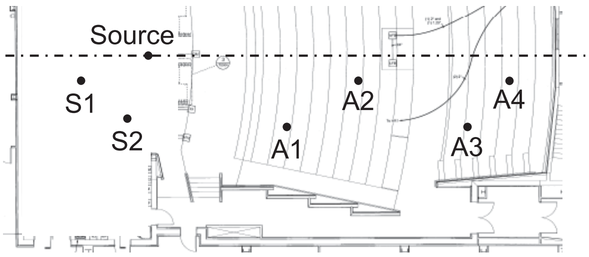

Figure 10: Measuring points in the hall.

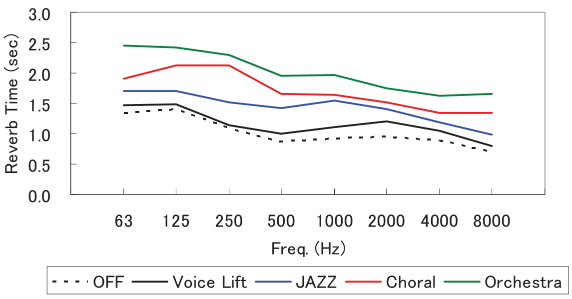

Figure 11: Reverberation time of each preset.

Figure 12 shows an example of the rms curves of each system and figure 13 shows the accumulated energy level within 80ms from the arrival of direct sound. We were able to confirm that the early reflection energy level is increased by the ER system. The system was able to independently control early reflection levels without significantly changing the RT that was set using the REV processor.

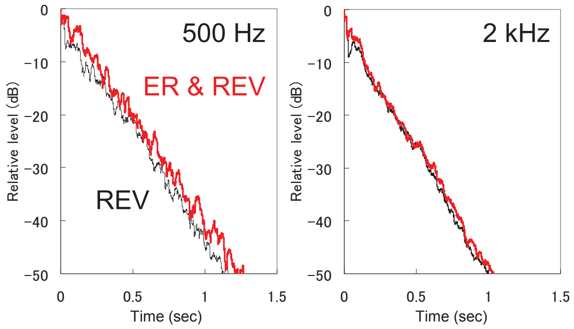

Figure 12: Example comparison of rms curves. (A2)

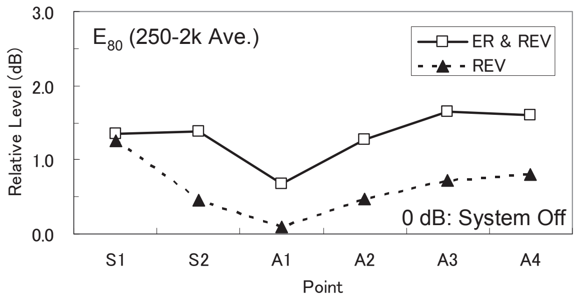

Figure 13: Comparison of early reflection energy level.

4. SYSTEM FOR IMPROVING THE UNIFORMITY BETWEEN THE STAGE AND THE AUDIENCE AREA

The third example is of a multi-purpose hall with 770 seats in which two REV systems were used for the stage and audience areas. With this system, we wanted to reduce the acoustic deviations between the stage area, the main area, and the area below the balcony. The system consisted of four boundary microphones that were installed in the proscenium and shared by both systems, 19 speakers that were installed in the ceiling of the audience area, 12 ceiling speakers that were installed below the balcony, and 18 ceiling speakers that were installed on the batons on the stage. Figure 14 and 15 show the system configuration of the hall and a block diagram of the system. Figure 16 shows the measuring points in the hall.

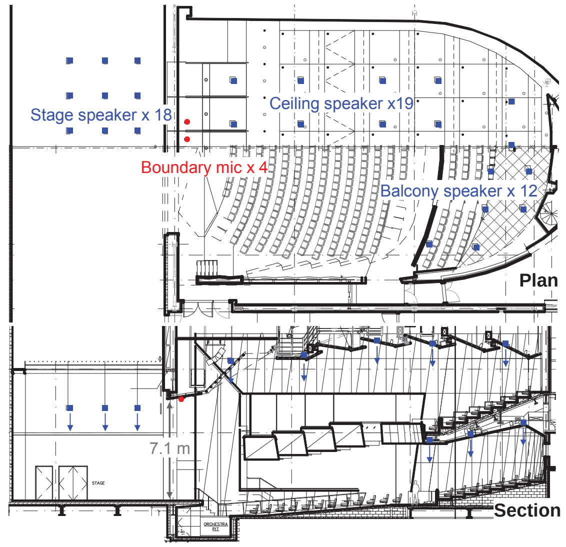

Figure 14: System configuration of the hall.

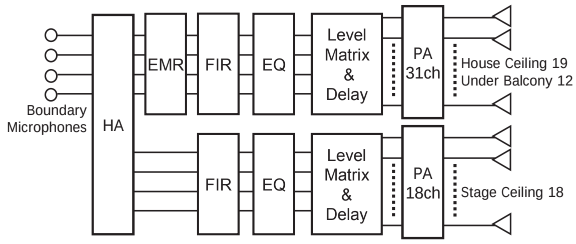

Figure 15: Block diagram of the system.

When the system was running, the energy level beneath the balcony rose by 2 dB as shown in Figure 17. The deviation of energy level throughout the hall improved by 0.6 dB.

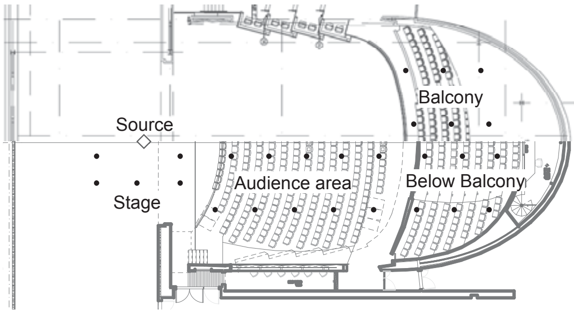

Figure 16: Measuring points in the hall.

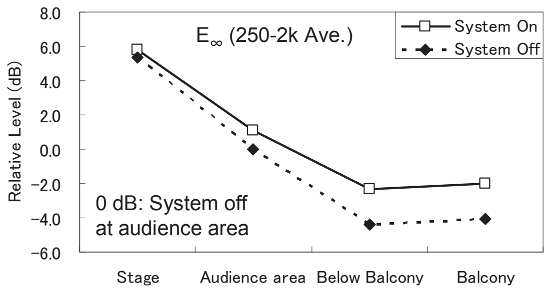

Figure 17: Comparison of energy level at each area.

5. CONCLUSION

Utilizing FIR processing and advanced feedback control techniques, the AFC system provides a stable and at the same time flexible tool to support solutions to a variety of acoustical challenges, using a regenerative approach to provide reverberation based on the existing acoustics of the room. The system can be used to compensate for architectural limitations, to make adjustments for various kinds of music, and to produce a variety of musical experiences. Three examples of how AFC was implemented in actual installations were presented: improving acoustic response on a stage without a stage shell, independently controlling early reflection and reverberation of a hall, and reducing the acoustic deviations in an auditorium.

6. REFERENCES

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