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2aAA6.  E-Venue - Affordable electronic acoustic enhancement for small venues

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The advent of modern digital signal processing has made altering the acoustical conditions in a venue using electro-acoustic tools practical. Such systems have been in constant use for many years in concert halls, opera houses performance spaces, houses of worship and a variety of other spaces and applications throughout the world. However, the cost associated with specialized nature of these systems has put them out of the reach of many small venues that stand to benefit most from use of this technology. This paper describes a new low cost integrated time variant electro-acoustic system designed specifically for use in small venues including but not limited to; performance venues, recital halls, rehearsal spaces, houses of worship, etc.

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Design Goals

The past generation of hardware and software developed for small pre-fabricated rehearsal rooms had several limitations that made it unsuitable for use outside of these integrated systems. The hardware could not support the full algorithm used in our professional systems. In addition, it did not provide professional audio termination or levels, and the system was comprised of several independent components. Thus, while it was very successful for the specific application it was designed for – the system could not be utilized for stand-alone applications in small venues. One of our primary design goals for any new system was developing hardware and software that did not present a compromise to using the same algorithm as our professional systems. Furthermore, this must be accomplished while maintaining a similar price point to the components that comprised the original system. In order to simplify set up and operation, the new system would need to integrate the functionality of the key components found in the original system (Figure 1); thereby allowing all critical functions to be manipulated form a single control point.

FIGURE 1. Signal flow for the previous generation system designed for practice rooms

Equalization settings for the previous generation system were determined by the pre-fabricated environment – a space that did not change significantly with each iteration. Thus it was essentially static. In general purpose applications, the Analog EQ indicated in Figure 1 sets the magnitude transfer function between the microphones and loudspeakers, and is critical to stable operation. We determined that the best way to ensure optimum stability and system performance is to integrate an automatic calibration routine as part of the new system. Even in small venues, however, the physics of the space may mandate the use of different loudspeaker types within the same volume. For example, a rehearsal room may have an acoustic tile ceiling; and the use of a baffle and back-can for the loudspeaker might be the best mounting solution. These devices may have a significantly different power response than the full range enclosures mounted in the lateral plane. Thus, additional system outputs with dedicated parametric EQ are needed to create a power uniform acoustic field prior to the automatic calibration.

For a small house of worship or small recital room with a pipe organ, the ability to change the dedicated parametric EQ’s to low pass filters for subwoofer operation is required. Thus, the new system needs to accommodate several modes of operation that are selectable as independent configurations.

The hardware for the previous generation system incorporated an IR receiver, and/or a serial communication port that required re-engineering a hand-held IR remote control to make it suitable for wall mounting with a hardwired interconnection. The new system would need to provide both a means for direct control, as well as the ability to integrate with industry standard control systems. Using human readable ASCII commands (SET param XX to XX and GET Param XX) over TCP/IP protocol) enables the new system to be controlled from both mobile applications as well as control systems on a dedicated network.
System Implementation

A new algorithm was developed to utilize Intel hardware developed for standalone appliance applications. The system provides professional audio interconnections; balanced microphone level inputs with integral phantom power, and balanced outputs with +4dBV nominal line level. The basic configuration is illustrated in Figure 2.

A second configuration that accommodates two loudspeaker types used in the same enclosed volume is illustrated in Figure 3.

The third system configuration provides low pass filters for additional outputs for subwoofers. This is utilized for mall houses or worship or small recital spaces that have an organ (Figure 4.)
The system is designed for use only in small venues – it cannot accommodate different delay zones, or multiple microphone configurations that are integral to our larger systems. However, the cost savings that are garnered from building an appliance that is targeted specifically for this application makes it affordable for small venues.

In conclusion, this system is a low cost appliance with all of the fundamental acoustics performance characteristic of our systems built for larger spaces. It provides both greater functionality, as well as superior sound quality to the previous generation system that it replaces.

REFERENCES