CASE STUDY

Variable Acoustics

Making it happen electronically

By Bernd Noack

Since its inception and subsequent opening late last year, the Herzliya Performing Arts Center (north of Tel Aviv, Israel) has provided an informative case study in variable room acoustics attained largely via the electronic realm.

From the genesis of the project, the idea was to create a hall with acoustics that could be tailored to suit the needs of all types of performances. In the "good old" non-electronic days, this was achieved largely with movable wall and ceiling elements.

The downside? Extremely high cost, a lot of extra space required, along with related fixed variations that can't be implemented without even more expenditure.

The more modern way to achieve variable room acoustics is with an electronic enhancement system. Two different approaches to this have been developed.

Non-regenerative systems are configured to pick up the stage sound, and, after appropriate signal processing, signal is distributed in a form of diffuse reverberant sound field into the hall. This approach uses high-quality reverb processors and can sound very good.

The disadvantage: the sound is fully artificially created and distributed. The audience itself is actually sitting in a different acoustical environment opposite of the impression of the stage sound.

In contrast, regenerative systems are able to alter the natural acoustics by "widening" the room using a certain number of high-quality microphones, special designed digital electronics and a lot of "invisibly" installed loudspeakers. These systems pick up the diffuse sound field in the hall, and using only a slight signal processing, re-distribute reverberant energy into the diffuse field of the hall.

Because the additional reverberant energy (and virtual early reflections as well) just add to the natural acoustics, the audience is sitting within the "new" acoustical environment and therefore are unable to detect whether it is hardware created (walls, ceiling) or software created (digital signal processing).

The Herzliya Performing Art Center offers a "medium-sized" concert hall with the dimensions of 80 feet (23.5 meters) wide by 54 feet (16.5 meters) deep and ceiling height of 62 feet (19 meters). Equipped with 764 seats, the hall hosts theatrical productions and speeches, chamber music and larger symphonic/orchestral events.

A stated goal of the project was an electronic-based regenerative variable acoustics system. Daniel Fichman, a noted A/V consultant and system designer in Israel, worked together with the theatre consultant, Uri Ofer,

Herzilya, subject of an intensive acoustical design, seen from the front and back.

to find a solution. They did intensive market research, looking into all available systems.

The overall goal was for different settings of the system to be programmed and stored, to be recalled by a set of push buttons. Patrons should not even be aware of the system working in the background. These criteria led the team to focus upon a VRAS system (which, appropriately enough, stands for "Variable Room Acoustics System") from Level Control Systems (LCS).

Tel Aviv-based A/V company Barkai Benny Brookstein Ltd. (BBB-Ltd.) installed the electronic room enhancement system, with Shy Kadmon of Barkai serving as project engineer.

MUCH PROPOGATING

Following construction of the hall, it was discovered that some of the materials in the roof area and on stage caused acoustical problems. Specifically, these were propagating too much natural low-frequency reverberation, a situation subsequently addressed before BBB-Ltd. could begin programming VRAS to meet the stated goals of the project.

Within a properly performing regenerative system, microphones and loudspeakers must be positioned within the diffuse field of the room. At Herzliya, 16 cardioid pattern microphones are installed in the permeable ceiling to capture reverberation, with six more hypercardioid pattern microphones positioned about 16 feet (5 meters) in front of the stage proscenium to capture early reflections.

On the output side, loudspeakers must blend inaudibly into the room's acoustic signature to create a smooth blend of natural room and virtual acoustics. Wide and even dispersion are optimal for this type of application, and at Herzliya, 22 ceiling loudspeakers presenting these characteristics were installed. They are joined by two subwoofers that are concealed within the ceiling structure.

These ceiling loudspeakers are combined into 14 loudspeaker groups, fed by 14 amplifier channels, with 16 of the 22 installed as a distributed grid. The remaining six divided into two rows, one to the left and right above the proscenium and directed at the audience.

Further, four groups of flat ML-Audio loudspeaker panels are integrated into the side walls (left and right). These are mainly used for early reflections, and they also distribute some additional reverberant sound energy from the ceiling system.

Maximum distance between loudspeakers and microphones was a priority concern in designing the ceiling grid, and as usual with projects of this nature, architectural restrictions were challenging. Once these elements are mapped and out and put into place, the key question becomes tying them all together into a cohesive group.

This is where the LCS components, including the VRAS system, come in. Different signal ratios are stored for each individual parameter setting in a matrix, and more specifically, two elementary system blocks were used: the reverberation processor unit for the corresponding microphones and loudspeakers; and the early reflections processor unit with the highly directive microphones pointing to the stage and wall loudspeakers.

24 FOR BOTH

Two LCS Matrix3 mainframes were specified by the project design team, outfitted with all necessary plug-in cards for inputs, outputs, communications and DSP power. The system is set up with 24 analog inputs for microphones and 24 analog outputs for the loudspeaker channels.

Both internal system blocks (reverberation and early reflections) are combined within a matrix of all incoming and outgoing channels. This matrix complexity is very useful to combine different signals into the same loudspeaker channels. Any system output can be routed to any loudspeaker channel in any chosen level, delay and equalization.

Because the Matrix3 system is actually a cue list controlled system, any of the created "virtual room" settings can be stored as a cue and recalled as an individual cue or be a part of a cuelist. Cues can have fade times for smooth changes. All programmed cues can be recalled by contact closures or any control system connected to the system via Ethernet or serial port.

Creating room acoustics in such a wide range doesn't mean just extend-

Some of the LCS gear that makes up the electronic variable acoustics system, with Michael Antek (left) and Shy Kadmon both active in its implementation.

ing the reverberation time. Much more complex parameters are involved in good sounding concert halls. (Any train station lobby gives a clear indication that just a longer reverberation time is not usually the singular goal.)

Further, speech and the various types of music each have distinct acoustical requirements to enable satisfactory performance. With electronic enhancement, the numerous microphones and loudspeakers work in tandem with sophisticated DSP algorithms to create the desired sound field by transforming the microphone signals into a new complex signal structure.

VRAS, which is controlled with Cuestation 4 software, provides distinct algorithms for both early reflections and reverberation for early and late energy enhancement. The room enhancement is not achieved by feeding artificially created reverb into the room, but by actually reinforcing the natural room reverberation itself.

Cuestation 4, released while the Herliya project was underway, incorporates input and new ideas in its algorithms supplied by several acousticians, particularly Michael Antek. (Based in Prague, Czech Republic, Antek also was involved in the tuning and optimization process on this project.)

One other interesting development of the VRAS Cuestation platform is that it is built as a server-client application. Any computer can be connected via Ethernet network to get data or to control the system, and further, this can also be achieved with a notebook on wireless LAN (local area network).

TASK FORCE

All VRAS systems installed in Europe and the Middle East are programmed and fine-tuned by a "VRAS task force" initiated by the German company MediasPro. Primary acousticians are Dr. Zdenek Kesner as well as the aforementioned Michael Antek.

This programming and tuning work is a service for the installing companies, lessening their need to learn programming and (most of all) acoustic measurement. Tuning of the system was done over a three-day period, with four different settings programmed: Dramatic Speech, Chamber Music, Classical Symphony and Romantic Symphony.

For final approval, Harvey Bordowitz, Herzliya musical director and conductor (and a veteran on several New York concert halls), was invited to listen to live musical performances in the hall.

Bordowitz did his evaluation from different positions and offered suggestions to the acoustical team. He then offered his formal approval.

Measurements performed following the tuning process included Reverberation Time, Early Decay Time and Clarity were evaluated (*see sidebar, below*).

On page 46 we show results of measurements done in the hall without enhancement as well as the same data plots gathered for the "Chamber Music" preset of the electronic system. ■

Based in Europe, Bernd Noack *is one of the longest serving contributors to* Live Sound, *consistently presenting interesting audio and acoustical concepts. He can be reached at mail@bnoack.de.*

Reverberation Primer

Reverberation time is the global quantitative criterion of the sound field in the room. Room reverberation gives fullness and a "singing tone" to the music. The reverberation time must be in the proper range depending on room size and the style of music.

But the acoustic quality of a room for musical performance cannot be evaluated by reverberation time alone. Two rooms with equal reverberation time can sound completely different!

Early Decay Time (EDT): This is the reverberation time derived from the initial 10 dB of decay. It's the length of time that it takes for the sound to decay 10 dB after the sound source is turned off. EDT more closely corresponds to subjective evaluation of the reverberation time that RT (below). EDT affects principally the hall's support to the voice and adds definition to the higher tones of music. The measurement is multiplied by the factor of six to make it comparable with RT60.

Reverberation Time (RT): The time it takes for a loud sound to decay into inaudibility after its source is cut off. It's defined as the level difference of -60 dB, and is normally evaluated over the -5 dB to -35 dB (RT30) decay of sound, and multiplied by the factor of two for conformity with RT60. These factors are necessary to make the measurements comparable with each other and with the more historical measurement of RT60, which is evaluated over a 60 dB sound decay.

Clarity: The ratio of the energy in the early sound compared to that in the reverberant sound, expressed in dB. Early sound is what is heard in the first 80 milliseconds ("C80") after the arrival of the direct sound. It's a measure of the degree to which the individual sounds stand apart from one another.

If the clarity is too low, the fast parts of the music are not "readable" anymore. If there is no reverberation in a room, the music will be very clear and C80 will have a large positive value. If the reverberation is large, the music will be unclear and C80 will have a relatively high negative value. C80 becomes 0 dB, if the early and the reverberant sound is equal.

For orchestral music, a C80 of 0 dB to -4 dB is preferred, but for rehearsals, a higher clarity of 1d B to 5 dB gives more control over the performances. For vocal performances, the clarity should be generally in the range between -4 dB and +4 dB.

For speech, in comparison to music, Clarity is measured as the ratio of the first 50 milliseconds (C50) instead of C80.

Other notable parameters: Definition D50, Strength (G), Interaural Cross-Correlation Coefficient (IACCA), Initial Time-Delay Gap (ITDG) and Envelopment.

Case Study

Data From Herzliya Performing Arts Center

The following data plots were generated by measuring impulse response with MLS signal – a steady pseudo-random test signal – in accordance with the ISO 3382 code. Data is shown for RT30, Early Decay Time (EDT) and Clarity (C80). Two omnidirectional measurement microphones located at different positions were deployed, and after several measurements, the average was calculated. (Type of software used in this process was withheld.)

Without Enhancement

Chamber Music Preset