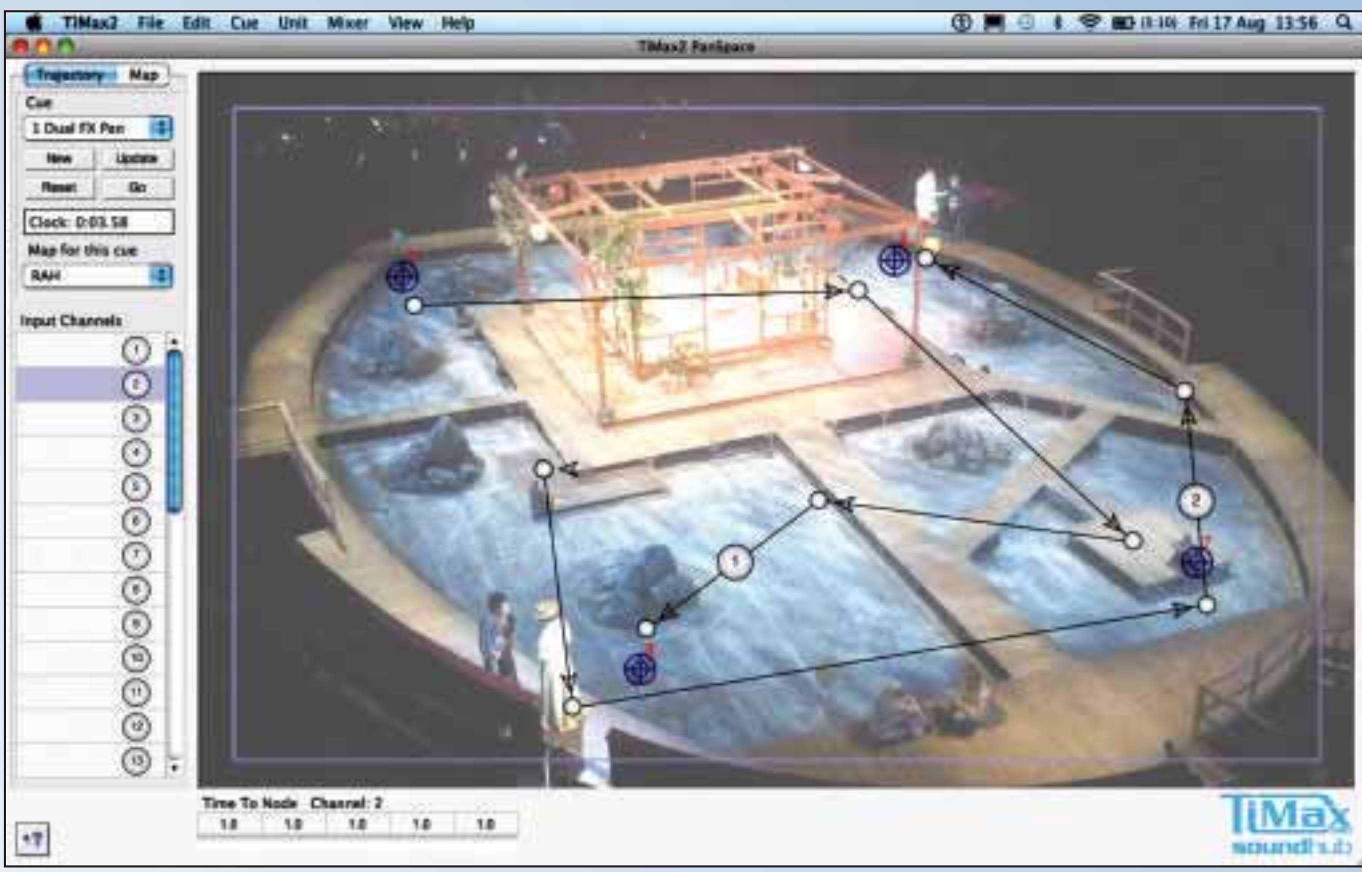
Focusing on the Speaker—not the Loudspeaker

SOURCE-ORIENTED REINFORCEMENT FOR HOUSES OF WORSHIP

by Alan Hardiman



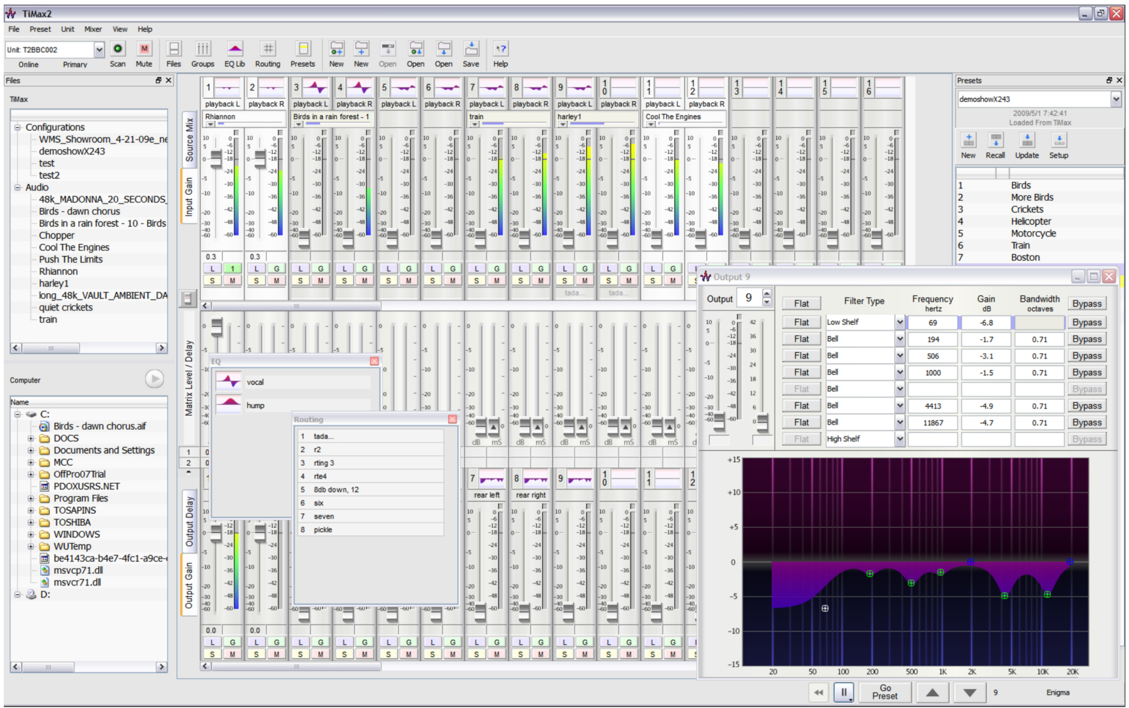
The PanSpace graphical object-based pan programming screen

You're sitting in church listening to the worship leader, and your attention wanders to the ceiling or the walls, following the speaker's voice to the sound system loudspeakers up there. You see the person speaking, the mouth is moving and the words are coming out, yet the voice seems somehow disembodied, disconnected from the person. You might say that the preacher is no longer one with the preaching, and that's a problem both technically and theologically.

To make matters worse, you realize that while you've been turning all this over in your mind, you've just tuned out the last few minutes of the service. The technology that's supposed to help you hear the Word has laid a speed bump in the middle of the road to salvation, and is not serving the purpose for which it was installed in the first place, which is to increase intelligibility.

This is usually accomplished by routing audio from a microphone through a number of distributed loudspeakers to bring the speaker's voice closer to the listeners' ears wherever they may be located throughout the seating area, so that it sounds pretty much as they would hear it in normal conversation seated at the front. Merely increasing the overall system level would make it too loud for people nearer the front. (Some components of the sound system are dedicated to assistive listening for the hearing impaired, but that's a separate issue.)

The trouble usually begins because somewhere in the process, someone neglected a psychoacoustic principle enunciated by the physicist Helmut Haas in 1949; namely, that we localize a sound in the direction from which it first arrives at our ears. This so called Haas effect (or precedence effect) is also known as the law of the first wavefront, and lies at the root of much of modern sound system design.



The TiMax GUI for PC and Mac provides a familiar environment for programming input, output, and matrix levels and delays

Consider the general shape of the human head—with two ears spaced on either side, and the barrier of the skull in between, any direct sound will arrive first at the ear that is closer to the source, and will arrive at the other ear slightly later. The miniscule difference in arrival times between the two ears allows our brain to determine the direction from which the sound emanated. Reflections arriving during the next 30 milliseconds (the fusion zone) from different directions off the floor, walls, and ceiling are not perceived as separate sounds, but are fused into one perceptual event, with all the sound appearing to come from the direction of the first arrival.

Since sound travels relatively slowly through air (at about 1.1 feet per millisecond) compared with an audio signal traveling at close to the speed of light through cables, listeners who are closer to a loudspeaker than to the worship leader will instinctively localize the sound of the voice to that loudspeaker rather than the worship leader, unless the audio signal is delayed electronically on its way to the loudspeaker.

The problem can be solved by inserting a digital delay in the signal chain ahead of the loudspeaker's amplifier, and adjusting the delay time by 1.1 ms per foot of distance between the delay loudspeaker and the sound source, in this case the worship leader. System techs usually add an additional 15-20 ms to the delay time in order to bury the delay loudspeaker well inside the fusion zone and ensure that the sound stays anchored to the original source. The level from the delay loudspeaker can be up to 8 dB higher than the original source at a listener's ears and still not be perceived as a distinct source.

The goal is to have the delay loudspeaker reinforce the original sound without appearing to be a sound source itself. In a properly calibrated system, the delay loudspeaker is not perceived at all. Larger houses of worship may require several delay loudspeakers at progressive distances from the front, as well as underneath and over the balcony, and in any other area that is acoustically shadowed from the front. The important thing is to mount the delay loudspeakers in line with the original source, and not firing in from the side walls, in which case it would be impossible to set a delay time that is right for every listener in that area.

If the pulpit is located at the front left, as viewed from the pews, then simple geometry dictates that for sound emanating from the pulpit, a loudspeaker over the 20th row on the right will need a longer delay compared with a loudspeaker mounted over the same row on the left, directly in line with the pulpit. The situation is reversed, however, for sound emanating from a lectern at the front right—in this case the delay to the loudspeaker on the left would need to be longer. Wider rooms will call for an even greater difference in these delay times.

There will also be a need for selective equalization in each of the delay zones, since the intelligibility deficit that you're trying to remedy in the first place is largely due to deficiencies in the audible spectrum, typically a lack of higher frequencies, but sometimes also too much muddy mid bass. This must be diagnosed with a high resolution spectrum analyzer, but on the plus side, it only needs to be done during the initial system set-up. The judicious application of parametric equalization will reduce these differences in the audible spectrum and eliminate what has been termed the sonic dissociation from one area of the house to another.

There are a number of ways to approach the installation of a sound reinforcement system in a house of worship with multiple custom equalized delay zones, and a number of individual source zones, including pulpit, lectern, the central area between them, soloist positions in the choir, and the locations of individual musicians. Typically, such a system includes a rack of digital delay units, parametric equalizers, distribution amplifiers, an input mixer, a mass of cables connecting it all together, and a hefty price tag.

One elegant, innovative and very cost-effective solution is the TiMax2 SoundHub from Out Board Electronics. It's a single device in a 2-RU package that combines 16 inputs, 16 outputs, and an input-output matrix. Any input can be routed to any number of outputs with programmable level and delay at each of the 256 cross-points (16 x 16). The level and delay programming of a cross-point is referred to as an image definition. Each output also features 8-band parametric EQ, as well as control of output level and additional delay. Four-band parametric EQ is available on each input.

To round out the SoundHub's capabilities, 16-channel replay from an internal hard disk is offered as standard, a useful utility for occasions when music, for example, is desired but musicians are unavailable, such as when the facility is open during the week for unstructured public drop-ins.

The TiMax Tracker radar tag, 1" square

Programming of levels, delays, EQs, and so on is done through a simple graphical user interface (GUI) on a PC or Mac connected to the SoundHub via a standard network cable.

For larger installations, the SoundHub is available with up to 64 inputs, 64 outputs, a 4,096 cross-point matrix, and 64 channels of hard-disk playback. For installations where the worship leader is more active and walks around among the congregation, Out Board offers the TiMax Tracker, a radar-assisted localizing device that permits a worship leader wearing a 1-inch square plastic radar tag to be precisely localized by radar sensors mounted unobtrusively in the building.

A software location engine analyzes the data from the sensors, generates a 3-D animated image of tagged individuals moving around the house and sends location information to the SoundHub via a stream of MIDI messages. These messages contain the tag numbers corresponding to SoundHub inputs, and their locations in the house correspond to the preprogrammed image definitions. When the worship leader or other tagged individual moves from one zone into another, the SoundHub dynamically interpolates between the pre-programmed zones to avoid any abrupt change in settings, which might be disconcerting to listeners. In this way, the mapping and crossfading of inputs to outputs in the matrix is slaved to the location data provided by the tracking system.

Events such as playback of prerecorded sound effects or music can be triggered automatically from the system's internal drive when a performer enters a particular zone. Obviously useful for theatre production, this feature can also be employed in interactive exhibits, for example; if patrons are given radar tags to wear on entering, then their movement through a zoned exhibit will trigger various preprogrammed events—great for immersive theatre or museum tours.

Out Board Electronics refers to their approach to sound system design as source-oriented reinforcement (SOR). Based on their extensive experience in theatre, they say that SOR allows more than 90 per cent of the audience to localize a performer's voice to within six inches in any direction, helps focus attention directly on that person, cuts through the reverberation and clatter typical of traditional hard-walled facilities, and, above all, increases intelligibility to a level unattainable with conventional sound systems that do not take advantage of our ability to localize a sound source in three dimensional space.

As a sound designer for large-scale public events, I can confirm that it works, and that it works well. I used a TiMax2 SoundHub in my design for the four-day Redpath Waterfront Festival in Toronto this past summer, which attracted up to 30,000 visitors a day to an outdoor interactive-immersive theatrical production celebrating the bicentennial of the War of 1812. Localizing the attention of such an audience to a group of wandering performers would otherwise have been a lost cause.

In fact, one of the event's producers was convinced that some of the loudspeakers weren't working, because she couldn't hear any sound coming out of them—until she marched up to within about 15 feet of the loudspeakers and finally could discern the direct sound coming from them. Until then, she had been absolutely certain that all the sound was emanating from the performers.

And that's exactly what's needed in a house of worship to keep attention focused where it belongs—on the speaker, not the loudspeakers. ◆

Veteran sound designer Alan Hardiman works with sound for large scale events and exhibits. Other recent projects include the original score for the dramatic film Joy, production of the documentary, Osprey Family Life on Georgian Bay, including original score, and interactive sound design for the Toronto Zoo's Tundra Trek.