Case Study – The Architectural Acoustics Digitool MX

As a Mixer for a Church with no Sound System Operator

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DRAFT 04-08-05 update 09-25-09

There are numerous churches whose method or style of worship includes the use of a full Praise and Worship Ensemble or Praise Band, of course the successful performance of these Praise Ensembles requires a quality sound system along with an experienced and skilled system operator. However, there are also many churches such as some Baptist, Catholic, Episcopal, Methodist, Presbyterian, and others that simply will not have a sound system operator. Some of these may be small churches and some may be regarded as large, there are a great number of churches that do not employ guitar, bass, and drums in their worship service. In many cases these are mainly Spoken Word churches that may utilize a little piano or organ, but generally have minimum requirements as to the number of mixer channels employed. However the lack of a system operator can result in less than ideal intelligibility due to the poor signal to noise ratio that is the result of having multiple open microphones at all times. What if churches like these could have a sound system that was fully automated in such a manner as to have all the benefits of a system that was indeed controlled by an experienced and skilled operator? Well the technology is now available to accomplish just this mission, via of a system that is controlled and managed by a state of the art digital signal processor.

A primary example of a sound system that was simply not up to task without the benefit of an operator existed in St. Patrick's Catholic Church here in Meridian Mississippi. St. Patrick's is of that architecture that is typical of the Cross or Crucifix shaped Catholic Church as it was built more than 100 years ago. It seats about 475 in the two rows of Pews within the Chancel, with overflow seats in the balcony for about 80 people. I was called in as a consultant in December of 2005. I evaluated the system components and re-configured the existing equipment. At that time they were using two wireless lapel microphones, one Pulpit microphone, one Lay Reader or Cantor microphone, and one Choir microphone.

The speaker system consists of 2 twelve inch low frequency transducers that are mounted into the apse of the ceiling above the chancel or sanctuary alter area. These two loudspeakers are mounted flush into the arch of the ceiling and are separated by a distance of about 10 feet. Since the chancel ceiling is vaulted in a conventional cathedral or arch type of architecture, the facet of the arch is such that the loudspeakers have a decent downward angle. However, the loudspeakers are significantly behind the front edge of the altar platform and perhaps as far as 30 (?) feet forward of the first section of pews in the nave. This is certainly not ideal, but to make things even worse is the fact that there are two horns that are a good 5 Feet forward of these two 12 inch low frequency transducers. Now the horns are of both Near Field and Far Field designs employing a 90 x 40 degree and a 60 x 30 degree horn respectively. The mouths or exits of the horns are lined-up, with the Long Throw (60 x 30) directly above the Short Throw (90 x 45), most likely for aesthetic or symmetry reasons. The result is that the compression drivers are not in the same plane and are therefore not aligned with each other as far as the acoustic centers or points of origin.

After verifying the performance of each component within the Sanctuary sound system, we replaced the Pulpit, Lay Reader, and Choir microphones with Peavey model PVM-480 electret condenser microphones, which have a very narrow super-cardioid pickup pattern. I then used a SIA Software Live Smaart measurement system to determine the needed calibration for the loudspeaker system equalization and some of the microphone inputs that had dedicated equalizers. Although the system exhibited improved intelligibility, it was still not as articulate, nor could we get it to be as loud as I would have liked. There just wasn't much gain before feedback. The environment of this acoustical space was extremely reverberant, as most would expect of a traditional church design such as this.

It was at this time that I decided to recommend an AA Digitool MX 8 x 8 matrix mixer as the solution to the remaining problems. Those problems being the number of constantly open microphones and the fact that the transducer components comprising the main system were spaced at extreme distances and were not able to operate as a true single point source.

The Architectural Acoustics Digitool MX matrix mixer is an 8 input by 8 output digital signal processor that has many features to allow for some of the most advanced applications of digital technology. Each of the eight inputs offers an input mode selection to set up the input for a Microphone or Line Level input. The Microphone input has Phantom Power, Phase Invert switch, and an input Gain control that is variable from +15 dB to +50 dB. The Line Input option offers a selectable gain structure with selectable input sensitivities of +12 dBu, +18 dBu, +24 dBu, and +30 dBu. There is a Trim control that has a range from –6.6 dB through +6.5 dB. There is also a channel Mute Switch as well as a channel Level control that has a range from –127 dB to 0 dB.

Each channel Input has a Filter function with the ability to choose one of the following Filters: Low Pass –12 dB per octave (LP_12), High Pass –12 dB per octave (HP_12). Low Pass –6 dB per octave (LP_6), High Pass –6 dB per octave (HP_6) Parametric Equalizer (PEQ) with a frequency range from 10 Hz to 20 kHz and Gain from –20 dB to +15 dB. The Bandwidth control has a designated range from 0.004 octaves to 100 octaves.

There are two types of All Pass Filters, A-P1 offers a filter that introduces –90 degrees of phase shift at a designated frequency followed by a continuing shift above the designated frequency that reaches –180 degrees. A-P2 offers a frequency dependent phase shift that performs a bandwidth definable phase shift through which the phase is shifted by 180 degrees. The All Pass filters are vastly under appreciated by most. The A-P2 is an excellent tool for overcoming the disparities caused in applications where adjacent drivers are overlapping in coverage, resulting in lobbing or comb filtering of the frequency response.

Each input also has a choice of a frequency selectable Low (L_SHF) and High (H_SHF) Shelving EQ, a Bandpass filter (BP), a Bandstop filter (BS), and Constant Directivity Horn EQ. Each Input has a Gating function, which offers adjustable Threshold, Attack, Hold, Decay, and Gate Floor. There is a mode switch associated with the Gate function that provides for a bypass and four modes of Automix functions. The EQ section of each input offers four more additional filters as outlined above in the Input Filter section. Each input also offers a Compression function with adjustments for Threshold, Ratio, Attack, Decay, and post compression Gain. As you can see that is certainly a lot of feature choices, and that is just on each of the eight inputs.

After the input section there is an assignable 8 x 8 Input to Output routing Matrix that allows you to choose which input you want to assign each output. The matrix section also provides for the ability to control the output gain of each input independently to each of the 8 outputs.

Each of the 8 outputs feature a Crossover module that allows for choices of Butterworth filter functions of -6, -12, -18, & -24 dB per octave, -12, and -24 dB per octave Linkwitz/Riley filters, and -12, -18, & -24 dB per octave Bessel filter functions. Each output offers 7 EQ filter functions as outlined above; LP_12, HP_12, LP_6, HP-6, PEQ, A-P1, AP-2, L_SHF, H_SHF, BP, BS, and Horn EQ. Each output also offers Compression, Delay, Phase reversal, Level, Mute, and adjustable maximum output levels in scales of +6 dBu, +12 dBu, +18 dBu, and +24 dBu. As you can see the Digitool MX has quite a powerhouse of signal processing capability within its' DSP engine.

Let's now look at how we solved the problems inherent in the existing sound system at St. Patrick's in Meridian. The very first thing we did is to employ the X-Over function of the Digitool MX to replace the analog electronic crossover that was in the system. Output 1 was designated to be Low Pass, so we created a Low Cut or High Pass filter set at 60 Hz using a –24 dB/Octave Butterworth filter function. 60 Hz was chosen after looking at the raw frequency response, and the system rolled off naturally below 60 Hz. The Low Pass or High Cut of Output 1 was set for 1260 Hz using a –24 dB per octave Butterworth filter function. Since a Butterworth filter is defined by the –3 dB down point, this setting actually placed the –6 dB down point of the filter at 1600Hz. However due to the natural roll off of the 12 inch loudspeakers the electro/acoustical X-Over point was more like 1200 Hz.

Output two of the Digitool MX DSP is assigned to the Near Field or short throw 90 x 45 degree horn.. The High Pass frequency is 1400 Hz and is a –24 dB Butterworth filter, which places the –6 dB down point at 1200 Hz The Low Pass (or High cut) frequency of output two is 18 kHz using a –12 dB per octave Butterworth filter. The delay setting to align this output with the 12" low frequency transducers is 5.113 milliSeconds or 5.8 Feet of delay.

Output 3 is assigned to the Far Field or long throw 60 x 30 degree horn. The High Pass frequency is 2.45 kHz (-3 dB) with a –24 dB per octave Butterworth filter, the –6 dB down point of this filter falls at 2 kHz. I chose to cross this second horn over higher to minimize problems in the acoustical summation between 1200 Hz and 2 kHz, as the Near Filed horn has a much greater vertical angle of coverage in this region. The Low Pass frequency or High Cut of output 3 is at 18 kHz with a –12 dB Butterworth filter. The delay setting is 6.476 milliSeconds or 7.3 Feet.

Output 4 of the Digitool MX is providing the Choir monitor mix, output 5 is for the Balcony, output 6 is the Cry Room, output 7 is for the Narthex or Foyer. These outputs (4 – 7) have each been high passed and equalized for a smooth response and also have appropriate delay settings to match the propagation time, along with an additional 20 milliSeconds of delay to account for the Hass Effect, to keep the sound image localized so that it appears to come from the Chancel or Alter area. Output 8 is designated as a recording output for such things as archiving wedding ceremonies. The interesting thing about the delay settings for both the Foyer and Cry Room is that with the doors closed, the sound appears to come from the local loudspeakers, but once the doors are opened, it is as if that speaker is shut off and the sound appears to come from the front of the sanctuary, The Digitool MX inputs are configured as follows; inputs 1 & 2 are for wireless lapel microphones, input 3 is the Pulpit microphone, input 4 is the Cantor or Lay Reader Mic, input 5 is the Choir Mic, and input 6 is a handheld wireless microphone. Each of the inputs have their respective gates set up to close or attenuate each microphone when no one is speaking into them. Each input has a High Pass filter (HP_12) set at 80 Hz with –12 dB per octave roll off and a Q of 0.707.

The Gates are generally set for a Threshold of –40 dB, with a 10 mS Attack time, a 800 mS Hold time, a 1000 mS Decay time, and a Floor of –100 dB. Since I was by myself working in the evening during this calibration, I had no one available to speak into each of the microphones while I adjusted the gates. I had given this some prior thought and my solution was to take a small FM radio and super-glue a threaded microphone mount to it. I put the radio on a Mic stand and set it to a Talk-radio program. I used an SPL meter to measure the loudness of my voice at the Mic position and set the FM radio level for the same sound pressure level. I was actually able to tune-in a Christian Talk-radio program, and interestingly enough (to me at least) they were having a discussion regarding the differences between the Douay Rheims, King James, New Jerusalem, NRSV, and other editions of the Bible. Under the circumstances, I thought this was better than some ranting talk host discussing politics.

The Digitool MX has a Priority function on channel 1. When the Priority function is engaged, as long as channel 1's Gate is open, then all other channels stay gated. In the future we are going to add an external Architectural Acoustics model D4S programmable wall-mount control panel, that offers 4-buttons for calling up system presets. Preset One will be for Saturday Mass when there is no Choir, and will have the Priority function engaged on the pastor's wireless lapel microphone with the Choir Mic muted. Preset Two will be the same as preset One, but with the Choir Mic un-muted for Sunday Mass. Preset 3 will be for the weekly children's Mass, where both the pulpit and lay readers microphones require more input gain due to the fact that some children do not speak out like an adult. Preset 4 will have the Priority function on channel one disabled, this is to allow for both the pulpit Mic and the lay reader Mic to be able to un-gate when necessary to allow for call and response prayers such as were done during Easter Vigil.

Over all the Digitool MX has greatly improved an otherwise marginal performing sound system. Yes the system is still actually too far from the nearest listener, but the DSP technology within the Digitool MX allowed us to correct the most inherent problems with the design. If you can, imagine the signal time-smear that the grossly un-aligned main system components had been causing. The Near Field horn fell on the congregation first, followed by the Far Field horn, whose acoustic center was effectively 1.5 Feet behind the Near Field horn. Then the low frequency information that started from two low frequency drivers (spaced 10 feet apart) arrived from a distance effectively 7.3 feet behind the Near Field information. Before the implementation of the Digitool MX and its' ability to spatially time align the four separate drivers, the multiple acoustical sources resulted in four discrete sources of sound in a highly reverberant environment in the first place. So the result was that the excitation of the reverberant filed was four times greater. The fact that the system is time aligned and the microphones are appropriately gated, results in a system with very defined intelligibility, and much less noticeable room reverberation. Yes, it is still a reverberant acoustic space, but compared to the amount of reverb perceived with the old system, it is a great improvement. Before the Digitool MX was added, with all of the microphones open at the same time, they each were each listening to the already excessive reverberation that resulted from the time smearing of the three different components of the original system with the four un-aligned drivers.