San Juan de Dios Sound System
Part 2
Implementation for English Mass

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Preface

San Juan de Dios in San Miguel de Allende is a colonial era church build around 1760. Like almost all such structures in Mexico it has terrible acoustic characteristics. The interior structure is almost entirely plaster or stone having virtually no acoustic absorption. The original sound system used non-directive loudspeakers. This caused the sound to be echoed and reverberated through the inside of the church making speech extremely difficult to understand.

This report describes the analyses, tests and changes that were made to improve the sound system.

Two software programs were used.

Ulysses is an acoustic simulation program where the interior of the church is modeled in three dimensions. The absorptive properties of all the interior surfaces can be independently specified. A data base is provided that includes the acoustic absorption values of hundreds of common construction materials. A speaker system can then be modeled, and each speaker independently located and aimed. The directional and frequency properties of each speaker are based on manufacturer's data measured in an anechoic (non-reflective) chamber and provided in another data base. The audience seating area can be defined, and sound calculations displayed over that area. These include the total sound level, the direct sound level, the difference between direct and reverberant sound levels and other properties, all of which vary with the frequency. Using this program, alternative wall treatments and speaker systems can be evaluated rapidly and without incurring any cost. Like any modeling program, however, it must be verified by test.

The second program used is called Smaart8, and it is an acoustic test data analysis program. It provides an excitation source that is fed into the sound system, and data is collected through a measurement microphone. The program then processes both data streams to calculate direct and reflected sound levels and many other acoustic properties, all of which again vary with the frequency. Smaart8's results can be used to update and verify the Ulysses model and to optimize the performance of the sound system through equalization, level adjustment, delay adjustment, etc.

The report is divided into parts. Part 1 documents the physical dimensions of the interior of the church and the development of the Ulysses acoustic model. Part 1 also analyzes the predicted performance of the existing sound system and the improvements possible using alterative loudspeaker configurations and wall treatments. Part 2 describes the implementation of the new sound system for the English Mass where the audience is limited to the front half of the church. It presents test data that allows comparison between the original system and the new one. It also compares predicted and measured data to confirm the acoustic model used in Ulysses. Changes required to expand the system to improve the performance in the back half of the church used for the Spanish Mass are also discussed. Part 3 explains a modification to the way clarity and direct sound is calculated and reanalyzes the measurements made in Part 2. It then presents four changes to the sound system to improve the performance at the back – replacement of the original loudspeakers with two additional HS1200's, delay compensation, level adjustment, and acoustic treatment of the rear wall. It also contains an expanded Summary and Conclusions that covers the results from all 3 parts.

This project has been successful based on both measured results and the opinions expressed by the church attendees. It is hoped that these results may be useful in making sound system improvements to other similar churches in Mexico.

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Introduction

Part 1 identified the need to replace the existing loudspeakers with more directional ones that would focus the sound on the audience area and away from the reflecting walls and ceilings. A new amplifier was required since the old one lacked sufficient power to drive the new speakers and had frequency response and distortion limitations.

In order to keep costs down, it was decided to initially implement a new sound system that would cover the front half of the church that is used for the English Mass. The old speakers could then be repurposed to cover the rear half of the church and the transepts for the Spanish Mass. Once installed and tested, the decision could then be made whether to expand the system to improve the rear half of the church.

This Part 2 report covers the initial amplifier and front half speaker replacement. Results are given for the entire church with no audience seating. (The presence of people makes a significant difference in the sound absorption, which is advantageous, but because of the way the sound measurements are made, only the empty pew case can be readily measured).

The new system works well – in the front half of the church, but as expected the rear half is lacking in sound uniformity and clarity. At the end of this report, the additions required to complete the project are identified. If implemented, these will be covered in a third part of the report.

Speaker System Concept

Based on the analyses in Part 1, the arrangement of the speaker system can be summarized as follows. The objective is to provide a uniform direct sound over the listening area and to minimize the sound reflected from the walls or ceiling.

The church can be divided into rectangular areas of roughly equal size, each covered by a loudspeaker. See Figure 1. More rectangles are better in that more speakers means that the audience is closer to a speaker and thus receives more direct sound. (In some churches pew mounted speakers are used to provide nearly personalized sound to each audience member). Of course, the cost increases with the number of loudspeakers.

For both aural and visual reasons, the speakers for each area should be located at the corner closest to the front of the church and nearest the outside walls. These are then aimed toward the diagonally opposite corner of that area. This keeps the sound away from the reflective walls. The speakers should be mounted at a height of approximately 2.5 meters (8 feet). Lower is better, but a height less than 2.5 meters is a potential hazard. The speakers should also be aimed down to a height of about 1.1 meters at the far corner of the rectangle. This is the normal head height when seated. By aiming the speaker at the farthest point, the direct sound level is made more uniform over the covered area.

Assuming the rectangular areas are not too elongated, the speakers should provide a horizontal beamwidth of about +/- 45 degrees. The vertical beamwidth should be even narrower to keep the sound focused on the audience and off the walls or ceiling. Achieving a narrow vertical beamwidth requires a line array type of speaker, and the narrower the beamwidth, the taller the array - which is costly. A good compromise is +/- 20 degrees. Speaker beamwidth is normally quoted as total rather than +/-. For example, a 40 x 90-degree speaker covers +/- 20 degrees vertically by +/- 45 degrees horizontal. Also, the beamwidth varies greatly depending on frequency, but for speech intelligibility the important range is between 1 and 4 KHz.

Church attendance affects the acoustic performance because the audience is the primary sound absorber in these old churches. The worst practical case is when the church is only partially filled. In that case, the audience would typically be near the front. The most important loudspeakers are of course those in the front that cover this audience. Those in the back should be switched off since they will only contribute to the reverberant sound field. When the church is filled, the speakers in the back should be turned on. These can be less directional (and less expensive) since the full audience will provide more absorption.

In Figure 1, the church is divided into 10 rectangular areas labeled A through J. Each is covered by a speaker mounted and aimed as shown. The most important (expensive) speakers are those covering areas A through D. All speakers are mounted at 2.5 m high except the ones at the front (A, B, I and J) which are mounted at 2.7 m. This looks better because the sanctuary (the unlabeled area at the top) and the two transepts have raised floors. Also, the rectangular areas covered (A, B, I and J) are more elongated, so the same tilt angle can be used for all the speakers if the front ones are elevated.

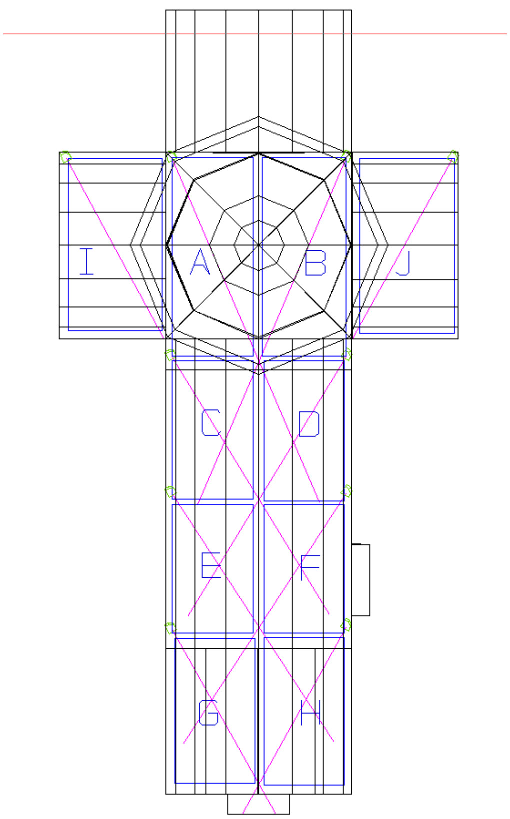

Figure 1 Church Divided into Rectangular Coverage Areas

The selected speakers for areas A – D are TOA model HS1200. These provide the desired 40 x 90-degree beamwidth and are of professional quality with excellent frequency response. The speakers used in E through H are the original Celestion KR1's which are not very directional. The speakers in areas I and J are the original Steren CS212 arrays. These are somewhat directional. Unfortunately, no data exists on either the KR1 or CS212 speakers.

All speakers contain built in line transformers that allow their volume to be individually set. Speakers for areas A and B which cover the largest area are set for 30 watts each. Speakers for areas C, D, E, F, G and H are set for 15 watts each. The transept speakers I and J are set for 12 watts each.

Zones

The new PAM245 amplifier allows up to 5 zones to be individually enabled. The speakers are grouped into zones as listed in Table 1.

Table 1 Speaker Zones

ZONE	Coverage Areas	Wire Color
1	A, B, C, D	BLUE
2	I, J	RED
3	E, F, G, H	GREEN
4	Outside	WHITE
5	Not used	

Wiring

AWG 18 wiring is used for all the speakers according to the color code in Table 1. In addition, a BLACK AWG 10 wire is used as the common ground conductor. All wiring is contained in new 20x12 mm (3/4 x ½ inch) plastic conduit. Junction boxes are placed near each speaker for connection using AWG 18 2-conductor speaker cable. The speaker cables connect to the 70.7 V zone terminals on the amplifier. The ground connection is split into 4 separate AWG 18 black conductors for connection to the amplifier terminal strip. See Figure 2 below.

NOTE: As discussed in Part 3, the use of a common return line proved to be a mistake because the stereo amplifier later used to provide delay compensation requires a separate return line for each zone.

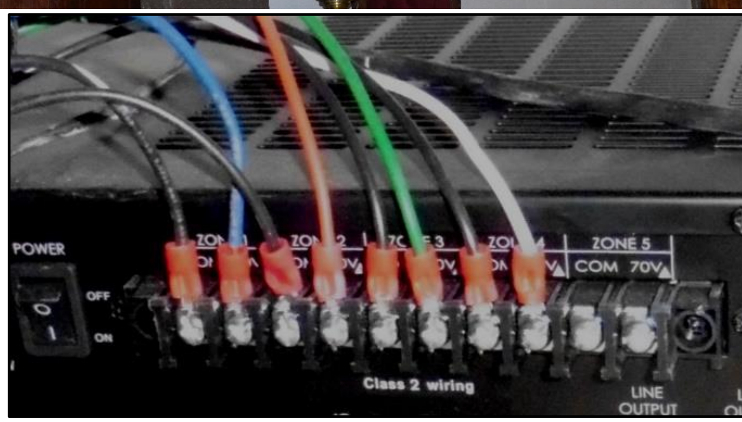

Figure 2 Channel Used for Speaker Wiring (top) and Speaker Connections to Amplifier (bottom)

Wall Attachment and Brackets

The conduit is held in place on the plaster walls using the adhesive backing on the conduit. Where short strips are used around the Cantera columns, the conduit is held using short masonry nails. Plastic anchors (*taquetes*) are used to hold the junction boxes and speaker brackets to the walls.

The 2-inch-long plastic anchors and lag bolts work well in the Cantera columns. Even the HS1200 speakers (which weight about 15 kg each including their brackets) secure well using 5 or 6 of these anchors. The plaster walls however require much longer anchors and lag bolts because the plaster is very thick but also very soft. It tends to crumble when drilled, and a long anchor is required to distribute the load. Both 80 mm (3-1/2 inch) and 100 mm (4 inch) anchors were used. All anchors required a 3/8-inch diameter hole. Lag bolts are ¼-inch hex.

The CS212 and KR1 speakers are mounted using their supplied brackets. The upper mounting bracket for the CS212 was extended using an aluminum strip so as to provide a 12-degree downward tilt of the loudspeaker.

TOA sells a 2-piece mounting kit for the HS1200 loudspeaker, but it is expensive ($289 each with shipping). Part numbers are HY-1200VW and HY-W0801W. In our case, a nearly identical bracket was fabricated and painted locally that provides a fixed 12-degree downward tilt angle and an adjustable pan angle. This bracket is shown below. The part numbers of the components required and cost with shipping are listed in Table 2.

Table 2 HS1200 Bracket Bill of Materials

Part	Part Number	Price US$
Speaker Yoke	100115B	13.41
Wall Mount Bracket	100116	17.16
Powder Coating	Beige	25.00
(4) Spacers (M6x26mm)	94669A356	9.20 + 20% = 11.04
(4) Bolts (M6x45mm)	91290A338	2.47 + 20% = 2.96
Hex Bolt ½-13x7/8”	92620A710	1.77 + 20% = 2.12
Hex Nut ½-13	94895A823	2.24 + 20% = 2.69
	TOTAL	74.38

Figure 3 HS1200 with Mounting Bracket

Measurement System

The measurement system used to test the sound system is shown in Figure 4. It uses the software program Smaart8 developed by Rational Acoustics and runs on a Windows 10 laptop computer. The hardware is connected through a Stereo Audio I/O device to the computer with a USB cable. Smaart8 includes a signal generator which for most of the tests was configured as a “synchronized pseudorandom noise source”. This drives the new amplifier which in turn drives the loudspeakers. It also is “wrapped around” as one of the inputs to the two-channel analyzer in Smaart8 and forms the reference signal. The second input comes from the measurement microphone which is positioned at various locations in the audience listening area.

Smaart8 has several operating modes which are used for these tests. Its “real time” mode provides a frequency response that can be used to adjust the amplifier bass and treble. This mode also measures the coherence between input and output signals which is an indication of the audio fidelity. The second operating mode is called the “impulse response”. This is not a real time mode, but rather it collects data over an interval which is subsequently processed to calculate not only the frequency response but also the response to an impulse such as would be produced by a balloon burst, pistol shot, etc. The impulse response separates the early arrival sound (direct) from the later arriving reflections and reverberations. This is especially useful in measuring the speech intelligibility as late arriving sound is the major reason for difficulty in separating sound consonants in speech.

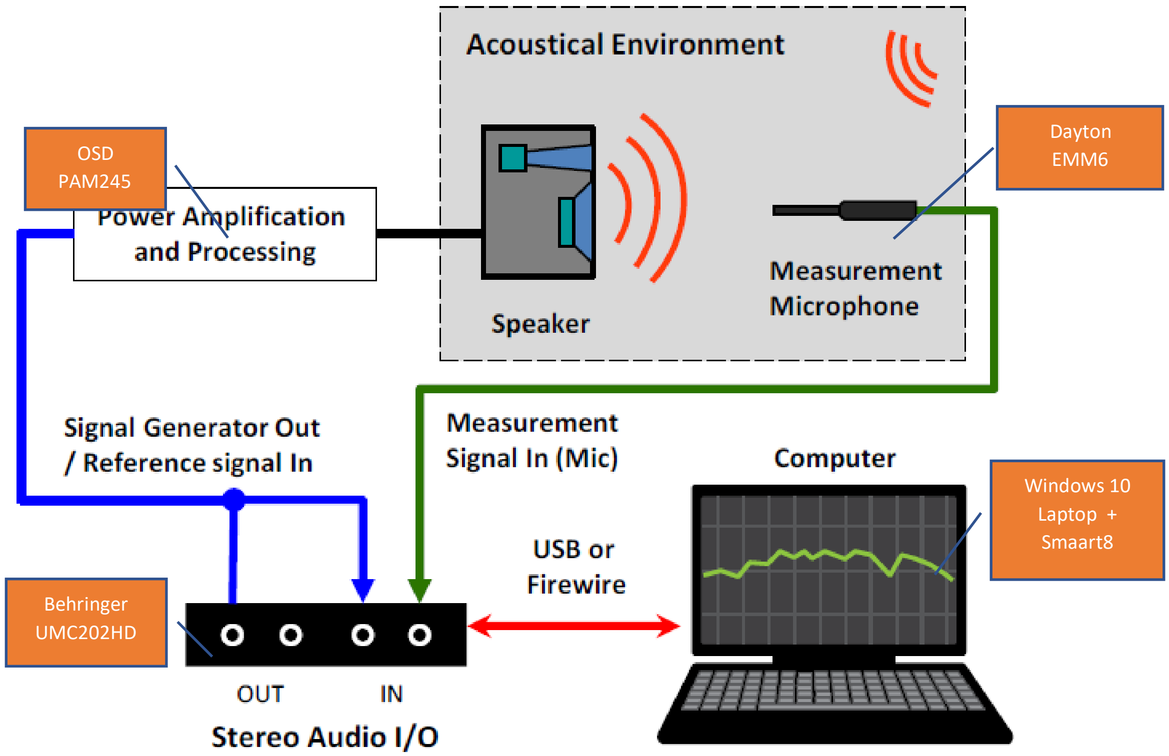

Figure 4 Block Diagram of Measurement System

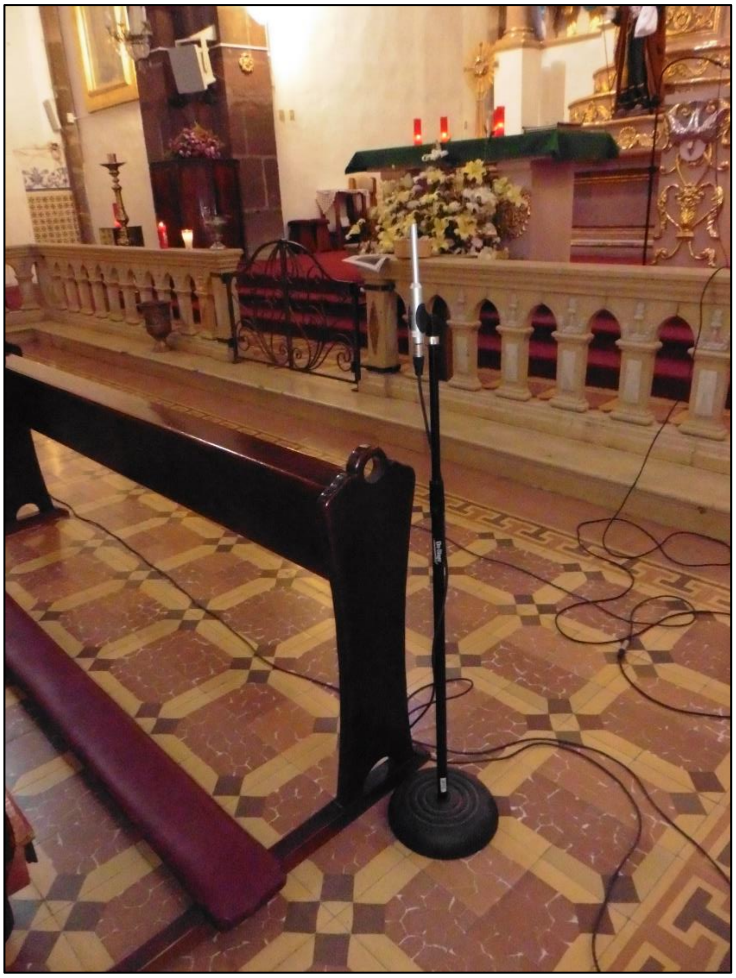

Figure 5 Measurement Microphone at Row 0 Right

Refer to Figure 6 below for the speaker and microphone measurement locations which will be used in the following discussion. The Zone 1 HS1200 speakers are labeled S1-S4. Zone 2 is the Steren speakers in the transepts, S9-S10. Zone 3 are the Celestion speakers S5-S8. The measurement locations are shown as the green squares and are identified by column location and row. For example, the center front measurement location is NC0 (Nave Center Row 0).

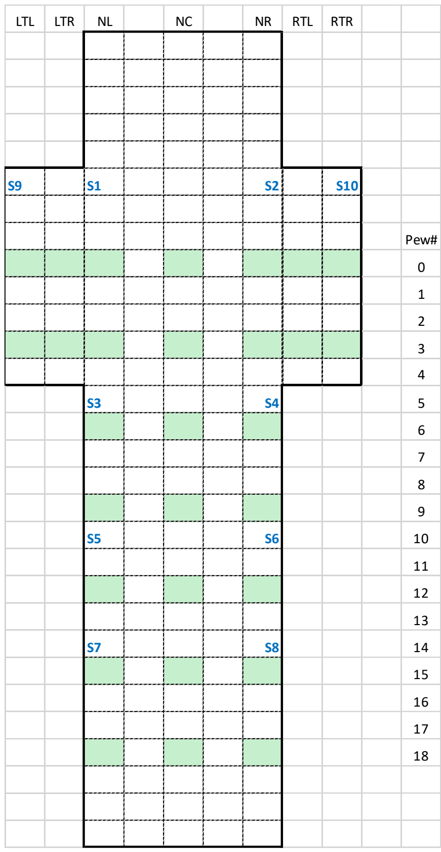

Figure 6 Plan View Template Showing Loudspeaker and Microphone Measurement Locations

Amplifier Replacement

The microphone was placed at the right side of row 6 where it is closest to a loudspeaker. In these first tests, this loudspeaker was a Celestion KR1. The frequency response was measured with the original Radio Shack amplifier and the new OSD PAM245 amplifier with nominal bass and treble settings on each. Figure 7 below shows that the new amplifier provides a much flatter frequency response.

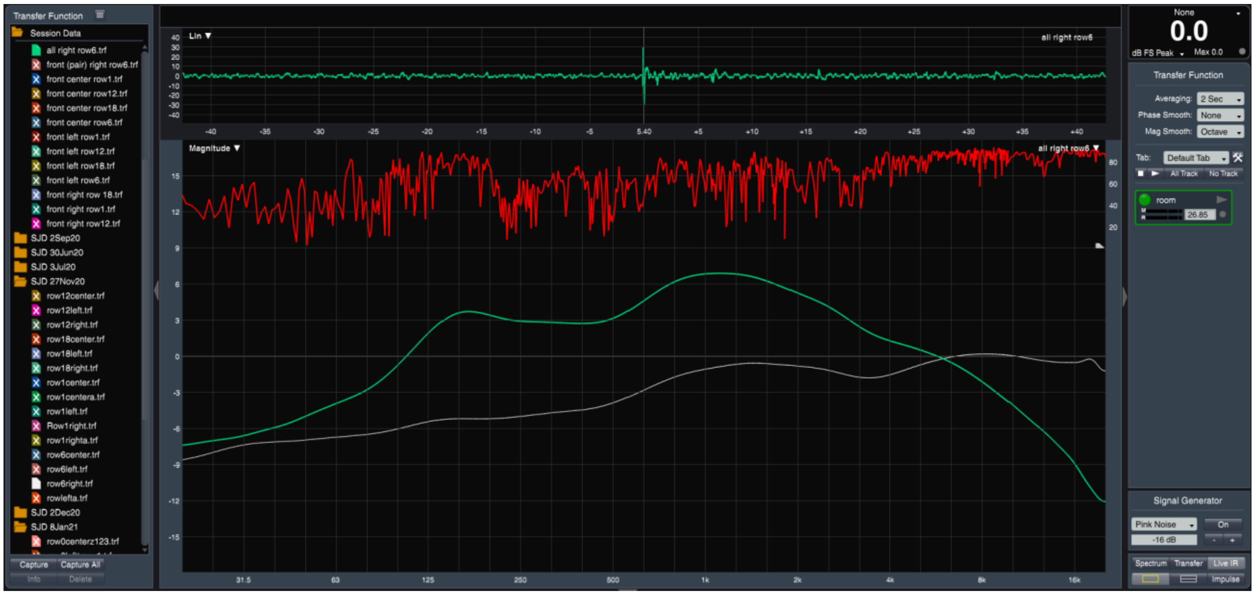

Figure 7 Frequency Response of KR1 Loudspeaker Using Old Amplifier (Green) and New Amplifier (White)

These tests were repeated at other locations including Row 1 which is closest to the Steren CS212 loudspeakers. Similar results were found at all locations; that is, the old amplifier lacked both high and low frequency response compared to the new one.

Loudspeaker Replacement

The new TOA HS1200 speakers were installed at the four corners of the crossing, the Celestions relocated to the rear sides of the nave, and the Sterens to the corners of the transepts. See Figure 8 and Figure 9.

Figure 8 HS1200's S1 and S2 at Front of Crossing, CS212 S10 in Right Transept

Figure 9 KR1's S5 and S7 at Back of Nave

Frequency Response and Coherence

The microphone was placed at the left, center, and right side of rows 0, 6, 12 and 18. It was also placed at the extension of row 0 in the middle of the left and right transepts.

With the microphone closest to the HS-1200 loudspeaker (row 6), the bass and treble were adjusted to obtain the flattest frequency response. This occurred when the bass control is at its 12:00 position and the treble is at maximum (5:00).

Figure 10 below compares the frequency response for the microphone at the right-side row 0/1 in front of the Steren loudspeaker (yellow trace) and the TOA loudspeaker (white trace). In both cases the frequency response is flat (+/- 3 dB) from 75 Hz to 16 KHz.

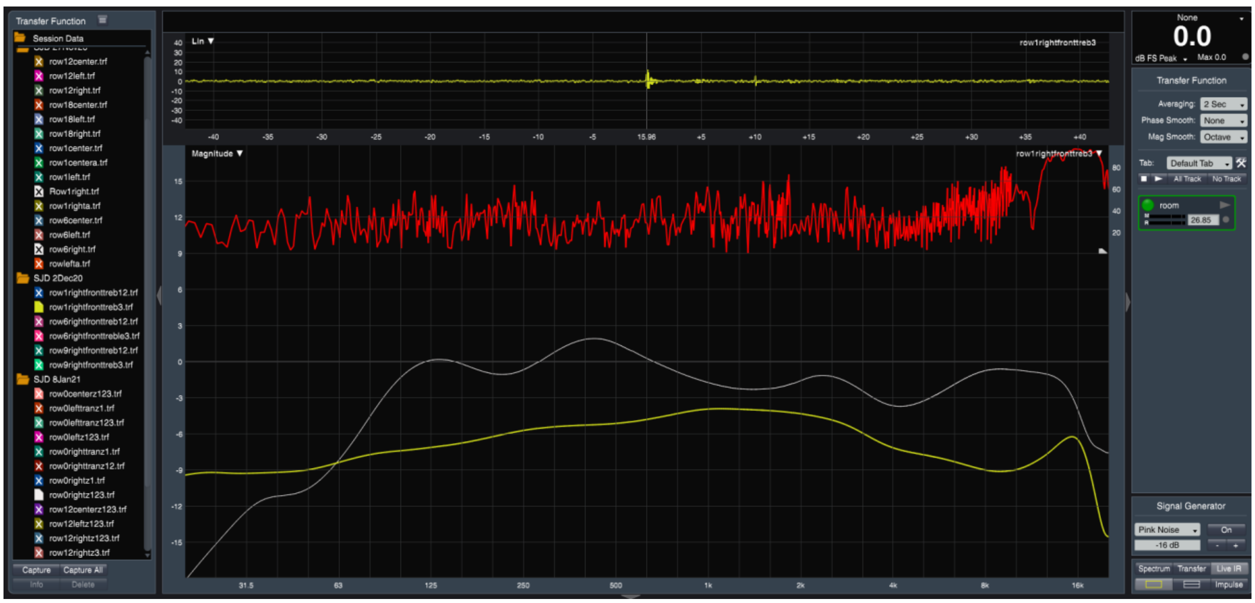

Figure 10 Frequency Response at Row 0/1 Right with CS212 Loudspeaker (Yellow Trace) and HD1200 Loudspeaker (White Trace)

In Figure 11, the frequency response is shown for all 12 locations. Since the volume control was not changed for these measurements, this plot also shows the variation in overall sound volume throughout the listening area. The level variation is about 6 dB (+/- 3 dB) which is considered acceptable.

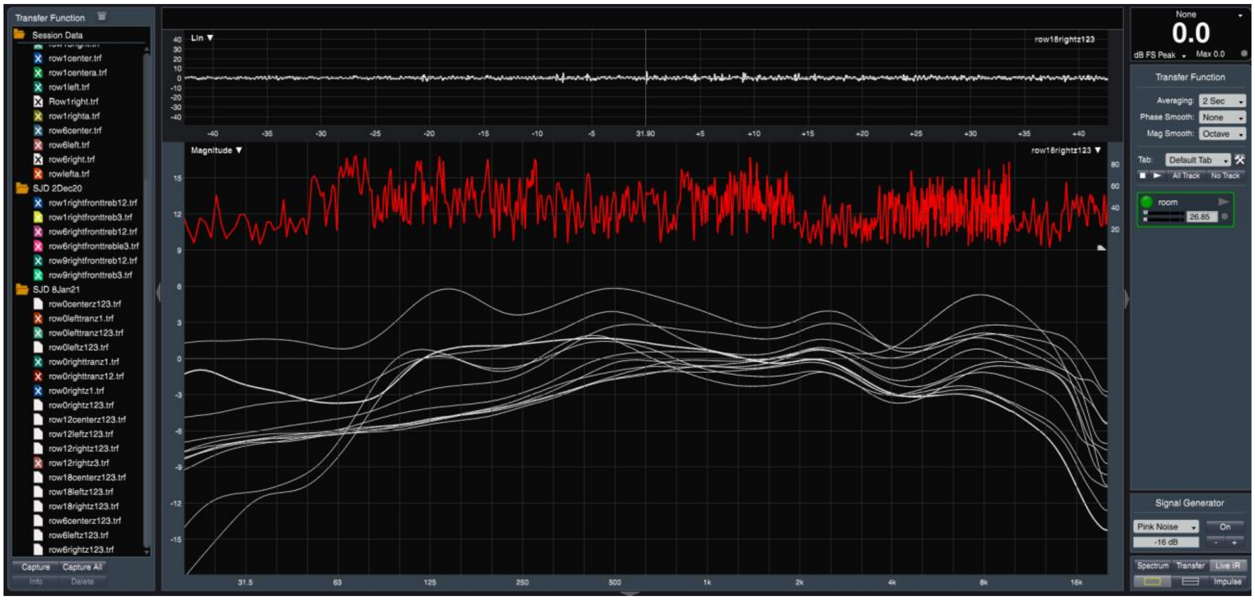

Figure 11 Frequency Responses Measured at Left, Center and Right Side of Rows 0, 6, 12 and 18 with New Speaker System

In addition to the frequency response, these plots also show the coherence (red trace). Since there is negligible noise with the church doors closed and very low distortion with the new amplifier, the coherence is a measure of the reverberation. It varies between 0 and 1 (100%) with the highest value indicating the best fidelity (lowest reverberation). In the frequency range 1 to 2 KHz, with all zones ON, the "average" coherence is as indicated in Table 3 below.

Table 3 Coherence with New Loudspeakers

	Left Transept	Left	Center	Right	Right Transept
Row 0	45%	70%	55%	70%	50%
Row 6		70%	50%	70%	
Row 12		45%	45%	45%	
Row 18		45%	40%	45%	

So, the coherence is better at the front where the new speakers are installed. It is lower at the center where the left and right-side speakers can interfere at certain frequencies.

For comparison, Table 4 below lists the average coherence in the 1-2 KHz range for the new amplifiers with the original loudspeakers.

Table 4 Coherence with Original Loudspeakers

	Left	Center	Right
Row 1	40%	45%	40%
Row 6	60%	50%	60%
Row 12	35%	35%	35%
Row 18	35%	30%	35%

The new speakers improve the coherence everywhere, even at the back.

Impulse Response Measurements

The same measurement setup is used as for the real time frequency response, but the data capture takes additional time and is processed afterwards (not in real time) to produce the impulse response, frequency response and audio performance measurements such as the clarity ratio. A typical impulse response is shown in Figure 12 below. This is for zone 1 ON with the measurement microphone at row 0 left.

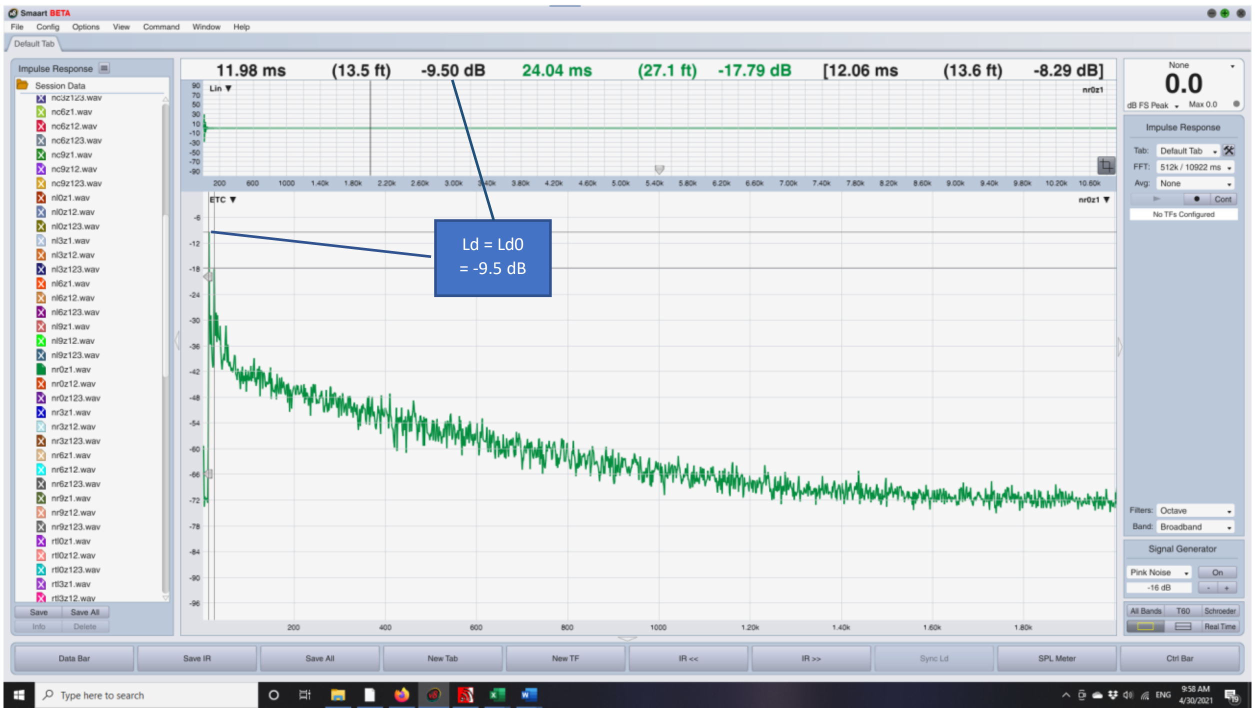

Figure 12 Impulse Response Row 0 Left with Zone 1 On

A great deal of information may be gleaned from the above impulse response. If the system were ideal (flat frequency response, no noise, no reverberation and only one loudspeaker), the impulse would be a single vertical spike on the left side of the plot at the time 12 msec. This is the delay time it takes the closest loudspeaker (front left) to reach row 0 on the left. In the actual plot shown, there is a second lower spike that occurs at 24 msec. This is the arrival of sound from the right-hand front loudspeaker. The closest speaker has an amplitude, Ld of -9.5 dB and the second front speaker has an amplitude of -17.8 dB. Additional pulses are visible that correspond to the sound coming from the back of the two loudspeakers mounted at row 5. These sounds all arrive within the first 35 msec or so of the first (direct) sound, and they are generally beneficial to hearing. There are a few additional spikes that occur later in time. These are reflections from walls, most likely the South wall of the transept and the South wall of the nave. These interfere with audio clarity. They could be reduced by placing sound absorption on the reflective walls; however, they are low enough that this is probably not warranted.

The sound level continues to decline with time, reaching the noise level of the measurement, Ln. In the case shown above, this is dominated by the noise generated internal to the microphone itself, but it is very low. (Recall that all data shown is on a logarithmic scale, so the ration of Ld to Ln which is 68 dB represents a power ratio of 6.3 million to 1). The time required for the reverberant sound to decrease 60 dB is known as RT60 and this is automatically calculated from the impulse response. RT60 does not depend on the location of the measurement, but on the physical volume, surface area and acoustic absorption of the building. For San Juan de Dios, it is about 2.5 seconds.

The program calculates the magnitude of the direct sound, Ld, and the ratio of sound energy that arrives early (within 35 msec of the direct sound) to the late energy. This is known as the clarity ratio, C35. It also calculates many other acoustic figures of merit.

Measurement of Direct Sound

Smaart8 automatically selects the largest received pulse and calls it Ld. Usually, but not always, this is the first pulse received. The exception occurs when the closest loudspeaker is mounted along the same row as the listening position. Then the largest pulse may come from a speaker mounted in a row closer to the front of the church.

The direct sound level can always be controlled up or down with the volume control, but it is important to have it uniform over the listening area. But as mentioned, the human ear integrates sounds received over an interval of 35 msec or so in a constructive manner, so we would really like to have the total sound level over this period. Smaart8 provides the largest peak and calls it Ld, but to distinguish it from the integrated direct sound, in the following the initial pulse will be called Ld0 and the integrated direct sound level Ld35.

Two methods are used to determine Ld35. The individual pulses in the impulse response can be combined using a root-summed-square (RMS) method as will be shown in the following examples. The second method used was to measure the total sound pressure level (SPL), with a calibrated meter, then use the ratio of direct to reflected sound calculated by Smaart8 from the impulse response to separate the direct from the reflected sound.

The newest version of Smaart8 provides an alternative method for measuring the direct sound. It uses a time windowed real time frequency response measurement rather than the impulse response, and it is known as the FTW option. From the description, it appears that it might produce a result similar to Ld35, but it has not been evaluated because it would require remeasuring all of the data in a different mode.

Row 0, Center Aisle

Row 0 is actually the kneeler that is normally centered directly below the dome. In this and the following plots shown in this section, the impulse response has been expanded to show just the first 180 msec.

With all speakers on, Figure 13 shows the initial response:

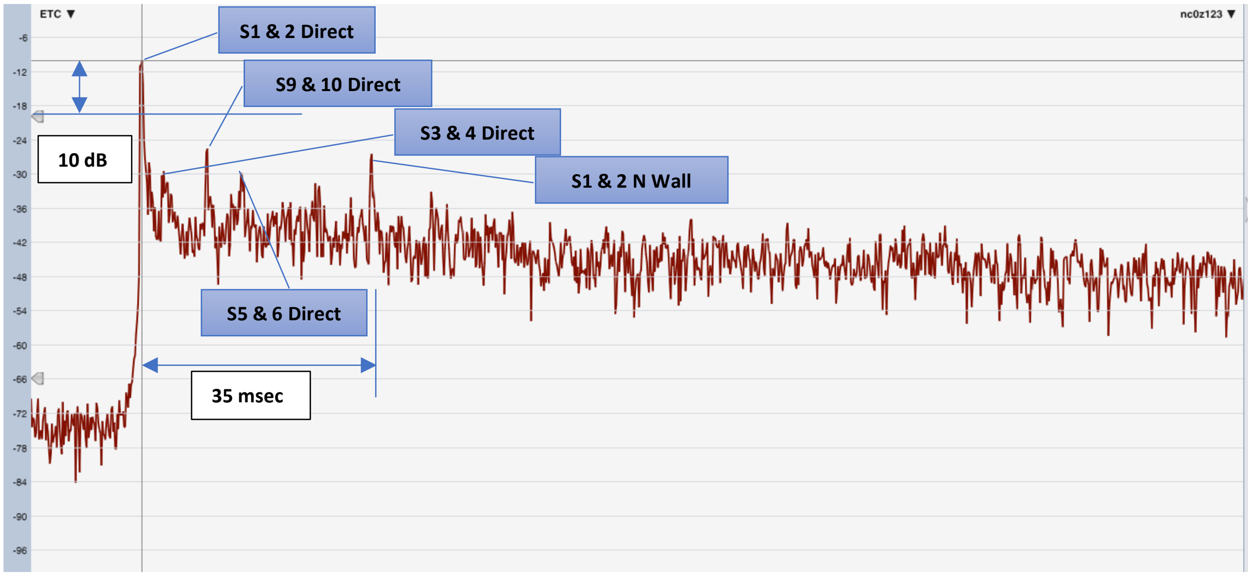

Figure 13 Expanded Impulse Response Row 0 Center with All Speakers On

The front two main speakers (S1 and S2) dominate. The direct signal can be seen from the transept speakers (S9 and S10), the main speakers at the rear of the crossing (S3 and S4) and even two of the rear speakers (S5 and S6). A reflection from the N wall of the church (the sanctuary) from S1 and S2 is apparent. But everything except the direct signal from S1 and S2 is more than 10 dB below the direct signal, so it does not contribute significantly to the direct signal. So Ld0 = Ld35 = -10.2 dB. The clarity ratio, C35 = 3.9 dB which is good.

Sound levels that are more than 10 dB below the highest peak can be ignored when calculating the RMS value for Ld35. This is because when two values 10 dB apart are combined, the result is only 0.4 dB different from the largest value. So, to simplify the calculations, both the 35 msec interval and the 10 dB levels are shown on the impulse responses.

The various direct and reflected sounds were identified by running the same case (speakers and microphone locations) in Ulysses using the Ray method of calculation with 1 reflection. This produces a very similar impulse response that allows each pulse to be easily identified on the Ulysses three-dimensional model. For example, the above case in Ulysses is shown below. Figure 14 is the impulse response and Figure 15 shows the corresponding ray for the S9 Direct signal. (It is the red colored ray).

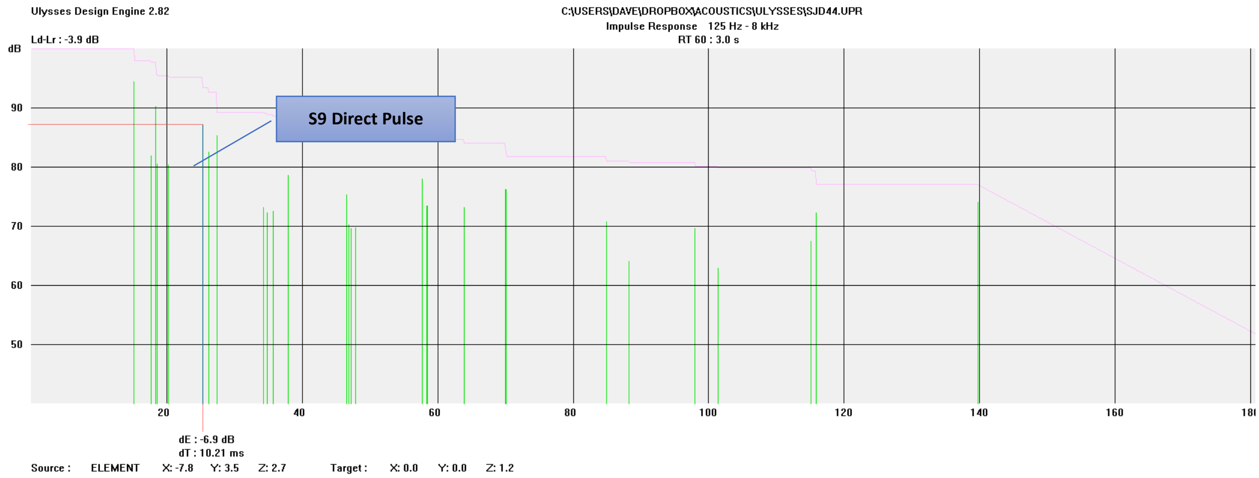

Figure 14 Ulysses Impulse Response Row 0 Center with All Speakers On

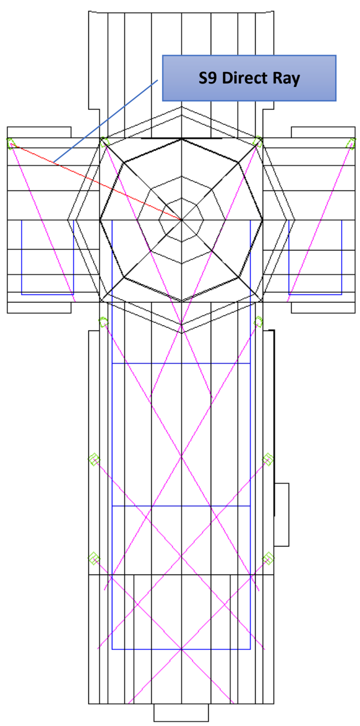

Figure 15 Ulysses Impulse Response Ray Display Row 0 Center with All Speakers On

Row 6, Center Aisle

This location is just behind the crossing, so it is picking up both the main speakers S1 and S2 at the front of the crossing and the main speakers S3 and S4 at the rear of the crossing (row 5). See Figure 16. These speakers contribute about equally at this location, but the front speakers S1 and S2 are delayed about 16 msec behind S3 and S4. This delay will not be noticeable because it is well within the 35 msec window. Speakers S9 and S10 are also heard, about 10 dB below the main speakers. Ld0 is -15.4 dB while the RSS total over the 35 msec interval is Ld35 = -12.6 dB. The clarity ratio is C35 = 4.0 dB which is good.

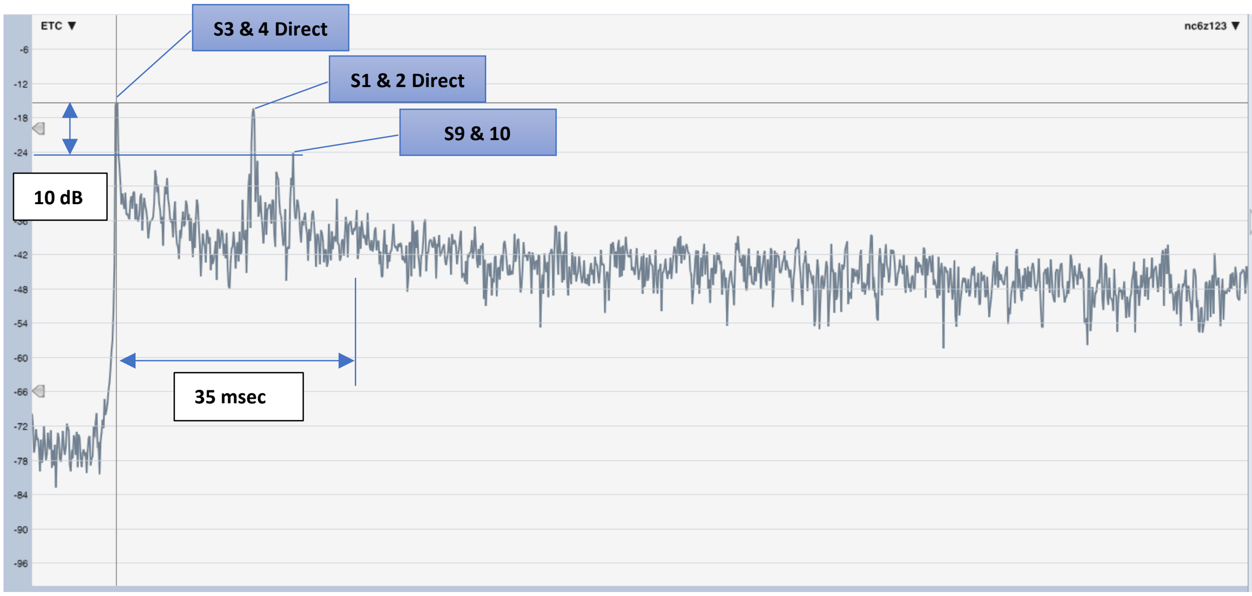

Figure 16 Expanded Impulse Response Row 6 Center with All Speakers On

Row 12, Center Aisle

This row is about midway between the rear speakers at row 10 (S5 and S6) and the additional rear speakers at row 14 (S7 and S8). The direct signal (Figure 17) from all of these speakers (except the transept speakers S9 and S10) can be observed. All the rear speakers arrive simultaneously at this location and produce Ld0 = -11.8 dB. The additional direct and floor bounce signals from S3 and S4 cause the RSS total to add up to Ld35 = -10.2 dB. Note that the direct and side wall bounce signals from S1 and S2 are 14 to 15 dB below the Ld0 signal, and they are outside the 35 msec window so they will add to the reflection. This makes the clarity ratio C35 only 2.6 dB which is considered only fair.

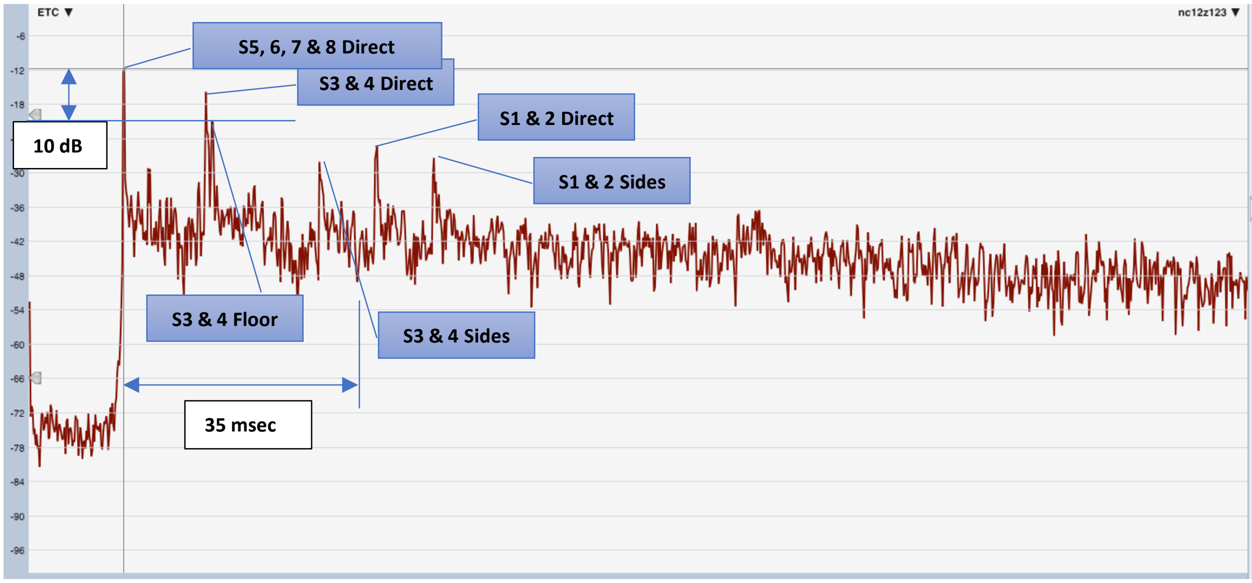

Figure 17 Expanded Impulse Response Row 12 Center with All Speakers On

Row 18, Center Aisle

This location is the extreme back of the church where all loudspeakers (except those in the transepts) are within the line of sight. The largest pulse (see Figure 18) is the direct one from S5 and S6 along with their floor bounce, but the first pulse comes from S7 and S8 along with their floor bounce. There is also a direct signal from S3 & S4 that is within the 10 dB range shown and within the 35 msec interval. Smaart8 identifies the largest pulse as Ld0 = -23.6 but the RMS total of all pulses within the 35 msec interval shown is Ld35 = -20.8 dB. The other large pulses from S1 and S2 and the reflections from S3 and S4 will add to the reverberation and decrease the intelligibility. C35 calculated by Smaart8 is -1.8 dB which is fair, but this is based on the 35 msec interval beginning at the S5 and S6 peak. Using the interval as shown below, C35 will be lower and most likely fall into the poor category.

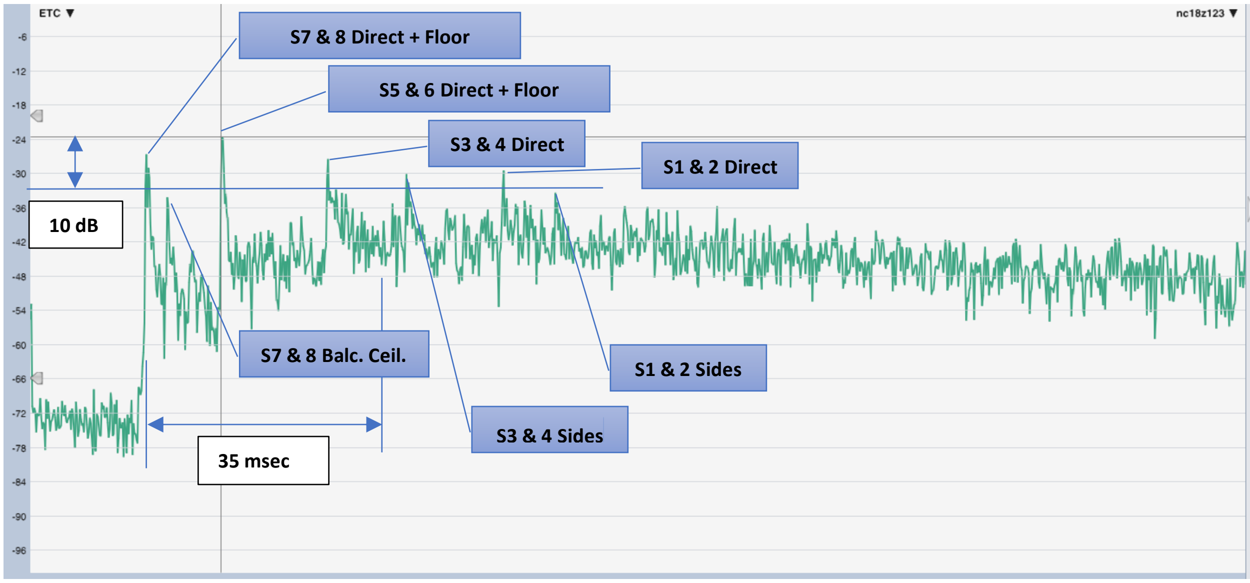

Figure 18 Expanded Impulse Response Row 18 Center with All Speakers On

Transept

In Figure 19 below, the listening location is left transept, row 3, right side. There is a significant sound from the two main speakers, S1 and S2 including the floor bounce from each. When the transept speakers are turned on, the direct signal from S9 (in the left transept) adds nearly as much as the S1 direct signal thus increasing the total direct volume by about 3 dB at this location. S9 and its right transept companion, do not propagate significantly beyond their respective transepts. Ld0 is -16.0 dB. The RSS total of all pulses above -26 dB (and within the 35 msec window shown) is Ld35 = -11.8 dB. The clarity ratio, C35 is 5.6 dB which is very good.

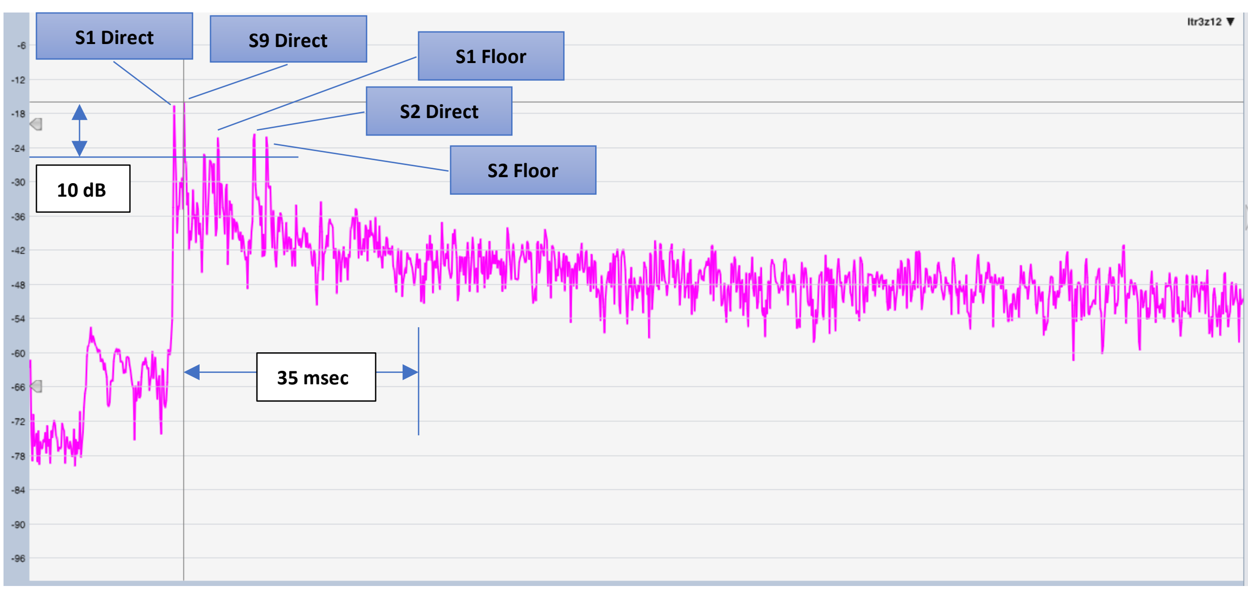

Figure 19 Expanded Impulse Response Row 3 Right Side of Left Transept with All Speakers On

Direct Signal Uniformity

The above process was continued for all the measured cases and the results were transferred to the following plan views where they are used to indicate the direct signal uniformity over the listening area.

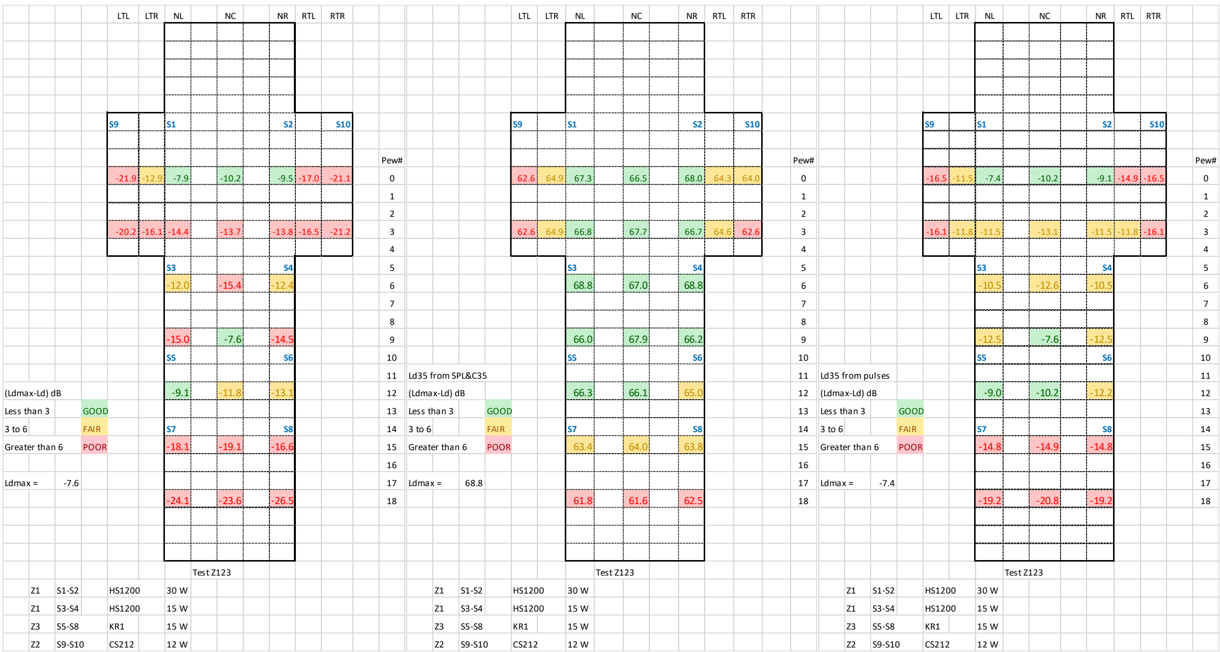

Figure 20 Direct Signal Uniformity Measurements Based on Initial Pulse (left), SPL and C35 (center), RMS of Pulses over 35 msec (right)

The left is Ld0, the center is Ld35 calculated from the measured total SPL and C35, and the right is Ld35 calculated by the RSS addition of the peaks on the impulse responses over the initial 35 msec interval. All three results are similar. The Ld0 is certainly the easiest to produce from the Smaart8 data, but the other two are probably more accurate. In any event, the conclusion to be drawn is that the uniformity is fair to good except at the far sides of the transepts (where coverage is not important) and at the rear of the nave. But as mentioned at the outset, the goal of the present effort is to improve listening in the front half of the church. The direct signal uniformity is fair to good now in that area.

Clarity - Direct to Reflected Sound Ratio

The ratio of direct to reflected sound with the crossover time equal to 35 msec is called C35, and it is calculated by Smaart8. These results were transferred to the plan views shown below.

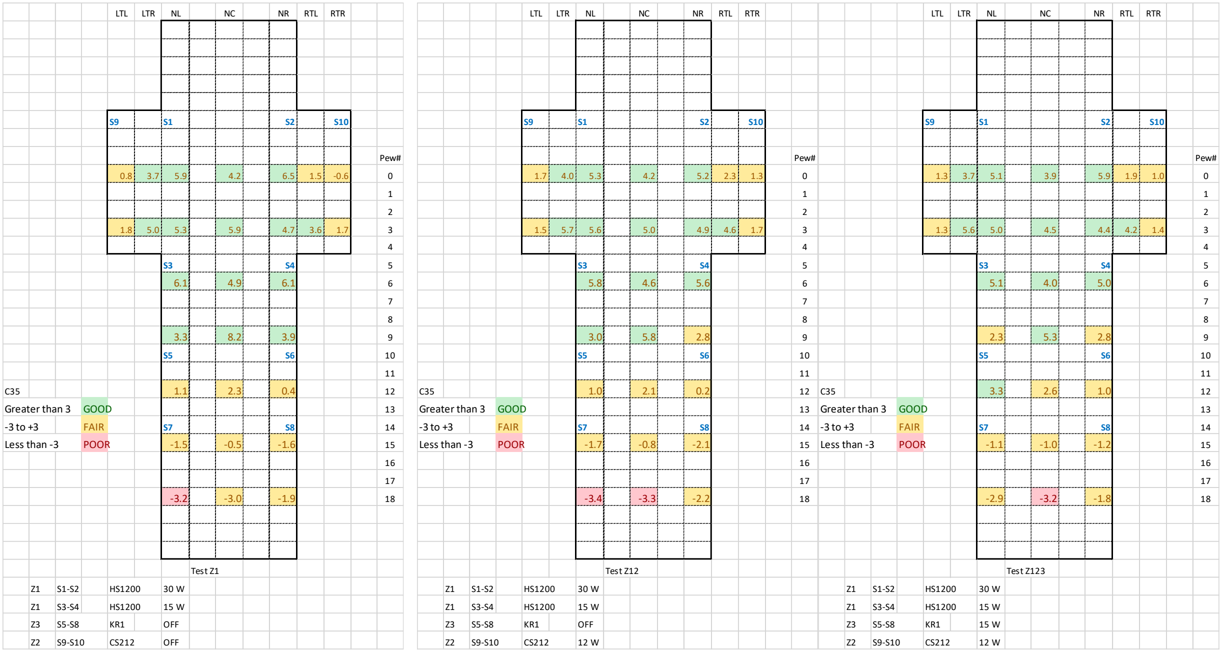

Figure 21 C35 Measured with Zone 1 On (left), Zones 1 and 2 On (center), Zones 1,2 and 3 On (right)

The three cases shown are zone 1 on only (left), zones 1 and 2 on (center) and all three zones on (right). Activating zone 2 makes a small but probably worthwhile improvement in the transepts. Activating zone 3 improves the clarity at the rear of the church but degrades the clarity in the front half. Therefore zone 3 should be switched off for the smaller English Mass. Clarity in the front half of the church is fair to good.

Speaker Delay

The lack of speaker delay compensation is most evident in the impulse response at the center of row 18 shown above in Figure 18. The direct signals arrive as listed in Table 5 below.

Table 5 Sound Propagation Delays Measured at Row 18, Center Aisle

Loudspeaker	Time to reach Row 18, center aisle	Delay relative to S7 and S8	Amplitude of Impulse
S1 and S2	70 msec	53 msec	-29.5 dB
S3 and S4	44 msec	27 msec	-27.5 dB
S5 and S6	28 msec	11 msec	-23.6 dB
S7 and S8	17 msec	0	-26.7 dB

If S7 and S8 were delayed 53 msec, and if S5 and S6 were delayed 53 – 11 = 42 msec, and if S3 and S4 were delayed 53 - 27 = 26 msec, the direct sound would arrive at row 18 simultaneously from all speakers. This should improve the clarity in the back half of the church significantly.

Improvement Relative to Original Sound System

The impulse response was recorded at selected locations before any of the changes were made to either the amplifier or the loudspeakers. This allowed the RSS direct sound Ld35 and the direct to reflected sound ratio C35 to be compared to the new system.

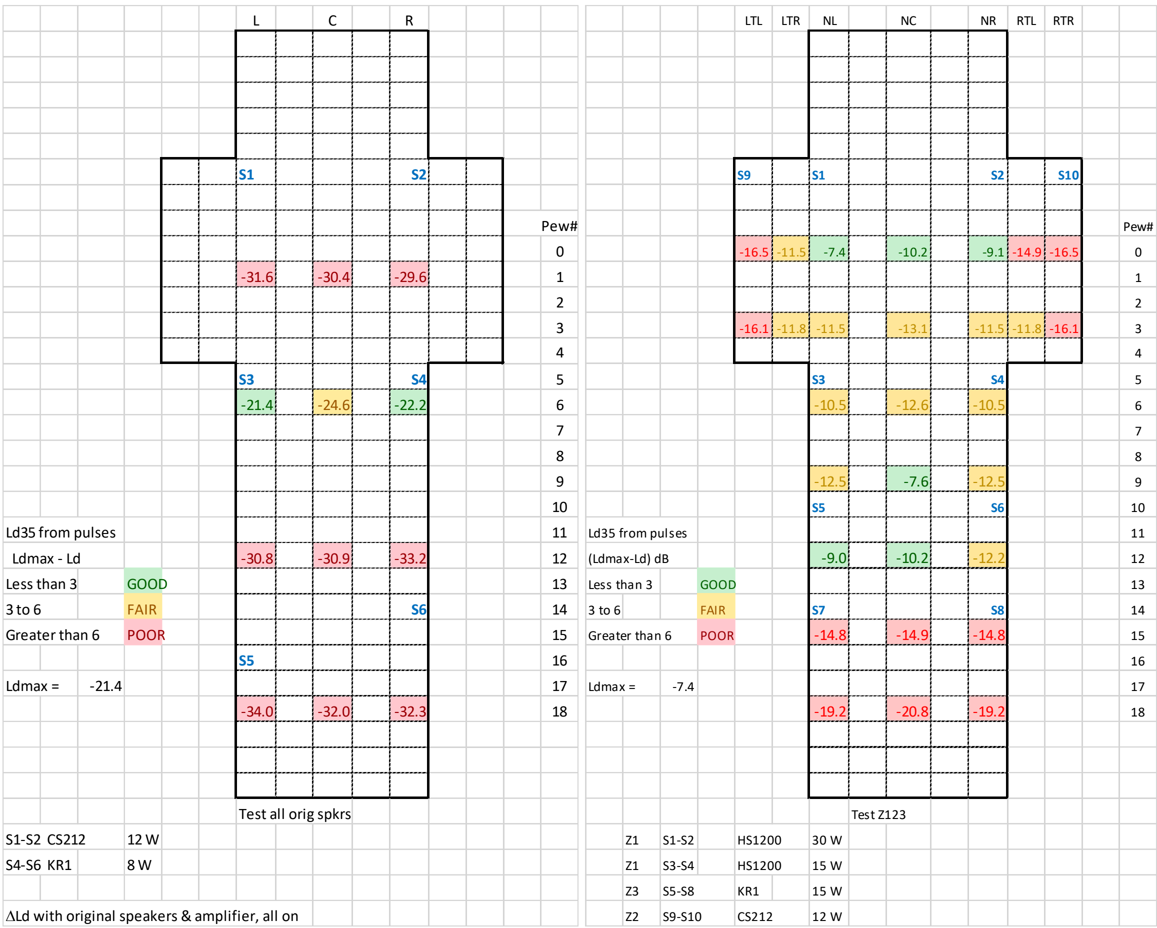

Figure 22 Direct Sound Uniformity with Original Sound System (left) and with New Sound System (right)

The comparison shown in Figure 22 indicates that the uniformity has improved a lot.

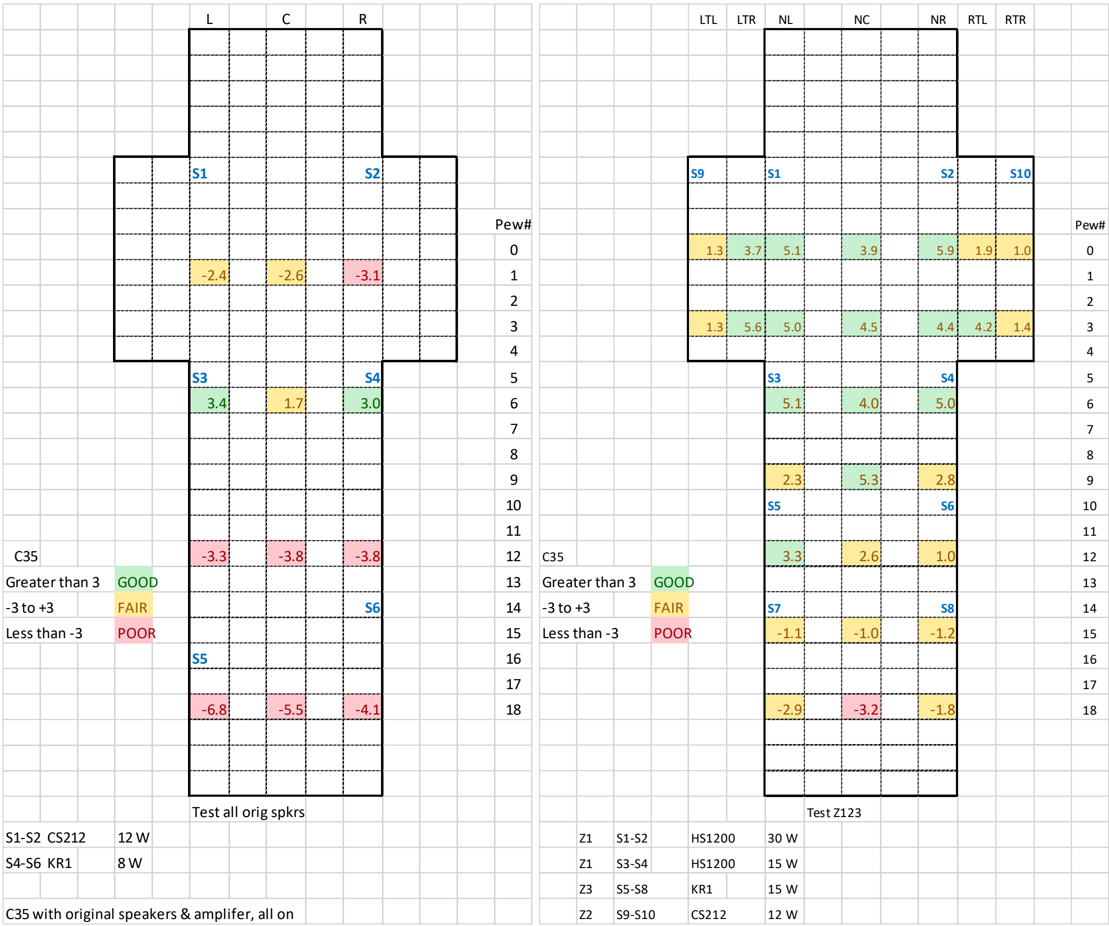

Figure 23 Clarity C35 with Original Sound System (left) and New Sound System (right)

Figure 23 shows that the clarity has also improved significantly.

Comparison of Ulysses Predictions to Smaart8 Measurements

These comparisons are difficult to make because Ulysses and Smaart8 define the direct and reflected sound differently. In Ulysses the direct sound Ld using the "level and time" calculation consists of the direct sound from all sources plus the first reflection, provided that the reflection occurs within the first 35 msec of the first direct signal. This is fairly close to the way Ld35 was manually calculated above from the impulse response measured by Smaart8. The comparison is shown in Figure 24.

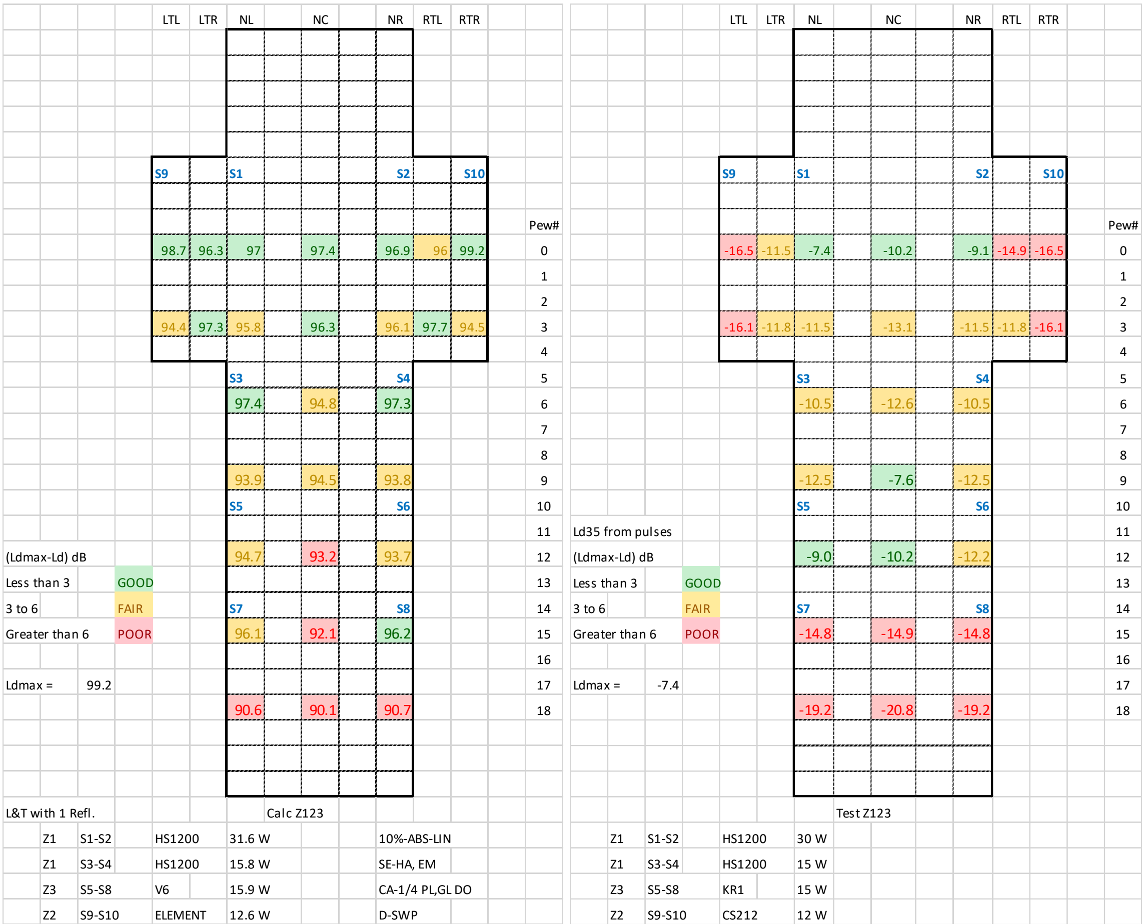

Figure 24 Uniformity Predicted with Ulysses Model (left) and Measured with Smaart8 (right)

Ulysses does not provide a measurement of C35, but it does calculate Ld-Lr which is similar. Ld is calculated the same way as described above, but Lr is calculated from the total reverberant sound field. Lr does not include any discrete reflections as it provides a different tool (the Ray calculation) for this. Still, the results are similar. Probably the biggest problem with the Ulysses model is the lack of manufacturer's data for the Steren CS212 speakers in the transepts and the Celestion KR1 speakers used at rows 10 and 14. The comparison is shown in Figure 25.

	LTL	LTR	NL		NC		NR	RTL	RTR	Pew#
	S9		S1				S2		S10	
	1.1	-1.2	-0.6		-1.3		-1	-1.7	1.7	0
										1
										2
	-4.0	1.35	-2		-2.3		-1.3	1.71	-3.7	3
										4
			S3				S4			5
			-1.1		-4.6		-0.9			6
										7
										8
			-5.6		-2.9		-5.6			9
			S5				S6			10
										11
Ld - Lr			-3.9		-6.8		-4.3			12
Greater than 3 GOOD										13
-3 to +3 FAIR			S7				S8			14
Less than -3 POOR			2.13		-6.5		-2.2			15
										16
										17
			-7.7		-8.4		-7.7			18

L&T with 1 Refl.			Calc Z123	
Z1	S1-S2	HS1200	31.6 W	10%-ABS-LIN
Z1	S3-S4	HS1200	15.8 W	SE-HA, EM
Z3	S5-S8	V6	15.9 W	CA-1/4 PL,GL DO
Z2	S9-S10	ELEMENT	12.6 W	D-SWP

	LTL	LTR	NL		NC		NR	RTL	RTR	Pew#
	S9		S1				S2		S10	
	1.3	3.7	5.1		3.9		5.9	1.9	1.0	0
										1
										2
	1.3	5.6	5.0		4.5		4.4	4.2	1.4	3
										4
			S3				S4			5
			5.1		4.0		5.0			6
										7
										8
			2.3		5.3		2.8			9
			S5				S6			10
										11
C35			3.3		2.6		1.0			12
Greater than 3 GOOD										13
-3 to +3 FAIR			S7				S8			14
Less than -3 POOR			-1.1		-1.0		-1.2			15
										16
										17
			-2.9		-3.2		-1.8			18

			Test Z123
Z1	S1-S2	HS1200	30 W
Z1	S3-S4	HS1200	15 W
Z3	S5-S8	KR1	15 W
Z2	S9-S10	CS212	12 W

Figure 25 Direct to Reflected Sound Predicted with Ulysses (left) and C35 Measured with Smaart8 (right)

The Ulysses model of the church that was used for these comparisons is called SJD44. This model is like the one used in Part 1 of this report except for the following changes:

- The walls were pushed out slightly to conform more closely with those of the actual building. In the previous models, the walls were aligned with the four columns used to support the dome.
- The listening area was enlarged to include the aisles where the microphone was located to make the measurements.
- The floor area was treated as empty hard seating. Wall or ceiling absorption is 10% at all frequencies.
- The mounting locations of the loudspeakers were moved to locate the actual speaker locations more accurately. The speaker types were also changed. HS-1200 speakers were mounted at the four columns, a generic column array was mounted in the corners of the transepts, and four V6 speakers were mounted to the walls in the rear of the nave to simulate the KR1 speakers actually used. These were mounted at rows 10 and 14.
- The speaker aim points were adjusted and the power to each of the speakers was set to match the power settings of the installed speakers.

The model for SJD44 is shown in Figure 26 below.

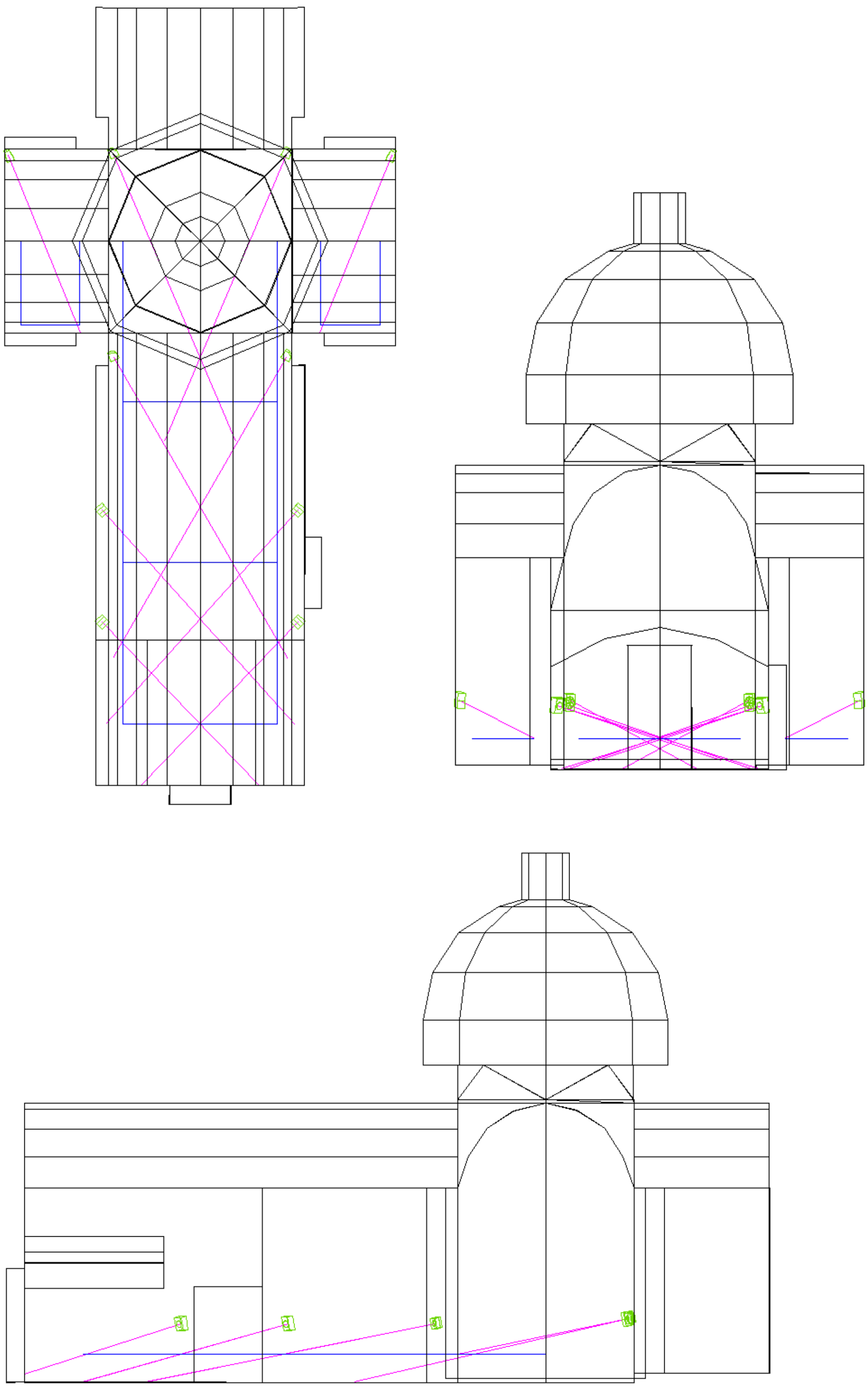

Figure 26 Ulysses Model SJD44

Changes Required to Improve the Performance at the Back of the Church

The performance is now adequate for the front half of the church but to improve performance at the back, the following changes should be made.

Replace the KR1 speakers at Rows 10 and 14 with a Pair of HS1200 Speakers at Row 11.

Actually, Row 12 would be better, but the presence of the side door and a large painting on the opposite wall prevents this. The Ulysses model for this case (6 HS1200 loudspeakers) is SJD045 and it is shown below as Figure 27.

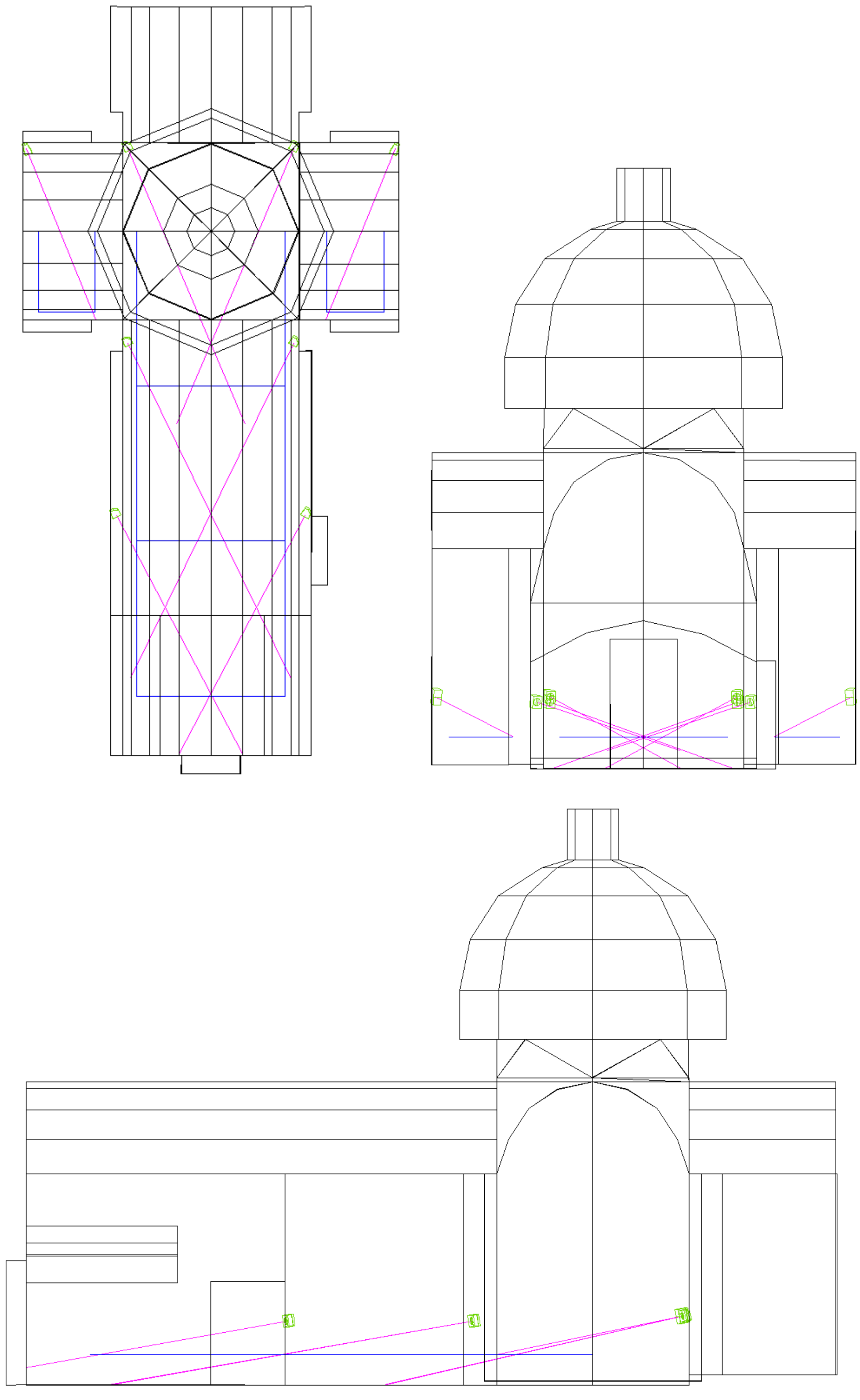

Figure 27 SJD45 Ulysses Model

The predicted performance for the Ld uniformity of the Ulysses models SJD44 and SJD45 are compared in Figure 28.

SJD44 (left)

LTL	LTR	NL		NC		NR	RTL	RTR	Pew#
S9		S1				S2		S10	
98.7	96.3	97		97.4		96.9	96	99.2	0
									1
									2
94.4	97.3	95.8		96.3		96.1	97.7	94.5	3
									4
		S3				S4			5
		97.4		94.8		97.3			6
									7
									8
		93.9		94.5		93.8			9
		S5				S6			10
									11
		94.7		93.2		93.7			12
									13
		S7				S8			14
		96.1		92.1		96.2			15
									16
									17
		90.6		90.1		90.7			18

(Ldmax-Ld) dB	
Less than 3	GOOD
3 to 6	FAIR
Greater than 6	POOR
Ldmax =	99.2

L&T with 1 Refl. Calc Z123

Z1	S1-S2	HS1200	31.6 W	10%-ABS-LIN
Z1	S3-S4	HS1200	15.8 W	SE-HA, EM
Z3	S5-S8	V6	15.9 W	CA-1/4 PL,GL DO
Z2	S9-S10	ELEMENT	12.6 W	D-SWP

SJD45 (right)

LTL	LTR	NL		NC		NR	RTL	RTR	Pew#
S9		S1				S2		S10	
98.7	96.4	97		97.4		96.9	96.1	99.2	0
									1
									2
94.4	97.3	95.7		96.3		96.1	97.7	94.5	3
									4
		S3				S4			5
		97.5		94.9		97.4			6
									7
									8
		95.2		95		95			9
									10
		S5				S6			11
		101		95.9		100			12
									13
									14
		96.4		96.1		95.8			15
									16
									17
		94		93.4		93.1			18

(Ldmax-Ld) dB	
Less than 3	GOOD
3 to 6	FAIR
Greater than 6	POOR
Ldmax =	100.8

L&T with 1 Refl. Calc Z123

Z1	S1-S2	HS1200	31.6 W	10%-ABS-LIN
Z1	S3-S4	HS1200	15.8 W	SE-HA, EM
Z3	S5-S6	HS1200	31.6 W	CA-1/4 PL,GL DO
Z2	S9-S10	ELEMENT	12.6 W	D-SWP

Figure 28 Direct Signal Uniformity of SJD44 (left) and SJD45 (right)

Replacing the zone 3 speakers with HS1200’s improves the direct sound uniformity by a couple of dB’s.

Similarly, the clarity (Ld-Lr) is compared in Figure 29.

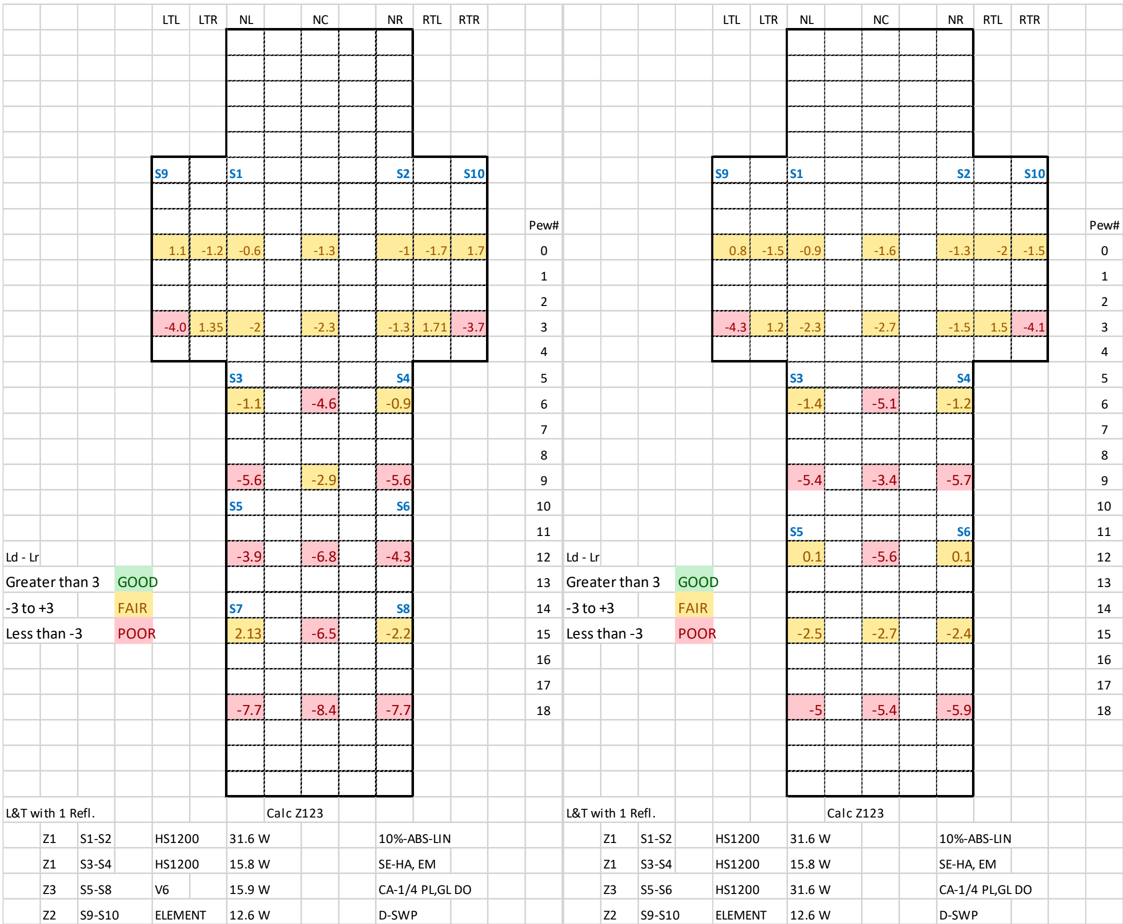

Figure 29 Comparison of (Ld – Lr) of SJD44 (left) and SLD45 (right)

The predicted ratio of direct to reflected signal in the rear of the church is improved by a couple dB's when the zone 3 speakers are replaced with a pair of HS1200's.

Add Delay Compensation to Rear and Middle Speakers

For the sound at the rear of the church to arrive simultaneously from speakers S1/2, S3/4 and S5/6, speakers S3/4 should be delayed by 23.6 msec and speakers S5/6 should be delayed by 43 msec.

When this is done in Ulysses, neither the direct sound level (uniformity) nor the direct to reflected sound ratio (Ld – Lr) is changed. This is a surprising result, and it is probably due to the way these values are calculated in the "level and time" mode in Ulysses. If, however, the "ray" mode is selected, and the ETC impulse response is calculated at row 18, center aisle, there is a noticeable change. If, for clarity, the reflections are limited to one each per loudspeaker, the impulse responses are shown in the following two figures for no delay (Figure 30) and with delay added (Figure 31).

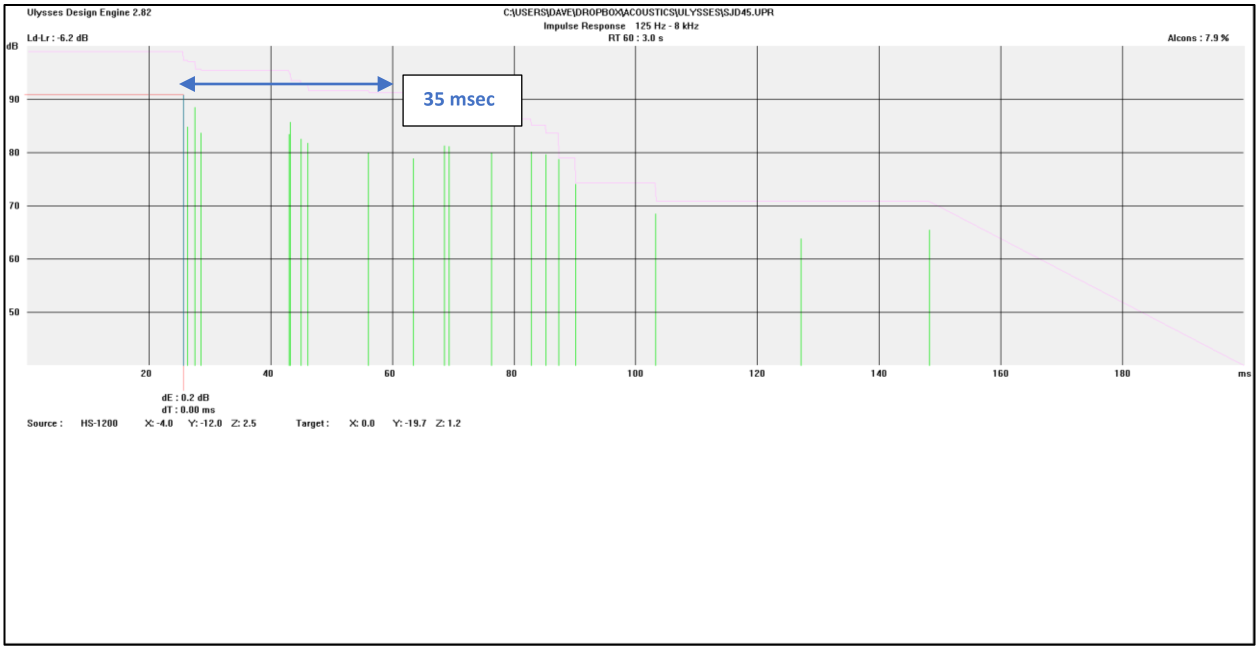

Figure 30 SJD45 Model, Row 18 Center Impulse Response, No Speaker Delays

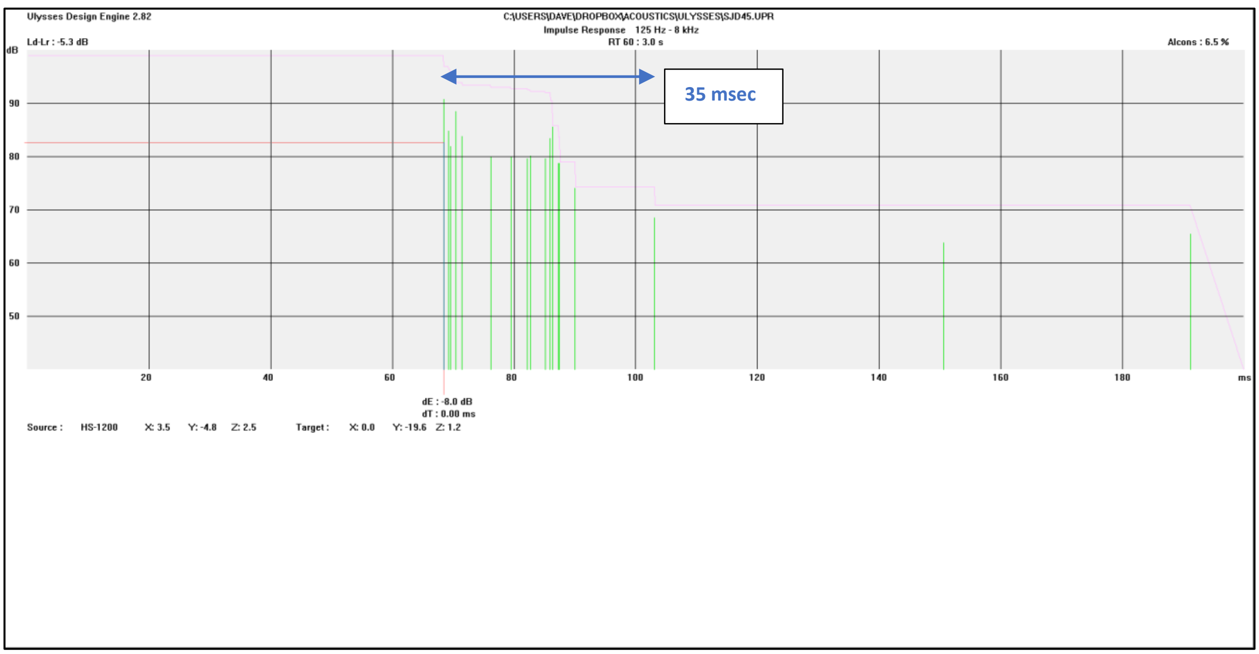

Figure 31 SJD45 Model, Row 18 Center Impulse Response, Speaker Delays Added

Note that with the delay added, the direct sound and nearly all the first reflections arrive during the 35 msec interval in which the human ear processes these constructively. Without the delays, these same impulses are spread out over a much larger time interval where they are heard as discrete echoes deleterious to speech comprehension.

The above is somewhat simplified as there are actually many more significant reflections than the ones shown, and with these added, the improvement is not so obvious as shown above. But the same principle applies, and the clarity should be improved significantly by adding the delays.

Wall Treatment

The walls and ceilings have an absorption of 10%. This means that any single reflection will only be attenuated by about 1 dB. Floor reflections and many of the side wall reflections will occur within the 35 msec interval of the direct signal, but long reflections from the front or back of the church may add echoes outside this interval. A good example is row 12 center. Looking at Figure 29 (right) this location has a poor direct to reflected ratio of -5.6 dB. Running Ulysses in the Ray mode results in the reflections shown in Figure 32 for all first order reflections. Figure 33 identifies the reflections by time of arrival. Speakers 5 and 6 are closest and the direct and reflected signals from these are noted. The largest impulse is from S5/6 direct and this is closely followed by the floor bounce. The side wall reflections occur at about 34 msec, then there is another reflection from the rear wall at 50.7 msec. There is a front wall reflection from S5/6 at 172 msec but that is probably not real because the model ignores all the statuary and circular structure in the sanctuary. There is however an S1/2 rear wall reflection at 106 msec.

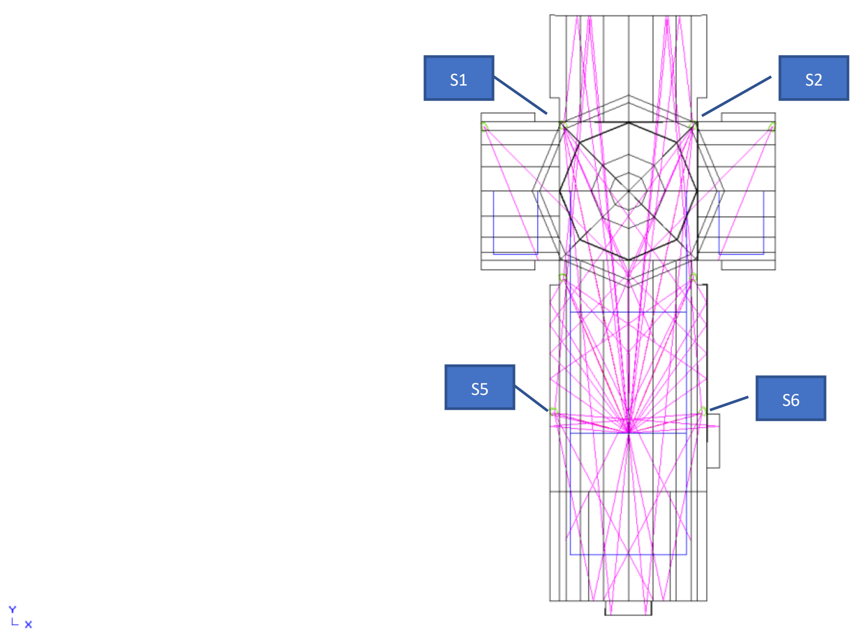

Figure 32 First Order Reflections of SJD45 Ulysses Model at Row 12 Center

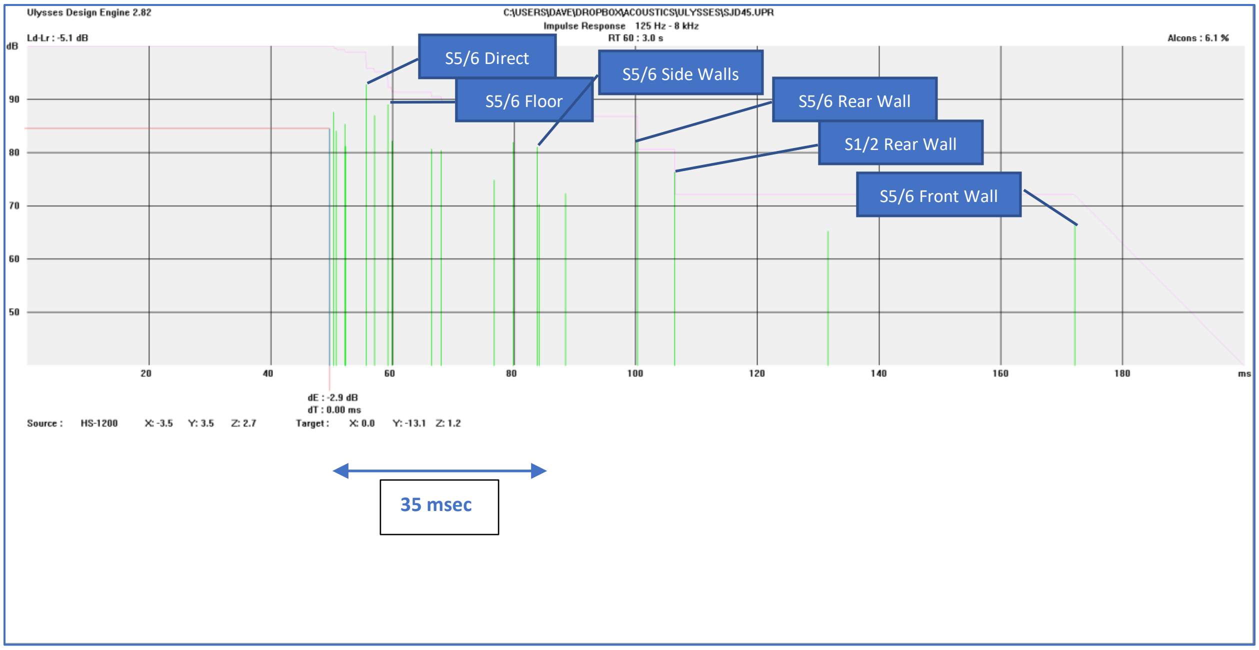

Figure 33 Impulse Response of Ulysses Model SJD45 for First Order Reflections at Row 12 Center

So, the rear wall reflections may be problematic, but fortunately this wall is amenable to treatment using the panels discussed in Part 1. Rather than rely on Ulysses to make this determination, the best approach would be to implement the rear HS1200 speaker system with delay compensation and measure its impulse response to see if rear wall reflections are significant. A simple experiment can be made to see the effect of rear wall absorption by temporarily hanging some furniture blankets on this wall and repeating the impulse response measurement.

Hardware Required

In addition to the two new HS1200 loudspeakers and brackets, two additional electronic items are required to provide the delay compensation. See Figure 34 and Figure 35.

Figure 34 Behringer DCX2496LE for Delay Compensation Zones 2 and 3

Figure 35 OSD XPA300 Stereo Amplifier for Zones 2 and 3

The way this equipment would connect to the existing OSD PAM245 mixer/amplifier is shown in Figure 36 below.

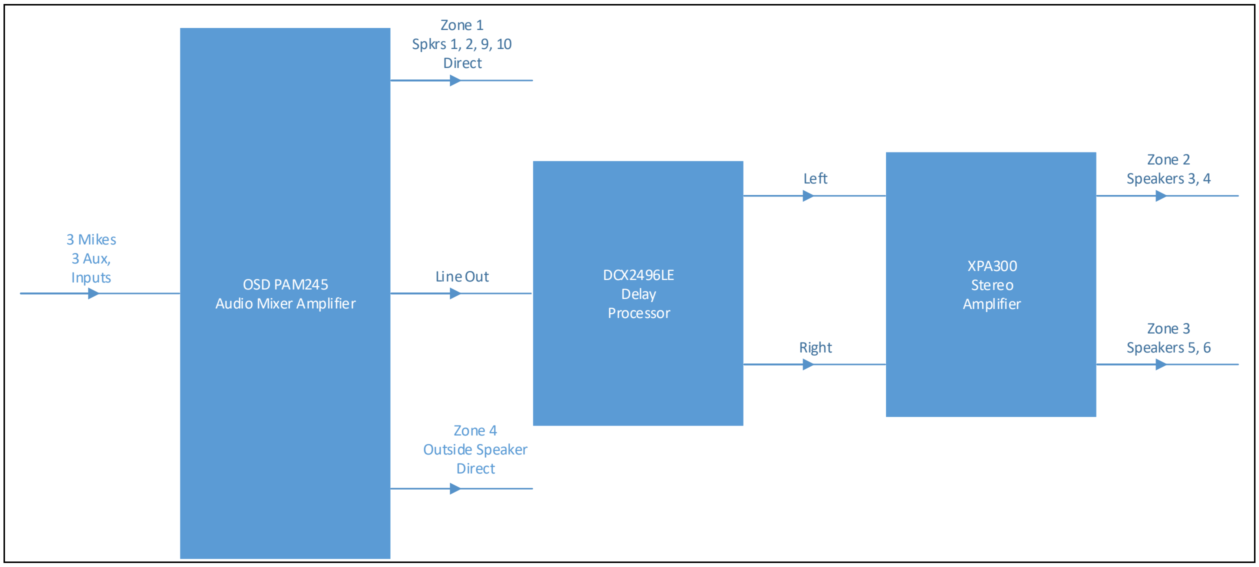

Figure 36 Block Diagram of PA Amplifiers with Delay Compensation

The zones will be changed slightly. Zone 1 will be rewired to contain the two main front speakers (S1/2) and the two transept speakers (S9/10). Zone 2 will contain only the two HS1200 speakers mounted at row 5. Zone 3 will contain the two new HS1200 speakers mounted at row 11. The existing wiring can be used for this arrangement with very minor changes.

Cost Estimate

The cost to complete the sound system design for the rear of the church is listed in Table 6.

Table 6 Cost to Complete Sound System for Rear of Church

(2) HS1200 Speakers with Shipping	$800
(2) Speaker Brackets	$150
(1) DCX2496LE Delay Processor with Shipping	$300
(1) XPA300 Stereo Amplifier with Shipping	$500
TOTAL COST	$1,750