San Juan de Dios Sound System
Part 3
Final Implementation and Testing

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April 2022

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

As discussed in Part 2, there is an advantage to disconnecting the speakers at the rear of the church when the size of the audience is small, and it sits near the front. That never happened for several reasons. People prefer to spread out, and some people just like to sit in the back. Also, the size of the congregation that attends to English Mass has grown from about 25 when this work began to over 50 now. So, efforts were made to improve the performance at the rear of the church with all speakers turned on.

After each of the changes described below, the performance was again characterized by measuring 21 locations in the Nave and up to 8 locations in the Transepts. Direct sound and clarity were then calculated at each of these locations and mapped over a plan view of the church with color coding to indicate good, fair and poor values for the uniformity of the direct sound and the clarity. As used in this report, the direct sound is that arriving within the initial 35 msec and reflected or reverberant sound is that arriving after 35 msec. The clarity is the ratio of direct to reflected sound. The human ear perceives all sound (including reflections) that arrive within the direct sound interval of 35 msec to be constructive to speech intelligence and those arriving later to be destructive.

Improved Prediction and Measurement of Direct Sound Uniformity and Clarity

Since Part 2 of this report was written, a better method has been developed for calculating the direct sound level (Ld35) and the clarity (C35) using the Schroeder integral in both Ulysses and Smaart8. The details are technical and therefore relegated to the Appendix. The new method is designed to include audible "pre-echoes" in the direct sound interval. The results are thought to be more accurate than those presented in Part 2 of this report. Those earlier results are not wrong – just less accurate.

Reanalysis of (2) CS212 + (4) KR1 Sound System Tested September 22, 2020

This is the original sound system. The impulse response was measured at the 12 locations shown in Figure 1. Loudspeakers S1 and S2 are Steren model CS212's mounted high on the front Cantera columns. Loudspeakers S3-S6 are Celestion model KR1's, and they were located as shown along the sides of the Nave.

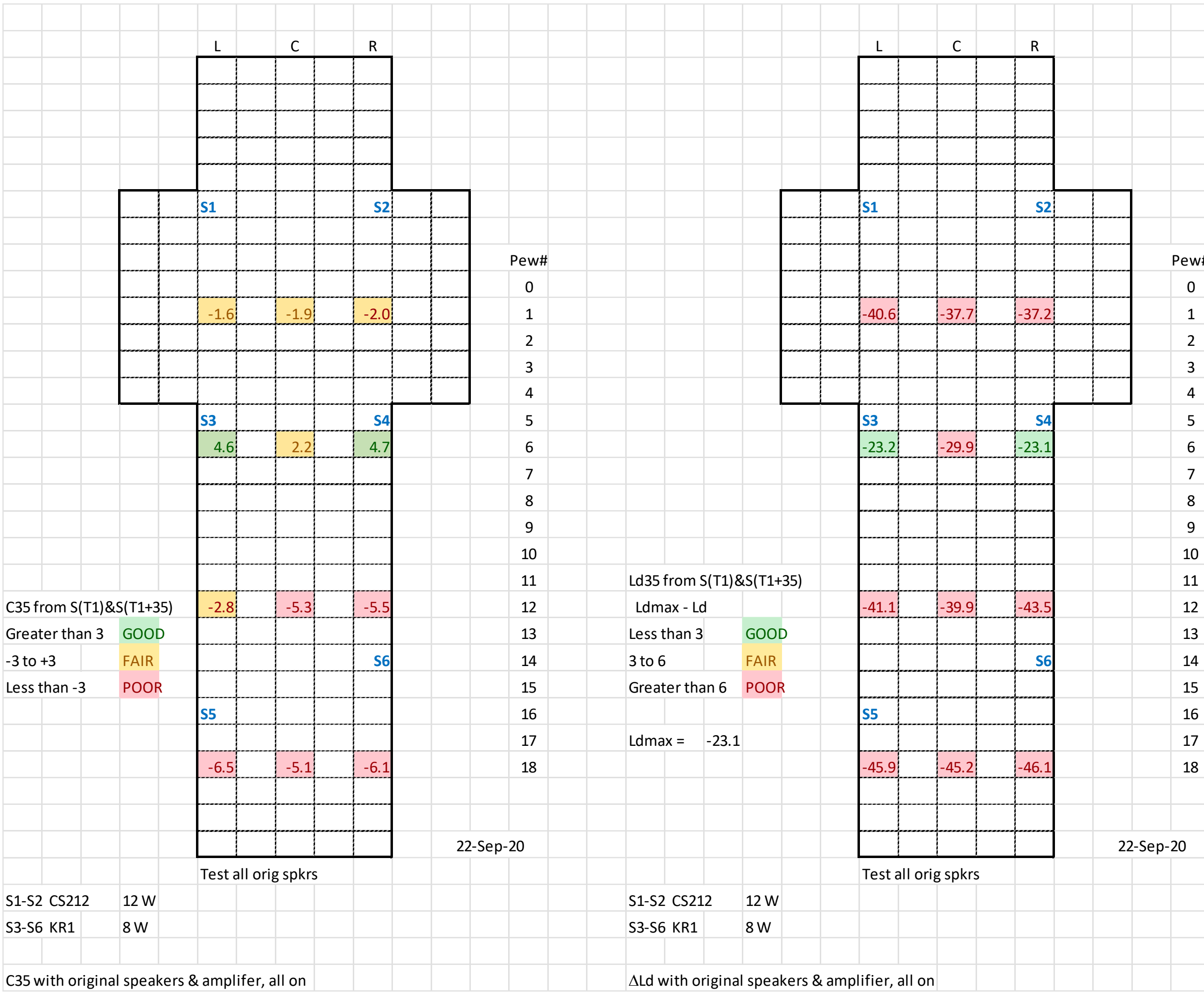


Figure 1 Original Sound System Clarity (left) and Direct Level Uniformity (right)

Reanalysis of (4) HS1200 + (2) CS212 + (4) KR1 Sound System Tested April 16-23, 2021

This system includes the new amplifier and (4) HS1200 speakers S1-S4, mounted on the Cantera columns. The (4) KR1 speakers are relocated towards the rear of the church and the (2) CS212 speakers are mounted in the front outside corners of the Transepts. Both clarity and direct sound uniformity have improved. But Figure 2 shows that the clarity is still poor at the back of the church.

The direct sound shown in Figure 2 (right) below can be compared to that in Figure 20 (left) in Part 2. The clarity shown below in Figure 2 (left) can be compared to that in Part 2, Figure 21 (right). Including the “pre-echoes” in the calculations makes a small but noticeable difference.

Figure 2 Clarity (left) and Direct Level Uniformity (right) with New Amplifier and (4) HS1200 Speakers at Columns

Installation of Third Pair of HS1200 Loudspeakers

These were installed at approximately Row 11 on each side of the Nave in order to improve the performance at the rear of the church. (Acoustically, row 12 would have been better but the presence of a side door on the East side and a Stucco painting on the West side prevented this). These speakers replaced the four Celestion KR1 speakers that were previously connected as Zone 3 to the amplifier.

Because the speakers were installed on the plaster walls and not in the Cantera columns, the mounting brackets used were extended vertically to provide additional mounting points for the wall anchors. The plastic anchors used were 100 mm long (4 inches). Because of the uneven depth and poor quality of the plaster walls, some of the holes drilled were unusable. But the brackets provided many predrilled hole locations, and we were able to get at least four holes that reached into the rubblestone under the plaster and provided a good grip on the inserts.

Figure 3 and Figure 4 below show the original brackets used for the front and center speakers and the extended brackets used for the rear speakers. The same yoke is used to tilt and pan the loudspeakers. (See Part 2, Figure 3).

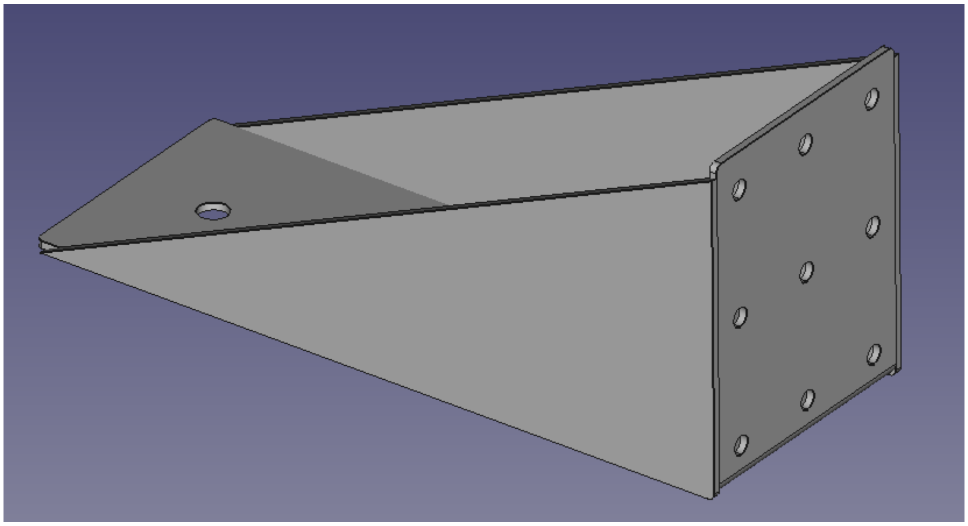


Figure 3 HS1200 Wall Mount Bracket Used on Cantera Columns

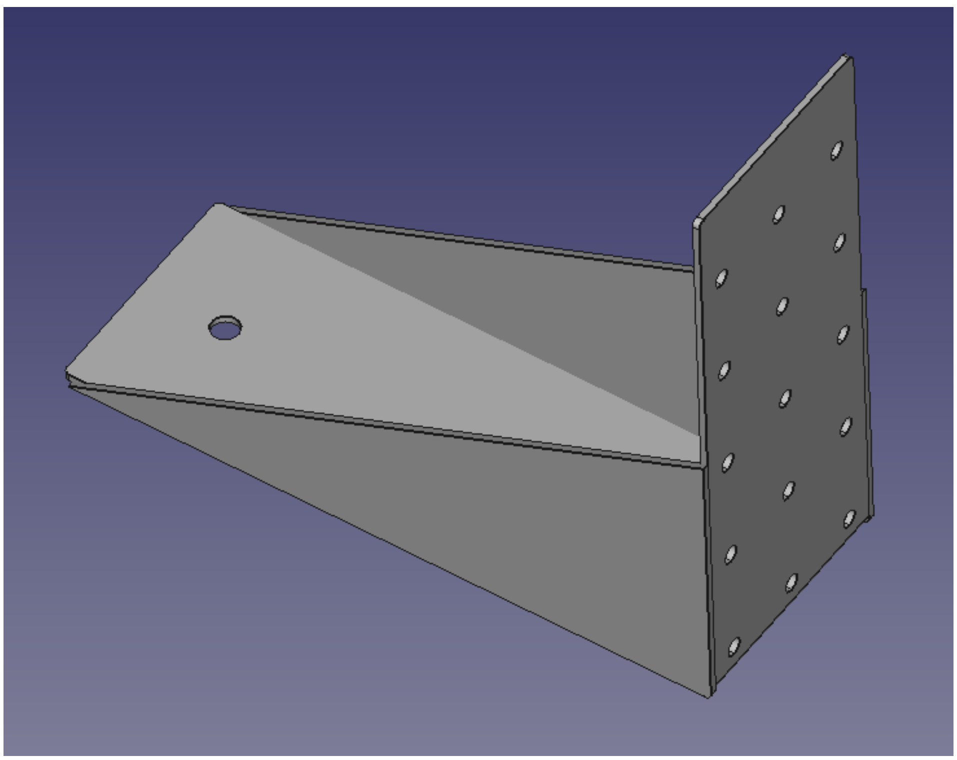


Figure 4 Extended HS1200 Wall Mount Bracket Used on Plastered Walls

Impulse Response Testing of (6) HS1200 + (2) CS212 Speaker System

The impulse response was measured with Smaart8 at the usual row and aisle locations and the clarity and direct sound uniformity were calculated from the Schroeder integral. These measurements were made on August 13, 2011. All speakers were turned on for these tests. Figure 5 shows that the clarity has improved, but the level uniformity is still poor in the middle and rear sections.

Figure 5 Clarity (left) and Direct Level Uniformity (right) with (6) HS1200 and (2) CS212 Speakers

Calculated Speaker Levels for Direct Sound Uniformity

The data in Figure 5 (right) indicates that the direct sound level is low in the middle and rear of the church. To determine whether the rear speakers alone should be increased in power or both the center and rear speakers increased, the Ulysses model was used.

Ld35 was calculated as described in the Appendix using the 5-reflection option for rows 0,3,6,9,12,15 and 18. The models were SJD45 and SJD46. SJD45 has the middle speakers set at 15 watts and the front and rear speakers set for 30 watts. SJD46 has all HS1200 speakers set to 30 watts.

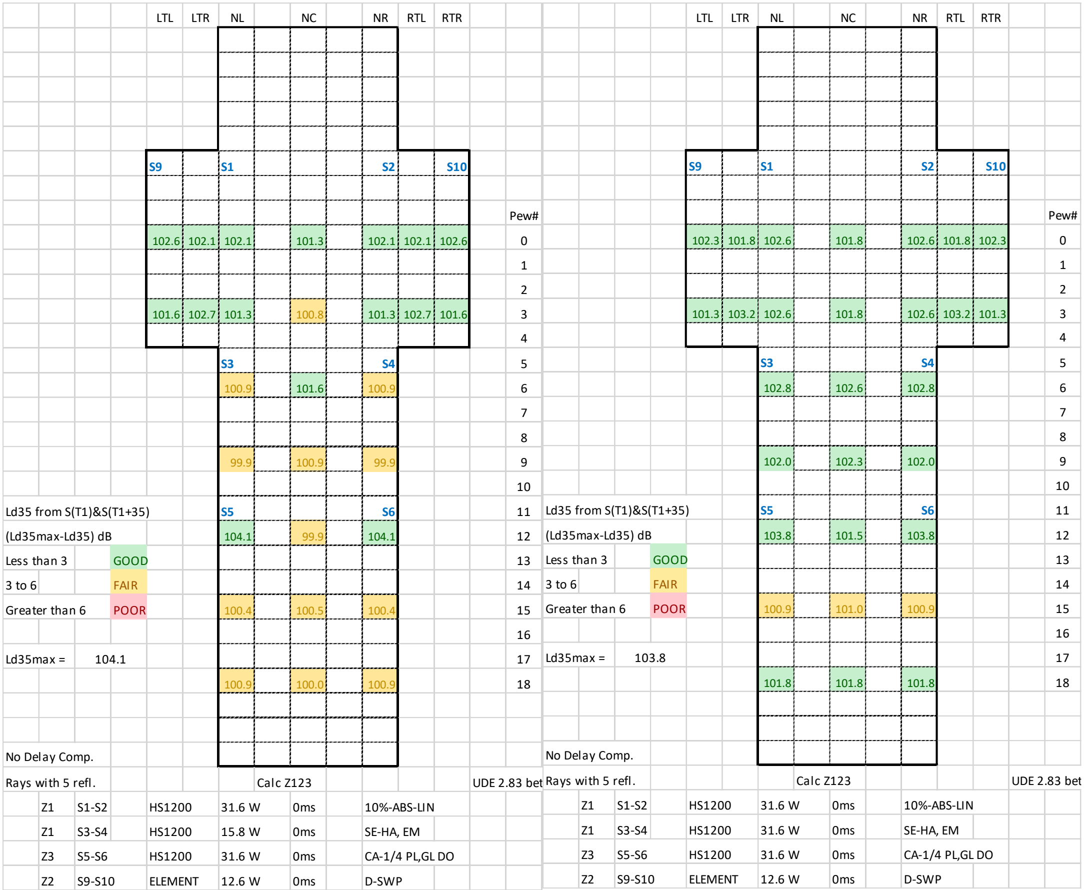


Figure 6 Calculated Direct Sound Uniformity with Middle Speakers at 15 watts (left) and All Speakers at 30 watts (right)

Figure 6 shows that all speakers set for 30 watts provides better direct sound uniformity.

Calculated Speaker Delay for Clarity

Ulysses was used with model SJD46 to see the effect of delay on clarity C35 using the method described in the Appendix. Ld35 was also calculated but the delay does not affect it significantly.

Three cases were considered: (1) no delay, (2) middle and rear speakers delayed 33.5 ms and (3) middle speakers delayed 23.6 msec and rear speakers delayed 43 msec. These cases are shown in Figure 7 below.

The third case (right) is the best but having just a single delay for the middle and rear speakers helps also. This is perhaps better shown by plotting the average clarity of the right aisle and center aisle for each row in the Nave. Figure 8 shows that the clarity improves past row 6 with either the two-zone delay of 33.5 msec or the three-zone delays of 23.6 and 43 msec.

I had planned to implement the 3-zone delay compensation, but for reasons discussed in the next section, it was decided instead to implement the 2-zone delay compensation.

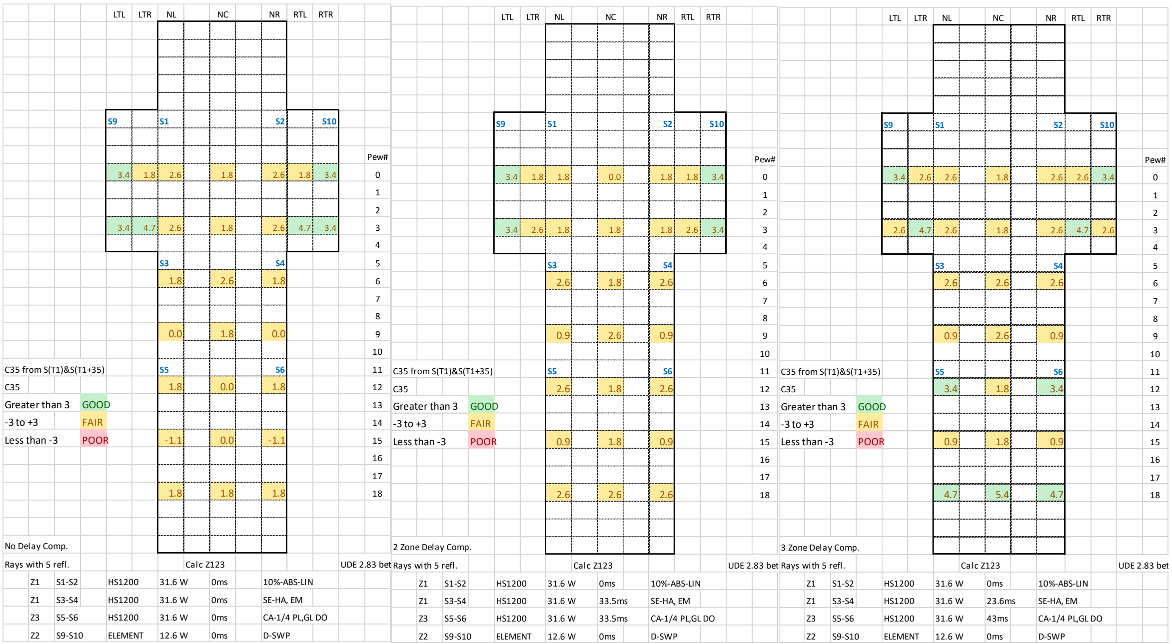


Figure 7 Predicted Clarity with No Delay (left), Single Delay (center) and Two Delays (right)

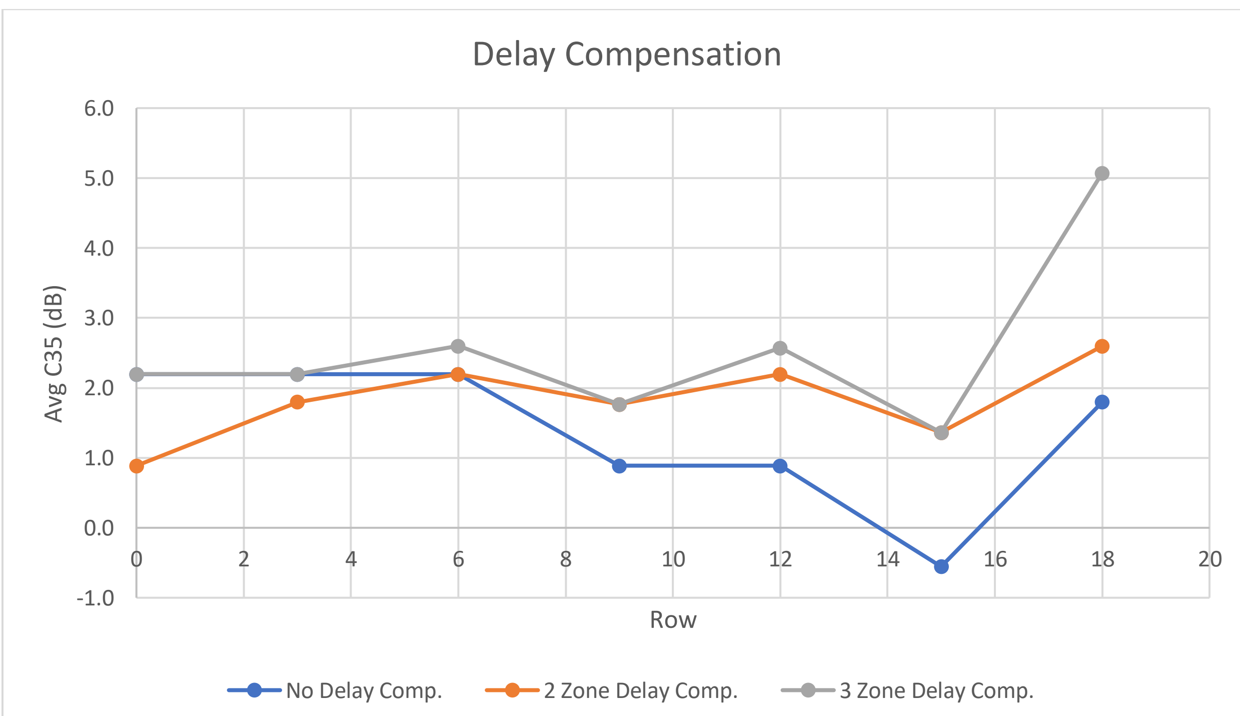


Figure 8 Average Predicted Clarity with and without Delay Compensation

Rewiring of Speaker Zones and Addition of Delay Amplifier

In preparation for the installation of the new amplifier and loudspeaker delay electronics, a change was made to the wiring. The new zoning assignment places the front two HS1200 and both CS212 loudspeakers in the same zone 1, while the middle pair of HS1200 speakers are now zone 2, and the rear pair of HS1200 speakers are zone

3. In addition, the new stereo amplifier intended for zones 2 and 3 required an isolated return, so all the AWG10 black common return wiring was removed and replaced by separate return lines for each of the zones. There is also the outside horn speaker which is connected as zone 5 to the amplifiers. Based on the Aug. 13 tests and the Ulysses analysis, the zone 2 and zone 3 speakers were changed to 30 watts. This was done and the resistance measured at the amplifier.

The new zoning is summarized in Table 1.

Table 1 Revised Speaker Wiring for Delay Compensation

Zone	Speakers	Wire Color	Resistance
1	(2) CS212 @ 12W + (2) HS1200 @ 30W	Red + Red/Blk	3.2 ohms
2	(2) HS1200 @ 30W	Blu + Blu/Blk	6.2 ohms
3	(2) HS1200 @ 30W	Grn + Grn/Blk	6.5 ohms
5	Outdoor Horn	Wht + Wht/Blk	15.6 ohms

When the new stereo amplifier (an OSD Audio Model XPA300) was received and tested, it was discovered that it had a serious overheating and overload problem. It was returned and replaced by a second model that had the same problems. In addition, the level controls did not retain their settings through power cycling and the unit required very careful level adjustment to avoid overload. It is a poor design for this application (maybe all applications), and the dealer agreed to replace it with an OSD Model PAM245 which is the same model we currently use to drive the entire sound system. Since the PAM245 is not a stereo amplifier, this meant there could be only one delay compensation. So, the middle and rear speaker pairs were delayed the same amount.

The DCX2496LE delay amplifier and second PAM245 power amplifier were connected and tested on December 3, 2021. Volume controls were adjusted to provide the same level at row 9 with only zone 2 activated. The delays were also verified as shown in the impulse response in Figure 9. The blue trace is for no delay, the green trace is 34 msec delay and the red trace is 43 msec delay.

Figure 9 Impulse Response at Row 9 Center, Zone 2 Only Activated.

Impulse Response Testing with and without Delay Compensation

All three zones were turned ON and data taken along the center aisle only (offset 1 ft to the right to help distinguish the arrival times). Ld35 and C35 were calculated from the Schroeder integral. These tests were conducted on December 3, 2021.

The measured values are shown in Figure 10 and Figure 11.

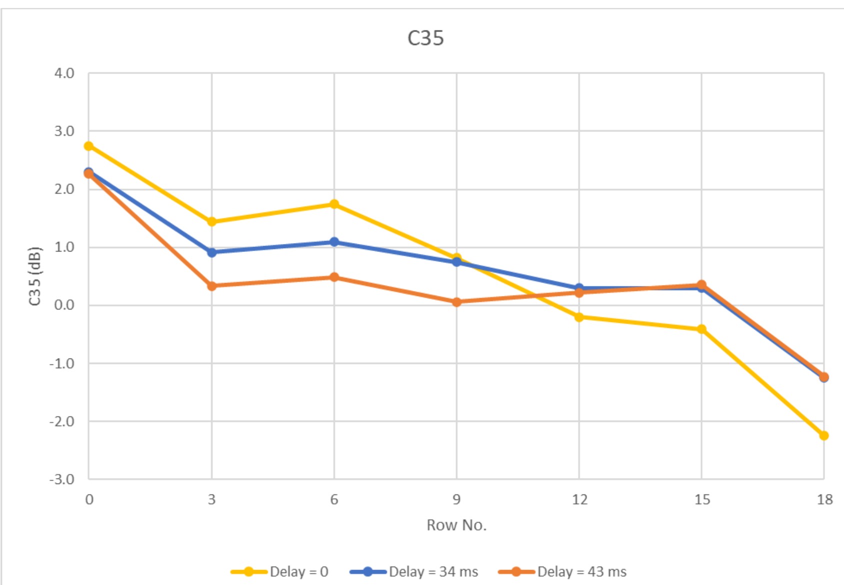


Figure 10 Clarity Along Center Aisle vs Delay

The delay(s) improve the clarity at the rear about 1 dB but make it worse near the front.

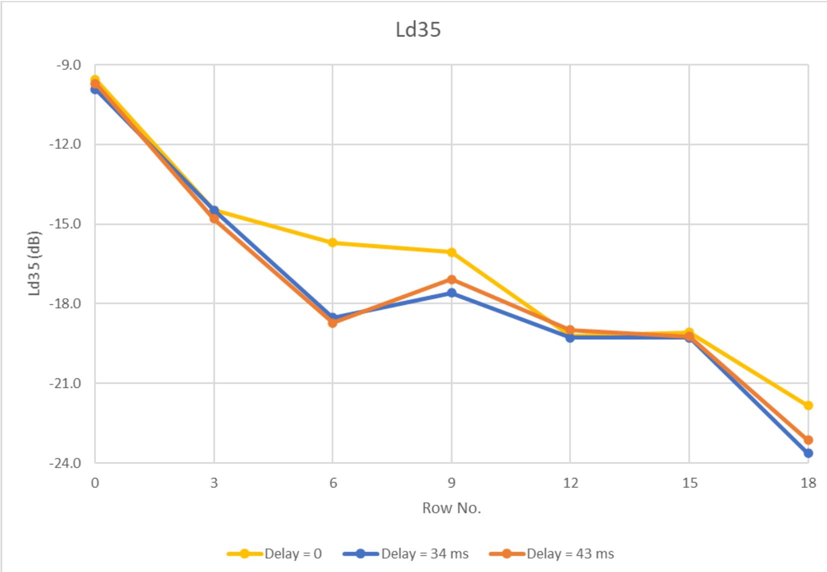


Figure 11 Direct Sound Level Along Center Aisle vs Delay

The delay(s) do not make much difference to the direct sound.

To see why the delay makes C35 worse near the front, consider the row 6 case with 0 delay and 43 msec delay.

Figure 12 Impulse Response at Row 6 Center with 0 Delay

Without the delay, Figure 12 shows there is a significant "preecho" from the middle speakers S3&4 at 11.52 msec where the Schroeder Integral (SI) is -13.44 dB. The value of the SI 35 msec later is -17.42 dB making the C35 = +1.8 dB. Note that the direct sound from all 8 speakers occurs within the 35 msec interval. There are also several reflections which are difficult to separate, but one of the larger ones is the reflection of S3 and S4 from the side walls. It occurs at nearly the same time as the direct sound from transept speakers S9 and S10.

Figure 13 Impulse Response at Row 6 Center with 43 msec Delay

With the 43 msec delay, Figure 13 shows that SI measures -15.93 dB at 31.90 msec and -19.19 dB at 67.02 msec, so C35 is only +0.5 dB. With this delay, the middle and rear speakers arrive after, rather than before the front speakers. But all 8 direct sounds still arrive within the 35 msec interval used to define C35. Note however that the S3/S4 side reflection now arrives outside the 35 msec interval thus contributing to the reverberant portion of C35.

The rational for adding delay compensation is based on the ideal situation where the sound travels from front to back without any reflections. If for example we had two loudspeakers that were aimed directly towards the back (with no backside radiation towards the front), separated by a distance that corresponded to 40 msec, with no wall reflections, there would be a very high C35 at all locations in front of the rear speaker and a much lower C35 behind it. Adding a delay of 40 msec to the rear speaker would not affect anything in front of it, but it would put the two sound waves in phase behind it, thereby improving the C35 at the rear.

In the more realistic case where the speakers do radiate somewhat towards the front and/or the walls reflect the sound from the rear speaker towards the front, the clarity in the middle (20 msec location) will be reduced with no delay compensation and the clarity to the rear (say, 60 msec location) will be worse yet. In the middle the sound will be the sum of the front speaker delayed 20 msec plus that of the rear speaker delayed 20 msec plus all the reflections.

With 40 msec delay now added to the rear speaker, the clarity in the rear will improve as the two direct signals will be in phase, but in the front and middle it will get worse. The sound in the middle will now consist of direct sound from the front speaker delayed 20 msec plus direct sound from the rear speaker delayed 20 + 40 = 60 msec plus all the reflections. So, the delay makes the clarity worse at the middle.

The measured data (and that shown by the Ulysses model in Figure 8) confirm that this is happening. As long as the speakers are similar to those used here and the wall reflections are relatively high, delay compensation can be expected to only improve the clarity at the rear at the expense of clarity at the front and middle.

At this point, I decided to abandon the idea of delay compensation, remove the delay electronics and second amplifier, and characterize the entire church for clarity and direct sound uniformity using all 6 HS1200 speakers at 30 watts with no delay compensation.

Impulse Response Testing with All HS1200 Speakers at 30 watts

These tests were run on December 28, 2021 with all speakers on, and no delay compensation.

Both the clarity and the uniformity are shown in Figure 14 and they have improved some more.

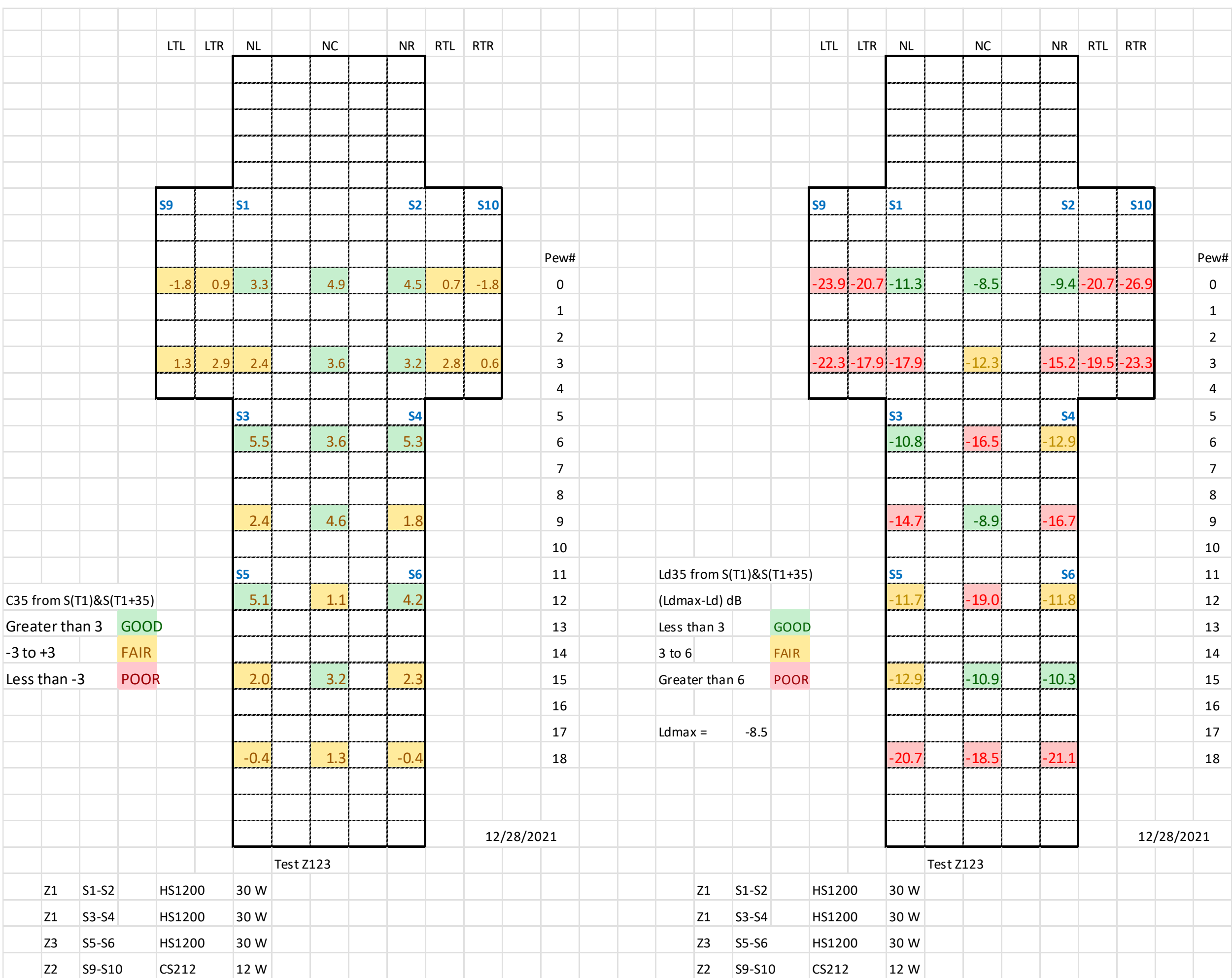


Figure 14 Clarity (left) and Direct Level Uniformity of (6) HS1200 Speakers at 30 watts

Calculated Effect of Adding Absorption to South Wall

The Ulysses models assume a wall absorption of 10% which matches the RT60 data very well. Model SJD47 was created to change the absorption of the South wall under the balcony to 100%. This is the wall at the back of the church with the front door, and it reflects sound toward the front from all the HS1200 loudspeakers.

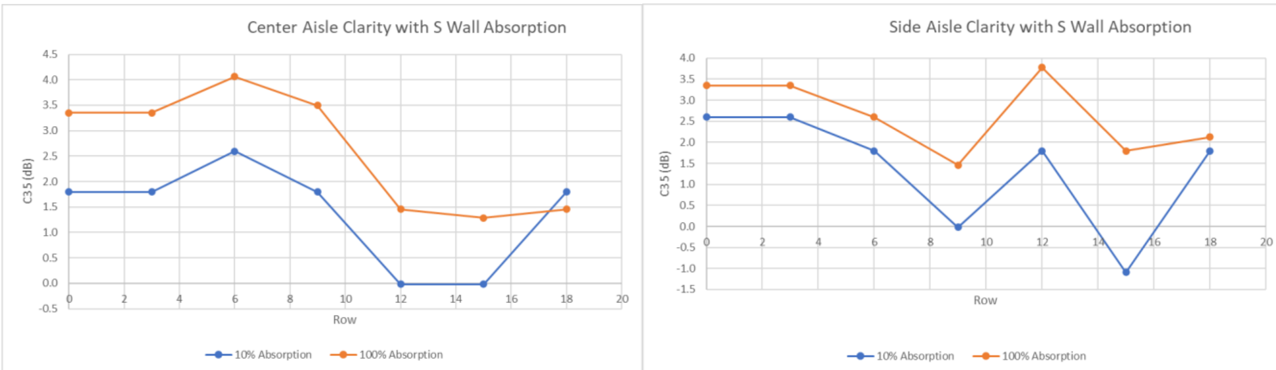


Figure 15 Calculated Clarity Improvement from S Wall Absorption along Center Aisle (left) and Side Aisle (right)

The clarity improvement shown in Figure 15 is between 1 and 1.5 dB which is significant. The South wall is not only the main source of troublesome reflections, but also the one most amenable to sound treatment because - unlike all the other walls – no art or statuary is mounted on this wall, and being behind the audience, it is not generally seen.

Optimal Delay with Absorption on South Wall

With less sound traveling from the back to the front of the church, it is more feasible to use delay to improve the clarity at the back without making it worse in the front. This is only partially true since there is still some backwards radiation from the loudspeakers.

Ulysses model SJD47 was run with zones 2 and 3 delayed first 33.5 msec then 23.6 msec. The 33.5 msec corresponds to the average between the ideal 23.6 msec for the zone 2 (middle) speakers and the 43 msec for the zone 3 (rear) speakers. However, this delay causes the backwards radiation from the middle speakers to arrive at aisle 0 center more than 35 msec behind the direct signal from the front speakers. Figure 16 shows that a single delay of 23.6 msec makes the clarity more or less uniform from front to back.

Looking at Figure 14, Figure 15, and Figure 16 suggests that a delay of 24 msec along with rear wall absorption provides the best balance between clarity at the front and at the rear.

Adding delay, however, is expensive and complicated. There are now three electronics to turn on and adjust rather than just the one amplifier. For these reasons, it was decided to not pursue the delay compensation.

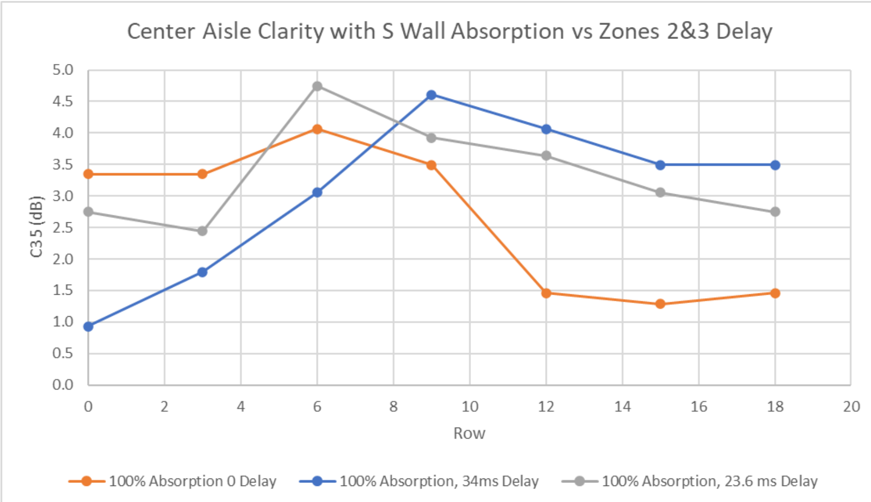


Figure 16 Clarity Along Center Aisle with Both South Wall Absorption and Delay Compensation

Impulse Response Testing with Rear Wall Absorption

Impulse response testing was conducted on February 2, 2022 by attaching furniture moving blankets to the rear wall as shown in Figure 17.

Figure 17 Moving Blankets Attached to Rear Wall to Absorb Reflections

Clarity and direct sound were calculated from the Smaart8 measurements using the Schroeder Integral method and are as indicated below for all of the Nave and most of the Transept positions. Figure 18 shows the results. Comparing these to Figure 14 shows that the clarity and uniformity have improved.

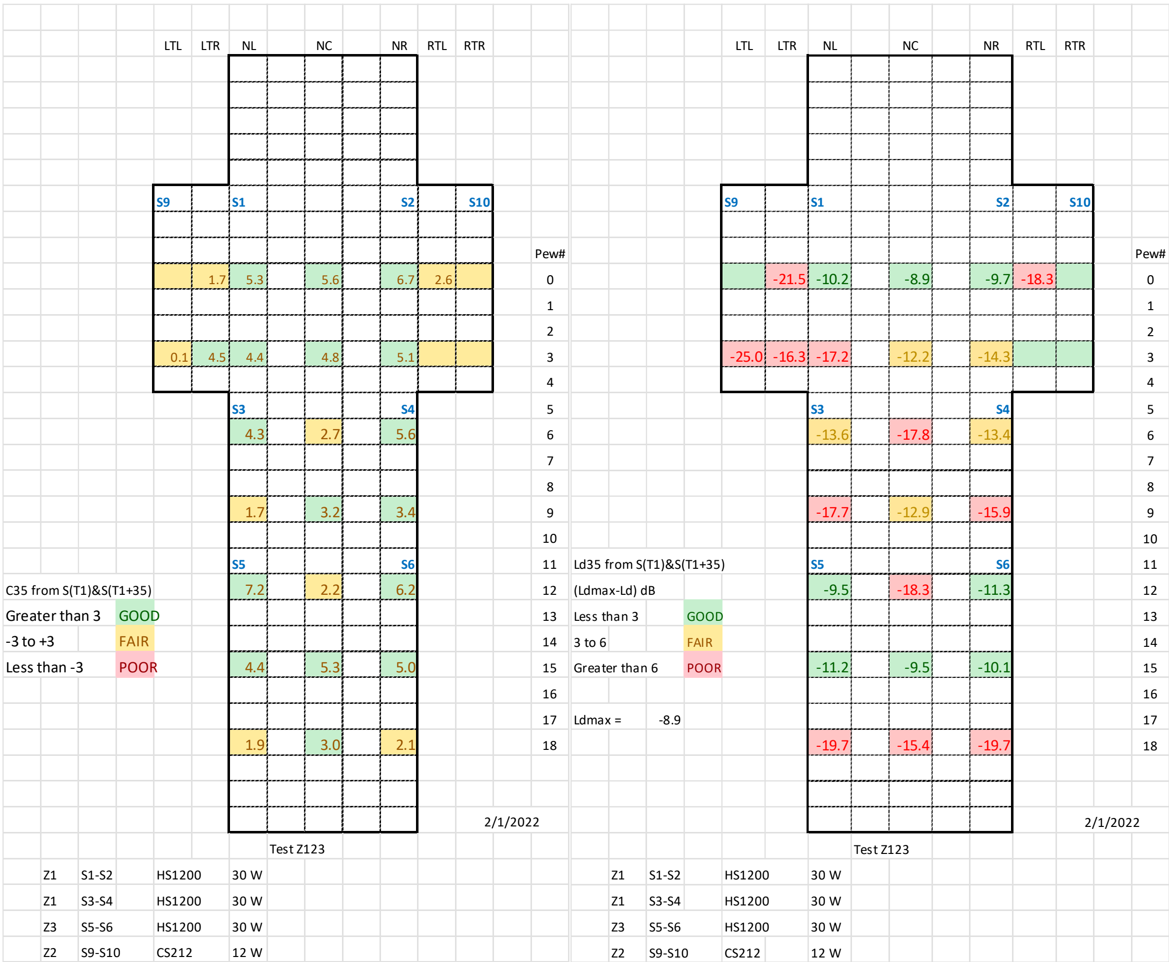


Figure 18 Clarity (left) and Direct Level Uniformity of (6) HS1200 Speakers at 30 watts and Absorption on Rear Wall

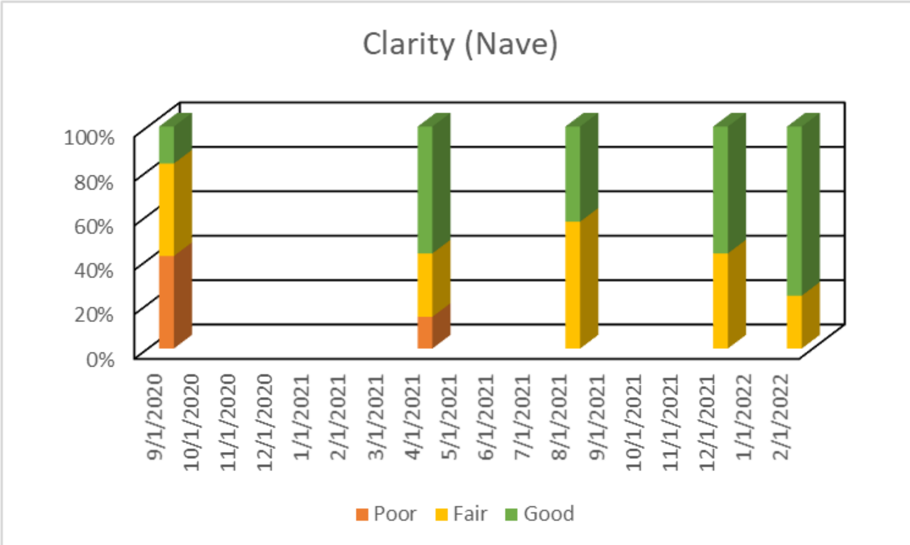
The next step is to install something esthetically attractive and permanent to the rear wall. One possibility is to use a low pile carpet matching the color of the other walls.

Summary and Conclusions

In this section, I am summarizing the conclusions reached over a period of 3 years analyzing, installing and testing the sound system at San Juan de Dios church.

Performance Improvement

A good idea of the performance improvement to date can be had by comparing the percentage of seating coverage in the Nave where the clarity or uniformity is good, fair or poor. This is shown in Figure 19 below. Considerable improvement has been made to both clarity and uniformity.



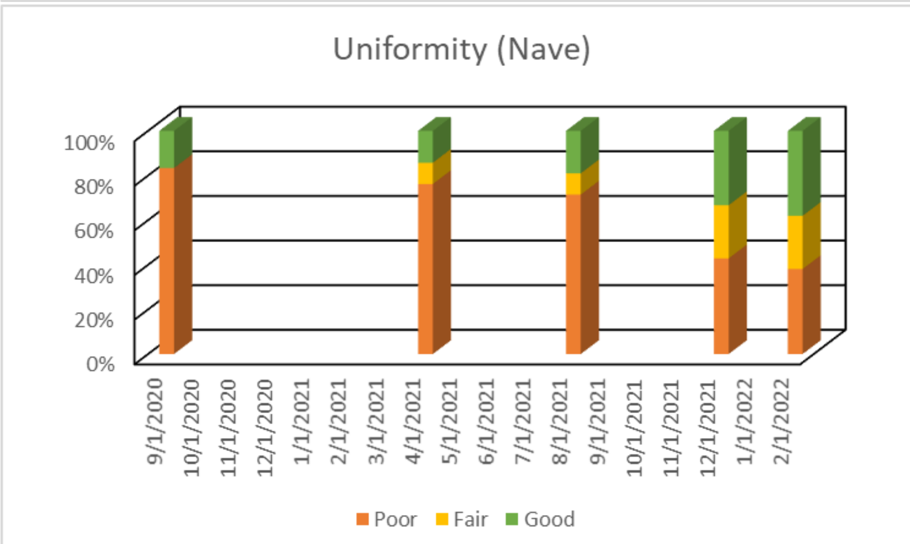


Figure 19 Percentage of Seating with Good, Fair or Poor Clarity (top) and Direct Level Uniformity (bottom)

Summarizing the cases tested and shown in Figure 19 above,

Table 2 Configurations Tested

22 Sep 2020	16-23 Apr 2021	13 Aug 2021	28 Dec 2021	2 Feb 2022
Radio Shack amp.	PAM245 amp.	PAM245 amp.	PAM245 amp.	PAM245 amp.
2 CS212 Front 4 KR1 Middle & Rear	4 HS212 Front & Middle 4 KR1 Rear 2 CS212 Transepts	6 HS212 Front, Middle & Rear 2 CS212 Transepts	6 HS212 Front, Middle & Rear 2 CS212 Transepts	6 HS212 Front, Middle & Rear 2 CS212 Transepts
			Equalized Power	Equalized Power, Added Rear Wall Absorption

Metrics for Measuring Sound Intelligibility

The Haas effect is a psychoacoustic phenomenon that states that when one sound is followed by another with a delay time of approximately 40 msec or less (below a human's echo threshold), the two are perceived as a single sound. Other time intervals are also used: 35, 50, 80 msec, etc. Early to late energy ratios are used to indicate the "clarity" of speech, music, etc. Clarity is often calculated by taking the impulse response, sending it through various bandpass filters, calculating the ratio of early to late sound energy for each band, then combining them so that the most important frequencies for speech are weighted more heavily. Other metrics can be used which take into account background noise. Articulation Loss of Consonants (ALCONS), Speech Transmissibility Index (STI), STI for Public Address systems (STIPA) and other metrics are all used, but they are much more difficult to calculate and interpret. In our case, noise is not a problem (except when Indians are setting off fireworks outside the front door), and only speech clarity is of interest. For simplicity's sake, the broadband (not band limited or weighted) early to late sound energy is calculated using a time interval of 35 msec. This interval is especially convenient because it is the value used in the Ulysses Energy Level calculations. So, C35 is the clarity metric selected. As discussed in the Appendix, there is reason to begin the 35 msec interval with the first significant (audible) sound arrival rather than the largest energy pulse.

The second metric used is the Direct Sound Uniformity because we want the level (for comprehension) to be about the same everywhere in the seating area. Total Sound Uniformity is useless in a highly reverberant building like San Juan de Dios Church because it measures the same everywhere. The Direct Sound Level Ld35 is the energy received in the initial 35 msec interval, and it can be calculated at the same time as C35 from the impulse response measured by Smaart8 or the Ray calculation in Ulysses.

Software Tools for Acoustic Design and Testing

Smaart8 is the measurement tool of choice. I am not aware of any other competitive program. It contains a sound generator and 1 or 2 channel Fast Fourier Transform (FFT) input analyzers. Three modes of operation are available. There is a sound pressure level mode which has a single microphone input. I did not use this mode. The second mode provides the instantaneous frequency response between the signal generator and the measurement microphone. This mode is very useful for adjusting the equalization on the amplifier. In our case, equalization consists of only the bass and treble adjustments. The third mode of operation calculates the impulse response from the frequency response data and stores it. It is not designed for real time analysis, but the program can recall the stored data, expand and display it as an impulse response in linear or logarithmic scales. Almost all the data measurements taken used the impulse response mode with the highest resolution (512K FFT bins) and 2 averages. The signal generator was set for period matched pseudo random pink noise. Smaart8 is sold and supported by Rational Acoustics. They provide good training videos and on-line product support.

Ulysses is an analysis tool used to predict the acoustic performance. It includes a 3-dimensional drawing tool that is used to create a model of the interior structure of a room or building. Each surface can be acoustically treated using an extensive data base of common construction materials. In the case of San Juan de Dios church, I found that using a constant value of 10% absorption (at all frequencies) provided the best match to the measured Sabine reverberation decay time RT60. The initial analyses which I made where various loudspeaker arrangements were considered used the "Time and Level" calculation mode. This is very fast and allows for parametric studies such as determining the best height to mount loudspeakers, evaluating different speaker types, orientations, level and delay settings, surface treatments, etc. These sorts of studies would be entirely impractical to do using actual hardware and testing. Once the initial design was selected (loudspeaker model and mounting) the Ray/Impulse Response calculation was used to predict C35 and Ld35. The results from Ulysses and Smaart8 compared reasonably well - at least well enough as to lend credence to the Ulysses predictions. Ulysses is sold and supported by IFBSoft in Germany. Support is slow but good. As discussed in the Appendix, the Ray calculation mode provides the Schroeder Integral required for calculating C35 and Ld35, but the plot saturates in the region of interest in the current released version of the program. IFBSoft provided me a beta version which corrected this problem, but it contained other bugs that limited the number of rays that could be used. This turned out not to be an issue, but the program needs updating. The biggest problem is that the data base for loudspeakers is out of date, making it difficult to find commercially available loudspeakers. The competitive program is called EASE, but I have not used it. Most loudspeaker manufacturers provide EASE data files for their products. These can be converted into the format used by Ulysses, but IFBSoft has not done this since 2013.

Measurement Hardware

A Behringer model UMC202HD is a 2 input, 2 output audio to USB interface that connects the laptop running Smaart8 to the main audio amplifier and the measurement microphone. It allows input and output levels to be adjusted and has a switchable phantom power source for ceramic microphones. LEDs on the front facilitate level setting for both the reference input (the excitation going to the amplifier) and the measurement microphone. This unit is inexpensive and simply worked.

The measurement microphone was a Dayton model EMM6. This is an inexpensive ceramic omnidirectional microphone that was mounted to a microphone stand set to 1.1 meters, which is about head height when seated. The first of these microphones had a serious deficiency in that its internal noise level started out very low but slowly increased over a 30-minute period to the point where it interfered with the impulse response measurements. Dayton replaced it, but it took about 4 months to make the exchange due to stocking problems. The new microphone works fine. The EMM6 microphone connects to the interface unit using (3) 50 ft long XLR(M) to XLR(F) extension cables.

Amplifiers

The sound system in the church is that of a public address system and not a theater style music system. That is, it is designed to provide monaural voice quality sound amplification to multiple loudspeakers from one to several microphones. The output drive can be 70.7 or 100 VRMS to minimize the wire size used for the loudspeakers. The original amplifier was a Radio Shack model MPA125 which had a rated power of 100 watts. This amount of power was adequate for the speakers then used. The output power should be at least equal to the sum of the average power requirement of all loudspeakers. For the HS1200 speakers selected here, the rated power is 60 watts each, but they are all connected with their 30-watt taps. There are 6 of them in addition to two CS212's (12 watts each) and an outdoor horn which is estimated at 30 watts. The total rated power is then 234 watts. There are surprisingly few amplifiers made which fall into this category. Outdoor Speaker Depot (OSD) makes a model PAM245 which was selected. It has a rated output of 240 watts, and it includes three microphone (XLR) inputs and three line-level (RCA) inputs – all with their own volume controls. There is a bass and treble control and a master volume control. All the controls are on the front panel, and all retain their settings during power off. The speaker outputs are provided through 5 front panel switchable zones. This amplifier is a class D type and runs very cool. It is possible to overload it by applying a too large input to an input with its volume control too high which triggers an overload condition. When it does occur, the amplifier must be power cycled using a back panel power switch. Fortunately, this is not a common occurrence. Normal power switching is from the front panel.

An alternative amplifier that was tested but returned is the OSD model XPA300. This is a 70/100 V Dual Channel amplifier which was selected to work with the delayed zones in the middle and rear of the church. It was a disaster both in terms of performance and operability. The amplifier runs extremely hot and goes into an overload condition unless the input is turned up very slowly. Worse yet, the input controls are on the back panel, and they are volatile, so they must be set every time the unit is turned on. The first of these amplifiers was returned because it was thought to be defective, but the replacement unit worked identically. It appears to be a poor design. OSD also makes a higher output version of this amplifier, the XPA500.

An equalization amplifier was tested also. This was a Behringer DCX2496LE. It provides 2 inputs and 6 outputs that can be matrixed and individually equalized. The only feature that I evaluated was the delay which worked fine.

Loudspeakers

Reverberation can be desirable for certain types of musical venues, but it is a huge detriment to voice comprehension. San Juan de Dios church has a great deal of reverberation because of its shape and construction. Since it is a classic Catholic church (with a dome, barrel ceilings, statuary, etc.), it does not lend itself to wall or ceiling treatments that would remedy this. So, the alternative solution is to focus the sound as directly as possible on the audience and avoid sound projection towards the walls or ceilings. Line array type loudspeakers provide a narrow beamwidth in one plane and are used to focus the sound in these situations.

A survey of line array loudspeakers was made, and several were identified that were affordable and which had models in the Ulysses data base. These two restrictions limited the choices severely.

A popular model loudspeaker used in many US churches is the TOA model HX-7 and its smaller version HX-5. These speakers comprise a set of four modules which can be configured to have vertical beamwidths of 60, 45, 30, or 15 degrees. The horizontal beamwidth is 100 degrees. Typically, this system is used overhead to cover the entire audience area. The best location in San Juan de Dios would be at the front of the crossing. Ulysses analyses showed that the best performance was with the vertical beamwidth set to 15 degrees and the height at 4 meters. However, this height blocks the line of sight to the sanctuary; the minimum height that does not do this is 6 meters. With the speaker array at this height, the direct sound uniformity predicted was good (Ld = 2.0), but the clarity was very poor (Ld-Lr = 9.6).

The second loudspeaker evaluated was a TOA model HS1200. This speaker contains a large (12 inch) woofer and a coaxial line array of six 1-inch tweeters. Unlike the other speakers evaluated, it has a built-in line transformer. Two arrangements of these speakers were analyzed using Ulysses. The speakers can be mounted along the sides of the Nave in a staggered cross wise pattern, or they can be mounted symmetrically and aimed towards the rear and center aisle. The staggered approach did not work well as well as the symmetrical design. Using a total of (8) HS1200's the clarity was poor (Ld-Lr = -3.5) when staggered and fair (Ld-Lr = -2.7) when symmetrical. It was determined that the speakers should be mounted as low as possible but no lower than 8 ft for safety. The coverage area should be divided into roughly equal rectangular spaces with the center aisle in the middle, the speakers located at the front outer corner of each of these rectangles, 2.5 meters high, tilted down and aimed at the diagonally opposite corner (the center aisle) at the seated listening height of 1.1 meters. Either 6 or 8 rectangles can be used, but 6 is adequate and was therefore chosen. The tilt angle works out to be 12 degrees.

The third loudspeaker option was recommended by TOA and is their model SR-H2S. This is a tall (26-inch), narrow (3-1/2 inch) line array speaker containing (9) 3-inch tweeters that has a narrower beamwidth (20x90 degrees). It costs about 60% more than the HS1200, but it is more attractive and would fit in better with the Cantera columns. Unfortunately, it was not included in the Ulysses data base and attempts to get it into this data base were unsuccessful. This speaker might have provided better clarity than the HS1200 but there was no way to predict this.

Speaker Brackets and Installation

The TOA bracket made for wall mounting the HS1200 loudspeaker allows for adjustment of both tilt and pan angle. It is expensive, however, and since a fixed tilt angle of 12 degrees is all that is required, a far less expensive version was designed and fabricated locally. Mounting the bracket to the Cantera columns is not difficult since this stone can be drilled readily and tightly grips a 3/8x2-inch plastic wall mount. At least (4) 1/4x2-inch lag bolts were used to support the speaker which weighs about 22 lbs. When the speaker is mounted to the plaster wall, it is much more difficult to get a good grip, and at least four plastic wall mounts 3/8x4-inch with 1/4x4-inch lag bolts were used.

Speaker Wiring

For convenience in testing and operation, the speakers were grouped into zones and each zone wired separately to the amplifier. Since the longest run was about 100 meters, AWG18 gauge wire was used in a 70 V system. A common return requires a larger gauge wire which is much harder to work with. 18-gauge wire is also available in a wide variety of colors to keep the zone wiring separate. Black tape on the ends was used to indicate the “low” side of each speaker’s polarity. Plastic channel that is 3/4x1/2-inch easily accommodated the wiring of 4 zones and makes an attractive and flexible installation. (Wiring can be added or removed easily in the future).

If a small audience is grouped toward the front, there is an acoustical advantage to disconnecting the zones at the rear, because they only contribute to the reverberant sound, i.e., the audience does not hear any direct sound from these speakers. But the normal situation is that the audience spreads out, and all zones are turned on.

Zoning also allows for the possibility of adding delay compensation to the middle or rear speakers. However, this did not prove effective at San Juan de Dios church because there was considerable sound moving from back to front due to radiation from the rear of the speakers and reflections from the walls. Adding delay compensation in this case did improve the clarity at the rear but made it worse at the front.

The HS1200 speakers have built in transformers that have taps for 60, 30, 15 and 7.5 watts. Even though the areas covered are unequal, it was found that the best uniformity and clarity was achieved if all of these speakers were set the same (30 watts). The Transept loudspeakers are CS212’s, and they were set for their maximum output of 12 watts.

Sound Absorption

The bare plaster walls absorb very little sound. Testing indicated that 90% if the sound was reflected from the walls and ceilings.

The rear wall of the church under the balcony is the source of a significant reflection from all the HS1200 speakers which are aimed in that general direction. When the sound system was tested with moving blankets attached to this wall, the clarity improved significantly. This wall is also the easiest to treat without upsetting the esthetics of the church as no decorations or statuary exists along it, and in general no one is facing it anyway. In fact, when the (ugly) moving blankets were left intact over the weekend, no one mentioned noticing them.

All testing was done with an empty church. The seating area for this case is very reflective as the pews have no cushions and the floor is tile. When the church is full, the reflections from the floor and seats will be reduced significantly, and the clarity will be much better.

Cost

The following are the costs which include shipment to Mexico. The rear wall treatment is an estimate. All prices are in US dollars.

Table 3 Sound System Cost

6	TOA HS1200 loudspeakers @ $400	$2400
6	HS1200 brackets locally fabricated and painted @ $75	$450
1	OSD PAM245 amplifier	$600
---	Wiring and conduit	$300
---	Rear wall acoustic treatment	$500
	Total	$4250

Appendix - Schroeder Integral Calculations

The Schroeder integral or energy decay curve S(t) is the total amount of signal energy remaining in the reverberant impulse response at time t: If h(t) is the impulse response,

$$S(t) = 10log\int_t^\infty h(\tau)^2 d\tau$$

$$s(t) = \int_t^\infty h(\tau)^2 d\tau$$

Upper- and lower-case letters are used here to distinguish between the dB value and linear value of the same variable.

Many useful audio characteristics can be found from the Schroeder integral. For our case, we are interested in the energy contained in two intervals following the arrival of the first significant pulse.

Normally (but not always) the first pulse is the largest. But with multiple loudspeakers, it is possible that some smaller pulses might arrive before the largest pulse. If these are more than 10 dB below the largest pulse, they can safely be ignored as they will add less than 5% (0.4 dB) to the total energy. But if an early arriving pulse is larger than that, it should be considered the first pulse. In any case, the first pulse will be delayed a few milliseconds from when the impulse is generated (time zero). Call the time of the first arrival T1.

We are interested in the direct energy Ld35 and the ratio of the direct to reflected energy, Ld35 - Lr35, where the 35 refers to the time 35 msec after the first pulse arrives, that is, T1 + 35 msec. The 35 msec is one of several intervals that have become standard in audio measurements. It corresponds to the time interval during which reflections are perceived by the human ear constructively. (This is known as the Haas Effect.)

Starting with the definition of Ld35,

$$Ld35 = 10log\int_{T1}^{T1+35msec} h(\tau)^2 d\tau$$

$$ld35 = \int_{T1}^{T1+35msec} h(\tau)^2 d\tau$$

$$ld35 = \int_{T1}^{\infty} h(\tau)^2 d\tau - \int_{T1+35msec}^{\infty} h(\tau) d\tau$$

$$ld35 = s(T1) - s(T1+35)$$

$$Ld35 = 10log\left[10^{S(T1)/10} - 10^{S(T1+35)/10}\right]$$

So, by reading off the dB magnitude of the Schroeder integral at times T1 and T1+35 msec, we can use the above equation to calculate the direct sound level L35 in dB.

The second parameter we want to calculate is the ratio of the direct to reflected sound energy, Ld35 – Lr35. This is also known as the clarity, C35.

$$C35 = 10\log\left[\frac{\int_{T1}^{T1+35msec} h(\tau)^2 d\tau}{\int_{T1+35msec}^{\infty} h(\tau)^2 d\tau}\right]$$

$$c35 = \frac{\int_{T1}^{T1+35msec} h(\tau)^2 d\tau}{\int_{T1+35msec}^{\infty} h(\tau)^2 d\tau}$$

$$c35 = \frac{s(T1) - s(T1+35)}{s(T1+35)}$$

$$c35 = \frac{s(T1)}{s(T1+35)} - 1$$

$$C35 = 10log\left[\frac{10^{S(T1)/10}}{10^{S(T1+35)/10}} - 1\right]$$

$$C35 = 10log\left[10^{\left(\frac{S(T1)}{10} - \frac{S(T1+35)}{10}\right)} - 1\right]$$

The calculation of C35 and Ld35 from S(T1) and S(T1+35) is implemented in a simple spreadsheet:

S(T1) dB	S(T1+35) dB	C35 dB	Ld35 dB
103.0	96.5	5.4	101.9

If S(T1) above is in cell C4 and S(T1+35) is in cell D4, the formulas are:

S(T1) dB	S(T1+35) dB	C35 dB	Ld35 dB
103	96.5	=10*LOG(10^(C4/10-D4/10)-1)	=10*LOG(10^(C4/10)-10^(D4/10))

Schroeder Integral in Ulysses

The released version of Ulysses is 2.82. Although it is not mentioned in the documentation, the Schroeder integral is plotted (very faintly) in the impulse response Energy Time Curve (ETC) that is calculated using the Calculate Ray feature. However, the plotted values are not much use in this version of the program because the automatic scaling used causes the Schroeder integral to limit during the time interval where we need to measure it.

After some discussion with Ulysses' customer support, their plotting code was modified to eliminate this problem and a new beta version 2.83 was provided. The integral is also now much more distinguishable.

There are still a few cautions to be watched for however that are probably bugs in the beta version. In the following examples, model SJD45 is used but with all speakers except the front two HS1200's turned off. The listening location is row 18 center (at the very back).

Ulysses' Ray calculations allow the user to specify how many reflections to include in the Ray or ETC calculations

1. The displayed Schroeder integral is not correct when the number of reflections is 1 or 3. For example, when 3 reflections are selected, the ETC is as shown in Figure 20.

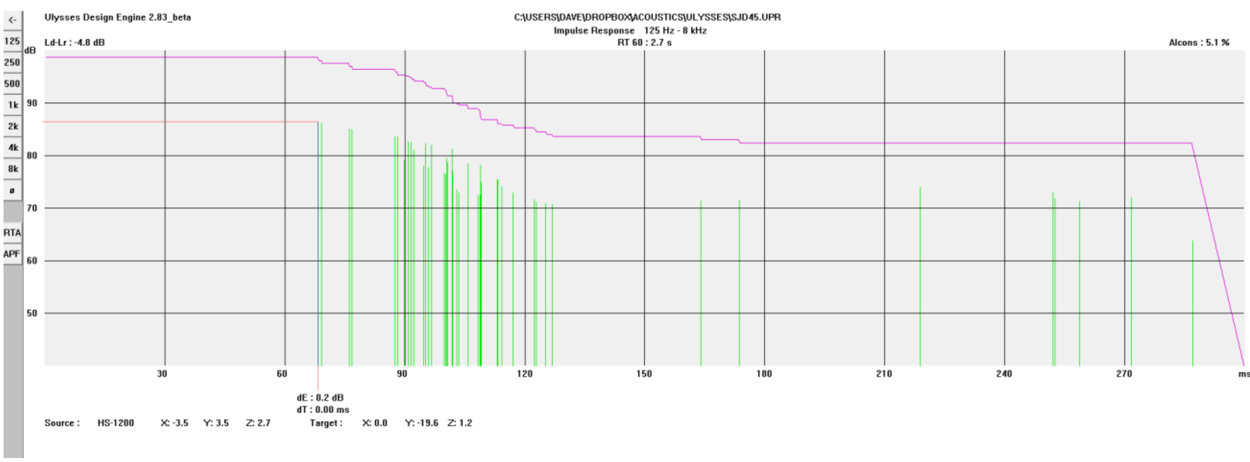


Figure 20 Ulysses Schroeder Integral "Flat" between 180 and 300 msec (3 Reflections Selected)

Note that the Schroeder integral does not change between 180 and 300 msec even though there are half a dozen pulses in this period.

Similarly, when 1 reflection is selected, the Schroeder integral does not change below 100 msec. See Figure 21.

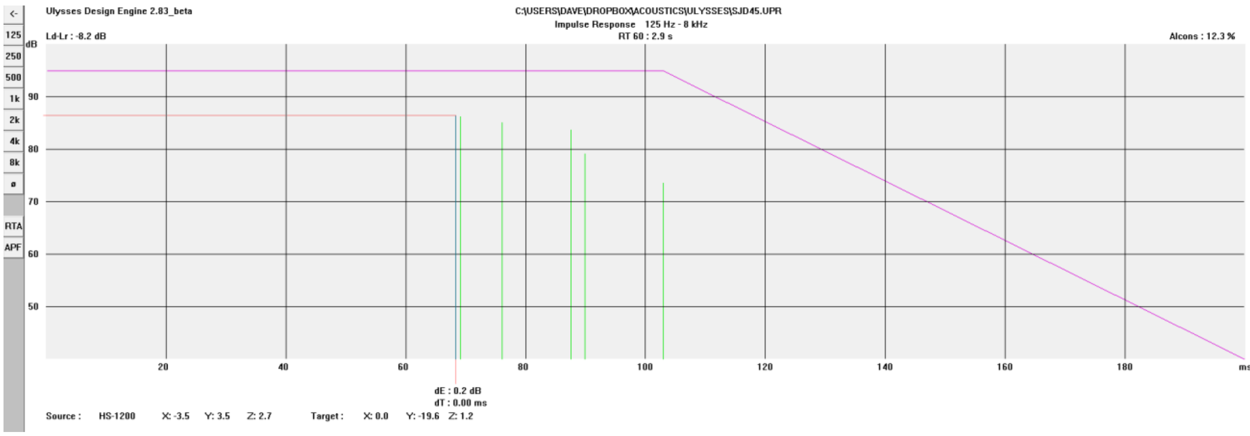


Figure 21 Ulysses Schroeder Integral "flat" below 100 msec (1 Reflection Selected)

Oddly, the Schroeder integral looks correct for 2 reflections (Figure 22).

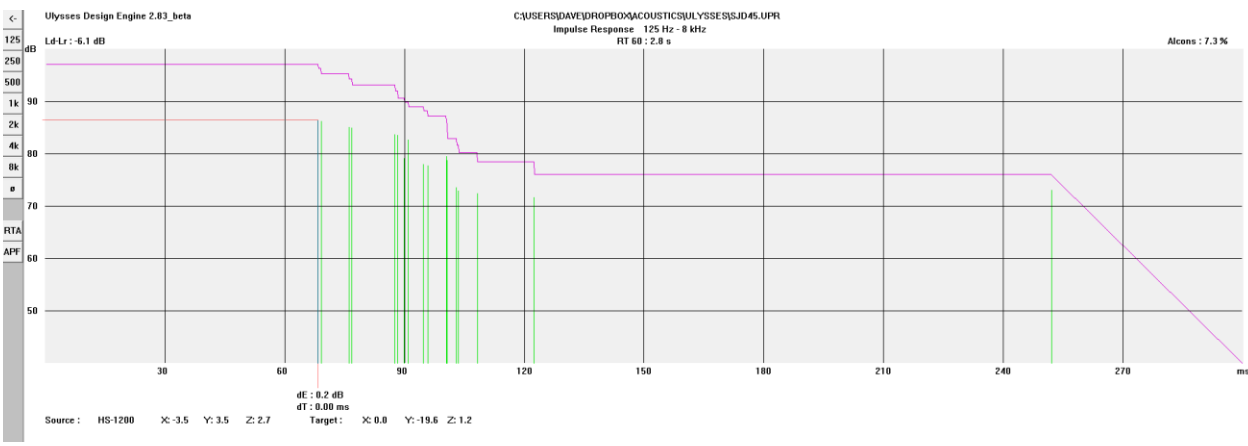


Figure 22 Ulysses Schroeder Integral Correct with 2 Reflections

2. There is an odd occurrence in the tail of the Schroeder integral for reflections greater than 5. Sometimes there are drops in the integral much larger than would be suggested by the pulses, and at other times there are no changes in the integral where they would be expected to occur. Fortunately, these do not appear to affect the calculation of either C35 or Ld35 since these anomalies are well beyond the 35 msec interval and where the integral itself is very small. Figure 23 below shows this. (This is the 10-reflection case). Note the large drop at 430 msec.

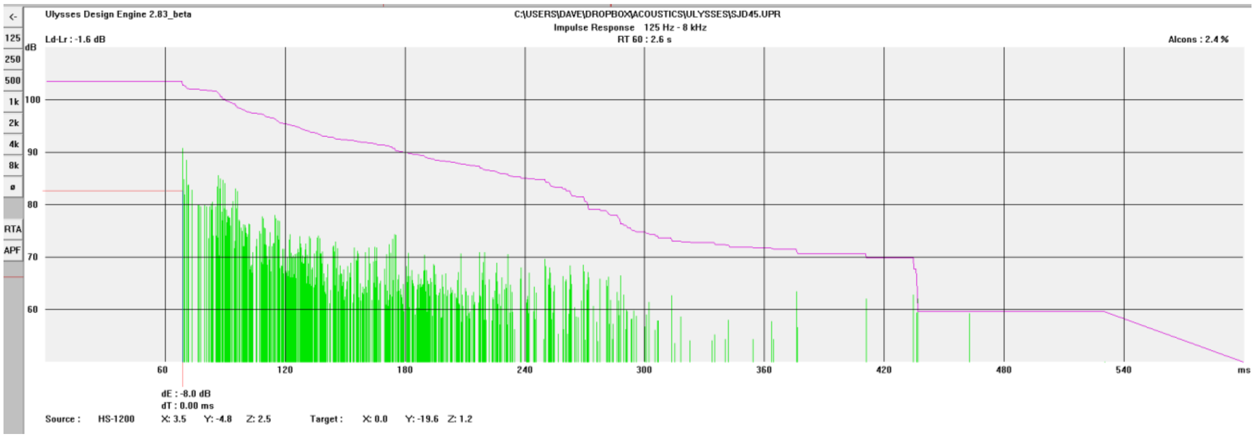


Figure 23 Ulysses Schroeder Integral Tail Problem with Multiple Loudspeakers and 10 Reflections

Presumable the more reflections included in the calculations, the more accurate the results, but the calculation time will be longer.

To evaluate this, the following calculations were made with all speakers on at nominal power and the delay compensated for the listening location at row 18 center.

Reflections	S(T1) dB	S(T1+35) dB	C35 dB	Ld35 dB	Time
5	103.0	96.5	5.4	101.9	2:56
7	103.5	97.0	5.4	102.4	3:02
10	103.5	97.5	4.7	102.2	3:11
15	103.5	97.5	4.7	102.2	3:25
20	103.5	97.5	4.7	102.2	3:39

The calculation time “penalty” is very modest. If more accuracy were available by running more reflections, it would be well worth doing. But this does not seem to be the case, because the results are the same within the resolution of the integral measurement (0.5 dB) for all reflections over about 7.

Ulysses does not provide for direct readout of the Schroeder integral values. The most accurate method found was to use the print command from the ETC plot, then measure the values using a cm rule because the printed plot scales to 1 dB per cm. Unlike Smaart8, Ulysses always selects the first arriving pulse as the zero time for the cursor offset dT. To get the correct 35 msec time offset, select the reflection using the > button that is closest to dT = 35 msec before printing the plot. See Figure 24.

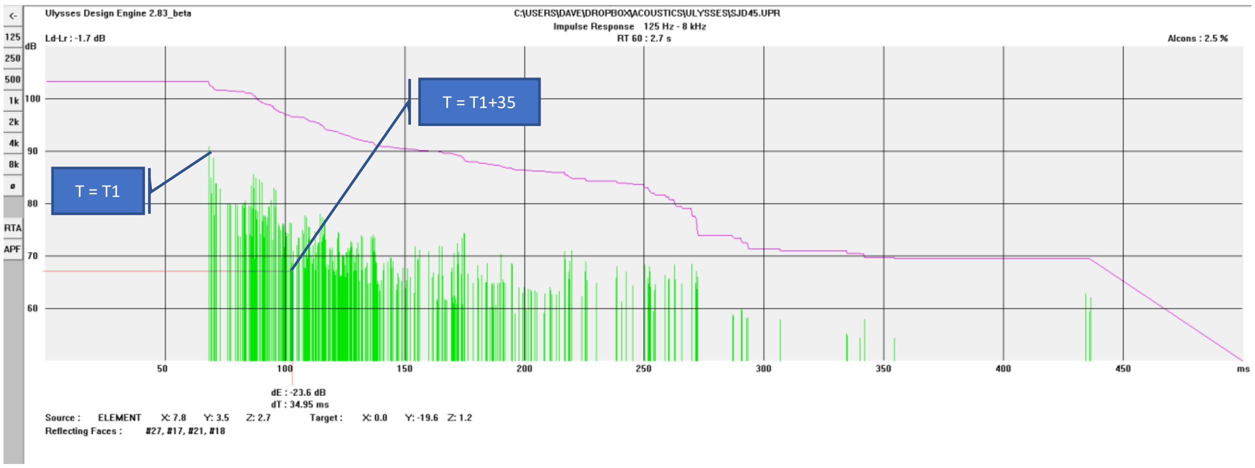


Figure 24 Ulysses Schroeder Integral with 5 Reflections Showing Marker at T1+35 msec

Schroeder Integral in Smaart8

The impulse response mode in Smaart8 includes an option to display the Schroeder integral. The version of the program used for these tests is beta version 8.5.8093.1 which is far better than the previous released version of Smaart8 in terms of the impulse response mode. The latest released version 8.5.2.1 also has these improvements.

Smaart8 automatically calculates the clarity C35 from the Schroeder integral, but not the direct sound level Ld35. Since both Ld35 and C35 require reading the Schroeder integral value at T1 and T1+35msec, it is more consistent to calculate these values as described above rather than use C35 calculated automatically. It is also important to view the Schroeder integral to be sure that the location of the marker Ln is correctly placed by the program. The beta version of the program does a very good job of placing this, but, if necessary, it can be manually adjusted.

Figure 25 shows a typical measured impulse response before expansion. The control buttons in the lower right corner marked T60 and Schroeder are set to display both the markers (Ld, Ln, etc.) and the Schroeder Integral. The position of Ln can be manually adjusted, but the position chosen by the program is about right for making the start of the Schroeder integral coincident with the “corner” of the impulse response data.

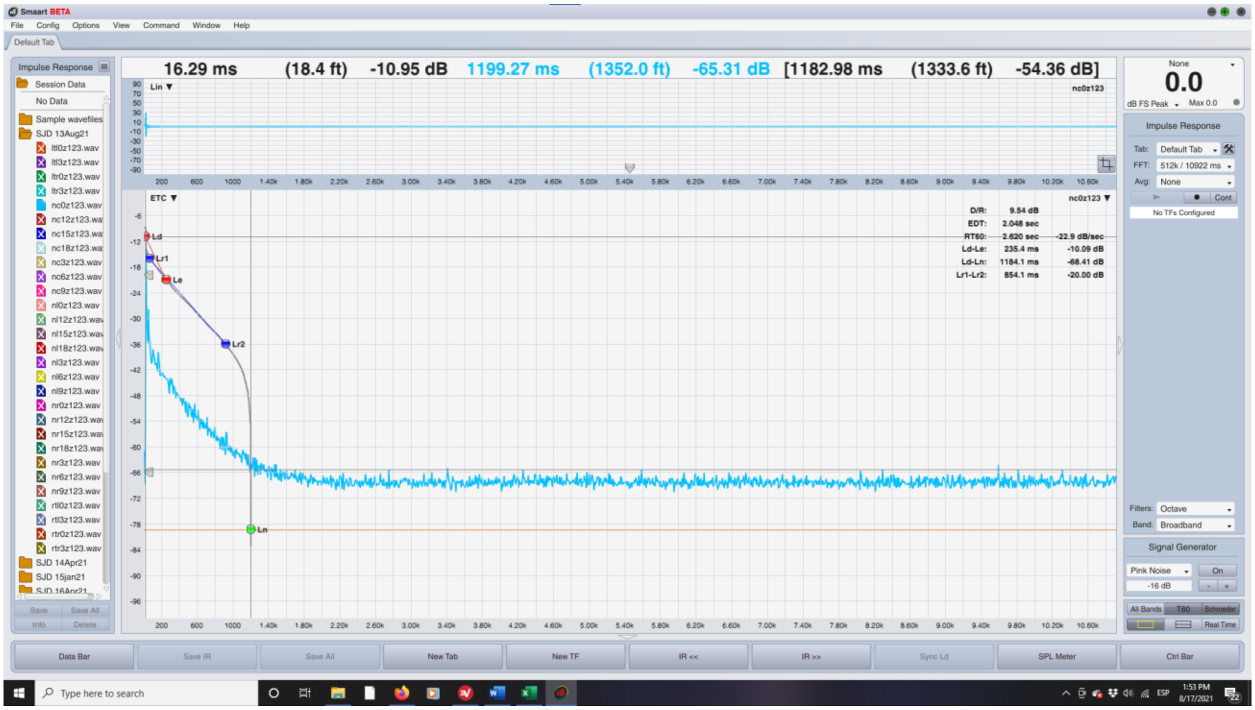


Figure 25 Smaart8 Full Trace Impulse Response

To accurately read the values S(T1) and S(T1+35), the impulse response must be expanded using the zoom tool in the lower right corner of the linear plot at the top. The marker display (T60) should be turned OFF and the vertical scale expanded for accuracy.

The data in the upper left shows that the peak occurs at 16.29 ms and has a magnitude of -10.95 dB. This is NOT the value of the Schroeder integral at the peak. The Schroeder values must be read using the variable cursor and are indicated in the center set of numbers at the top. For example, the cursor in Figure 26 is set to 52.65 ms and the Schroeder integral is -15.71 dB.

The impulse shown in Figure 26 has one clearly defined large peak which the program properly identifies by the fixed cursor. However, to read the Schroeder value at this peak, we need to position the variable cursor slightly to the left of this peak where we read S(T1) = -8.85 dB at 16.0 msec. To find the value 35 msec later, we position the cursor at 51 msec and read S(T1+35) = -15.64 dB.

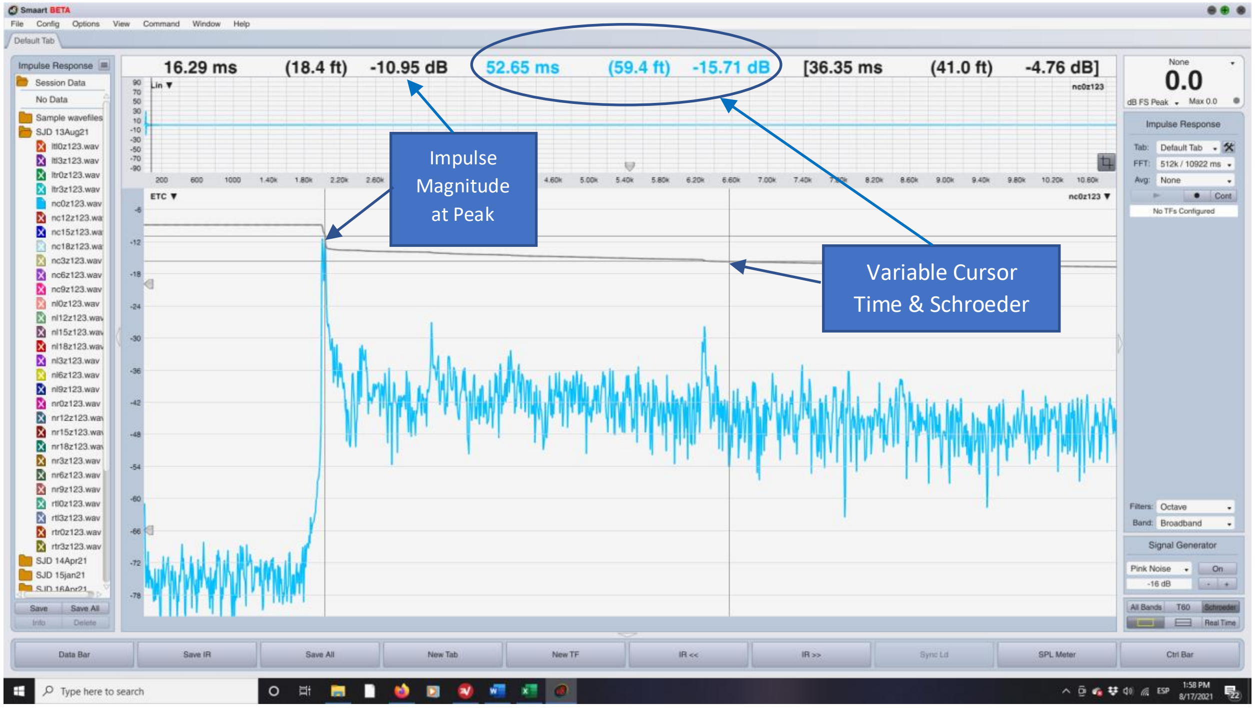


Figure 26 Smaart8 Impulse Response Expanded

If we enter these values into the spreadsheet calculator, we get:

S(T1) dB	S(T1+35) dB	C35 dB	Ld35 dB
-8.85	-15.64	5.8	-9.9

For comparison, Smaart8 calculates the value of C35 for this case as C35 = 2.71 dB.

Figure 27 shows another impulse response, this one having a large "pre-echo". Smaart8 selects the larger (and later) of the two peaks as T1 = 32.54 ms but we can manually choose the first peak to be T1 = 12.33 ms instead. So, S(T1) = -14.75 dB and S(T1+35) will be -19.16 dB. Entering these values in the spreadsheet calculator we get:

S(T1) dB	S(T1+35) dB	C35 dB	Ld35 dB
-14.75	-19.16	2.5	-16.7

For comparisons, Smaart8 calculates C35 to be -0.02 dB

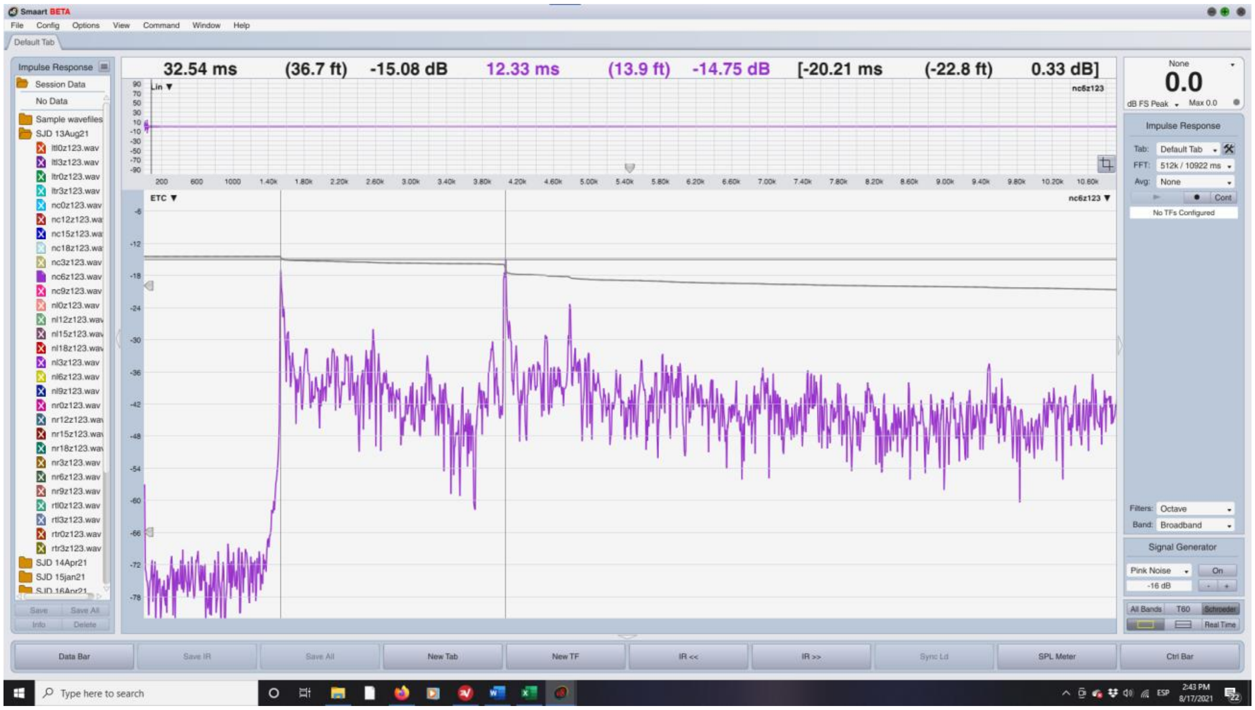


Figure 27 Smaart8 Impulse Response Having "Pre echo""

The usual approach appears to be not to include "pre-echoes". But if they are audible, it would seem logical that they should be included in the clarity calculation.