Auditorium Acoustics

Foellinger Great Hall

Krannert Center for the Performing Arts

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Physics 406: The Acoustical Physics of Music

University of Illinois at Urbana-Champaign

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I. Introduction

Musicians and concertgoers alike know the importance of good concert hall acoustics. Concert hall design is complicated, and much of the calculations made involve approximations, especially in terms of room volume, area of “holes,” absorption properties of seats (and the audience), and reflection properties of walls and ceilings. Sound travels at 344 meters per second, so it typically takes anywhere from 20 to 200 milliseconds for sound from, say, a loudspeaker on stage or an orchestra, to reach the audience, depending on where each member of the audience is seated. This direct sound is followed by a first reflection from walls or ceilings, and then the reverberant sound dominates the arena. Our focus is on this reverberant regime of auditorium acoustics.

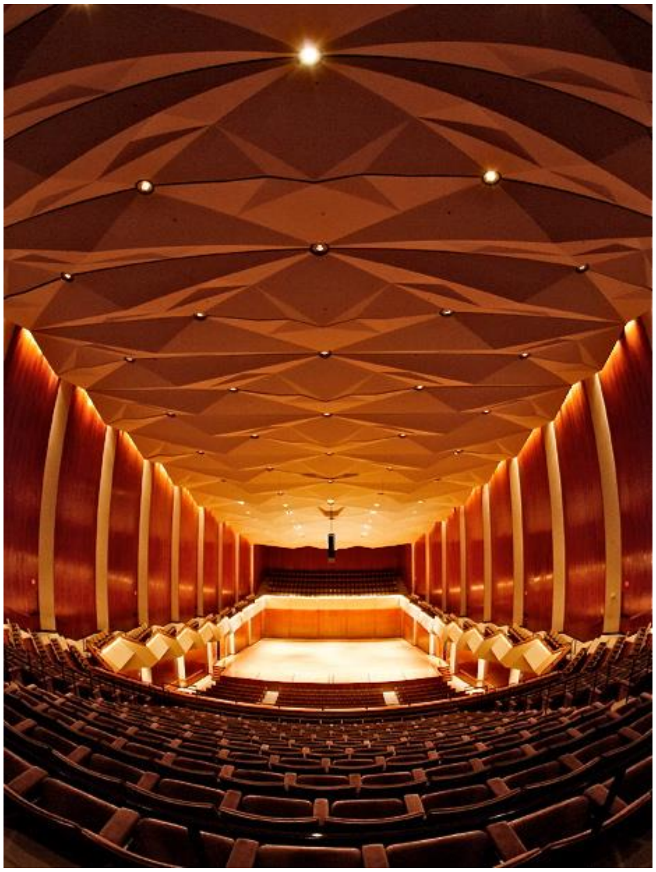


Figure 1: Foellinger Great Hall (Unoccupied).

In the Krannert Center for the Performing Arts, the Foellinger Great Hall proudly touts a reputation for being one of the most acoustically perfect concert halls in the world! There are very, very few parallel surfaces in the venue—if any at all—so the sound reflects directly toward the audience and is prevented from being trapped in corners. The room has such superior sound that amplification is strongly discouraged. Designed by professional acoustician Dr. Cyril Harris in the early 1960s, and with the help of a then state-of-the-art computer system, the hall is renown for its versatility, hosting string quartets, operas, drum concerts, ballets, bagpipes, and thousands of other musical performances over the four decades of its existence. With 2,066 variable-absorption plush seats, a huge 60- by 48-foot hollow white oak stage, beautiful Indiana butternut wall panels, and a remarkable tiered acoustic ceiling made of heavy-duty springs and plaster, the Foellinger Great Hall will continue to be an extraordinary place to see great live performances for many years to come.

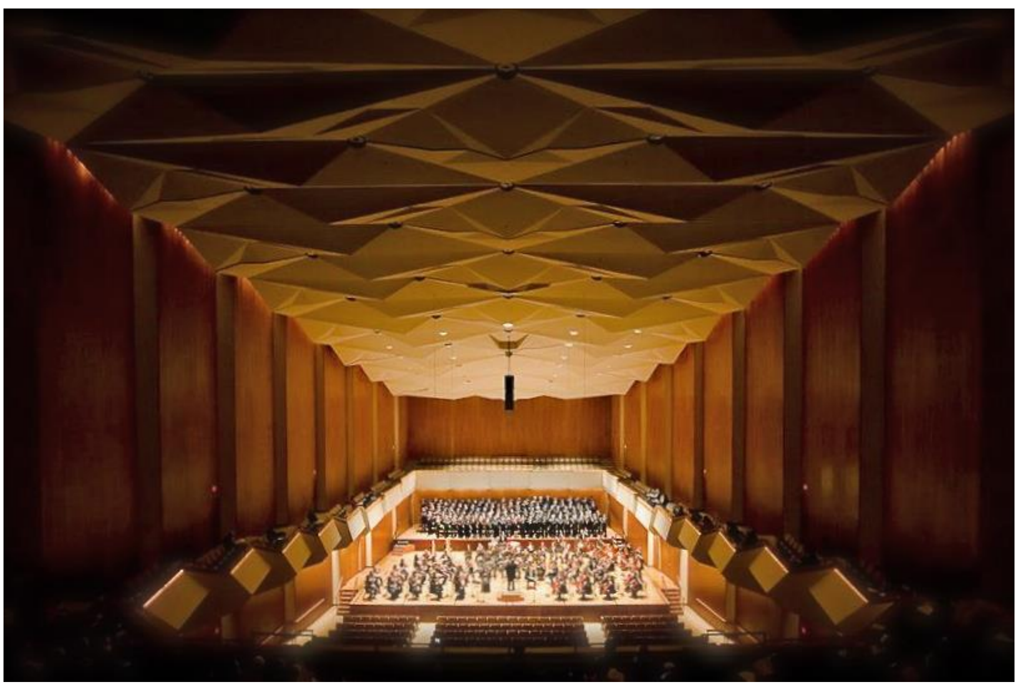


Figure 2: Foellinger Great Hall (Occupied).

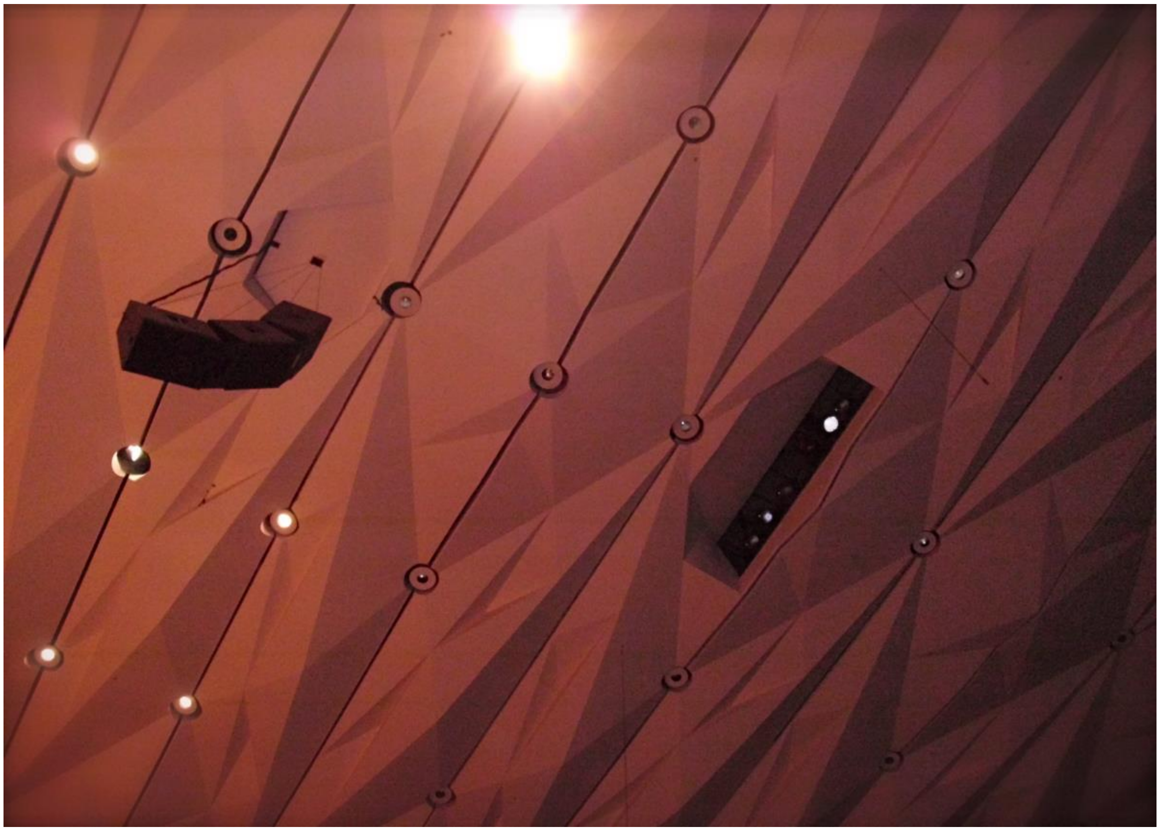


Figure 3: Tiered Ceiling (Plaster Suspended From Springs).

As a world-famous music venue, the Foellinger Great Hall is an excellent place to perform auditorium acoustics measurements. In particular, my concern was with gathering information about the reverberation time at various places in the hall. Reverberation time is one particular characteristic of auditoriums that people are most familiar with; it often gives a good indication of “liveliness” of an auditorium. The reverberation time is defined as the time required for the stored or reverberant sound to decrease by 60 decibels, or to a millionth of its original intensity. This is also known as T-60. Similarly, T-30 is the time required for the reverberant sound to decrease by 30 decibels, or to a thousandth of its original intensity. Our analyses will look at both T-60 and T-30 reverberation times.

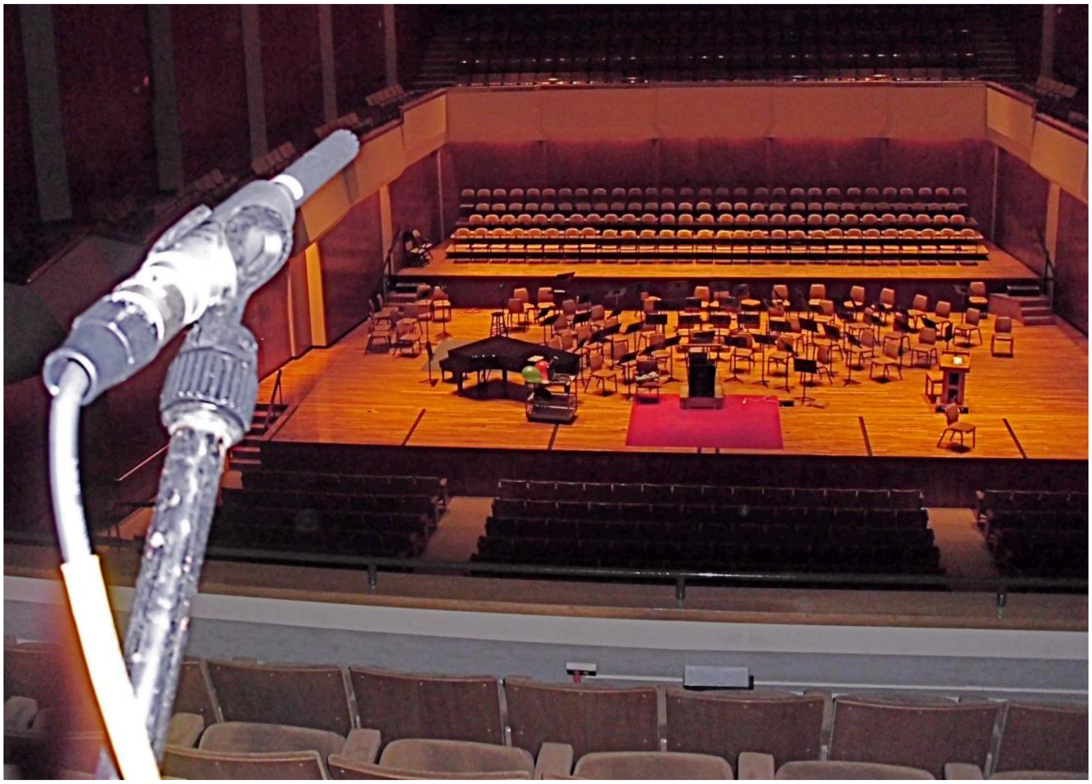


Figure 4: Balcony Microphone (Center) and View of Balloons and Gear on Stage.

Reverberation time can be calculated by hand using the Sabine equation, which relates the volume of the room, the total surface area, and the absorption coefficient. Unfortunately, the exact dimensions of the Foellinger Great Hall were not attained, so a theoretical calculation regarding the expected reverberation time was not accomplished. However, I heard from Eric Bisgyer, a fellow student in PHYS 406 and an employee at the Krannert Center for the Performing Arts, that the reverberation time of the Foellinger Great Hall was on the order of 1.5 to 2.5 seconds, which is in agreement with what is considered to be an ideal range for music halls. It would be a great project in the future to make detailed measurements of the dimensions of the Foellinger Great Hall. That way, a theoretical value for the reverberation time can be documented. I searched long and hard for this information online, but to no avail.

II. Equipment

The function generator used was an Agilent 33220A. It is capable of square waves, pulses, and custom waveforms. (We chose to use 1-millisecond square wave pulses; we realized that anything longer than 1-ms resulted in undesirable behavior, so we will not analyze the 2-ms to 64-ms pulse files.) The power amplifier used was a Marantz Model 510 (120 VAC, 60 Hz, 8 A). The loudspeaker used was an Electro-Voice S-152 two-way stage system (8-Ω nominal impedance, 200-W power capacity). The digital recorder used was a Marantz PMD671 (solid state, 24-bit, WAV, one-touch, portable). The microphones used were two identical Behringer ECM 8000s (electret condenser, omni-directional, 600-Ω impedance, -60 dB sensitivity). We used 15-inch latex balloons to burst at stage center.

Figure 5: Agilent 33220A Function Generator.

Figure 6: Marantz Model 510 Power Amplifier.

Figure 7: Maranz PMD671 Digital Recorder.

Figure 8: The Complete Sound Source Setup.

Figure 9: Behringer ECM 8000 Condenser Microphone.

III. Measurements

We recorded 40 sound files at three different locations in the auditorium. We placed a pair of microphones at stage left and stage center for each of these three locations. First, we placed the microphones in Row F (7^{th} row), one directly in the center of the main floor and another at the fourth seat from the stage left aisle. The microphones were then moved to Row T (20^{th} row) and placed in the same position as mentioned above. Finally, the microphones were moved to Row H (8^{th} row) of the balcony. For each of the three locations, three types of measurements were made: 1) white noise (filled the room with "white noise" of "all frequencies" and "quickly" turned off the source), 2) 1-ms square impulses (generated by the function generator's square wave option), and 3) balloon burst (punctured fully-inflated 15-inch latex balloons at approximately 6 feet above the stage floor at the front of stage center).

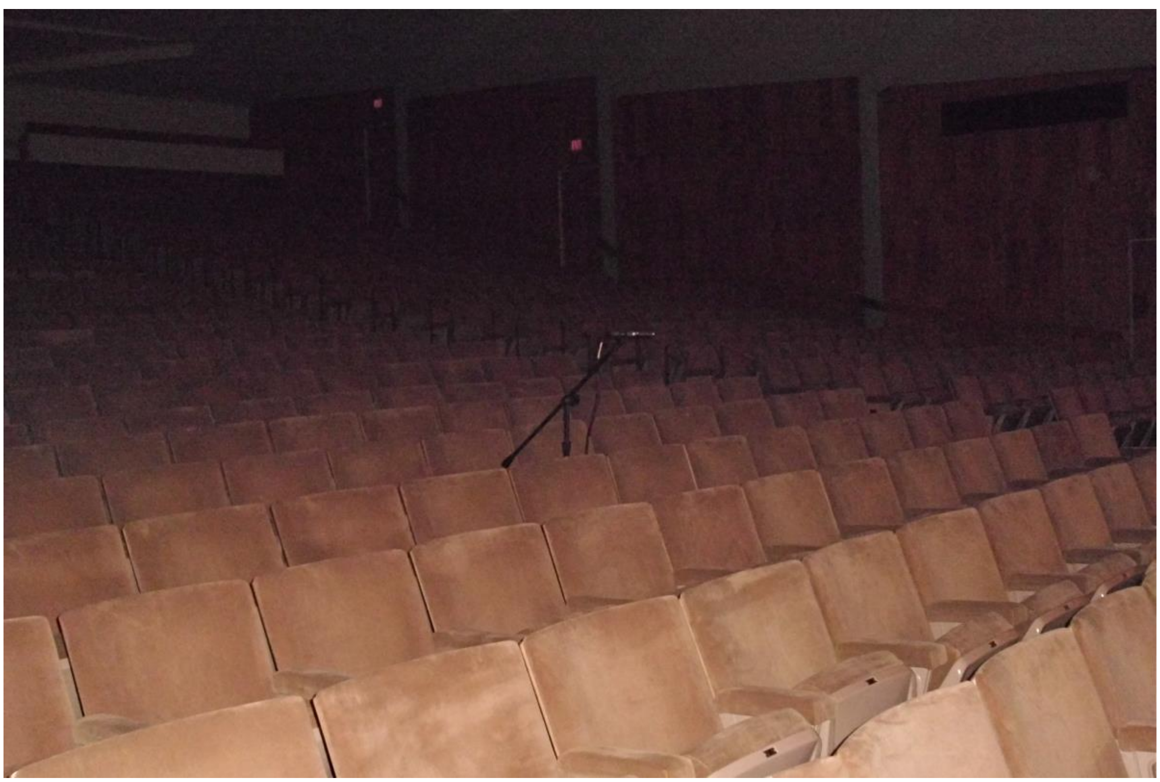


Figure 10: Microphone Placement (Main Floor, Center).

Professor Errede was in charge of generating the sounds with the function generator, the power amplifier, and the loudspeaker. He also performed the white noise "cut-off" by quickly turning down the level knob on the amplifier after the room was filled with white noise (i.e., noise in a wide range of audible frequencies). Eric Bisgyer was the recorder. He was in charge of capturing each of the sound trials using the digital recorder situated between the microphones, and he did an excellent job of adjusting the recorder to accommodate the varying sound intensity levels. In particular, he lowered the input on the recorder if the levels were too high (i.e., in the "red zone"). Also, Professor Emeritus Leland E. Holloway helped with the setup and teardown and in making some dimensional measurements of the hall.

Figure 11: Professor Errede Setting Up the Recorder.

IV. Data

As previously mentioned, we obtained 40 sound files in all. These were then imported into MATLAB and manipulated with a series of files written by Professor Errede. The MATLAB code was originally capable of performing analysis specifically on the white noise sound files; however, Professor Errede spent a tremendous amount of time updating and revamping the code in order to accommodate the impulsive sound files, too (i.e., the 1-ms square wave pulses and the balloon bursts). The code is very versatile in that it allows for several tests to run simultaneously. Unfortunately, though, the computers running MATLAB in the PHYS 406 laboratory do not have a tremendous amount of RAM, and these tests are quite complicated, what with the heavy computations, so only one or two tests can be run at a time.

I performed a total of 18 unique tests on the data. The breakdown is as follows: 9 tests for each channel, left (i.e., stage left microphone) and right (i.e., stage center microphone). Of the 9 tests per channel, there were 3 tests performed for each of the 3 microphone locations (i.e., front, middle, and balcony). Finally, for each of the 3 locations, 1 white noise, 1 1-ms square wave pulse, and 1 balloon burst test was run. It was very important to be careful and diligent in selecting the "area of interest" in the original graphs that appeared when a sound file was loaded and a particular test was selected. The user had to carefully select the region just to the right of the initial (step-like) increase in the sound intensity. The code was written in such a way as to detect that region of interest and then select an appropriate sampling interval.

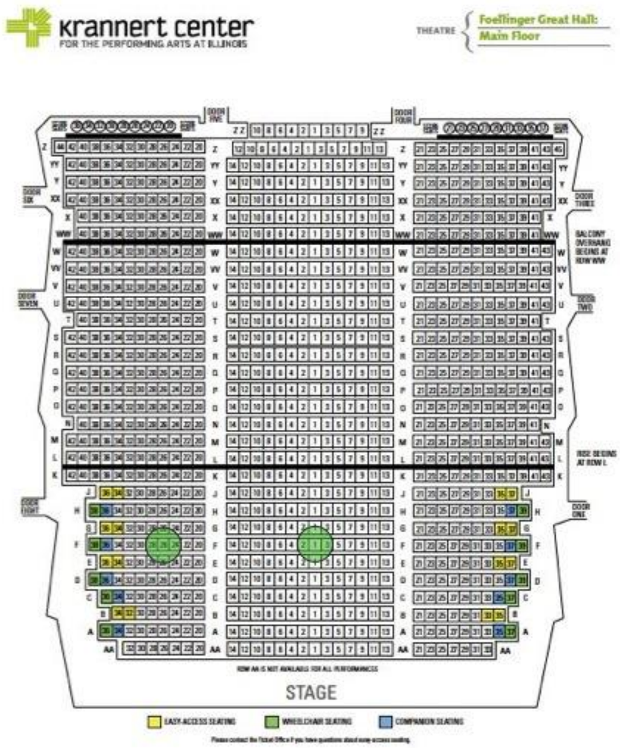


Figure 12: Microphones Placement (Front).

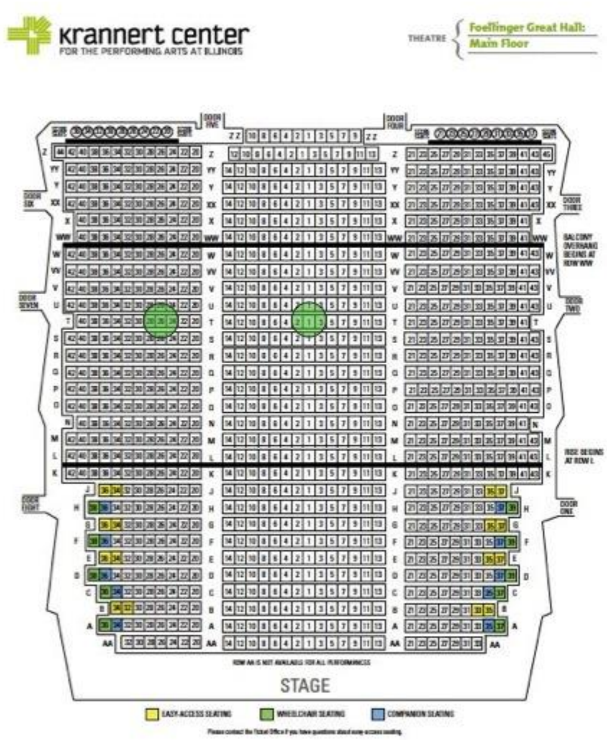


Figure 13: Microphones Placement (Back).

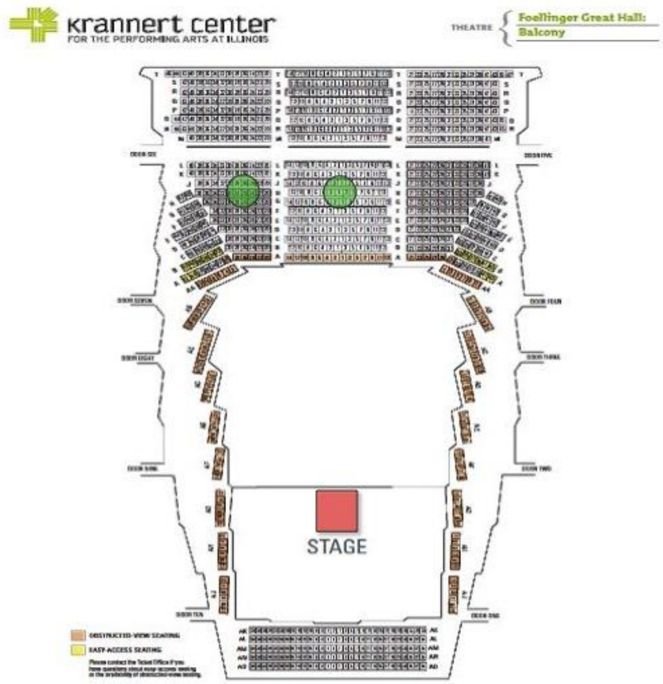


Figure 14: Microphones Placement (Balcony).

All of the calculations were then run on the selected interval. In particular, the reverberation time calculations were performed, and figures and numerical data were generated. The figures and numerical data were then relocated to strategically named folders and backup locations for safekeeping and easy access during analysis time.

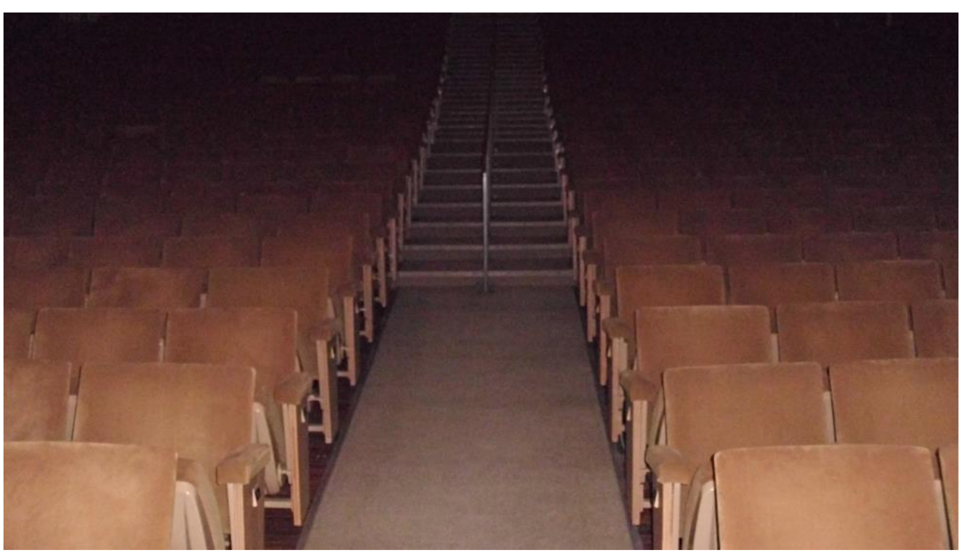
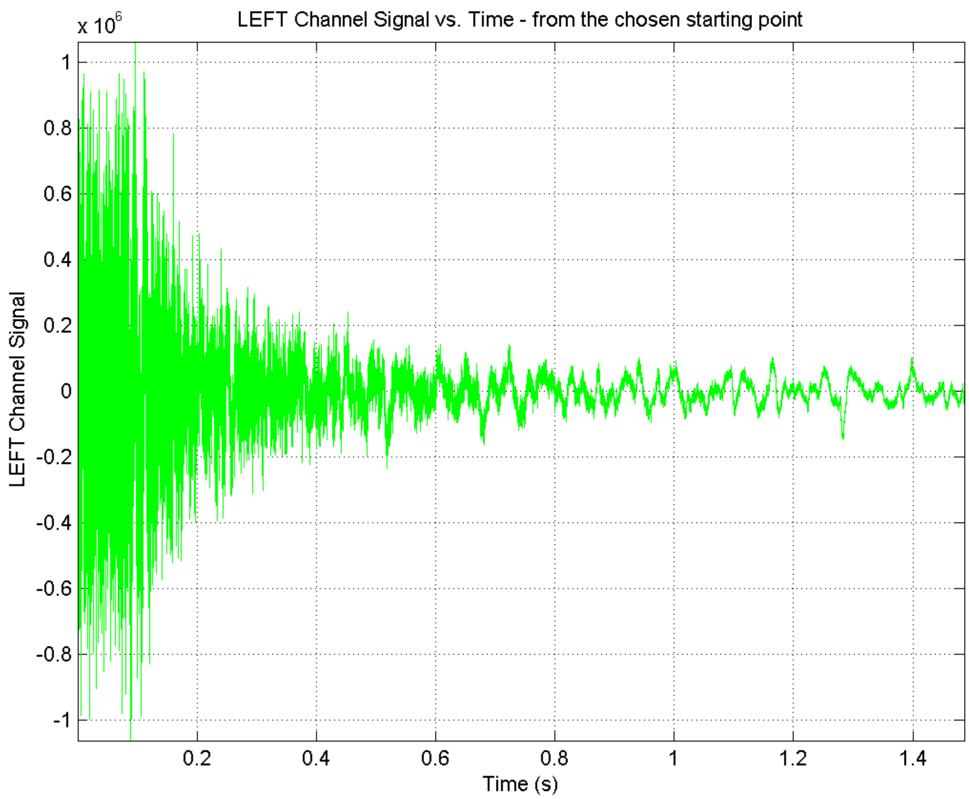
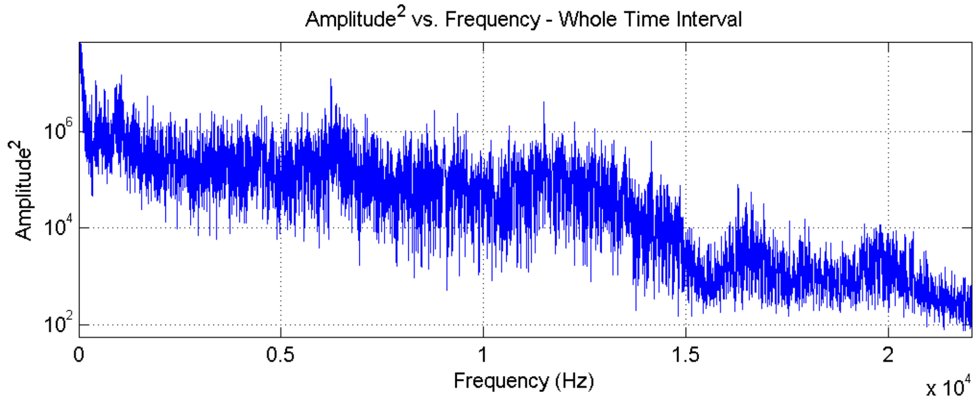


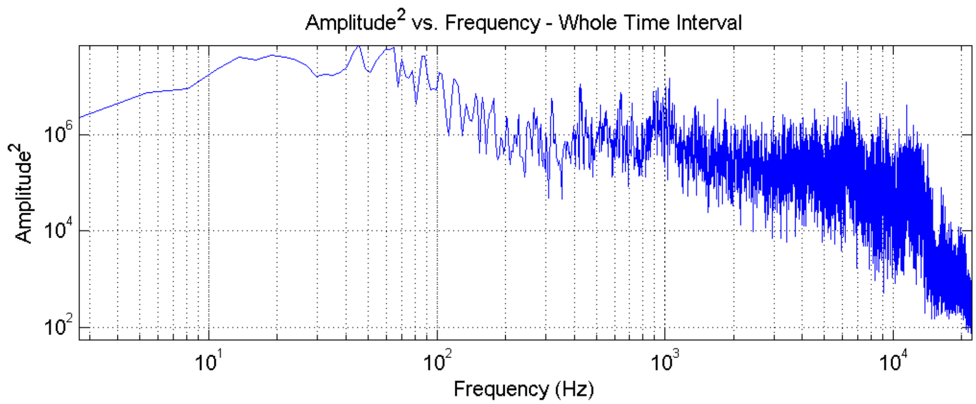
Figure 15: Main Level Aisle.

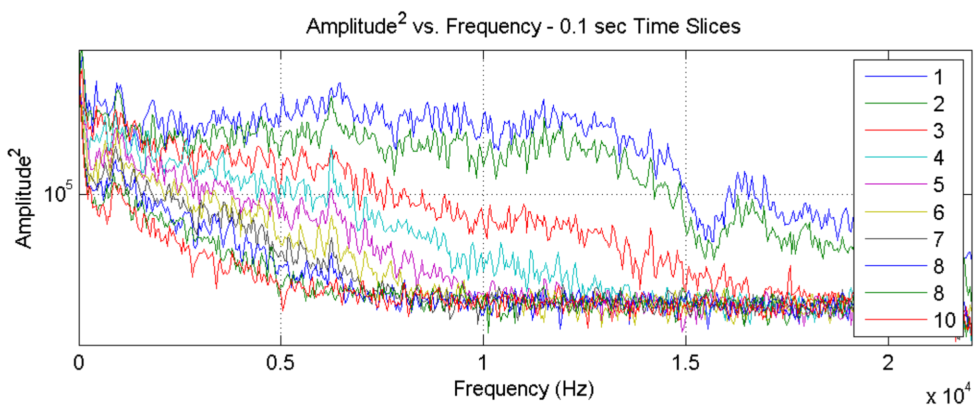
V. Analysis

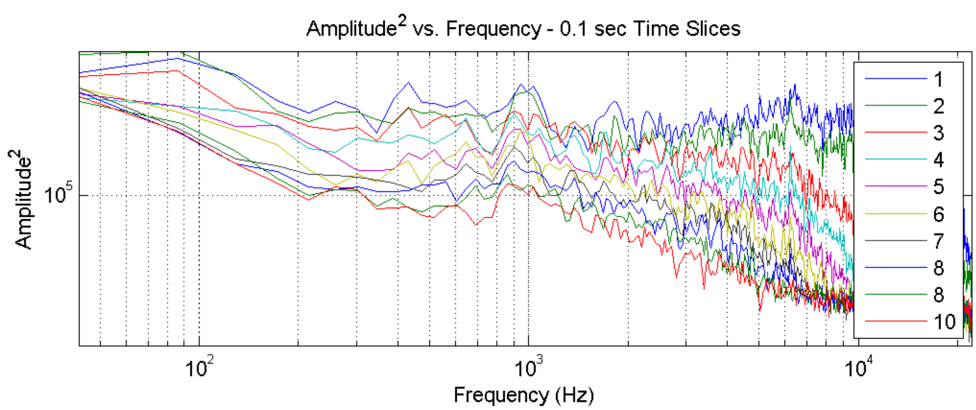
There were many figures and numerical tables generated for each test, so identifying which ones were of interest was quite a daunting task. The following are plots created by the MATLAB program. They all correspond to the left channel (stage left microphone) that was situated in the front configuration (i.e., 7th row). The sound analyzed was white noise, and the test was reverberation time.

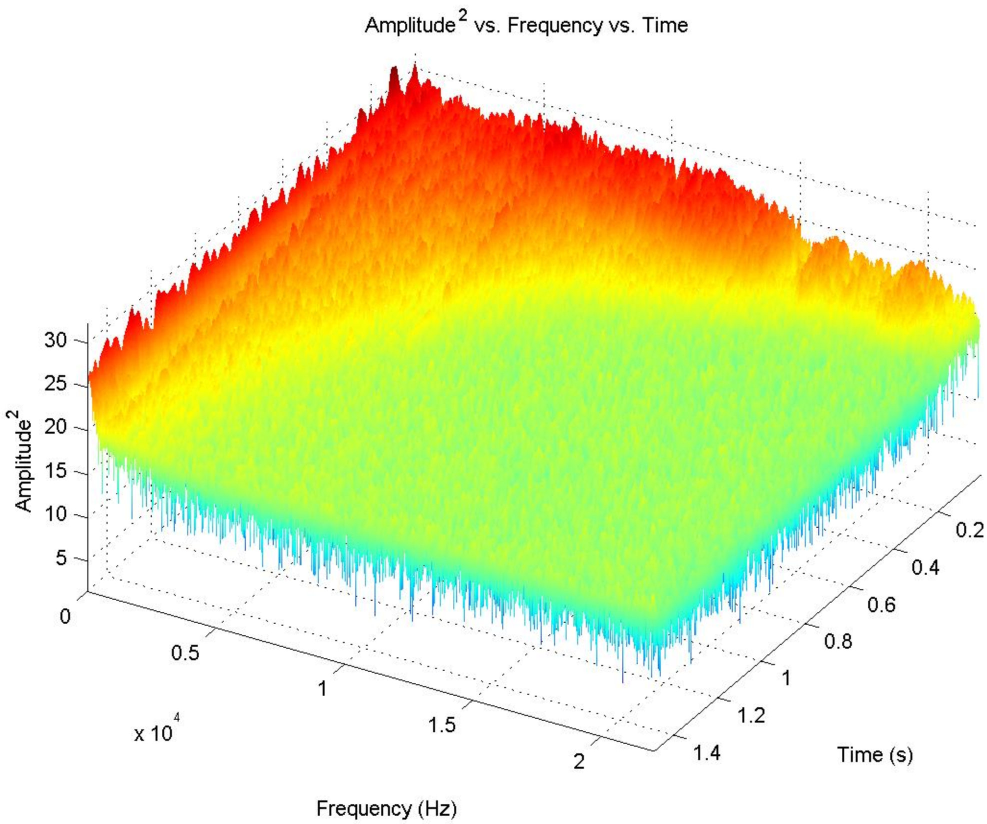



Amplitude^2 vs. Frequency - Whole Time Interval
Amplitude^2
10^6
10^4
10^2
0
0.5
1
1.5
2
Frequency (Hz)
x 10^4

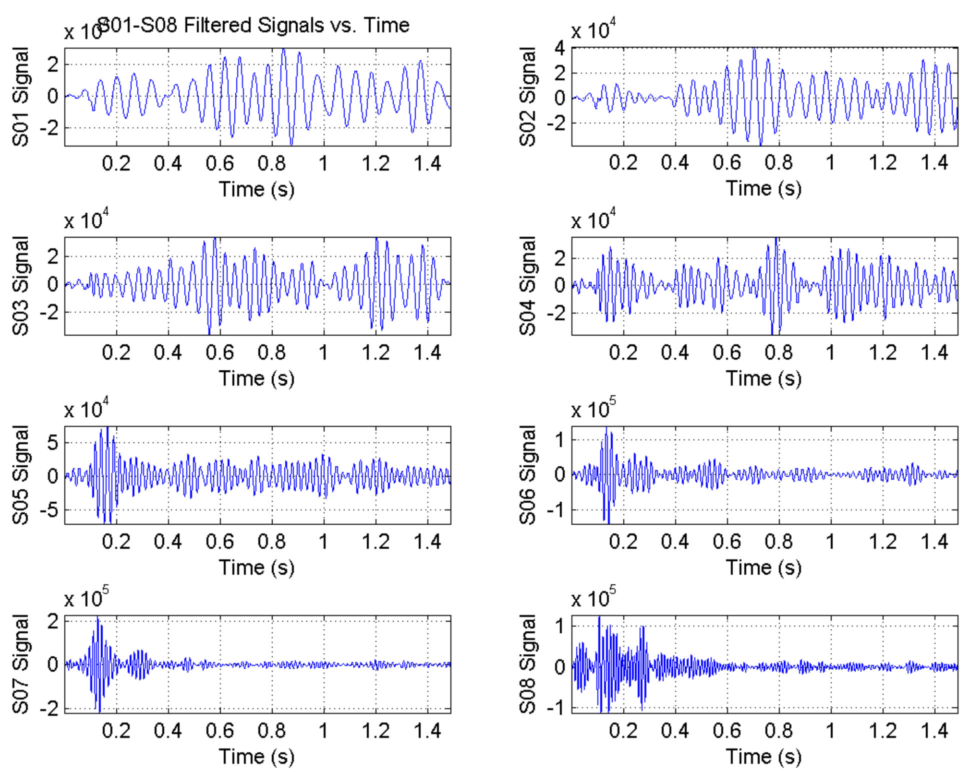

Amplitude^2 vs. Frequency - Whole Time Interval
Amplitude^2
10^6
10^4
10^2
10^1
10^2
10^3
10^4
Frequency (Hz)


Amplitude^2 vs. Frequency - 0.1 sec Time Slices
Amplitude^2
10^5
0
0.5
1
1.5
2
Frequency (Hz)
x 10^4
1
2
3
4
5
6
7
8
8
10

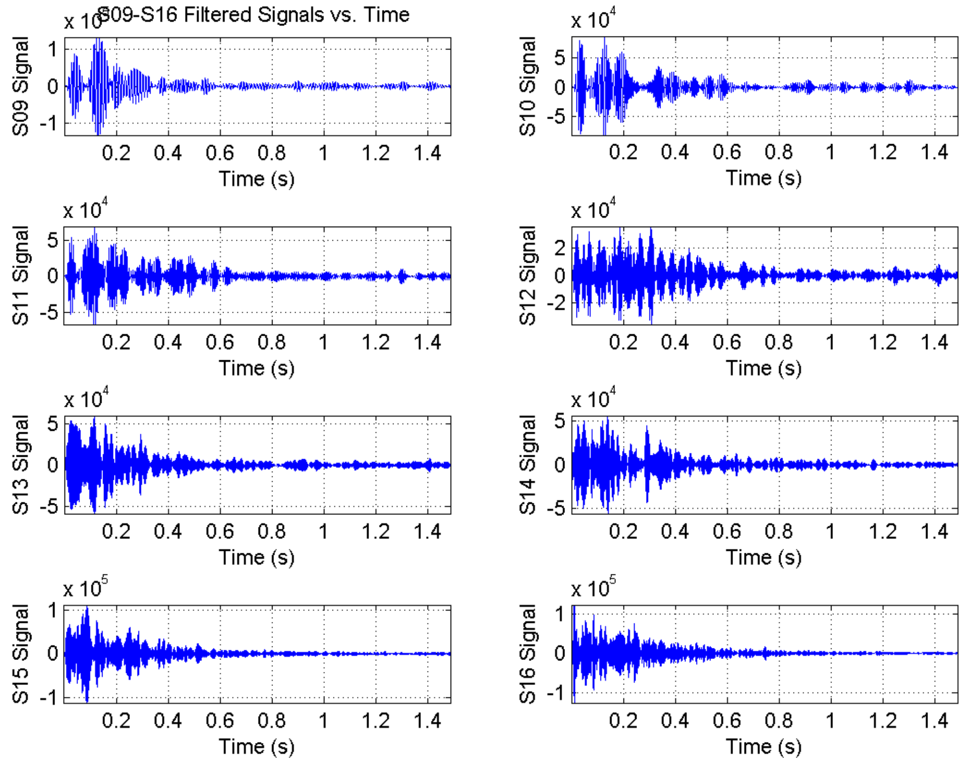

Amplitude^2 vs. Frequency - 0.1 sec Time Slices
Amplitude^2
10^5
10^2
10^3
10^4
Frequency (Hz)
1
2
3
4
5
6
7
8
8
10

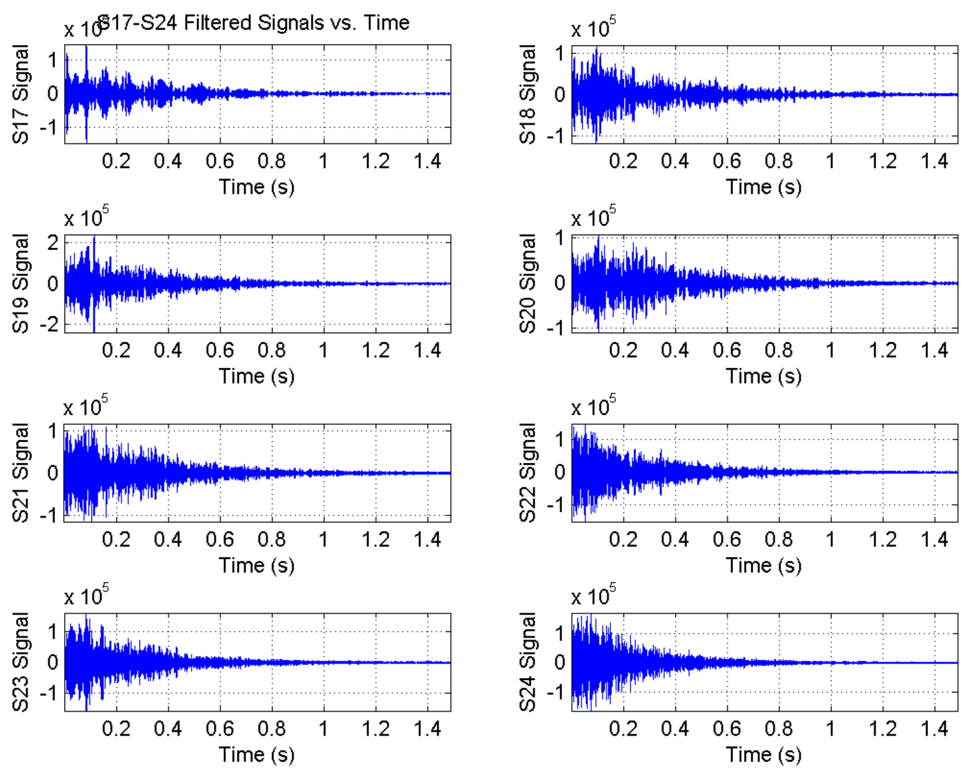


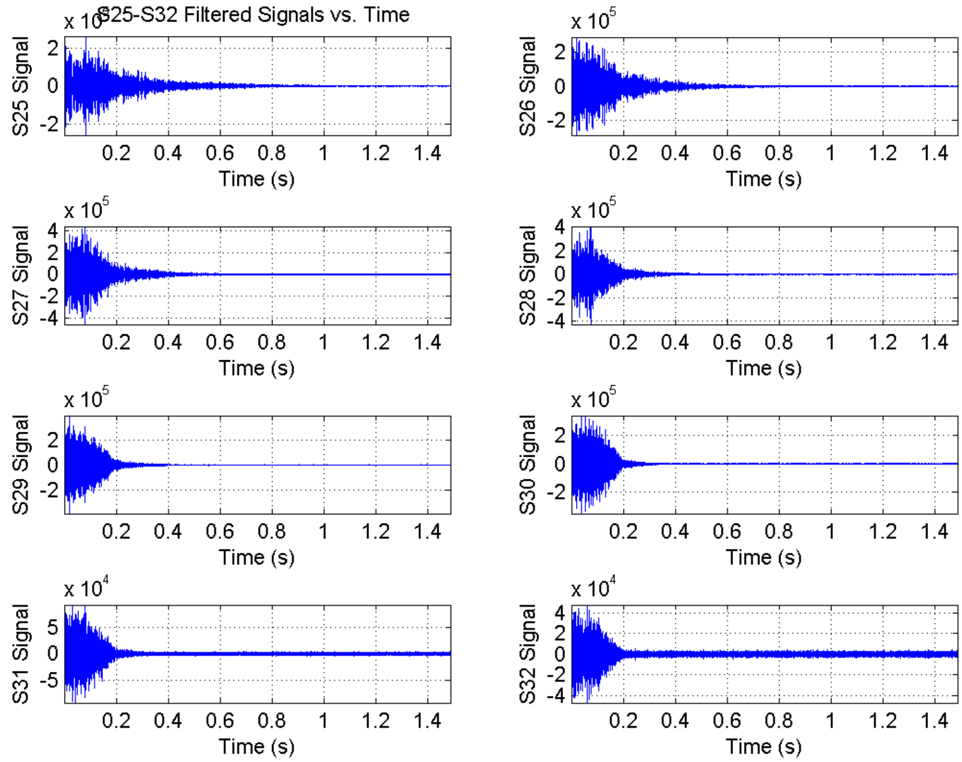
Amplitude^2 vs. Frequency vs. Time
Amplitude^2
Frequency (Hz)
x 10^4
Time (s)



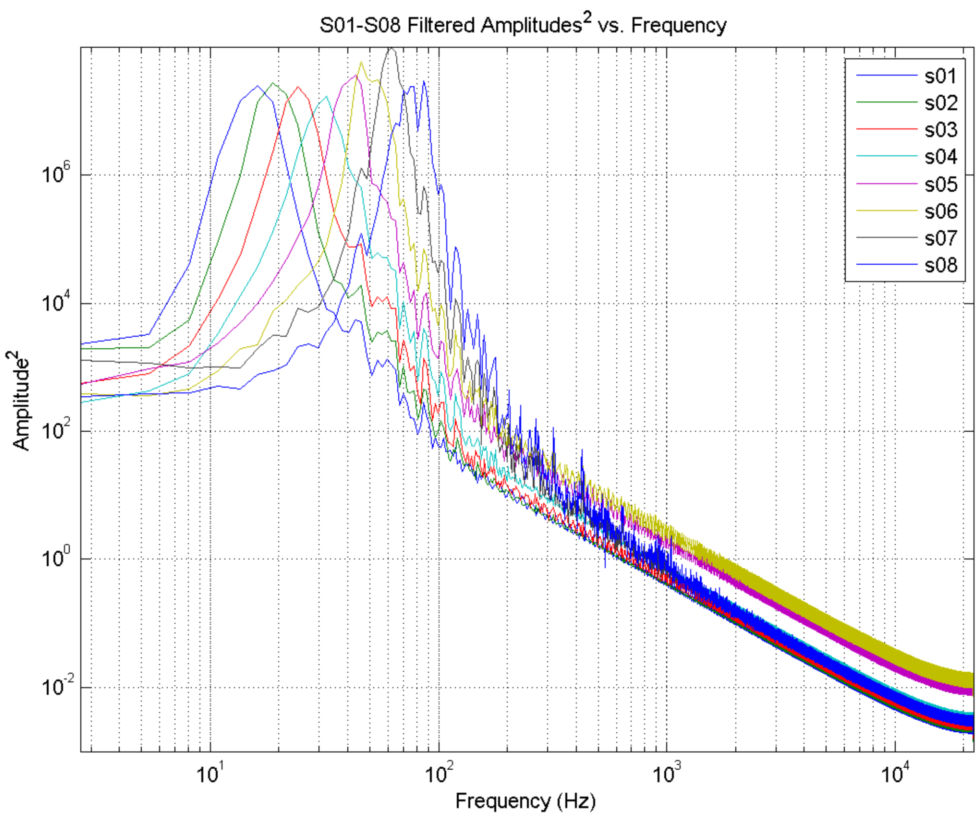
S01-S08 Filtered Signals vs. Time
S01 Signal
S02 Signal
S03 Signal
S04 Signal
S05 Signal
S06 Signal
S07 Signal
S08 Signal
Time (s)


S09-S16 Filtered Signals vs. Time
S09 Signal
S10 Signal
S11 Signal
S12 Signal
S13 Signal
S14 Signal
S15 Signal
S16 Signal
Time (s)

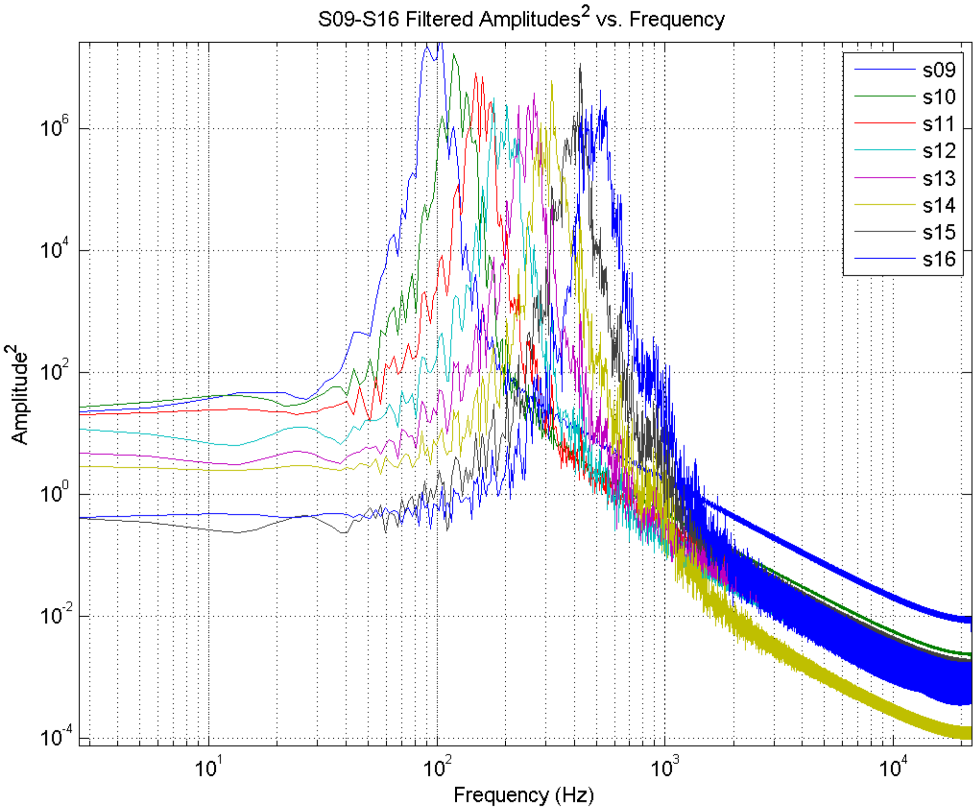

S17-S24 Filtered Signals vs. Time
S17 Signal
S18 Signal
S19 Signal
S20 Signal
S21 Signal
S22 Signal
S23 Signal
S24 Signal
Time (s)

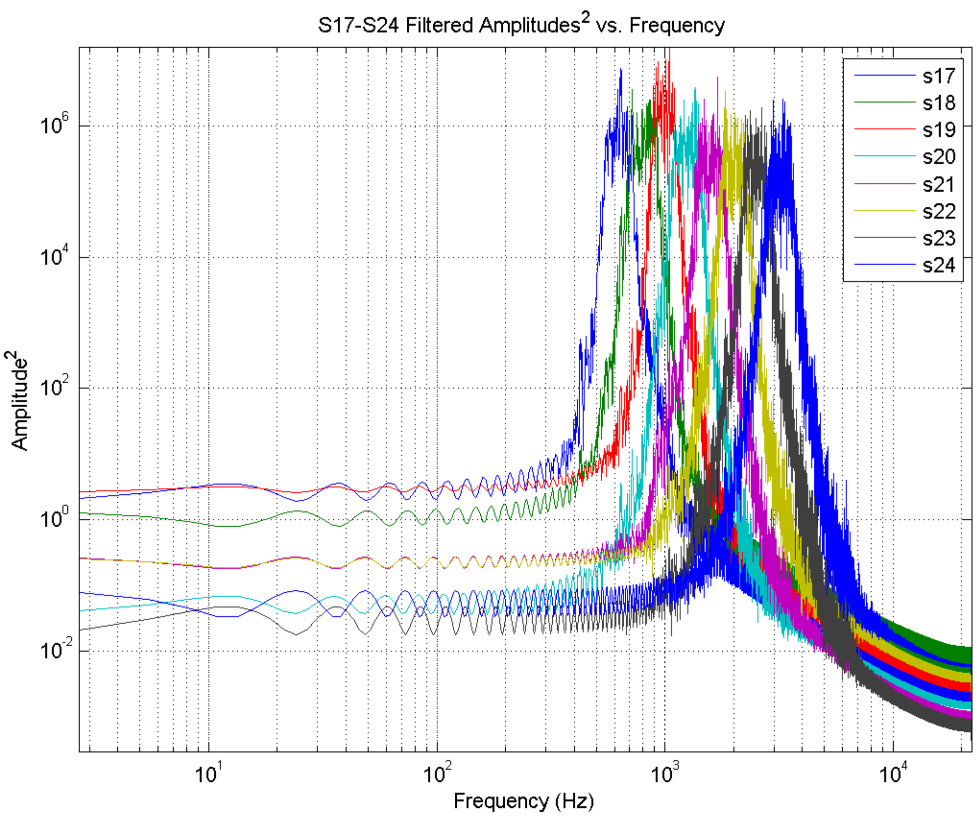


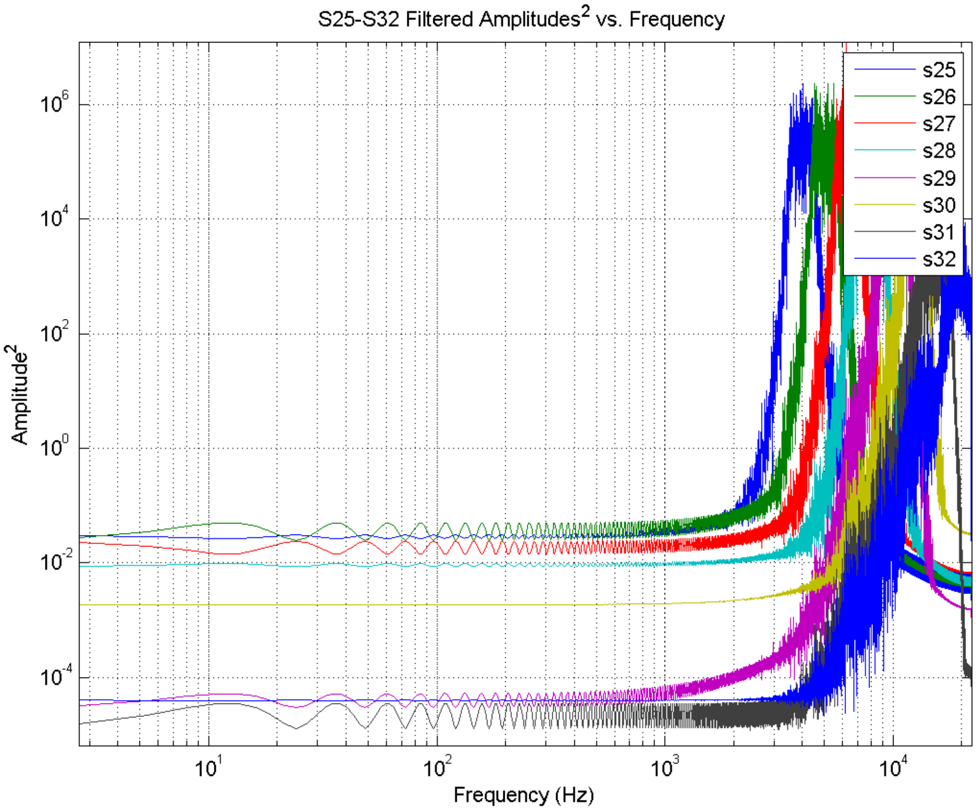
S25-S32 Filtered Signals vs. Time
S25 Signal
S26 Signal
S27 Signal
S28 Signal
S29 Signal
S30 Signal
S31 Signal
S32 Signal
Time (s)



S01-S08 Filtered Amplitudes^2 vs. Frequency
Amplitude^2
Frequency (Hz)
s01
s02
s03
s04
s05
s06
s07
s08


S09-S16 Filtered Amplitudes^2 vs. Frequency
Amplitude^2
Frequency (Hz)
s09
s10
s11
s12
s13
s14
s15
s16


S17-S24 Filtered Amplitudes^2 vs. Frequency
Amplitude^2
Frequency (Hz)
s17
s18
s19
s20
s21
s22
s23
s24


S25-S32 Filtered Amplitudes^2 vs. Frequency
s25
s26
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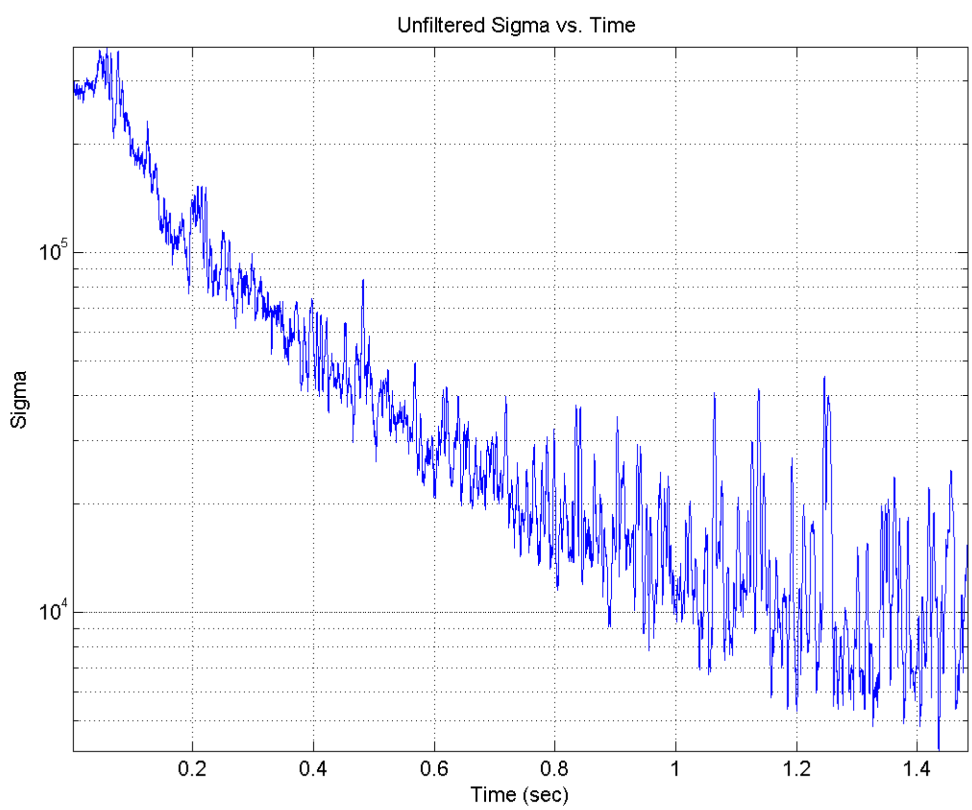
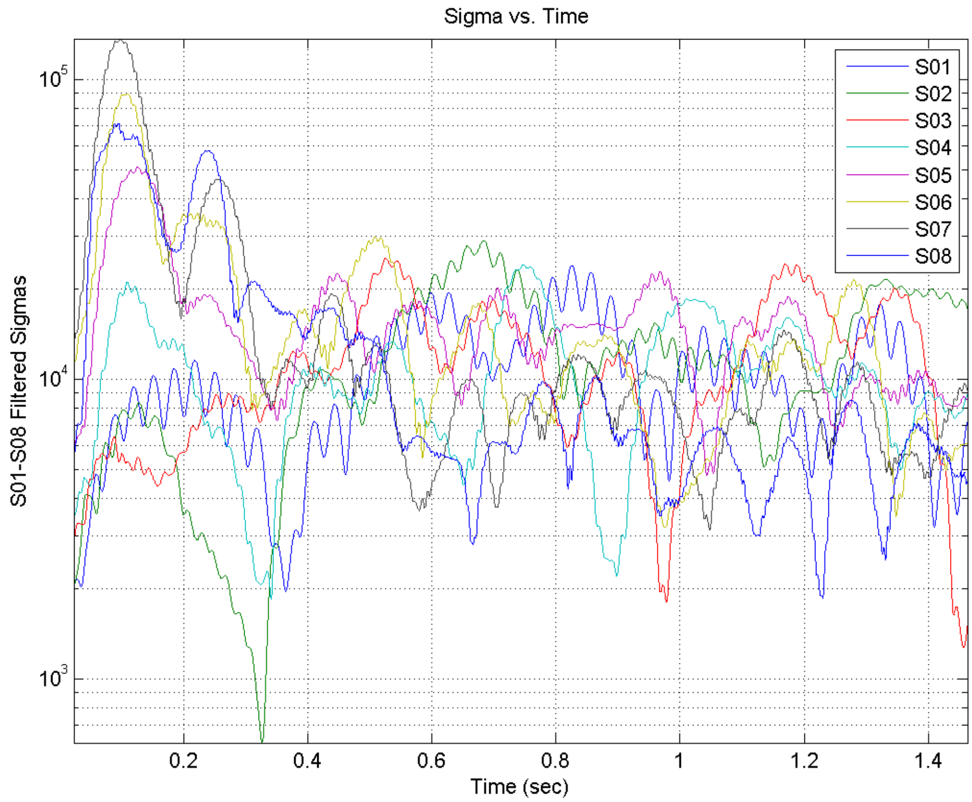
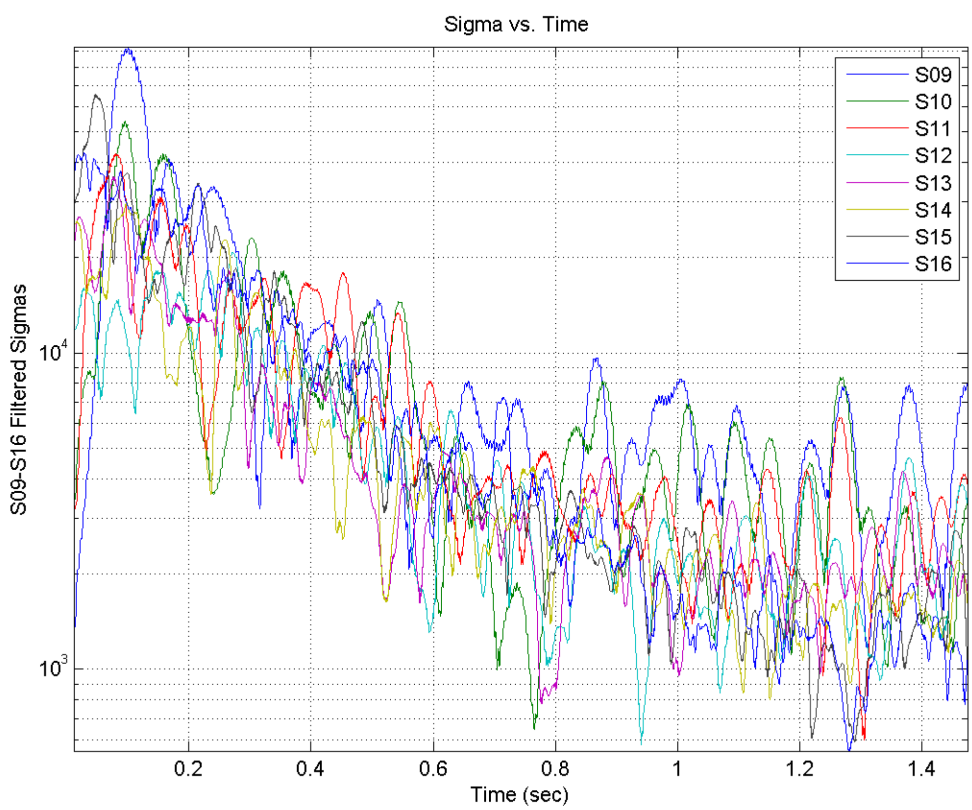
s31
s32
Amplitude^2
Frequency (Hz)


Unfiltered Sigma vs. Time
Sigma
Time (sec)


Sigma vs. Time
S01-S08 Filtered Sigmas
Time (sec)
S01
S02
S03
S04
S05
S06
S07
S08
0.2
0.4
0.6
0.8
1
1.2
1.4
10^5
10^4
10^3


Sigma vs. Time
S09-S16 Filtered Sigmas
Time (sec)
S09
S10
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S16
0.2
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0.6
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1
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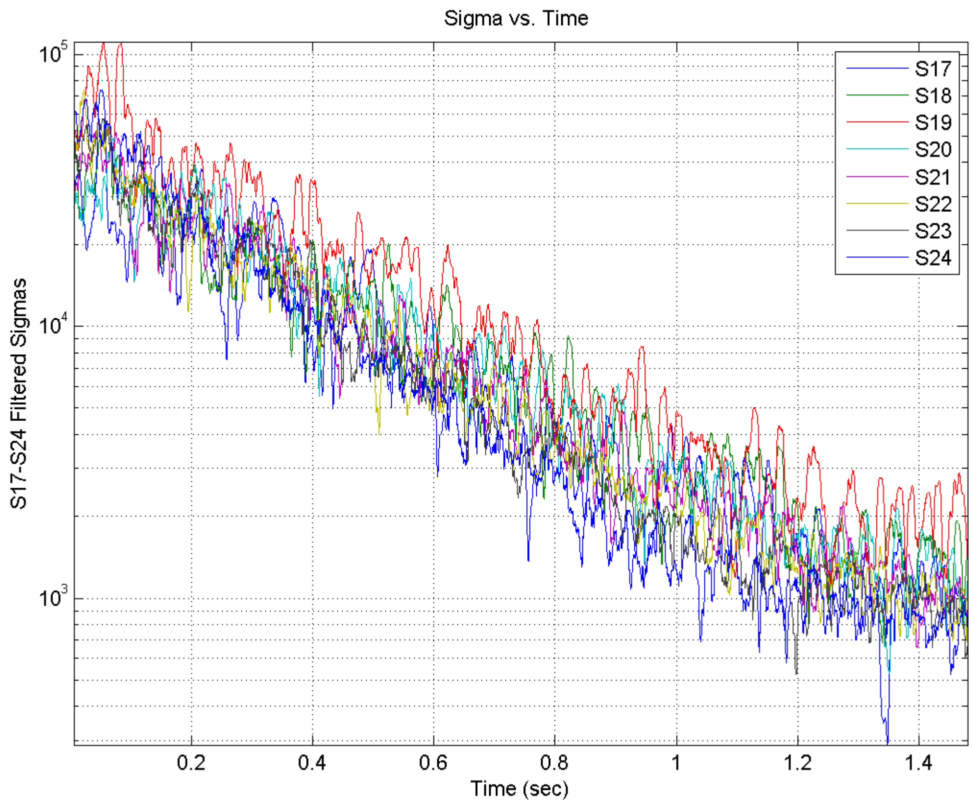
10^4
10^3


Sigma vs. Time
S17-S24 Filtered Sigmas
Time (sec)
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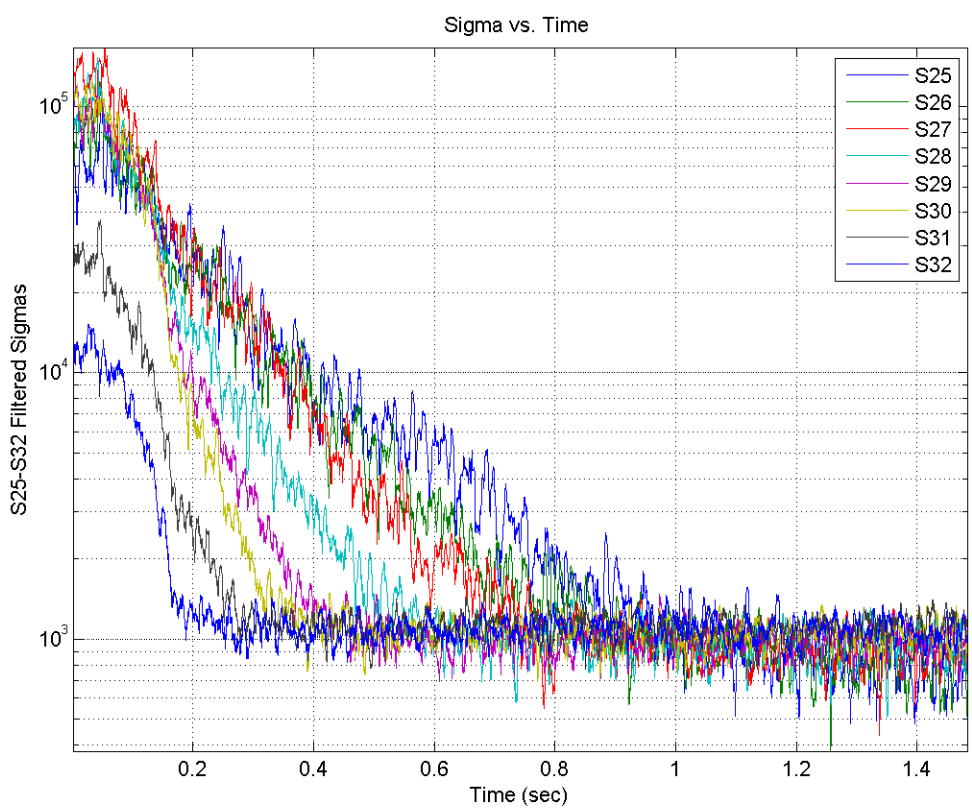
Sigma vs. Time
S25-S32 Filtered Sigmas
Time (sec)
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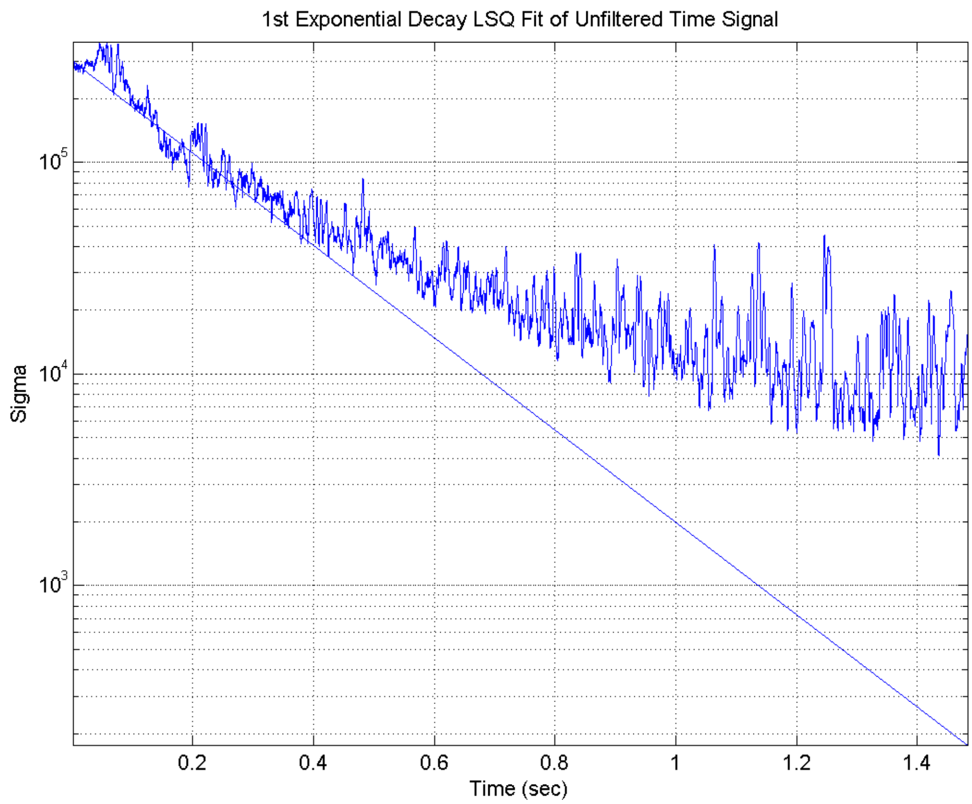
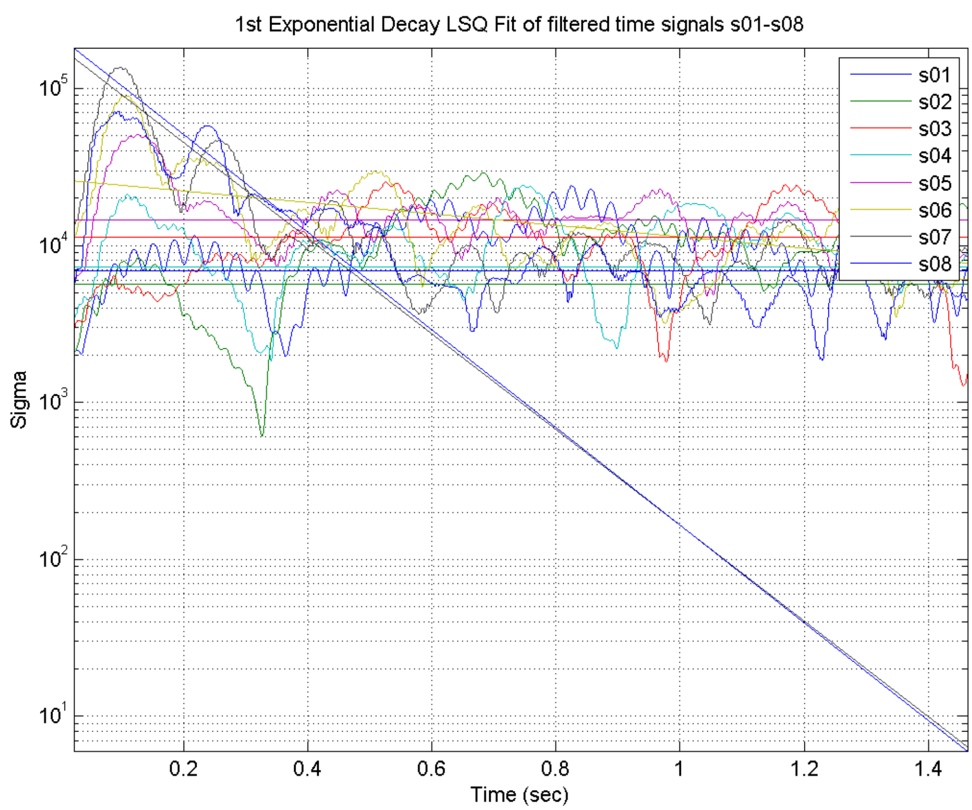
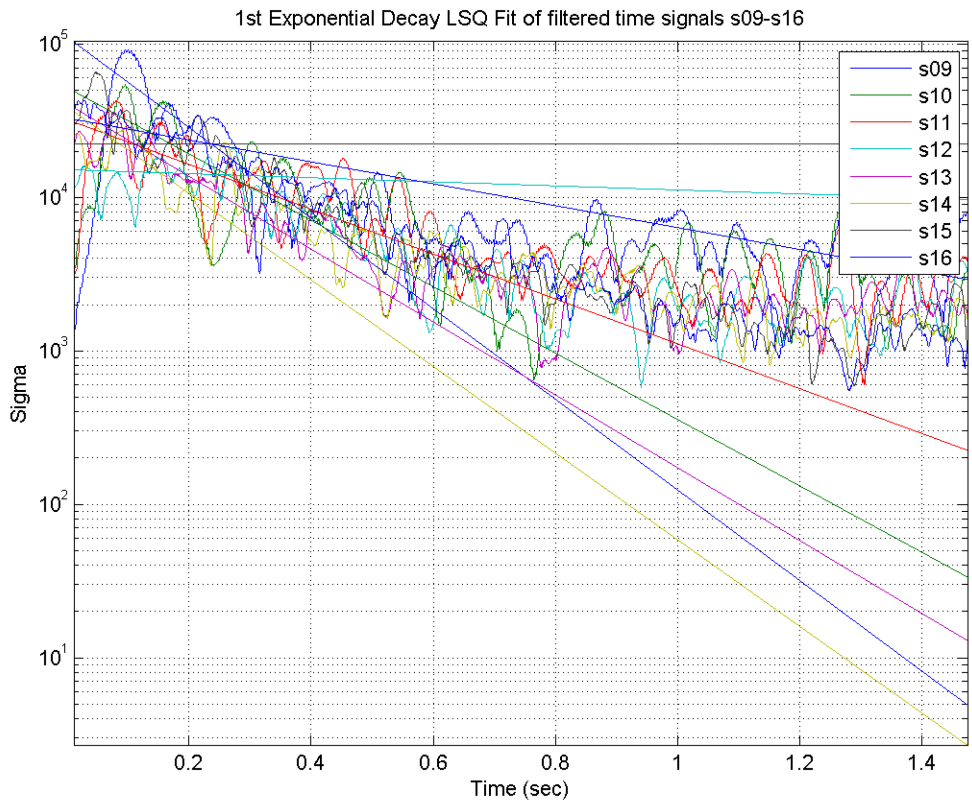
S32


1st Exponential Decay LSQ Fit of Unfiltered Time Signal
Sigma
10^5
10^4
10^3
0.2
0.4
0.6
0.8
1
1.2
1.4
Time (sec)


1st Exponential Decay LSQ Fit of filtered time signals s01-s08
s01
s02
s03
s04
s05
s06
s07
s08
Sigma
10^5
10^4
10^3
10^2
10^1
0.2
0.4
0.6
0.8
1
1.2
1.4
Time (sec)


1st Exponential Decay LSQ Fit of filtered time signals s09-s16
Sigma
Time (sec)
s09
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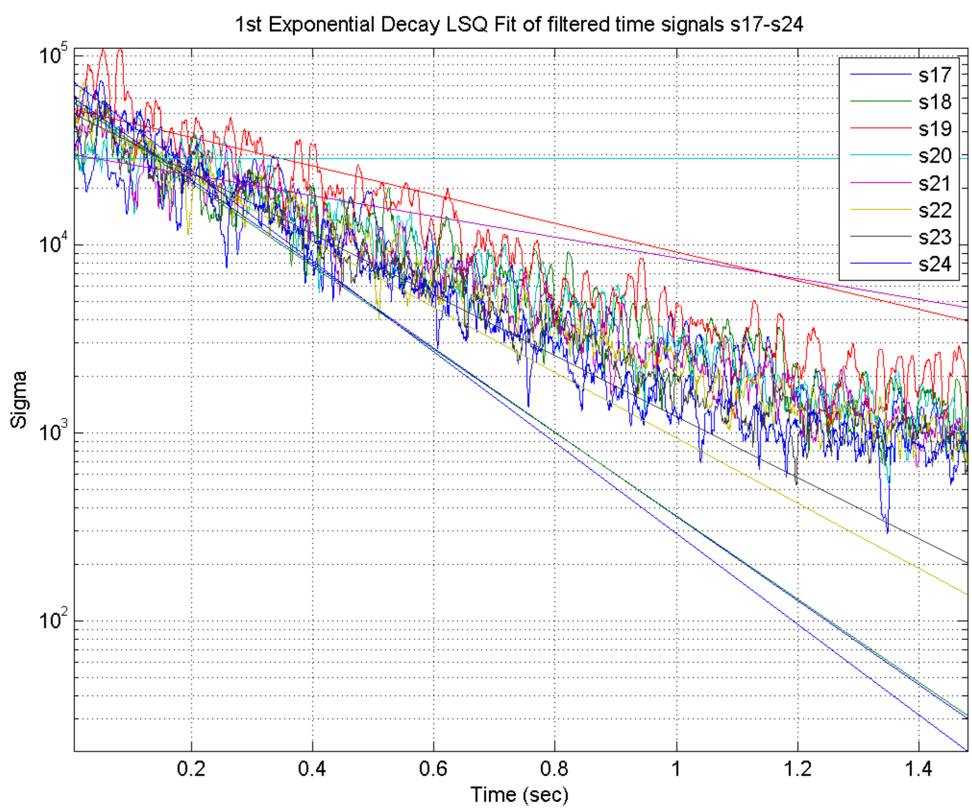
s16


1st Exponential Decay LSQ Fit of filtered time signals s17-s24
Sigma
Time (sec)
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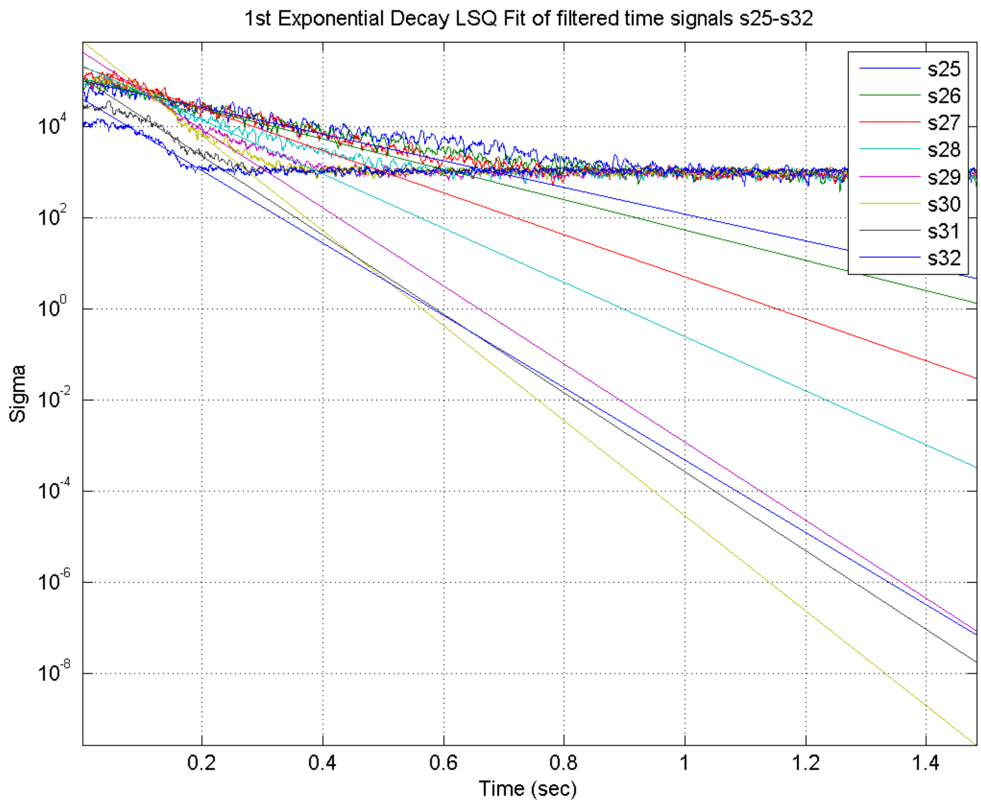
1st Exponential Decay LSQ Fit of filtered time signals s25-s32
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s26
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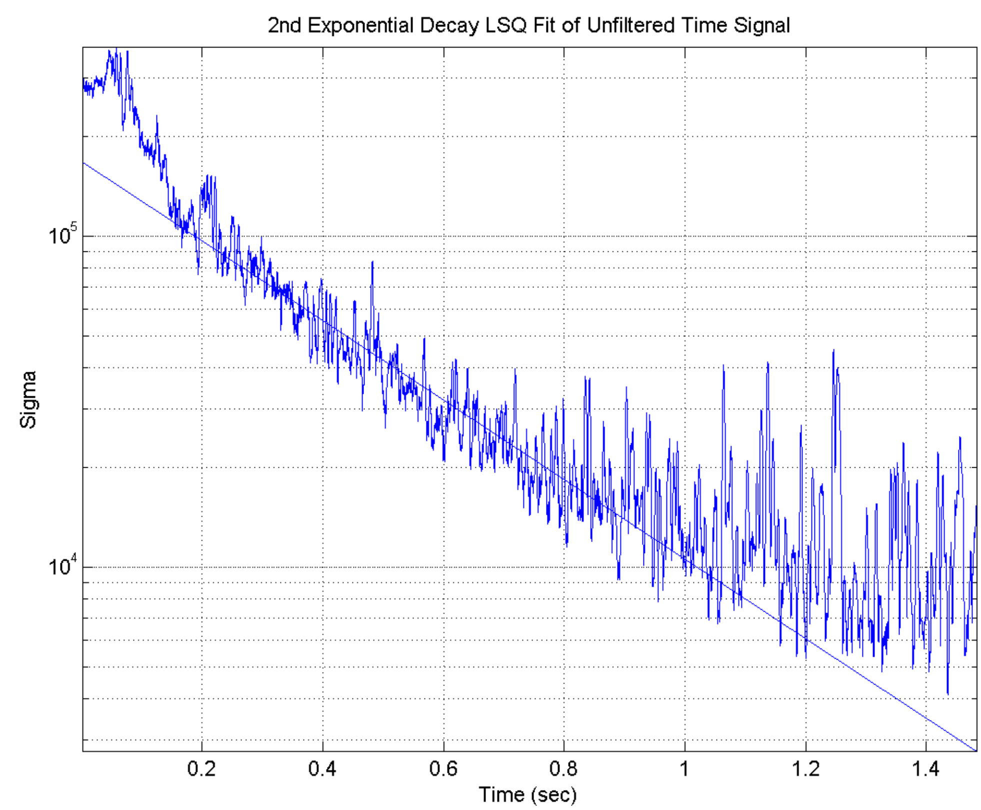
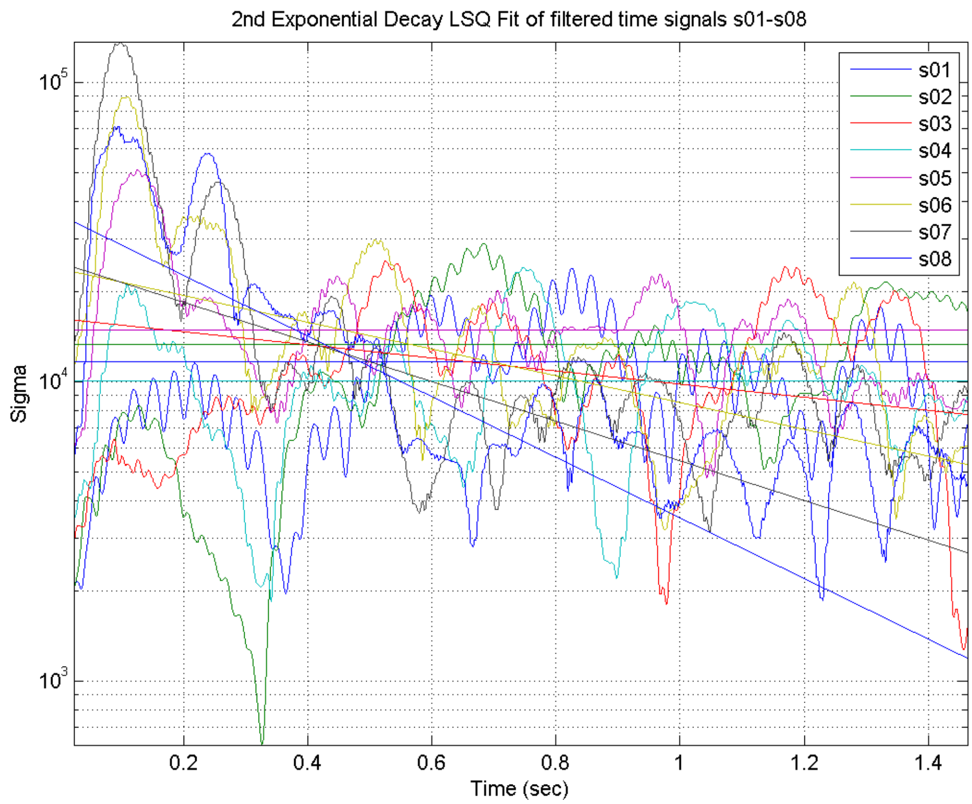
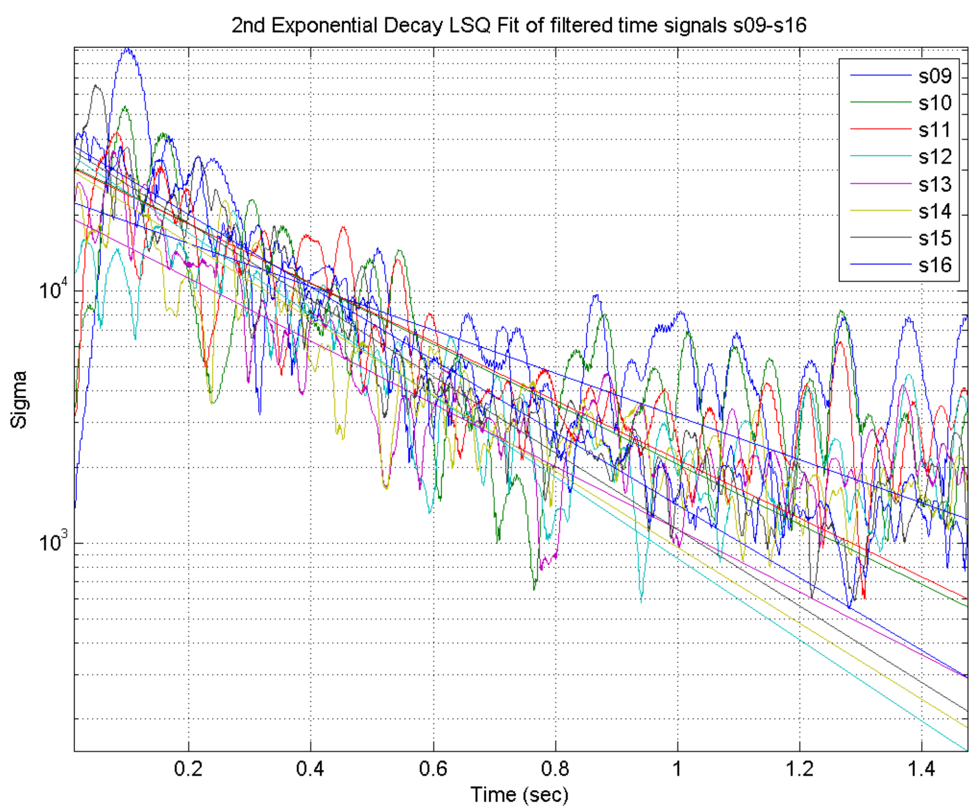
s29
s30
s31
s32
Sigma
Time (sec)


2nd Exponential Decay LSQ Fit of Unfiltered Time Signal
Sigma
Time (sec)


2nd Exponential Decay LSQ Fit of filtered time signals s01-s08
Sigma
Time (sec)
s01
s02
s03
s04
s05
s06
s07
s08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Sigma
Time (sec)
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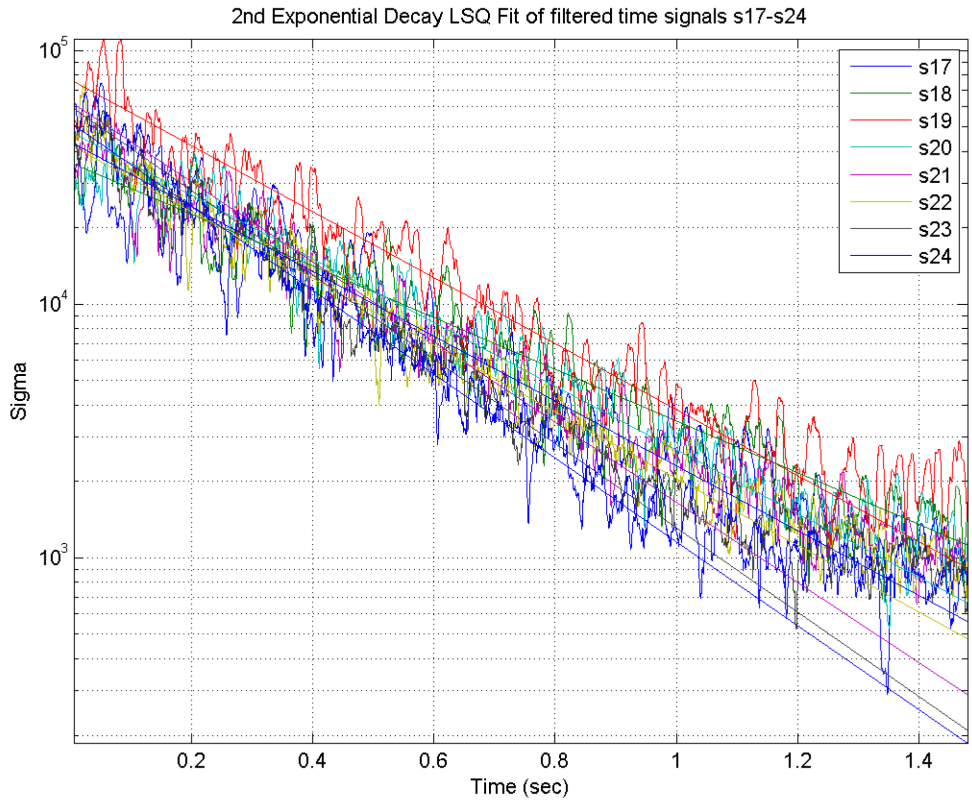
2nd Exponential Decay LSQ Fit of filtered time signals s17-s24
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s18
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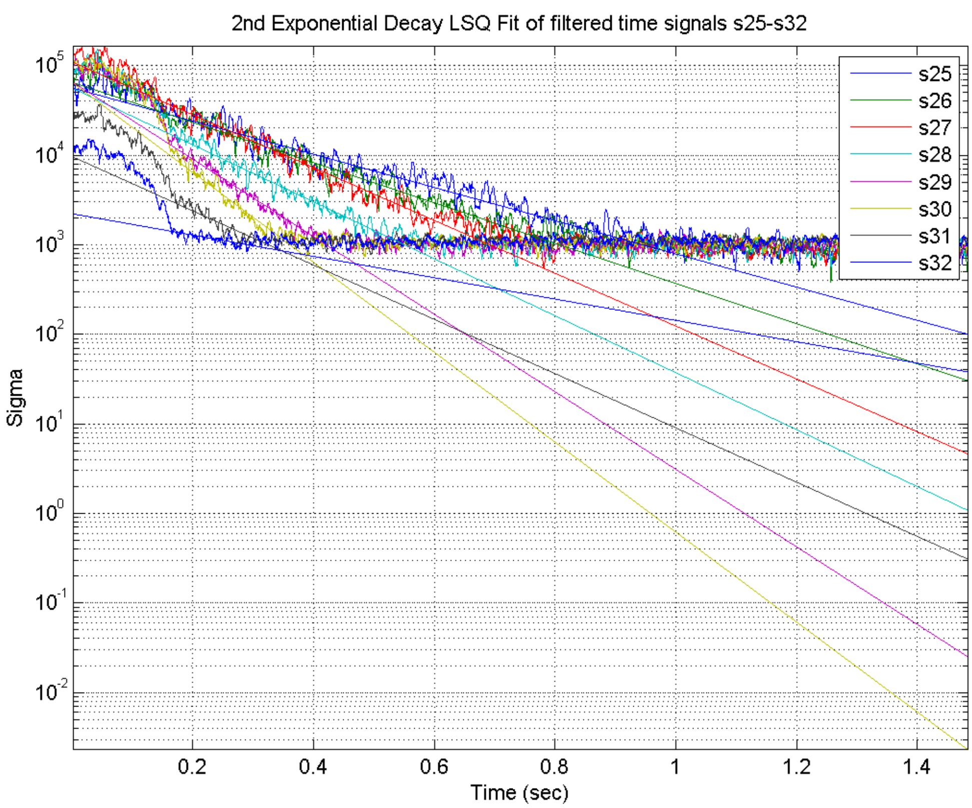
s22
s23
s24
Sigma
Time (sec)


2nd Exponential Decay LSQ Fit of filtered time signals s25-s32
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s26
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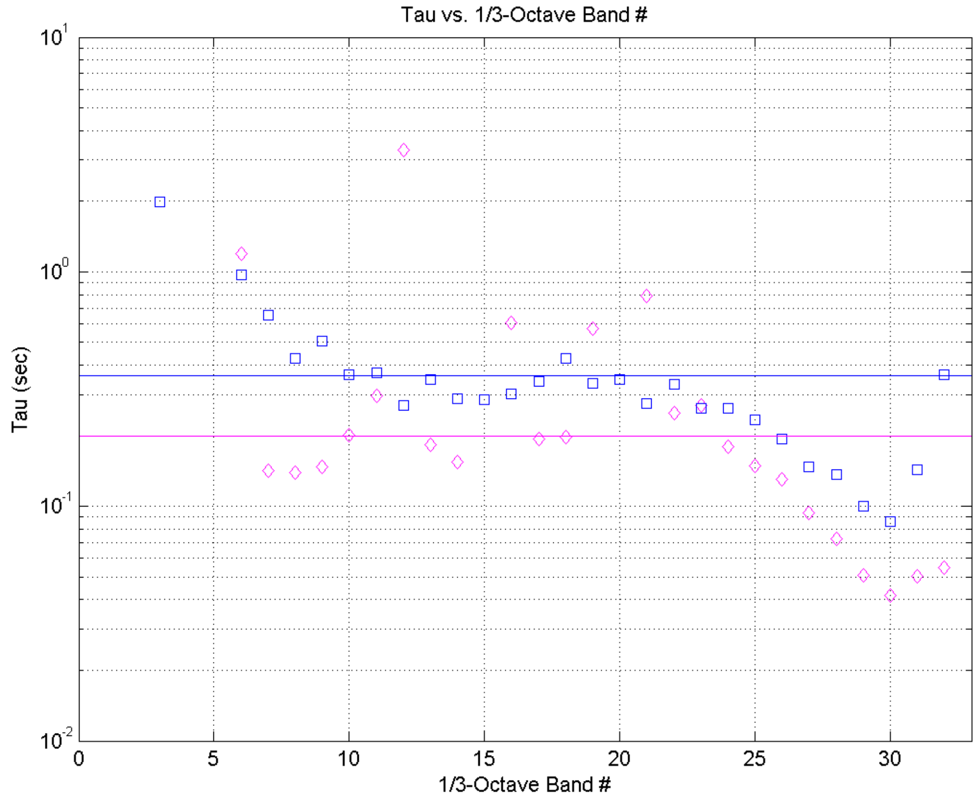
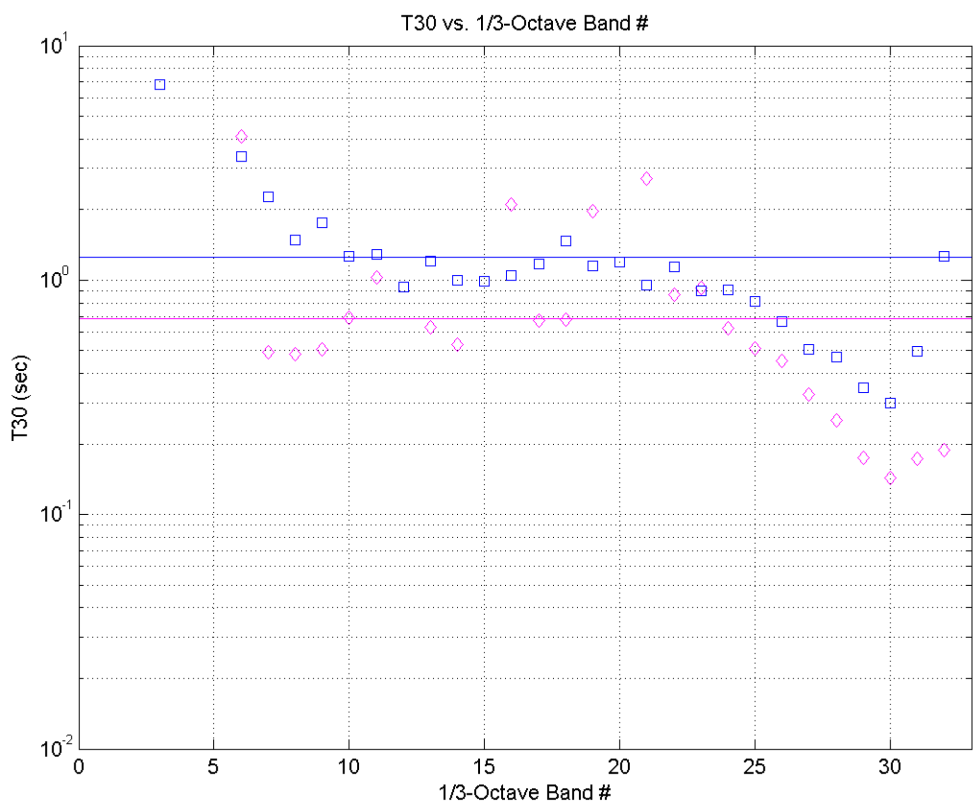
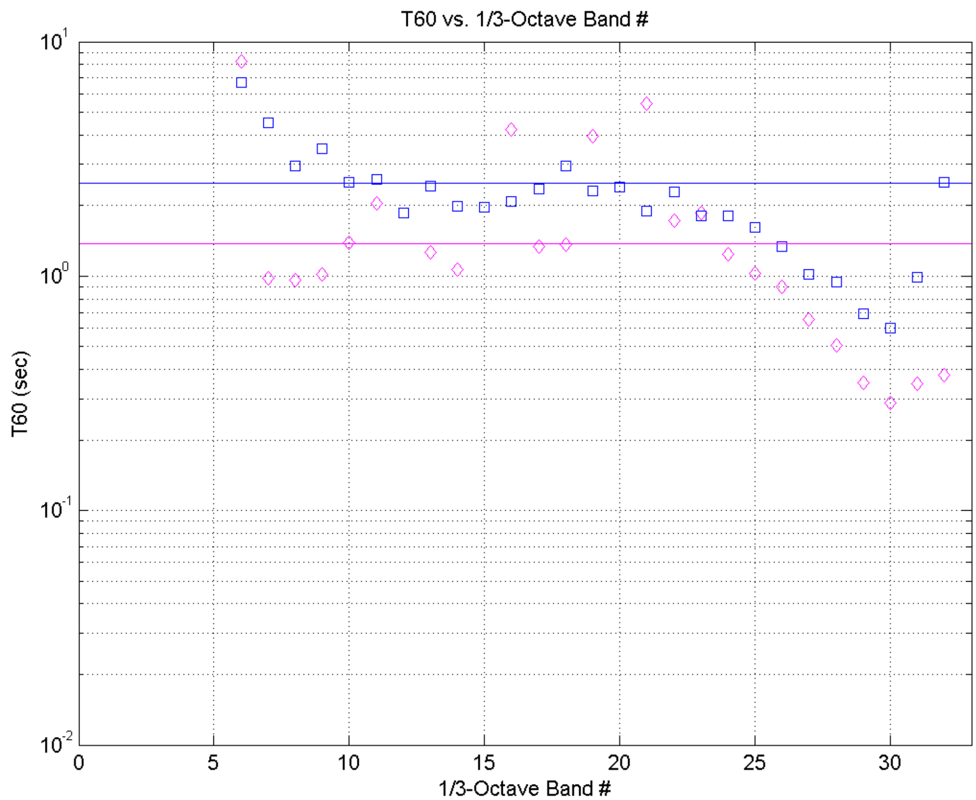
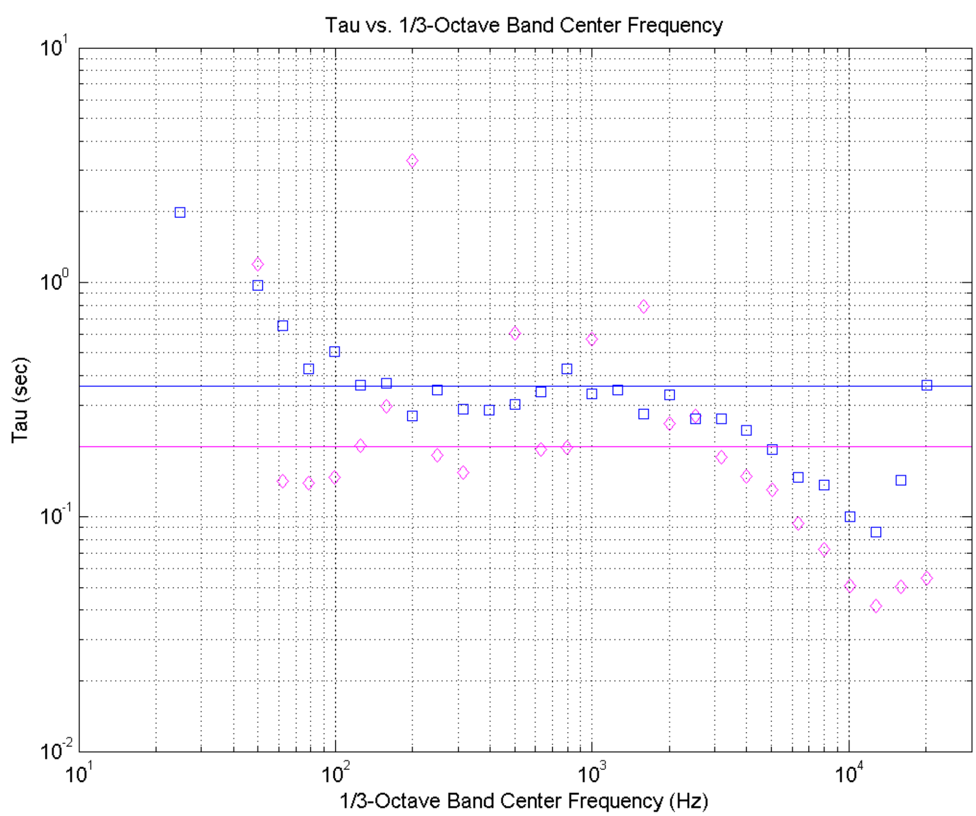
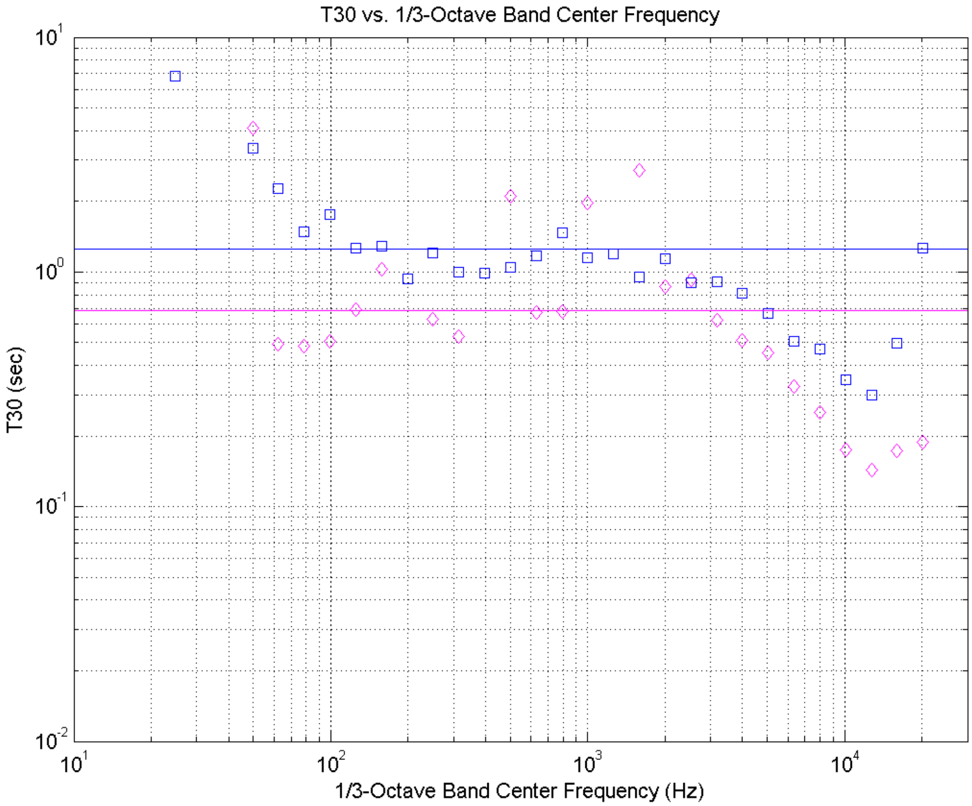
s29
s30
s31
s32
Sigma
Time (sec)

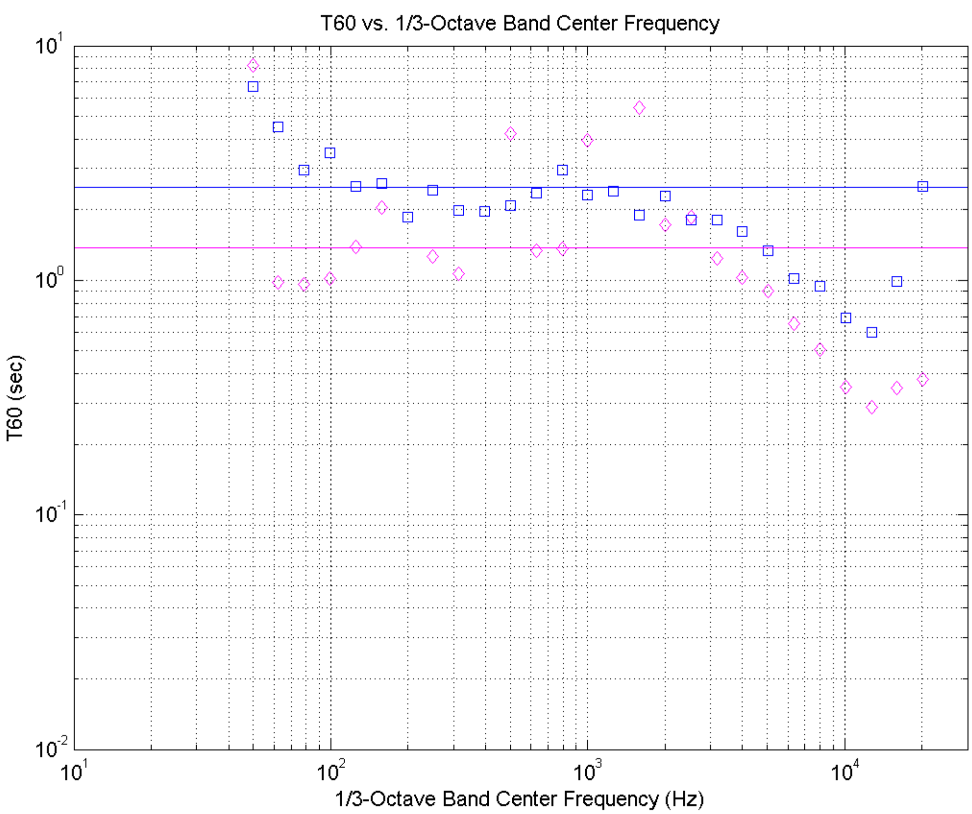

Tau vs. 1/3-Octave Band #
Tau (sec)
1/3-Octave Band #
10^1
10^0
10^-1
10^-2
0
5
10
15
20
25
30


T30 vs. 1/3-Octave Band #
T30 (sec)
1/3-Octave Band #
10^1
10^0
10^-1
10^-2
0
5
10
15
20
25
30

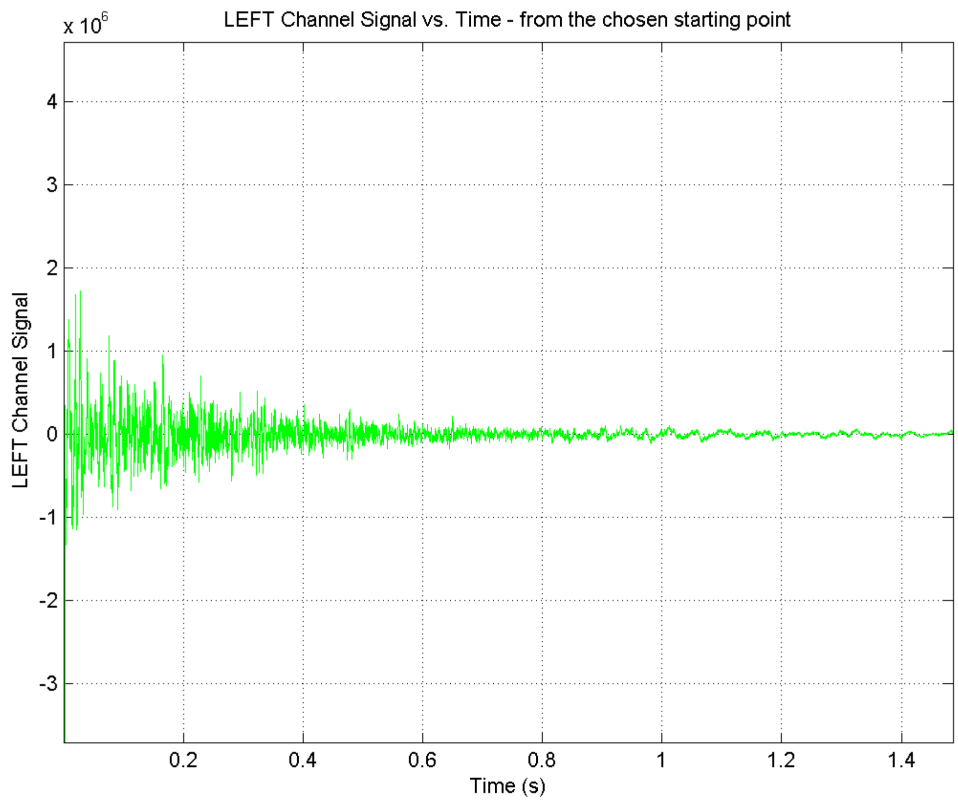

T60 vs. 1/3-Octave Band #
T60 (sec)
1/3-Octave Band #
10^{1}
10^{0}
10^{-1}
10^{-2}
0
5
10
15
20
25
30

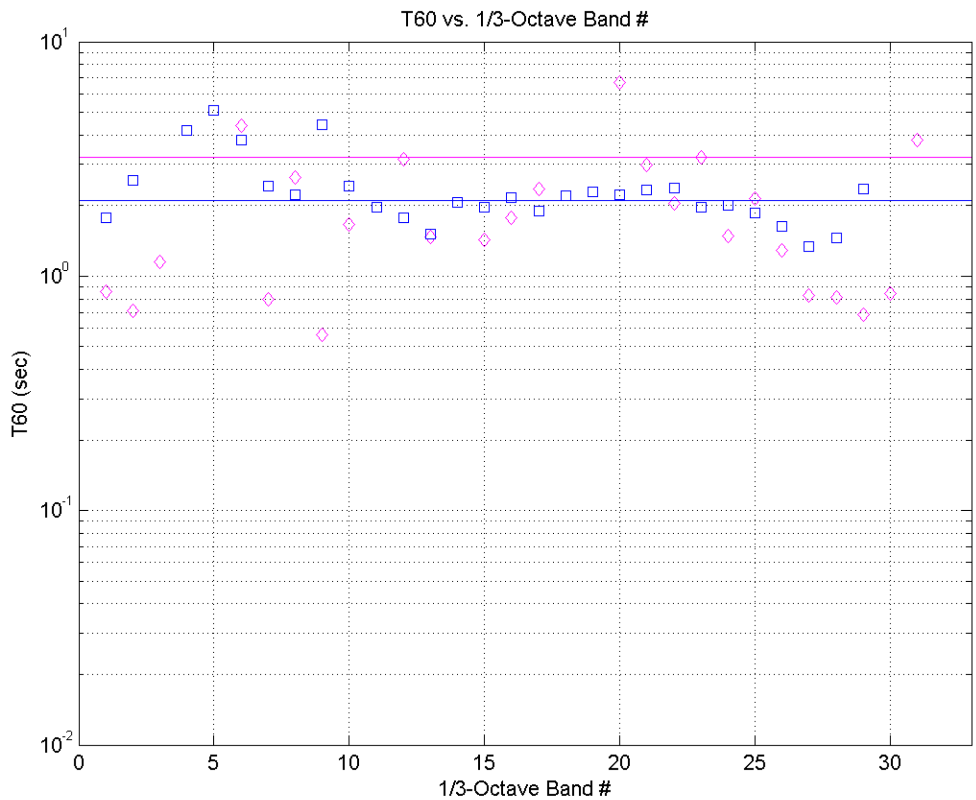

Tau vs. 1/3-Octave Band Center Frequency
Tau (sec)
1/3-Octave Band Center Frequency (Hz)
10^{1}
10^{0}
10^{-1}
10^{-2}
10^{1}
10^{2}
10^{3}
10^{4}

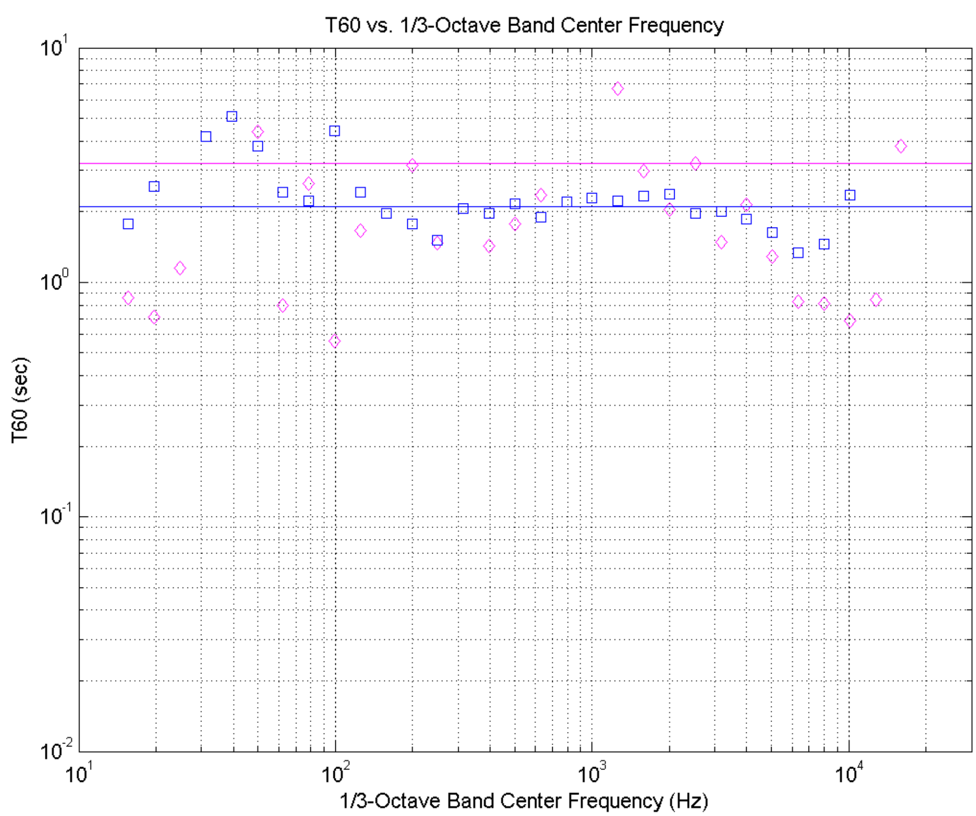

T30 vs. 1/3-Octave Band Center Frequency
T30 (sec)
1/3-Octave Band Center Frequency (Hz)


T60 vs. 1/3-Octave Band Center Frequency
T60 (sec)
1/3-Octave Band Center Frequency (Hz)

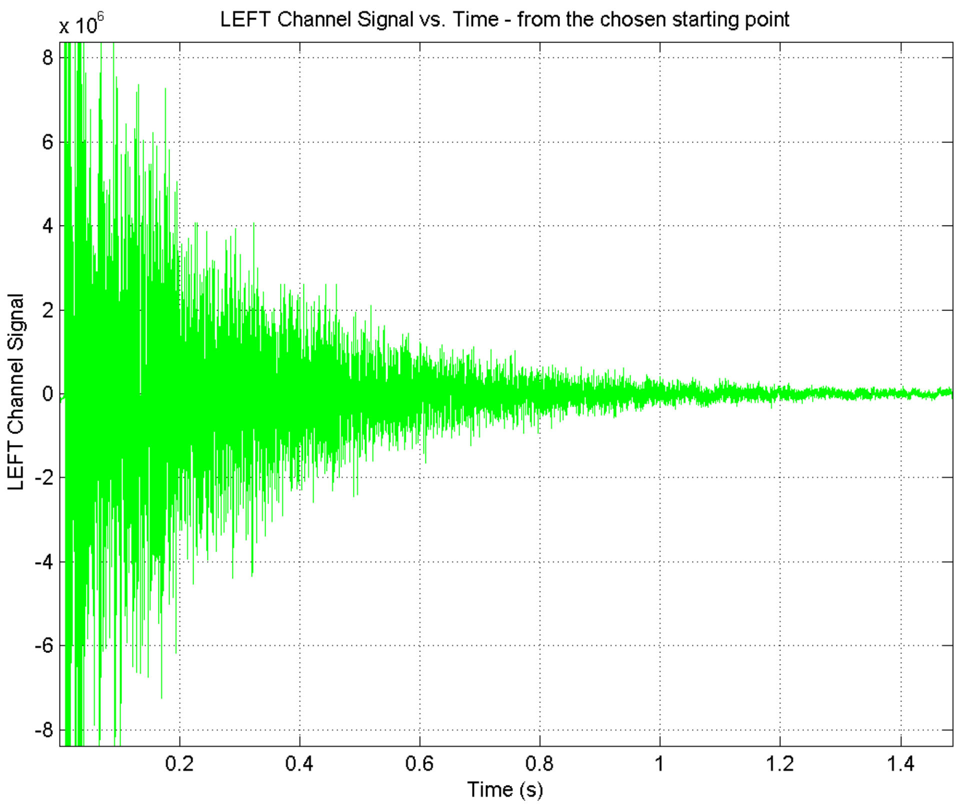
As we can see from the T-60 vs. 1/3 Octave Band Center Frequency plot directly above, the reverberation time is approximately 2.4 seconds, which is within range of our predictions. Let us now take a look at a 1-ms square wave test. I will not show all of the plots, only the original (cut) signal and the T-60 plots. This test was the stage left channel for the 7th row, the same location as the previous plots. Notice in the second T-60 plot below that the reverberation time is approximately 2 seconds.

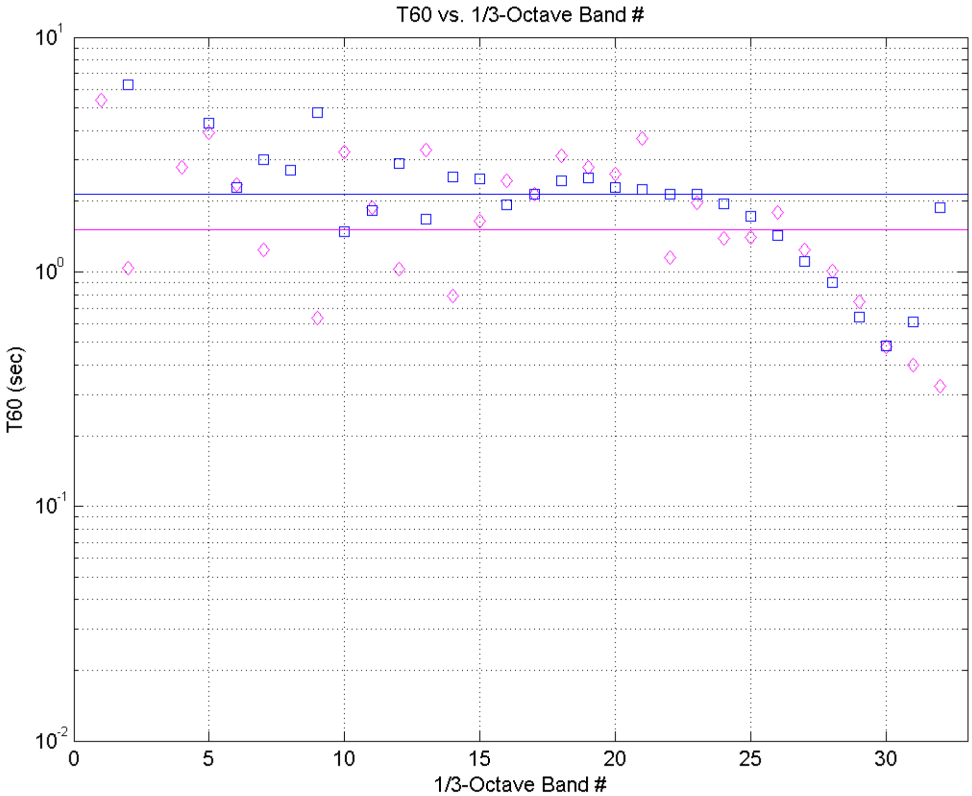


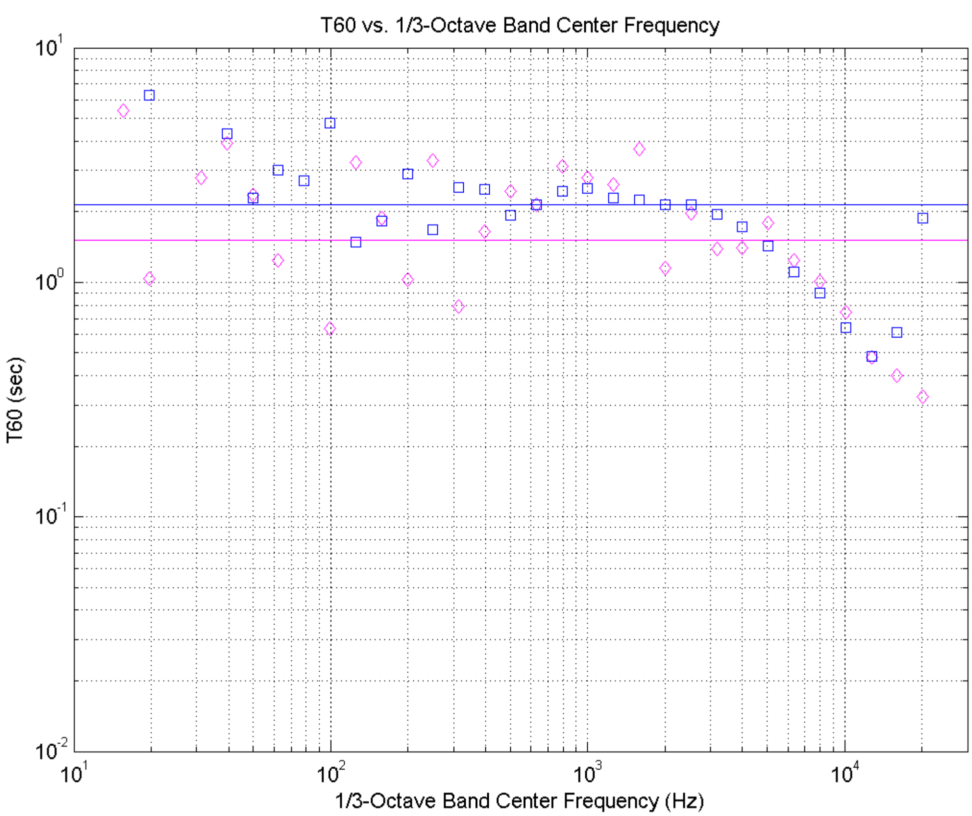

T60 vs. 1/3-Octave Band #
T60 (sec)
1/3-Octave Band #
10^{1}
10^{0}
10^{-1}
10^{-2}
0
5
10
15
20
25
30


T60 vs. 1/3-Octave Band Center Frequency
T60 (sec)
1/3-Octave Band Center Frequency (Hz)
10^{1}
10^{0}
10^{-1}
10^{-2}
10^{1}
10^{2}
10^{3}
10^{4}

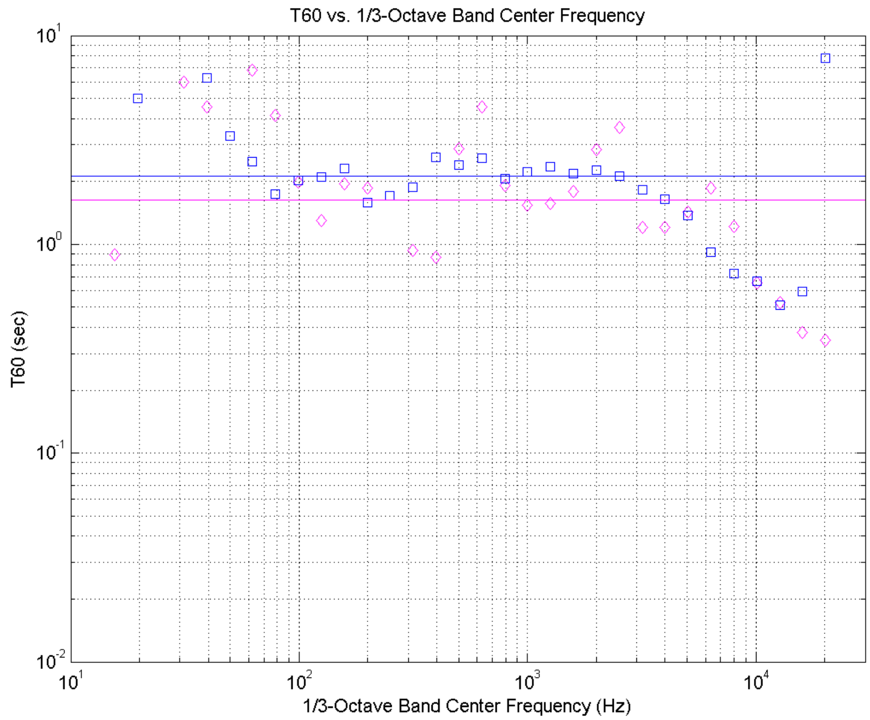
Let us now take a look at a balloon burst, as recorded at the same location (i.e., left channel, front). Notice again that the T-60 reverberation time is just over 2 seconds, once again confirming our predictions.

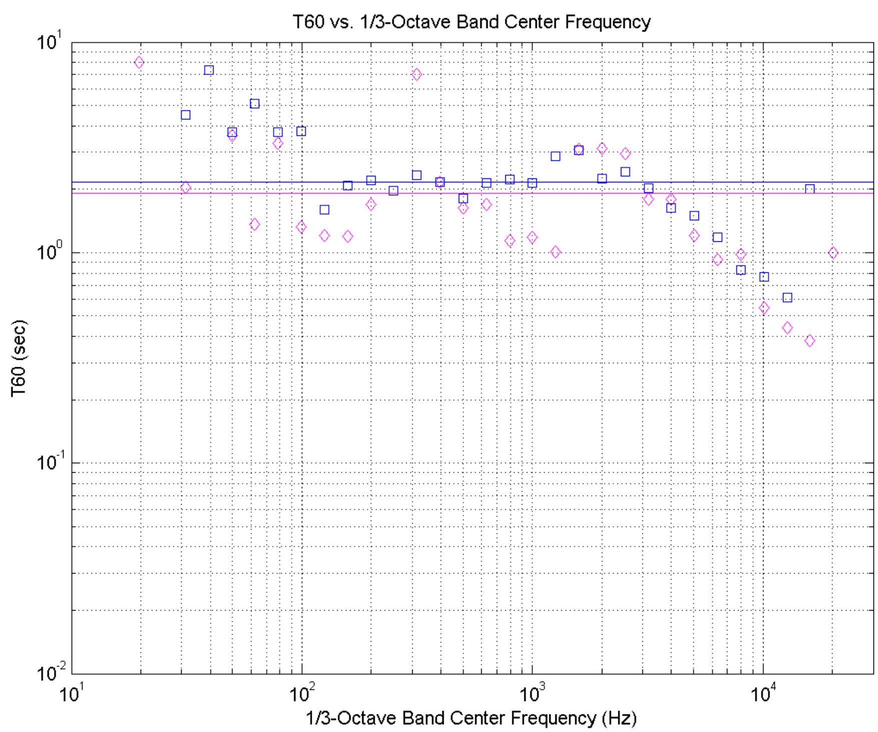



T60 vs. 1/3-Octave Band #
T60 (sec)
1/3-Octave Band #


T60 vs. 1/3-Octave Band Center Frequency
T60 (sec)
1/3-Octave Band Center Frequency (Hz)

Let's now compare the T-60 plots for the left channel balloon bursts in the back of the main floor and on the balcony. Again, we see a T-60 time of about 2 seconds.





VI. Future Projects and Conclusion

There are many other testing options, but I simply cannot comment on any more of them here for the sake of clarity. We isolated the reverberation time; in particular, T-60, the time it takes for the sound to reach a millionth of its original intensity. Other data analyses can be performed, for example, on the data regarding the clarity and duration of the sound interval, among other things. Professor Errede enabled the code to give the user the option to run several interesting types of analyses on the sound files. So, an excellent project for the future would be to use the MATLAB code to compare various sound characteristics for different venues. Additionally, there is still a project that can be done at the assisted living home in Iowa. That might be a great place to look at the reverberation time, clarity, and duration. If it is difficult for people to hear each other speak, even at a reasonably close range, there ought to be something done to improve the quality of voices in the room. Doing the acoustics measurements will hopefully allow for an isolation of the problem(s). For example, if the reverberation time is too long, say, on the order of something like we saw for the Foellinger Great Hall (approximately 2 seconds), then spoken word might be very difficult to discern there, even at short range. Additionally, if it is discovered that the clarity or duration or not optimal, perhaps some means of damping or reflecting the sound at various locations could be modeled and simulated in order to help those folks out.

In conclusion, this was quite an interesting project. We were lucky enough to get into the Foellinger Great Hall to perform the measurements. A big thank you goes to Eric Bisgyer and the Krannert Center for the Performing Arts. Also, thanks to Professor Emeritus Leland Holloway for helping with the setup and teardown, as well as with measurements of the room. Finally, a huge thanks goes to Professor Errede for having the great idea to perform these measurements in the first place. Also, without his expertise and dedication, the MATLAB endeavor may never have come to fruition. The code is robust and will do great things in the future. It can be used for a number of room acoustics measurements. Furthermore, Professor Errede supplied all of the gear and transported it himself round-trip, not once but twice (due to a small communication error early on). I appreciate all the help I received, and I hope to check back next semester when more room acoustics analyses are performed.

VII. Bibliography

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