Acoustic Characterization of 41 Cooper Square Academic Spaces

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***Abstract*—The acoustics of Cooper Union's new academic building were tested for two different acoustical parameters: reverberation time and ambient sound levels. These measurements were compared with ANSI/ASA S12.60-2010. Based on the overall ambient sound pressure level and reverberation time measurements, Cooper Union's new academic building does not meet ANSI standards for academic acoustical characteristics. Although most rooms were just below the maximum limit for reverberation times of 0.7 seconds, almost every room that was tested failed to fall below the ANSI standard for the ambient sound levels of the room. In many cases, this was due to external noise from the city, and in almost all cases, the HVAC system seemed to be at fault.**

I. Background

A. Sound Level Measurement

The intensity of sound can be quantified by measuring the pressure of the sound wave, commonly measured in Pascals. A microphone is an instrument which converts a changing sound pressure into a voltage. This voltage can then be read by a sound level meter, computer, or other device which then displays the voltage as a sound pressure. The sensitivity of a microphone is commonly expressed in units of mV/Pa, which describes the voltage output of the microphone compared with the pressure of the sound wave.

Sound levels are often measured with the units of decibels (abbreviated dB). The decibel is a logarithmic scale, which works well for sound measurements, since our ears also sense sound logarithmically. To convert a sound pressure into logarithmic scale, the following equation is used:

$$SPL = 20\log_{10}\left(\frac{p_{rms}}{p_{ref}}\right) \tag{1}$$

where p_{rms} is the root mean square pressure (measured in Pascals), p_{ref} is a reference pressure of 20×10^{-6} Pa (roughly equal to the faintest sound that the average human can hear), and SPL is the resulting Sound Pressure Level, expressed in decibels.

Human hears are not equally sensitive to all frequencies. Generally, most people cannot hear frequencies lower than 20 Hz, or higher than 20 kHz. Weighting schemes have been developed which attempt to correct sound pressure level measurements to account for the variable sensitivity to different frequencies. The most common weighting scheme is the A-weighting. Because we are not as sensitive to low frequency noises, these frequencies have a lower weighting, while the most sensitive frequencies (in the 2 kHz range) are weighted higher. This weighting scheme was developed to more accurately represent the way that we actually hear different frequencies, although it is a very simplified version. Figure 1 shows the standard A-weighting curve[1].

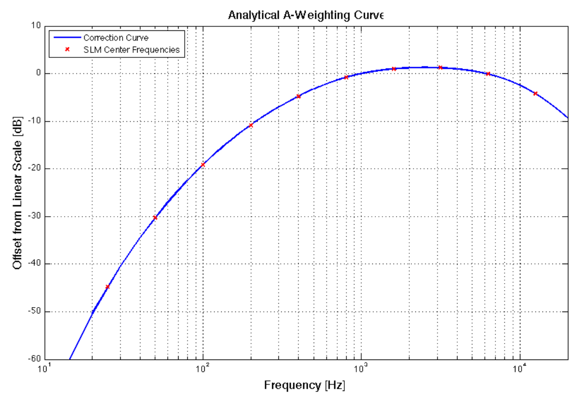

Fig. 1. A-Weighting Curve

The chart shows how to adjust sound pressure level measurements according to frequency. If, for example, a 100 Hz tone was measured to have a sound pressure level of 60 dB (unweighted), then the A-weighted sound pressure would be:

$$SPL_A = 60 - 19 = 41dBA$$

Often, when referencing standards for maximum allowable sound pressure levels on job sites or in schools, the value is expressed as an A-weighted measurement. Other weighting schemes (such as B, C, and D weighting) also exist; however, they are generally for specific industries, and are generally not commonly used for standard room acoustics. For example, D-weighting was specifically designed for measuring aircraft, and has frequency adjustments specifically designed for jet engines.

A.1 Octave Band Levels

Sounds can be broken up into different frequency bands. One common method of dividing frequency bands is into "octave bands". An octave band consists of a range of frequencies surrounding a center frequency, f_c. The limits of the band can be calculated using the following equations:

$$f_{low} = \frac{f_c}{\sqrt{2}}$$

and

$$f_{high} = f_c \times \sqrt{2}$$

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[1] Everest, F. Alton; Pohlmann, Ken C. "Master Handbook of Acoustics." 5th ed. 2009: McGraw Hill. pp. 30-31

Commonly, center frequencies begin at 31.5 Hz, and each successive center frequency is equal to twice the previous center frequency. When plotted on a logarithmic scale, the center frequencies will be equally spaced. Octave band levels are often split further into *third octave bands*, which split the octave bands into three sub-bands.

B. Reverberation Time

A key measure of the sound quality of a room is the *reverberation time* of the room. The reverberation time (expressed in seconds) is the amount of time that it takes a sound to decay by 60dB on an unweighted scale. For this reason, the reverberation time is often abbreviated as T_{60}.

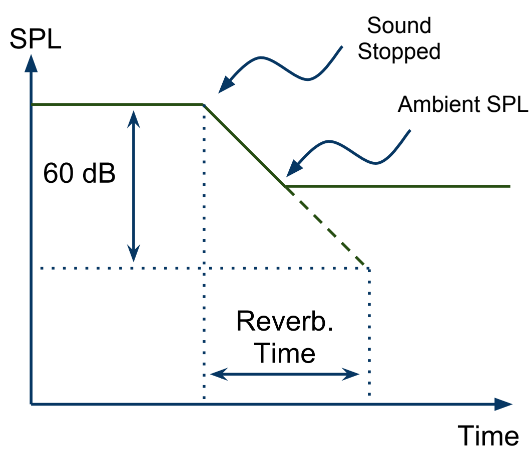

Fig. 2. Non-ideal Reverberation Time Measurement

Because of equipment limitations, it is not always possible to measure time it takes for a sound to decay by exactly 60dB. If the background sound level of the room is approximately 60dB, then the sound would have to be 120 dB in order to decay by a full 60dB,which would be roughly as loud as an air raid siren. If measuring in an indoor space, this is obviously not practical. Therefore, it is common to measure a smaller drop (30-50dB), and extrapolate the slope of the decay to determine how long it would have taken to decay by a full 60dB, as shown in Figure 2.

High reverberation times can be desirable in some situations, such as music performance. In this case, reverberation adds fullness to the music, and can make the performance sound better. Often, reverberation times of large concert venues can be as high as 1.5-2.5 seconds[2]. For academic spaces, however, reverberation should be kept to a minimum. Longer reverberation times make it far more difficult to understand speech, and can be detrimental to learning. ANSI S12.60 recommends that reverberation times in academic core learning spaces be no longer than 0.7 seconds. Furthermore, it states that any academic space should have the ability to reduce reverberation times to 0.35 seconds, if necessary.

[2] *ibid. p. 378*

II. Testing Methods and Procedure

A. Ambient Sound Pressure Level Measurement

Ambient sound pressure levels were measured with both A and C weightings. This was achieved using LabView software with the Sound and Vibration Suite. A microphone was placed near the center of the room, away from any reflective objects such as walls or desks. The A and C weighted sound pressure level was recorded for ten seconds, and the average value of that period was recorded. The sound pressure level was recorded with a sample rate of 51.2 kHz, to effectively capture the entire audible frequency range. A schematic of the setup is shown in Figure 3.

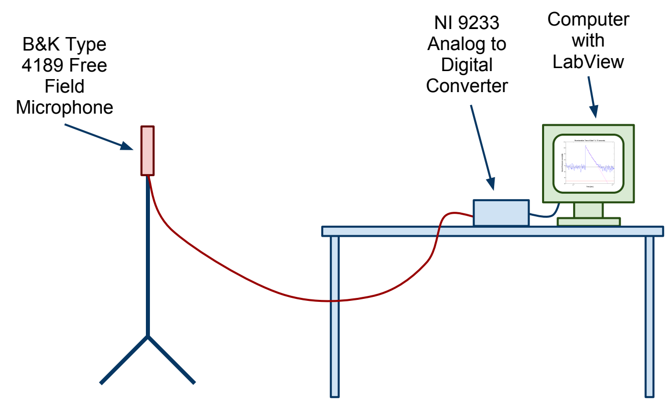

Fig. 3. Ambient Sound Pressure Level Measurement Setup

Sound pressure levels were recorded with A and C weightings to compare with the standards set by ANSI S12-60, which states that core learning spaces with a volume of less than 20,000 ft^3 should not have ambient sound pressure levels which exceed 35dBA (55dBC), and core learning spaces with a volume of greater than 20,000 ft^3 should not have ambient sound pressure levels exceeding 40dBA (60dBC).

B. Reverberation Time

The reverberation time of various rooms was measured using two different methods. The first method used an impulse sound source, and the decay of the impulse was measured. The second method used a sustained white-noise signal. The signal was suddenly stopped, and the decay time was measured. Both methods used a combination of Matlab and LabView to perform the measurements.

B.1 Impulse Measurement

The easiest method of measuring the reverberation time of a room is using the impulse method. With this method, an impulse sound source is introduced, and the decay time of the impulse is measured. To be effective, the impulse should be very loud (greater than 120dB), and should contain a broad spectrum of frequencies. Ideally, the impulse source would contain all audible frequencies; however, this is rarely possible with common sources. Common impulse sources include gunshots, large capacitor discharges, and balloon pops. For convenience and safety, balloons were used as impulse sources for this experiment.

During the testing process, a microphone was set up in the center of the room, away from any tables or walls (to reduce the effects of direct reverberations). A LabView VI was used to record the sound pressure level at a sample rate of 51.2 kHz. Two different sized balloons were used during the testing procedure: a small water balloon with a 5" diameter, and a standard sized "party" balloon with an 11" diameter.

After recording the sound pressure from the balloon pop, custom Matlab scripts were used to convert the sound pressure to an instantaneous sound pressure level. The peak value of the sound pressure level was identified, and a linear regression was automatically fit to the decay. Since all measurements were less than 60dB above the ambient sound pressure level of the room, the regression line was extrapolated to calculate the final reverberation time of the room. This procedure was repeated using a smaller balloon, so that the two values could be compared.

B.2 White Noise Measurement

Although the impulse method of measuring reverberation time is the easiest, it is extremely difficult to excite all audible frequencies equally. Generally, only a small range of frequencies are excited. Although this may be acceptable for a rough estimation of the reverberation time, it does not fully characterize the performance of the room.

To excite a broader range of frequencies, white noise can be used instead of an impulse source. With this method, white noise is played loudly, and is allowed to fill the room with noise. Ideally, the sound would be loud enough that the sound pressure level would be the same regardless of where the measurement was taken. This situation is known as a diffuse sound field, and is ideal for reverberation times measurements. After the sound filed has become diffuse, the noise is suddenly stopped, and the decay is measured.

A microphone was setup in the center of the room, and a pair of studio monitors were placed on one side of the room. Matlab was used to generate a white noise signal at a frequency of 48 kHz, which ensured that all audible frequencies would be excited. The noise was generated while simultaneously recording the sound pressure in the room using LabView. After a period of 5 seconds, the white noise was stopped, and the decay was recorded. The setup was similar to the schematic shown in Figure 3

After recording the decay with LabView, the data was imported into Matlab, where the decay was automatically identified, and a line was fit to the curve. The maximum sound pressure level achieved using this method was approximately 95dB, with an ambient sound pressure level of approximately 65dB. Since the sound pressure level was not 60dB above ambient, the slope was automatically extrapolated to calculate the reverberation time.

B.3 Octave Band White Noise Measurement

White noise reverberation time measurement ensures that all frequencies are excited; however, it does not indicate which frequencies are contributing the most to the reverberation in the room. To find this information, an octave band reverberation time measurement was performed. Although the general setup was the same as previous reverberation time measurements, instead of playing pure white noise, octave filters were applied to the noise. Each set of filtered noise was played individually, and the reverberation time for each octave band was recorded.

The octave band filter set was generated using Matlab, as shown in Figure B.3, such that the upper cutoff frequency of the first band intersected the lower cutoff frequency of the second band at -3dB. This was done for a total of 9 center frequencies, covering the entire audible spectrum. White noise, generated at a sample rate of 48kHz was filtered through each octave band. Each band was played individually, and the reverberation time was recorded. This process was repeated ten times, and an average reverberation time for each octave band was recorded.

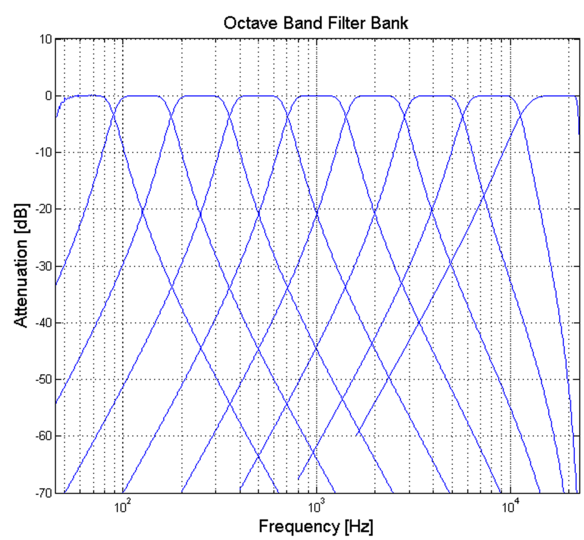

Fig. 4. Octave Band Filter Bank

Because the original white noise signal was filtered so many times, the resulting octave band signals had far lower overall acoustic power. Therefore, larger speakers were necessary to produce high enough sound levels to accurately measure the reverberation time. Rather than using the Mackie HR824 Studio Monitors, which were used for previous white noise experiments, larger JBL MP418S speakers and a Crown XLS402 power amplifier was used. Because of the large size of the speakers, amplifiers and other equipment, the procedure was only performed in the acoustics lab.

C. Software

Extensive Matlab scripts were written to help with the data acquisition and analysis process. The goal was to provide a set of scripts that would allow any user to perform acoustic tests with little or no knowledge of Matlab. For additional information about the Matlab scripts, the following commands can be used:

```
>> help balloon_pop_analysis
>> help octave_band_reverb_time
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III. Results and Discussion

A. Ambient Sound Pressure Level

The ambient sound pressure level of various classrooms, laboratories, study spaces, and common areas is shown in Figure 5.

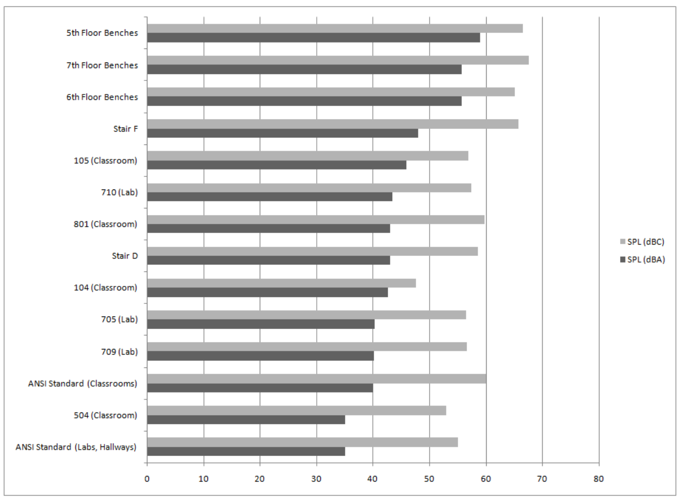

Fig. 5. Ambient Sound Pressure Level of Various Rooms [dB]

From this information, it is clear that almost none of Cooper Unions core academic spaces or ancillary spaces conform with the ANSI standard for ambient sound pressure levels. For many of these rooms, especially those with a large number of windows, the ambient sound pressure level was heavily influenced by noise from the outside. These recordings were take late at night, when traffic is relatively light; however, in the middle of a day, sound levels may rise considerably.

Another significant source of noise was the HVAC system. This noise tends to be quite variable, and the ambient sound can change by as much as 10-15 dBA with the cycles of the HVAC. For example: room 504 (which had the lowest ambient sound pressure level) did not have any HVAC turned on during the measurement. All other rooms did, which caused the ambient sound pressure level to rise.

Finally, although all rooms were tested without any other people present in the room, students' activities in other rooms were clearly audible, and influenced the average sound pressure levels. Music and speech could be clearly heard through the walls, indicating that improper attention was given to sound transmission during the design of the building.

There are two possible methods of lowering the ambient sound pressure level: add absorptive materials on the inside of the room, or reflective materials on the outside of the room. The first method would not reduce the amount of noise that gets into the room; however, it would reduce the reverberations of that noise, reducing the amount of time the sound stays in the room, and also the overall ambient sound pressure level. This method would also reduce the reverberation time of the room, if necessary.

The second method would tend to reflect any sounds from outside of the room back at the source, reducing the amount of noise that ever reaches the room. Although effective, it may be difficult to do this approach, since the building has already been constructed. Furthermore, this method would not help reduce reverberation times in the room.

B. Reverberation Time

The reverberation times of various room and common areas was measured using the impulse method. A balloon was used as the impact source. Often, two different size balloons were used. This is due to the different frequency components of the impulse of each source. Larger balloons generally have a wider range of frequency components, while smaller balloons have smaller ranges. Furthermore, larger balloons have higher total acoustical energy, and are able to produce louder impulses[3], which made them suitable for larger spaces (such as the Grand Stiarcase and the Rose Auditorium).

The reverberation times various rooms in the building are shown in Figure 6, where "Broad Band" excitation corresponds to the large balloon pops, and "narrow band" excitation corresponds to the smaller balloon pops. More narrow band measurements were taken, since smaller balloons were more readily available.

Figure 6 shows that although most core learning spaces fall within ANSI standards for reverberation time (with the exception of rooms LL210 and 801), most rooms are on the upper end of the standard, and are nearly beyond the acceptable maximum reverberation time. The rooms which were beyond ANSI standards could be considered non-standard classrooms (although they are still commonly used as core learning spaces).

Room 801 is officially considered a conference room, and has glass windows covering approximately 50% of the walls. Since glass is much harder than drywall, it reflects sounds much more, leading to an increased reverberation time. Room LL210 also had higher than recommended reverberation times. Again, this may have been due to the floor to ceiling windows on one of the walls of the classroom.

Reduction of reverberation time is a key component in speech intelligibility, and therefore, academic success. It is essential that reverberation times do not exceed ANSI

[3] Patynen, Jukka; Katz, Brian F. G.; Lokki, Tapio. "Investigations on the Balloon as an Impulse Source." *Journal of the Acoustical Society of America* **129** (1), January 2011.

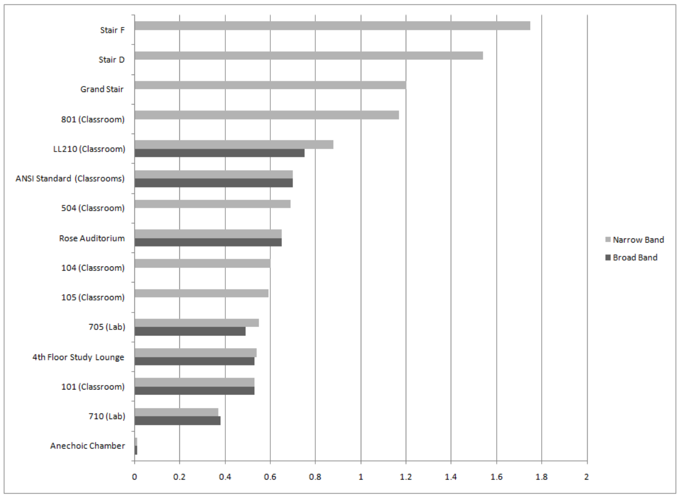

Fig. 6. Reverberation Times of Various Rooms [seconds]

standards; therefore, it is highly recommended that sound absorbing treatment be applied to the interior of highly reverberant rooms. This treatment will not only reduce the reverberation time, but may also reduce the overall ambient sound pressure level in the room as well.

C. Octave Band Reverberation Time

An octave band reverberation time analysis was performed in the acoustics laboratory (Room 710). In this analysis, the reverberation time of the room was calculated for seven octave bands with center frequencies at 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz, 8 kHz, and 16 kHz. The frequency dependent reverberation times are shown in Figure 7.

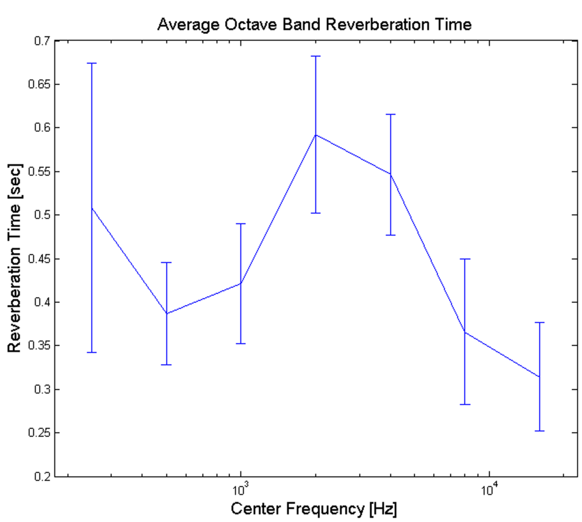

Fig. 7. Average of 10 Octave Band Reverberation Time Measurements

The purpose of this study was to determine the frequencies which effected the reverberation time of the room the greatest. This information can also be useful when determining how easy it will be to hear and understand speech. A spectrograph of human speech reveals that most of the sounds that we can produce lie between 100-4000 Hz, with most of the sound concentrated between 1000-3000 Hz[4], although some speech can even reach frequencies of up to 8000 Hz[5]. Therefore, if speech intelegibility is the primary concern in classroom acoustics, it is important to have low reverberation times for these frequency ranges.

Figure 7 does not show any reverberation times that are drastically above the standard of 0.7 seconds; however, the peak reverberation occurred for the octave band centered at 2,000 Hz, which directly corresponds to the average frequency of human speech. Ideally, the opposite trend would be observed, with the lowest reverberation time at this octave band.

Reverberation times were originally also recorded for the 31.5 Hz, 63 Hz, and 125 Hz, octave bands; however, these measurements were found to be extremely variable, and unable to produce consistent results. Even at the 250 Hz band level, the deviation from the mean is extremely wide, ranging from 0.35 seconds to nearly 0.7 seconds. Additional measurements would have reduced this error; however, most of the error was due to imprecise curve fitting and extrapolation of the decay slope. This is likely due to the fact that the low frequency signals were only able of raising the overall sound pressure level of the room by approximately 20 dB, requiring a large amount of extrapolation to determine the full reverberation time. Because of the overall low sound level, the signal to noise ratio was also lower, increasing the chances of random error in the measurement.

D. Anechoic Chamber Status

The anechoic chamber was recently completed, and is now available for use by the Cooper community. A brief timeline of the of the progress is outlined below:

Installation of Floor and Wall Wedges: Installation of a large portion of the floor and wall wedges was completed during the summer of 2010. This was the fastest stage of the process, since the floor wedges did not require any special mounting brackets, and the wall brackets had already been purchased and installed. At this point, the chamber was also being used for storage of some of the remaining sound absorbing wedges, preventing access to the chamber.

Finish Installation of Wall and Ceiling Wedges: Construction of the chamber ceased for approximately 6 months while waiting for the delivery of special mounting brackets for the remaining ceiling and wall wedges. These brackets were manufactured to work with the existing mounting system that the sound absorbing wedges used. These wedges

[4] Everest, F. Alton; Pohlmann, Ken C. "Master Handbook of Acoustics." 5th ed. 2009: McGraw Hill. p. 74

[5] Lord, Harold W.; Gately, William S.; Evensen, Harold A. "Noise Control for Engineers." 1987: Krieger Publishing Company. pp. 47-53

were moved from the anechoic chamber in the old engineering building at 51 Astor Place. Although the ceiling is sloped, allowing the wedges to be higher at one side than at the other, the wedges were all installed at the same elevation.

Modify Existing Wall: Before a door could be installed in the chamber, the existing opening for the door was widened to allow for the non-standard size of the chamber door. Additionally, portions of the opening were closed off, since the door was installed approximately 2.5' above ground level.

Installation of Doors: Two doors are used on the anechoic chamber. The outer door is constructed with a metal shell with insulating material on the inside, and is approximately 2" thick. This door forms a tight seal with the frame, preventing outside noises from entering the chamber. An interior door is also installed, which is constructed with a light metal frame, and has sound absorbing wedges installed on the face. This door reduces internal reflections that would occur off of the metal exterior door, and further attenuates sounds entering from the exterior.

Lighting in the anechoic chamber is provided by a single incandescent bulb (shown in Figure 8), hanging from the ceiling at the center of the chamber. Incandescent lighting is important, since florescent bulbs tend to produce an audible hum. Other forms of lighting, such as LED lighting, are sometimes used in anechoic chambers; however, these lighting systems are often more expensive and complicated to install. Additionally, LED lighting requires transformers to provide DC power, which can also produce audible hums.

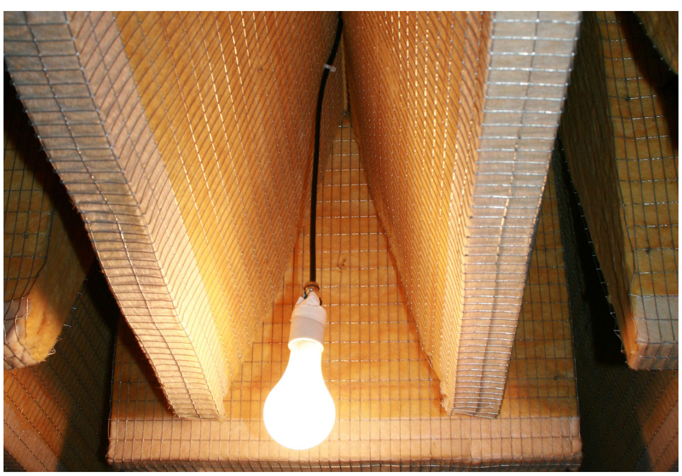

Fig. 8. Anechoic Chamber Lighting

A fire suppression system was also included in the new anechoic chamber. Although the sound absorbing wedges are not flamible, a sprinkler was installed in the center of the room, near the light. The sprinkler (shown in Figure 9) will cause small reflections of high frequencies; however, since the sprinkler is mostly hidden inside a sound absorbing wedge, the effect should be minimal.

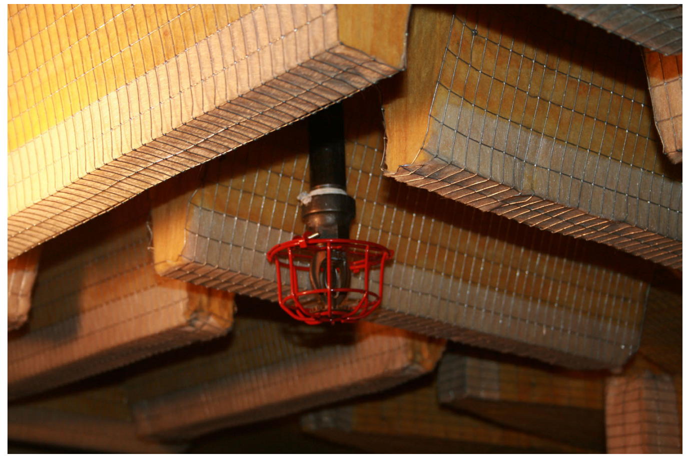

Fig. 9. Anechoic Chamber Fire Suppression

Data lines and power outlets were also included inside the chamber, allowing for computers and other equipment to be used inside the chamber. The single data connection is currently hanging from the ceiling of the chamber, while the power is supplied on two lower corners of the room.

The anechoic chamber was finished in mid April. Although extensive testing has not yet occurred, initial reverberation time measurements were performed in the anechoic chamber. Ideally, the anechoic chamber should have zero reverberation time (since no echos or reverberations should occur in the chamber). Initial measurements produced reverberation times which were less then 0.01 seconds. This is the lowest detectable reverberation time with the current calculation algorithm, which calculates instantaneous sound pressure levels in 0.01 second intervals. Therefore, the actual reverberation time of the room was unmeasurable using current equipment and software.

IV. Conclusions

Based on the overall ambient sound pressure level and reverberation time measurements, Cooper Union's new academic building does not meet ANSI standards for academic acoustical characteristics. Although most rooms were just below the maximum limit set for reverberation times, almost every room that was tested failed to fall below the ANSI standard for the ambient sound levels of the room. In many cases, this was due to external noise from the city, and in almost all cases, the HVAC system seemed to be at fault.

In order to bring the classrooms, laboratories, and study areas into compliance with ANSI standards, it is recommended that sound absorbing panels be installed in the most reverberant rooms. This would help lower the reverberation time in the room, and also reduce the overall ambient sound levels.

V. Suggested Future Work

The goal of this project was not only to characterize the building's acoustical properties, but also setup a system for future testing. This project focused on developing a software system that would allow acoustical measurements to be made by users without significant knowledge of acoustics or programming.

In order to get an accurate representation of the acoustics of the building, it is important that all core learning spaces be tested for both reverberation time as well as ambient sound pressure level. This can be accomplished using the software that has been created for this project, or by using sound level meters. Current sound level meters available in the Cooper Union Acoustics Lab have not been calibrated recently, and do not have the capability of measuring below 60dB. Therefore, in order to make accurate and meaningful measurements, it would be necessary to purchase a new sound level meter capable of performing the measurements described in the report.

Additionally, it would be useful to perform octave (or even third octave) band frequency reverberation time measurements on each of the rooms. This process could shed light on the specific frequencies that are problematic, and determine whether these frequencies interfere with human speech. It can also help with the selection of sound absorbing panels (if necessary), since the effectiveness of most of these products is frequency dependent.

VI. Acknowledgements

I would like to thank my advisors: Professor Wei and Professor Baglione for their guidance and assistance with this project. I would also like to thank Jody Grapes and the staff of buildings and grounds for their help with the completion of the anechoic chamber.

VII. References

[1]ANSI/ASA S12.60-2010/Part 1: Acoustic Performance Criteria, Design Requirements, and Guidelines for Schools.

[2] Everest, F. Alton; Pohlmann, Ken C. “Master Handbook of Acoustics.” 5th ed. 2009: McGraw Hill.

[3] Patynen, Jukka; Katz, Brian F. G.; Lokki, Tapio. “Investigations on the Balloon as an Impulse Source.” *Journal of the Acoustical Society of America* **129** (1), January 2011.

[4] Lord, Harold W.; Gately, William S.; Evensen, Harold A. “Noise Control for Engineers.” 1987: Krieger Publishing Company. pp. 47-53