Acoustic
Final Project Report

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Introduction

This essay will give a detailed overview of the four group practical Workshops completed for the module MD4003.

REVERBERATION

Introduction
The aim for this project was to decipher the reverberation time within closed environments, a closet, a classroom and a café. This method is especially useful when designing and or equalizing a studio live or control rooms..

Background
Reverberation is how long sound continues after the sound has been generated. The sound creates reflections from the walls and other material in the room and we measure these reflections on how long they take to decay to 60db. [1]

The RT60 is how long it takes for the reverberation to decay to one-millionth of the source sound, 60dbs. [2]

The equation:
0.161 x **(V)**olume / **(A)**bsorption

Volume = the length, height and width of the room.
Absorption = the materials absorption co-efficiency

Procedure
Firstly the rooms were chosen and then measured in width, height and length, following on to measure the surface area. Furthermore the materials of the walls in the rooms were recorded for the absorption co-efficiency. In each location balloons were popped and the sound was recorded. The sound files were then imported in to software. The results from the software showed the time of decay for each sound and room. As the sound is generated the software calculates the time it takes for the sound to decay to 60dbs.

Results
The largest space with the most volume was the Café and this had the longest reverb time.

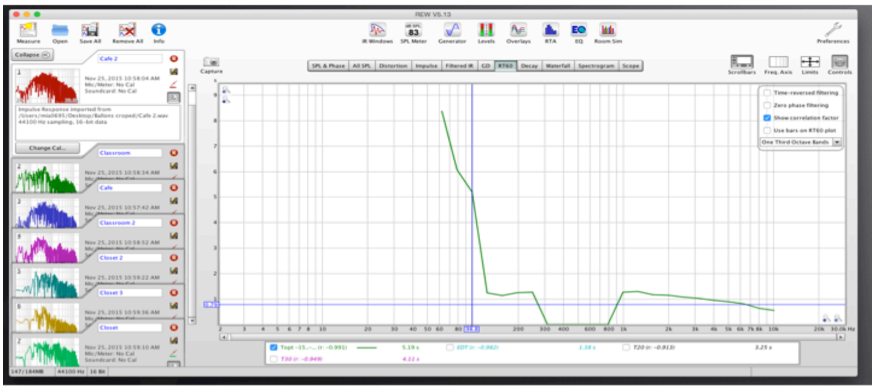

Figure 1 shows the café with has the highest time of 3.39ms

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[1]Llewelyn S Lloyd, *Music And Sound* (Freeport, N.Y.: Books for Libraries Press, 1970).

The smallest space with the least volume the closet had an average reverberation time.

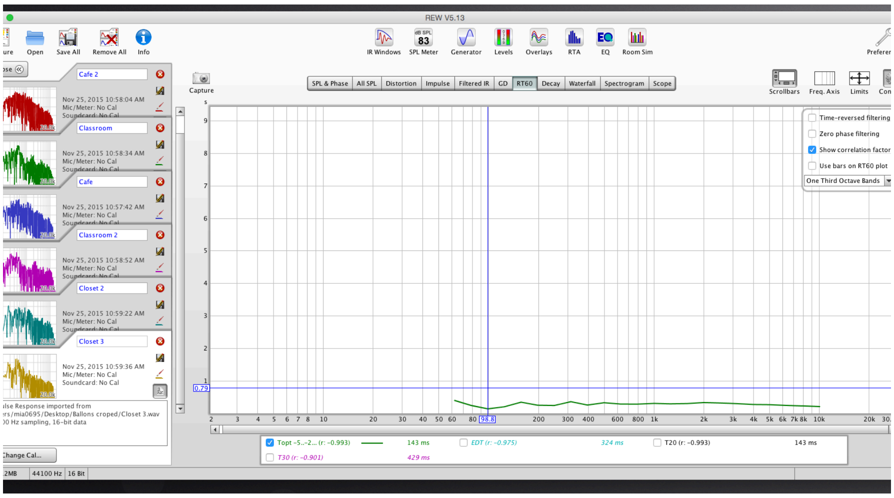

Figure 2 analysis showing 143ms of the time taken for the sound to decay to 60dbs.

Conclusion and review

The mid sized space (the classroom) had the least reverb time but also shows a high rt60 of 3.09ms and a low rt60 of 823ms making the data slightly inconclusive, as an average cannot be established. *(See figure 3 & 4)*

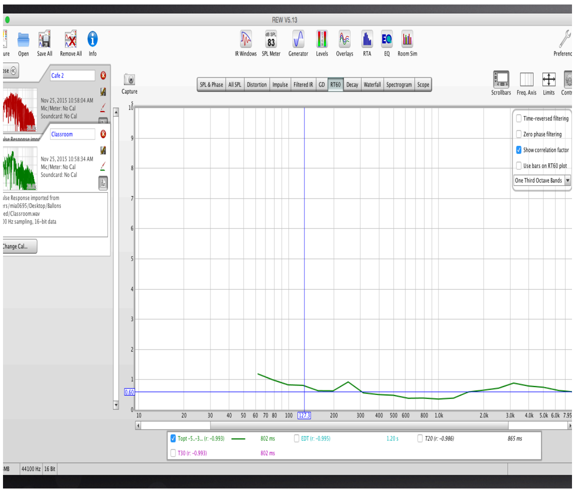

Figure 3 RT60 of 823ms

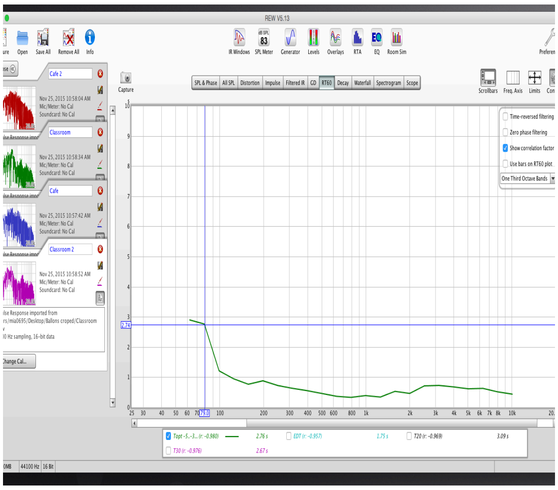

Figure 4 RT60 of 3.09ms

This might be due to the material absorption coefficients in each space and insulation of the walls. The problem with this equation is that it implies that the decay of sound is exponential, that the sound dies down in even increments of dbs. Which is not the case of a real life situation.

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ACOUSTICS OF A GUITAR

Introduction

For this project the aim was to investigate the acoustics of a guitar, by finding out if there are similar harmonics generate when similar notes on different strings are plucked. The group used two different guitars and recorded the same notes on different strings on the guitar.

Background

When a string is plucked on a guitar it creates a harmonic series, the series of this harmonic phenomenon differs depending on factors such as where the string is plucked, what string is plucked and the body of the guitar i.e. type of wood used.

With the plucking of each string on a specific note, the guitar will produce harmonics. We can say that if two or more frequency peaks occur at the same frequency on different strings or notes, then we determine this as Formants; the same harmonics resonating regardless of the string plucked. [3]

Procedure

Using an Acoustic Spanish Guitar the group recorded each note from each of the strings, from the open string to the 12th fret.
The use of a closed studio environment was used and to record the group used a TASCAM recorder.
Once these had been, the group transferred these sound files to software called Audacity, where they were chopped into more manageable and organised analysis.
By using guitar tuner sifter we were able to organise these individual sound files into notes.
For further analysis the notes were then transferred to software AudioXplorer where the parameters:

1. Spectrum Analysis 2. 32768 size points 3. Window for Hamming.

Results

Comparing the data from two of the same note but on different strings. The findings show that on note E the 2nd, 3rd and 4th string contain harmonics in the frequencies of 320Hz, 690Hz and 1,000 Hz *(see figure 5, 6 &7).* Meaning that on the soundboard of the guitar there are formants at these frequencies.

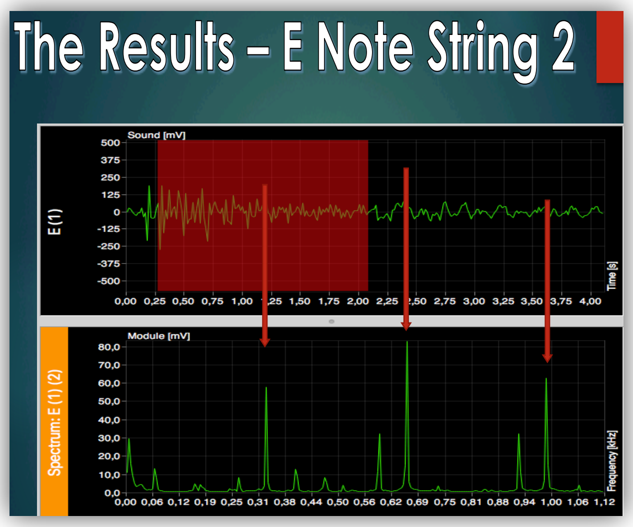


Figure 5 harmonics at 320Hz, 690Hz and 1,000 Hz

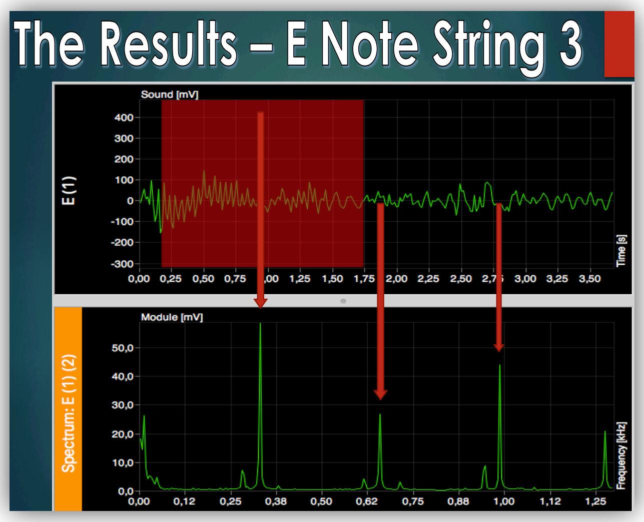


Figure 6 harmonics at 320Hz, 690Hz and 1,000 Hz

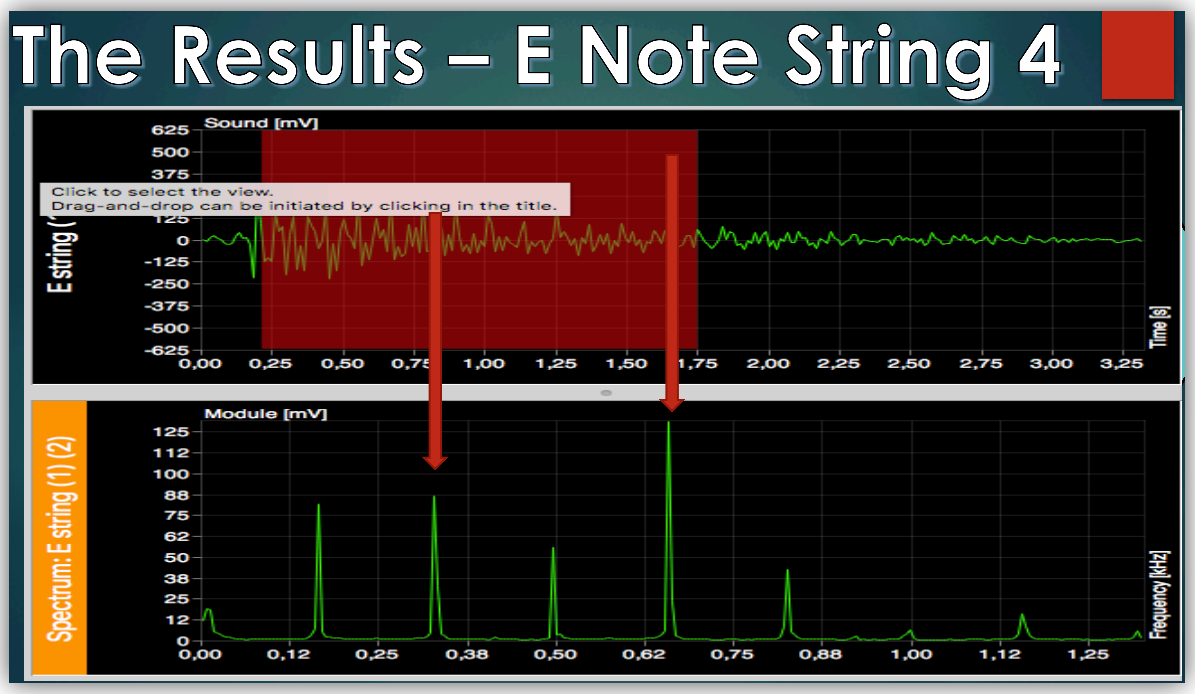


Figure 7 harmonics at 320Hz, 690Hz and 1,000 Hz

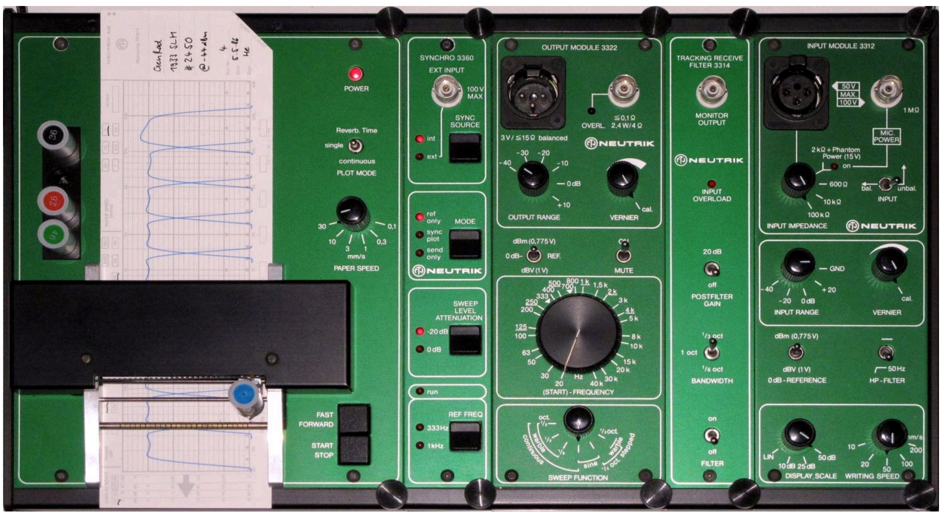
The next note was the A note, where the frequencies 440Hz and 880 Hz were amplified when played on the 1st, 3rd and 5th strings. Determining that on there are formants at these peaks on the soundboard of the guitar.

On the note of B, there were significant boosts in the 500hz and 1000hz range. The 3rd and 4th strings had a boost in the 750Hz range but the first string at that frequency spike did not share this.

Part 2

A Neutrik Audiograph system is a digitally controlled precision audio analysis system. It allows you to analyse the acoustic sound of guitars, but rooms as well as noise-level. [4]

Using a Neutrik Audiograph System *(see figure 8)*, the group connected a small node from the Neutrik Audiograph to the guitar, which is a sensitive accelerometer, creating areas of no vibration on the guitar. Also connected to the Neutrik Audiograph was the vibrating ascending frequency generator that created a series of tones increasing in frequency.



As the tones are being played the Neutrik Audiograph collects the information from the vibrations and the results show the peak resonant frequencies that are within the body of the guitar.

This shows us that when the node is placed underneath the bridge to the right of the guitar that the resonant frequencies peak at around 900hz and 1900kHz, when placed on the upper bout of the guitar the peak frequency was 2250khz.
Placing the node underneath the bridge gave a high frequency peak of 600hz and 1600khz and finally positioning the node on the upper side of the bridge gave frequency peaks of 450hz, 1000khz and 2000khz. *(See figure 9)*

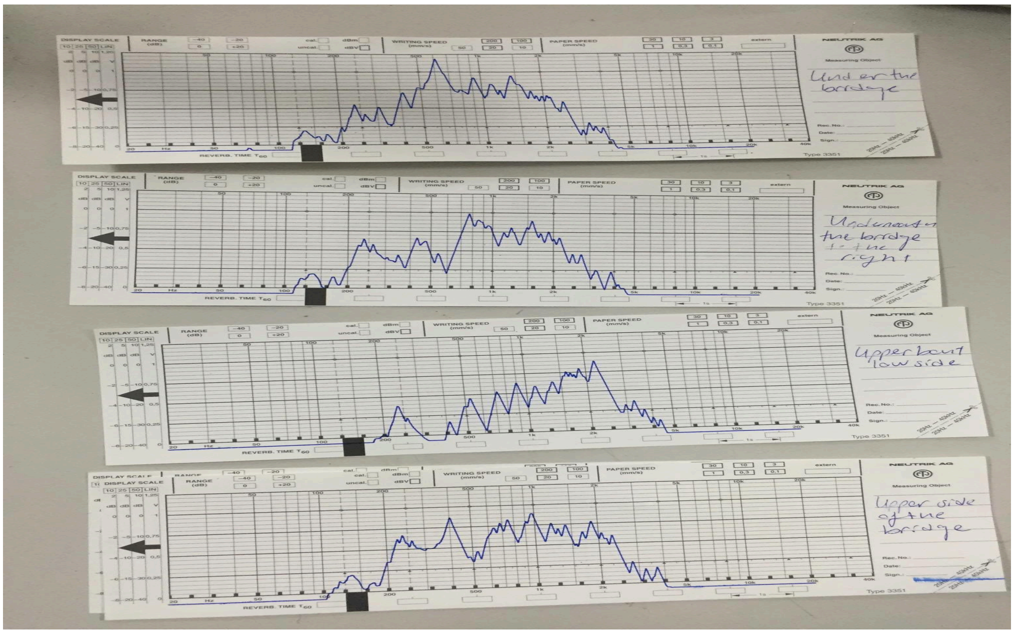


Figure 9 Results of the Neutrik Audiograph on the right is where the node was positioned. The centre shows the peaks in frequency.

Conclusion
This project demonstrated how the vibrations of the strings and notes when played on the guitar would be boosted in certain ranges of frequency. The soundboard of the guitar has its own resonate frequencies and so when the strings are plucked the soundboard vibrates in sympathy creating formants within these regions.

AN INVESTIGATION IN PERCEPTUAL MASKING

Introduction
The human ear splits sound into frequency bands based on critical bands. The critical band is the group of frequencies whereby a second tone can interfere with the perception of the first tone by masking. The aim in this project was to see how the brain masks different frequencies, dependant upon their amplitude and their distance between each other.

[4]Mr Andrew Brakhan, *Neutrik Connectors*, 1st edn (Stamford: Neutrik Products, 1982), p. 1

Background
Often two different frequencies can be heard when played at the simultaneously, instead of a whole tone. When the frequencies are within the same critical band they are perceived as a tone. This is due to the filtering of the sound by the cochlea within the ear. Combinations of sound are divided into distinctive frequency sections and cause a peak within inside the cochlea. They are then autonomously selected on the auditory nerve and then transmitted to the brain. This only happens if the sounds are so different in frequency that they are then perceived as separate. Instead they will be considered as within the same critical band and therefore perceived as a whole tone.[5]

Masking is how much of the frequency will be heard. When a masker masks a sound with a different frequency our auditory system is unable to separate the two frequencies.[6]

Procedure
Firstly using MAX software the group created stimuli by creating parameters within the MAX software. Selecting the IOSC bank in MAX creates a tone and selecting the HZ using a number command allows the user to choose which frequencies are desired. Once the frequencies have been chosen, parameters for the amplitude are entered, amplifying the frequency at 0.01khz increments and the frequency spectrum was visually displayed by a diagram *(See 10)* and then to finally record these as .WAV sound files using the sfrecord command.

On the first experiment three stimuli were created, the fundamental of the first frequency was 300hz with its first 5 harmonics. The second had its 5th harmonic removed and the third had the 5th removed and the 12th harmonic additional.

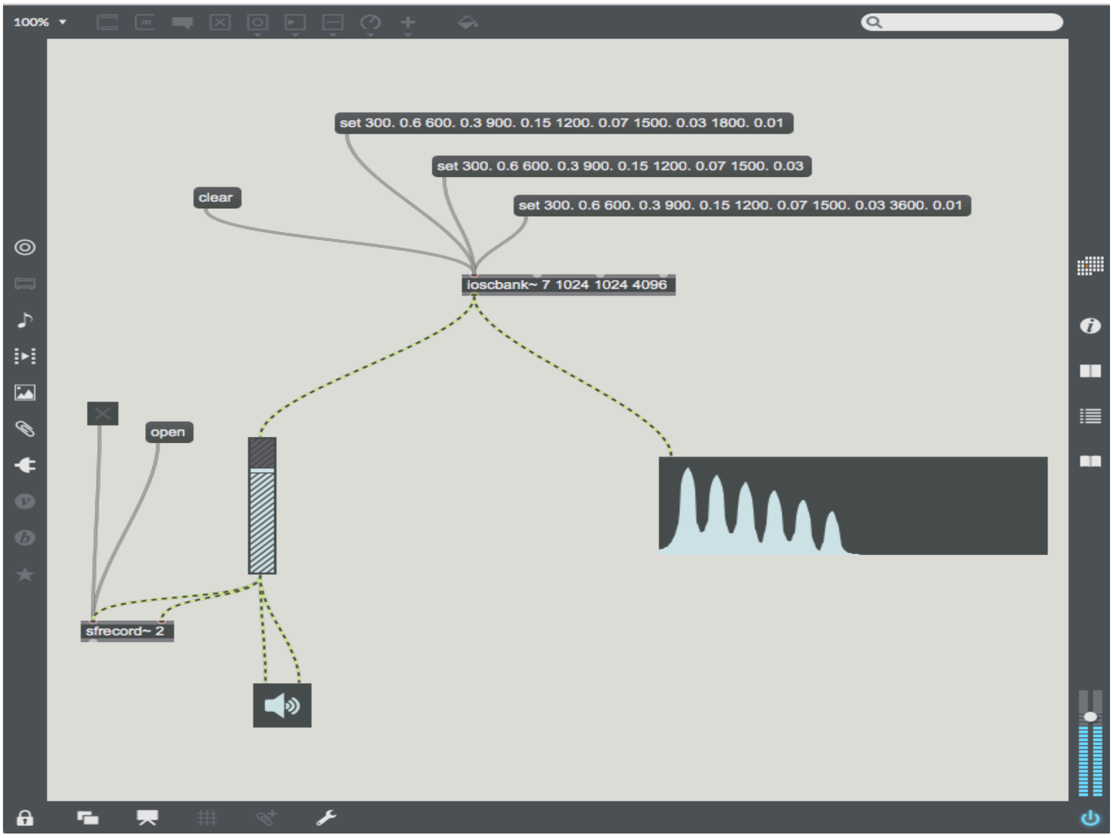


Figure 10 Max patch with frequencies and amplitudes, frequency spectrum and IOSC bank.

Once the stimulus had been created the next step was to introduce these to 20 subjects. The way in which the stimulus were played to the subjects was important, they were asked firstly to see if they could hear a difference in the stimuli or not?

Hypothesis
Expectations were that subjects would hear a difference between stimuli 1 and 3 but not 1 and 2. This was due to the greater distance between the last two harmonics.

Results
The results show that the hypothesis was correct with more people hearing a difference between stimuli 1 and 3 than between 1 and 2. As there is only one record of the statistics taken for the tests, the graph below will be based on the 9 subjects taken by a group member.

Subject (9)	Heard Stimulus?	First Test Difference?	2nd Test Difference?
Valerie Hines – 47 No music background. Test Taken at home in front room with noise cancelling headphones.	Y/Y	Subject noted difference in all sound files.	Subject noted a difference in all sound files, in particularly said that sound file 1f got bigger?
Oliver Sims – 26 Works in television, no music background. Test Taken at home in front room with noise cancelling headphones.	Y/Y	Noted that all sound files were different, stating stimulus b is lower and C had a high pitched ringing.	Subject noted that all sound files were different apart from the 1f & 1g that the subjected noted, sounded very similar if not the same.

no music background. Test taken in TV studio gallery with noise cancelling headphones.		and noted a Difference in A and C1	Difference in D & E same.
Keith Goodyer – 49 Works in television as a technician has many years of experience in audio and video as well as radio. Test taken in TV studio gallery with noise cancelling headphones.	Y/N	Noted no difference in A & B, but noted difference in A & C.	Subject noted that 1a & 1b had a difference in amplitude. 1c had difference. 1d & 1e were the same. 1e+1f. 1g had 2 separate frequencies.
Ophelia Dennis– 34 Works in television, no music background. Test taken in TV studio gallery with noise cancelling headphones.	N/Y	A & B no difference, Noted C had a 'deeper' sound.	1c high pitched 1c+d sound the same. 1g higher.
Natalie Thomas – No music background. Test taken in TV studio gallery with noise cancelling headphones.	Y/N	A & B no difference. A & C difference noted.	1d Lourder 1e Bit of difference 1f big difference 1g different.
Tasty - 47 Works in television, Has a background in radio and radio hamming. Test taken in TV studio gallery with noise cancelling headphones.	N/Y	A & B similar C difference.	1b louder 1c difference 1d+1e shifted slightly. 1f – Emptier/higher 1g – 2 tones louder than the other.
Lucy Hillier – 30 Works in television, no music background. Test taken in TV studio gallery with noise cancelling headphones.	N/Y	A & B no difference. C1 – Difference.	Difference in 1f – 1g.
Nick Stoppani – 37 Works in television, no music background. Test taken in TV studio gallery with noise cancelling headphones.	N/Y	A& B C – No difference. C a little higher pitched.	Difference in – 1g a little – faster

Figure 11 the information noted down at the exact time of conducting the listening test. (These words (432) should be discounted from the word count, as this is a from previous recorded data)

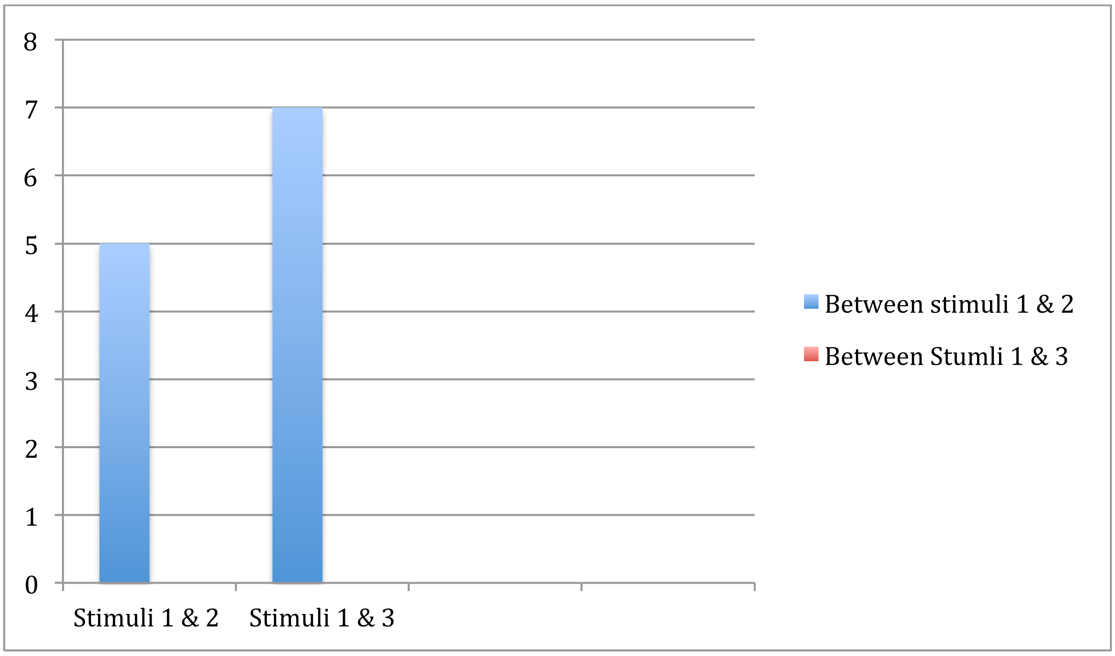


Figure 12 Details 5 people hearing a difference in between stimuli 1 & 2 and 7 people hearing a difference between stimuli 1 & 3.

Conclusion

Concluding that the hypothesis was correct that because stimuli 1 and 3 had the furthest distance between the last two harmonics.

This project is especially useful for the mixing and production of music due to the understanding of how the general public hears frequencies and music in general through the critical band and auditory masking.

Test 2

Procedure

With the 2nd experiment the subjects were asked to listen to a series of 7 tones and tell whether they could hear a difference.

Stimuli (1A) had a fundamental frequency of 300Hz with the first 5 harmonics.

The other stimulus (1B-1G), were the same with the difference that the 5th harmonic has a gradual increase in amplitude from 0 dBs to 0.1 dbs.

Hypothesis

The hypothesis is that people will hear differences between stimuli 1F and 1G, as this is where the biggest change in amplitude happens between these two stimuli from 0.6 – 0.1dbs.

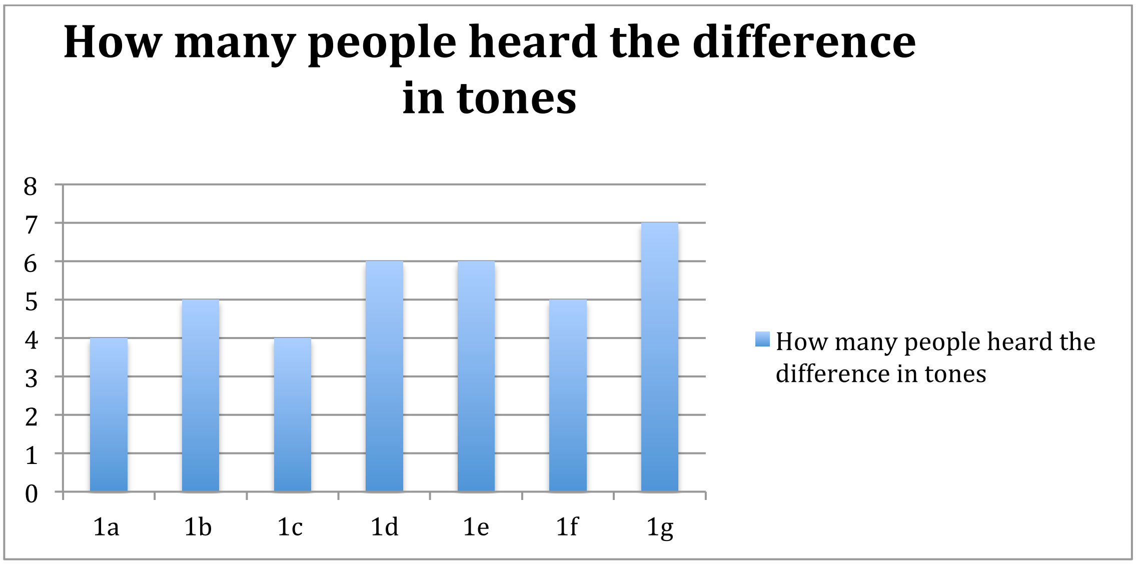


Figure 13 Graph detailed the amount of people heard differences

Conclusion
These results show that the most heard difference in stimulus 1g as this had the highest amplitude of 0.1dbs on its 5th harmonic.

SOUND INSULATION

Introduction
This project investigated how much leakage of sound from one studio control room to the adjacent studio live room. The experiment's goal was to calculate how much transmission loss there was between the control room and the live room in the recording studio.

Background
There are different sound absorbents and reflectors within the studio rooms and any given time. Transmission loss is the reduction of sound as it passes through acoustic barriers. Materials cause different transmission loss due to their absorption co-efficiency.

Hypothesis
The predicted hypothesis was that frequencies higher in the spectrum would have more transmission loss and the lower frequencies would have less transmission loss. This is due to high frequencies having a short wavelength and lower frequencies having a longer wavelength and so the higher frequencies would lose energy quicker.

Procedure
Using a decibel meter, in both the live and studio rooms, the group recorded the ambient level at 6 different points within each room and played increasing frequencies, measuring the amplitude at the six points of the control room and live room.

The logarithmic value of the decibel requires an equation in order to locate the amplitude for individual frequencies in each of the rooms.

The equation used:
10xlog (10 db1/10 + 10 db2/10 +…10 db6/10)

Db(#)- References which measurement 1-6

	Measurements			
	90.4	1096478196		
	80.7	117489755		
	88	630957344		
	94	2511886432		
	90.9	1230268771		
	89	794328235		
		6381408733	1063568122	
	Logarithmic average		90.27	dB

Figure 14 indicates the amplitude level for each 6 point in the control room with a frequency produce at 1000 kHz and its logarithmic average.

	Control Room	Live Room
Ambient Level	49.5	49.23
200Hz	94.12	46.47
500Hz	85.32	45.35
1000Hz	90.27	47.92
2000Hz	95.73	47.08
5000Hz	88.31	46.73

Figure 15 indicates the ambient level in each room and then the amplitude level of the room at these frequencies.

Each room required having the average of the ambient level subtracted from the equation so to calculate the loss in transmission. The control room's subtractions did not change the results significantly enough to warrant any change.
Within the live room the readings indicated that they were higher than that of the recorded frequencies, giving negative results that could not be calculated.

To calculate the transmission loss, there requires a change in the averages from logarithmic to non-logarithmic values to subtract them. Once the averages have been converted, the live room averages are subtracted from the live room averages in-order to calculate the transmission lost.

	Control Room	Live Room	Transmission Loss
Ambient Level	49.5	49.23	
200Hz	94.12	46.47	47.65
500Hz	85.32	45.35	39.97
1000Hz	90.27	47.92	42.35
2000Hz	95.73	47.08	48.65
5000Hz	88.31	46.73	41.58

Figure 16 Average amplitude levels plus transmission lost, with an irregular reading at 200hz for transmission loss.

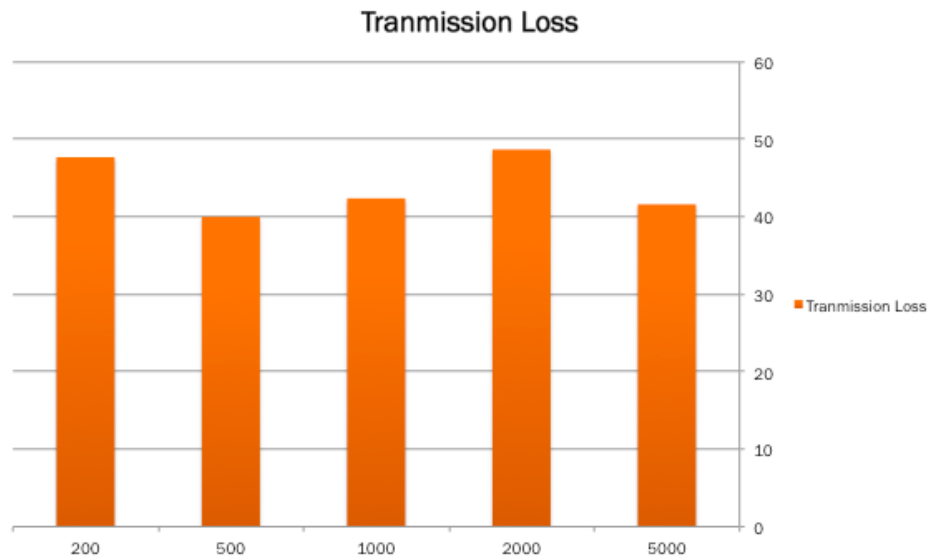


Figure 17 detailing the loss in transmission at what frequencies within the spectrum.

Conclusion
In conclusion our hypothesis states that the higher frequency = a higher transmission loss and lower frequency = a lower transmission loss

The results have shown that the theory is correct especially within the central area of the spectrum but the low end and mid-high regions of the spectrum had high levels of transmission loss, which were not expected.

Initially this could be due to large amounts of people within the rooms, but a more academic approach would be due the resonate frequencies that are present in the room, as such the sound vibrations could be vibrating materials in the room that are boosting frequencies in specific frequency bands.

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Word Count: 2636 – (432) for figure = 2204

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