Measurements of Acoustic Systems with Maximum-Length Sequences

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I. ABSTRACT

Using a maximum-length sequence technique, which is a pseudo-random generated random sequence with properties that help in acoustic measurements of system, such as its random cycling makes it appear spectrally white, but the sequence is deterministic and periodic and it cycles though all possible binary values. This allows for accurate and repeatable measurements on a system, and an MLS impulse can be easily transformed into a frequency response, giving a complete representation of the system. This can be used to describe systems such as Low and High pass filters, anechoic measurements of a loudspeaker in a normal room, as well as the reverberation time using the Schroeder method of backward integration to transform the an impulse response plot, into a decay plot. We were able to analyse High and Low Pass filters and compare their operation with theory, and found a correlation with a difference of only 13-22% and a delay time between 0.02-0.03 ms for the signal response through the filter. Also analysing the frequency response and impulse response of a loudspeaker we were able to determine the speed of sound to be $396\frac{m}{s}$, which only differed by 16% from the accepted of $340\frac{m}{s}$. The reflection responses were also observed in the MLS analysis, it can be seen that higher frequencies reflected more than lower frequencies. It was found that lower frequencies have a greater reverberation time than higher frequencies within a room.

II. INTRODUCTION

Acoustics is a subject with pervades many aspects of scientific and social aspects of life. It is often taken for granted and gone overlooked in day to day life, yet it is something that effects people's lives every day, be it directly for entertainment through the use of speakers or sound systems, or indirectly though engineered products that reduce the sound they emit or systems that block sound from external sources.

We will be investigating the method of using a maximum-length sequence (MLS) computer generated sequence which acts like random noise, but is deterministic and periodic to make quasi-anechoic measurements. This is an efficient time-domain method to identify a system. Using Fourier transformations the periodic impulse of the MLS can be transformed to the frequency domain where it becomes the frequency response of the system.

The MLS method is popular because it's properties along with computers make it easy to measure the acoustic properties of a system, quickly and efficiently. Since the MLS is periodic and deterministic these measurements can be repeated with a high degree of accuracy and repeatability, which is important for such measurements in scientific and commercial applications. Along with the cross-correlation, this method isn't affected much by extraneous noises which may be present in non-ideal environments, so this could be potentially used in noisy environments.

Using this method we will measure the response of a High and Low pass filter, make quasi-anechoic measurements of a loudspeaker in a normal room (rather than an anechoic chamber), and measure and identify the reverberation decay of a room for specific frequency ranges (octave bands).

III. THEORETICAL BACKGROUND

The maximum-length sequence (MLS) is a type of pseudo-random binary sequence which can be inexpensively generated using maximal linear feedback shift registers (LFSR) which is periodic and produce every binary sequence that can be represented by the shift registers. MLS exhibits a random noise as it cycles through its values, which is important because it makes it spectrally white (like white noise) and MLS also has a property which the autocorrelation function closely approximates the train of a Kronecker delta function, or an impulse train.

$$R_{xx} \approx \delta(k) \tag{1}$$

Using this property the cross-correlation can be easily found by cross-correlating the noise input (which is easily calculated for the MLS as the autocorrelation) with the output.

$$R_{xy} = R_{xx} * h(k) \tag{2}$$

So the impulse response $h(k)$ can be easily found for an MLS with:

$$R_{xy} \approx \delta(k) * h(k) = h(k) \tag{3}$$

quite efficiently since it only requires addition and subtraction which is an extremely simple operation on binary bites compared to multiplication or division.

The impulse response of a system can be calculated using the Fast Hadamard Transform (FHT), which is a form of generalized Fourier Transforms, which is equivalent to a multidimensional Discrete Fourier Transform, and like the Fast Fourier Transform (FFT) can quickly and efficiently transform from the time to frequency domain and vice-versa.

The frequency domain is often characterized by a magnitude and a phase angle, where the magnitude is defined in decibels (dB), a logarithmic scale:

$$dB = 10 \log \frac{Power}{Power_{ref}} \tag{4}$$

$Power_{ref}$ is often standardized to the sound pressure of 20 micropascals in air, or 1 micropascal in water, however we will determine the unity gain of the system using a loopback method, which is the gain needed to have an equivalent input to output power ratio of 1:1 and us this as our reference level for our system.

Filters

Filters are used to attenuate signals as they pass through them. An electronic filter, such as a Low Pass (LP) filter, attenuates a signal for frequencies *higher* than a specific cut-off frequency, similarly a High Pass (HP) attenuates signals for frequencies *lower* than a cut-off frequency.

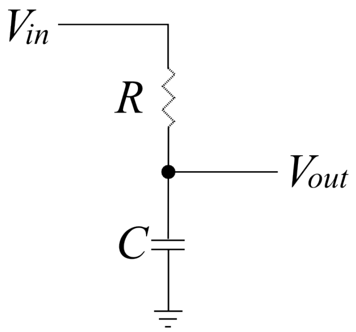

FIG. 1: The circuit diagram of a first order, analogue Low Pass electronic filter.

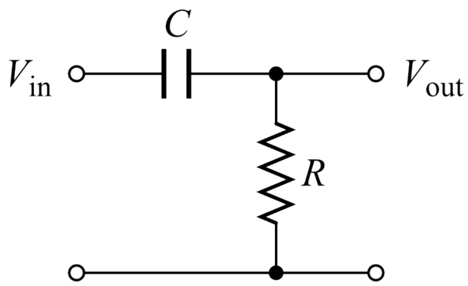

FIG. 2: The circuit diagram of a first order, analogue High Pass electronic filter.

The cutoff frequency can be given by:

$$f_c = \frac{1}{2\pi\tau} = \frac{1}{2\pi RC} \tag{5}$$

Or for angular frequency breakpoint in radians:

$$\omega RC = 1 \tag{6}$$

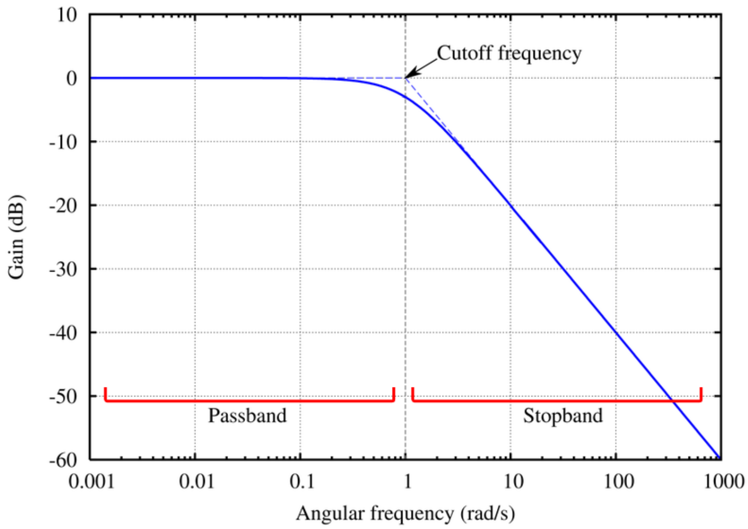

FIG. 3: The frequency response of a signal processed by a Low Pass filter, where the passband and stopband regions are shown, and the cutoff frequency labelled.

Reverberation Time

The definition of reverberation time, is the time it takes for the sound energy density to decay 60dB, which is often referred to RT_{60}. The decay rate of sound is exponential, so the rate of decay per dB should be constant, but it depends on the absorbing surfaces within the room. By using reverse time integration, also known as Schroeder Backward Integration, a impulse response over time plot can be transformed into a decay plot (Schroeder Plot). By using linear regression on the decaying part of the plot at specific bounds, the reverberation time can be found quite easily and accurately.

IV. EXPERIMENTAL DESIGN AND PROCEDURE

Normalisation of Gain and Time Delays

To find the data required for normalisation of the gain and the time delay we attached the output of the CLIO to the input. This allowed us to measure the loop-back. We then used the maximum length sequence analysis mode to record the minimum time delay from the impulse response. Using the frequency response graph we measured the response that represents unity gain for the feedback system.

Time and Frequency Response of Filter

Given a high and low pass filter networks we measured the attenuation of the signal over a range of frequencies. For each network we plotted the frequency response over a range of 20Hz to 20 kHz. As well as the impulse response for the first 9ms of operation. From this data it is possible to measure both the breakpoint and the time constants for each network.

Frequency Response of Loudspeaker

To measure the frequency response of a loudspeaker and microphone set up we connected the input and output cables from the CLIO to an amplifier box. This amplifier powers both the loudspeaker and an active microphone. We placed the microphone 0.5m from the speaker, on the same axis as the tweeters. Ensuring that the speaker and microphone were far from walls and other obstacles, maximising the reflection free time, we performed an analysis using the program. The Frequency response for the whole time range was recorded as well as truncated impulse responses from 1-5ms.

This test was repeated with the speaker and microphone placed close to a wall so that strong reflection signals could be observed in the impulse response.

Reverberation of a Room

To measure the amount of time taken for the sound in a room to decay by a certain amount we arranged the speakers and microphone so that they were quite far from each other. We also made sure that the door was closed and minimised the absorbers present. The time we were interested in was the RT60, defined as the time taken for the sound to decay by 60dB. We set up the CLIO system to measure the reverberation decay and selected all the frequency ranges from 63Hz to 8kHz. With this data we were able to analyse the reverberation time as a function of frequency.

We repeated the experiment with the door open and a large foam absorber placed behind the microphone, allowing comparisons of reverberation in closed reflective surroundings and open absorbing surroundings.

V. ANALYSIS

Normalization of the Gain and Time Delays

The frequency response of the loop-back measurement was a straight line at $-5.09dB$, this relates to unity gain. The flat line is expected as there is no input signal to create a transient waveform.

The impulse response of the system is a time delay followed by a spike in the voltage. This delay is the minimum possible for the system, as it occurs from just the imput and output wires and the components within the CLIO. The time delay, as measured from the impulse response graph, is 0.23 ms.

Time and Frequency Responses of a Filter

The frequency and impulse response graphs are given in Figures 4 and 5 respectively. The frequency response graphs have straight line extrapolations superimposed on them, the intercept of these two lines is the frequency breakpoint.

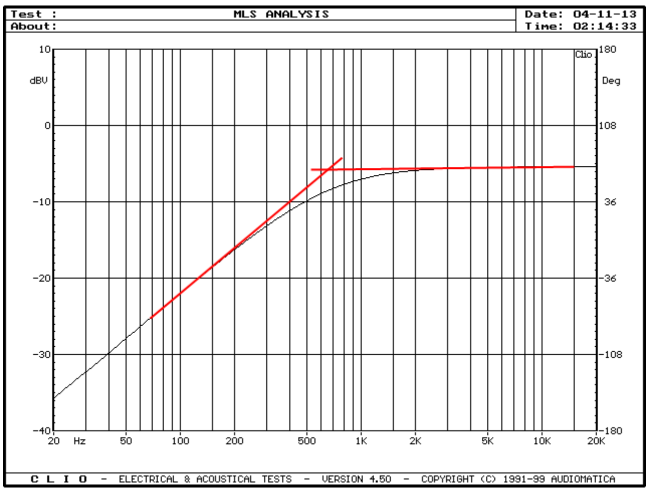

FIG. 4: Frequency response of a high pass filter

From inspection of the graph it is determined that the frequency breakpoint for this system is $650 \pm 50Hz$. The theoretical value for this is calculated using equation 5

$$R = 15k\Omega$$
$$C = 20nF$$
$$f_c = \frac{1}{2\pi \times 15 \times 10^3 \times 20 \times 10^{-9}} = 531Hz$$

This gives our value a difference of 22% from the theoretical value. It can be seen that in the pass band for this filter the gain is $\sim -5dB$ this is close to the value associated with unity gain.

The time constant of the filter is related to the breakpoint frequency by equation 7.

$$\tau = \frac{1}{2\pi f_c} \tag{7}$$

using the measured value for f_c from the graph gives

$$\begin{aligned} \tau_H &= \frac{1}{2\pi f_c} \\ &= \frac{1}{2\pi \times 650} \\ &= 0.24ms \end{aligned}$$

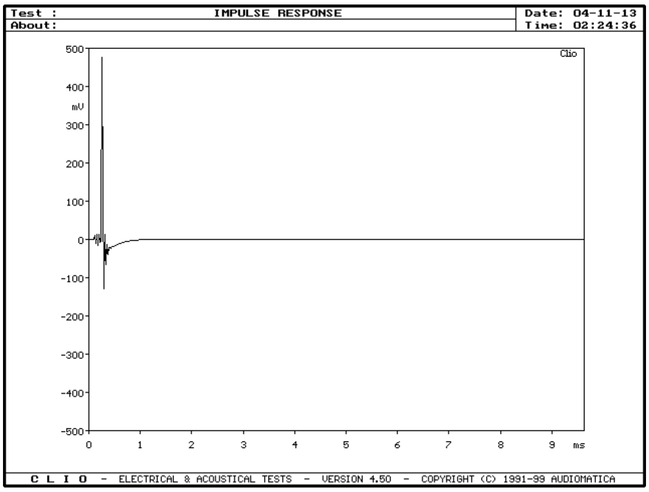

FIG. 5: Impulse response of a high pass filter

The impulse response shows the delay between the signal being generated and then received. The value for this as determined using the cursors on the software is 0.25ms. When the minimum time delay is subtracted from this value we get a 0.02ms time delay associated with the filter alone.

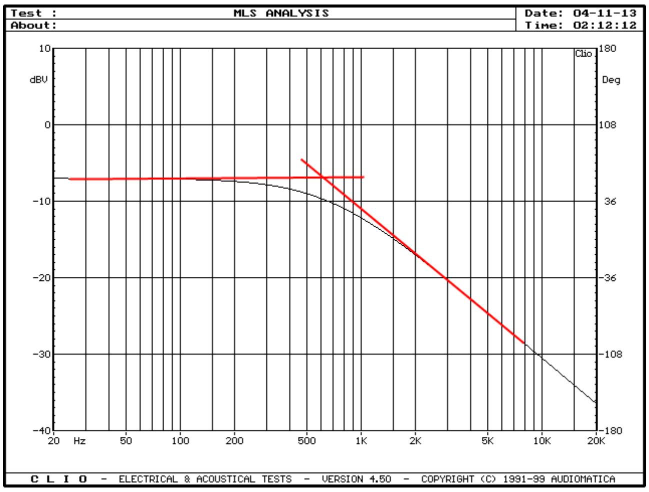

FIG. 6: Frequency response of a low pass filter

From inspection of the intercept from this graph it can be seen that the frequency breakpoint of the low pass network is $600 \pm 50Hz$. As the resistor and capacitor values are the same as in the high pass filter the theoretical breakpoint is the same ($f_c = 531Hz$). This gives our measured value a difference of 13% to that of the theory. It can also be seen from the frequency response graph that the gain in this filter is $\sim -7dB$, given that the decibel value associated with unity gain is $-5.09dB$ this shows an attenuation in the pass band of the filter. This attenuation is caused by the input resistance of the CLIO.

Substituting the breakpoint value into equation 7 in order to calculate the time constant

$$\begin{aligned}\tau_L &= \frac{1}{2\pi f_c}\\ &= \frac{1}{2\pi \times 600}\\ &= 0.27ms\end{aligned}$$

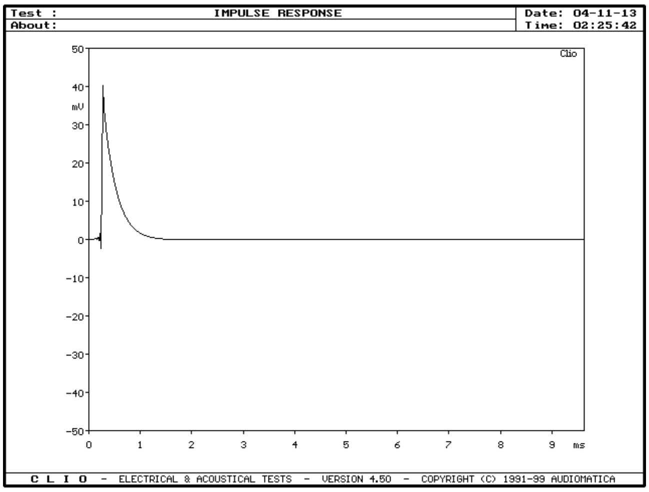

FIG. 7: Impulse response of a low pass filter

The delay in this network, as determined with the cursors on the software, is $0.25ms$. When the minimum delay is subtracted we have a delay of $0.03ms$ associated with the low pass filter alone.

1. *Calculating theoretical time constant*

The time constant of both of the filters is equal as they both have the same associated resistance and capacitance values. Time constant can be calculated using eq. 8.

$$\tau = RC \tag{8}$$

Substituting values gives

$$\tau = (15 \times 10^3) \cdot (20 \times 10^{-9})$$
$$\tau = 300 \times 10^{-6}s = 0.3ms$$

The difference between the measured value for the high pass filter, τ_H, and the calculated value is -20%. The difference between the measured value for the low pass filter, τ_L, and the calculated value is -10%.

Frequency Response of a Loudspeaker

The impulse and frequency response graphs for a microphone places 0.5m from a loudspeaker, away from walls and obstructions, are given in figures 8 and 9 respectively.

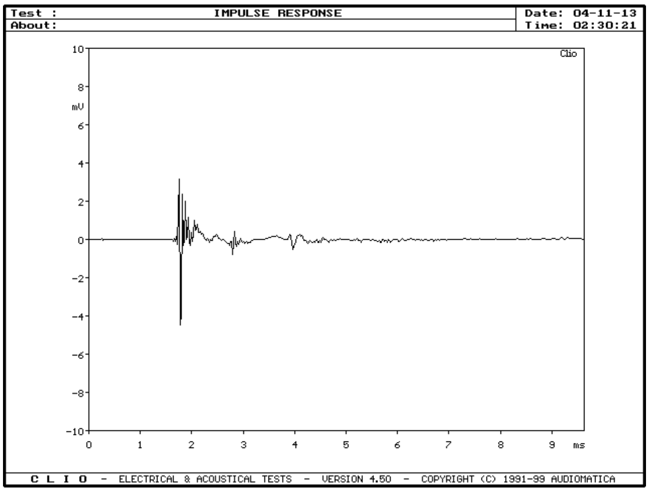

FIG. 8: Impulse response for a microphone adjacent to a loudspeaker, minimal reflection.

By reading the time until the first spike on the impulse response and the distance of the microphone from the speaker, we are able estimate the speed of sound. Taking the time for the impulse, t, the minimum time delay, t_0, the corrected time for the impulse, t_c, and the distance of the microphone from the speaker, D, the speed of sound can be determined as follows

$$t = 1.72ms$$
$$t_c = t - t_0 = 1.72 - 0.23 = 1.49ms$$
$$D = 0.591m$$
$$s = \frac{D}{t_c} = \frac{0.591}{1.49 \times 10^{-3}} = 396ms^{-1}$$

given that the accepted value for the speed of sound at sea level is $340ms^{-1}$, our estimated value has a difference of 16%.

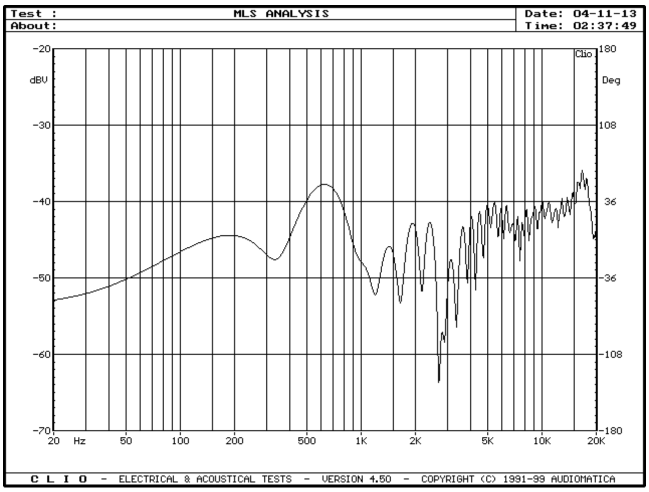

FIG. 9: Frequency response for a microphone adjacent to a loudspeaker, minimal reflection.

It is not possible to quasi-anechoically measure the lowest frequencies in a normal room as the reflection-free time in which quasi-anechoic measurements may be performed is so short. This means that for low frequency signals only a portion of the wave is captured in this window due to the time period of the wave being larger than that of the window.

The impulse and frequency response graphs for a microphone places 0.5m from a loudspeaker are given in figures 10 and 11 respectively. In this case the speaker and microphone were placed close to a wall so that a strong reflection peak could be observed

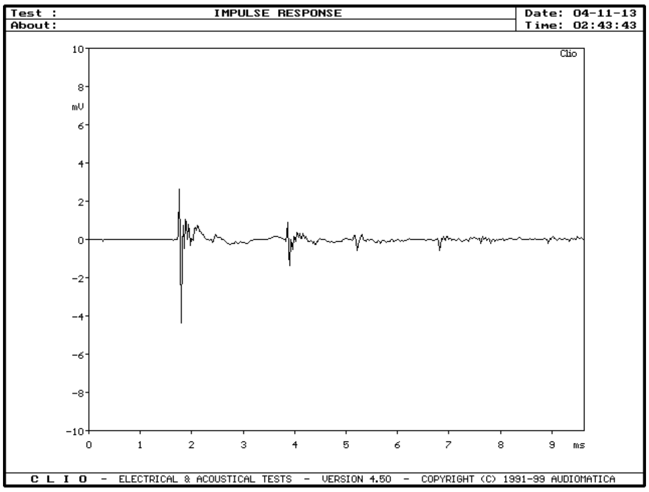

FIG. 10: Impulse response for a microphone adjacent to a loudspeaker, strong reflection

The time delay for the initial peak, t_i, and the reflected peak, t_r, as well as their distances, D_i and D_r, are

$$t_i = 1.74ms$$
$$D_i = 0.598m$$
$$t_r = 3.85ms$$
$$D_r = 1.324m$$

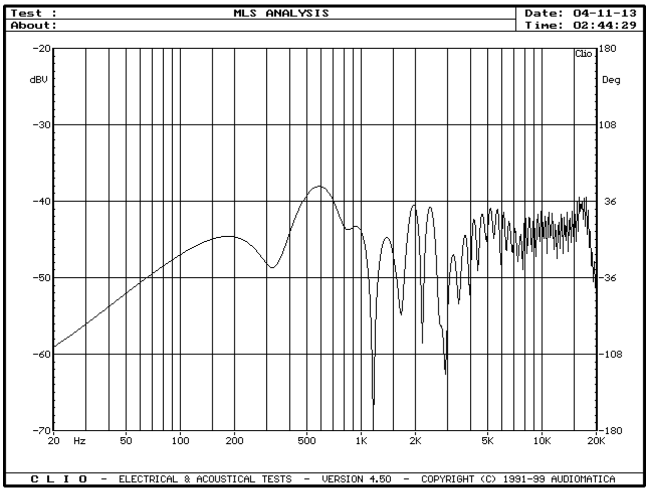

FIG. 11: Frequency response for a microphone adjacent to a loudspeaker, strong reflection

The frequency response graphs for both the minimal reflection case and the strong reflection case have been truncated to between 1 and 5ms. It can be seen from comparison between the two that in the case of the strong reflection, there is a greater attenuation at the lower frequencies. Both graphs seem to follow similar patterns although for lower and mid range frequencies it is seen that there is a greater variation between the peaks and troughs in the case of the speaker placed close to a wall. In the reflection case it is also seen that there is a very sharp negative peak at $f \approx 1250Hz$ where the response drops off completely. The case of strong reflection also shows closer spaced peaks and troughs in the high frequency region of the graph.

Reverberation of a Room

Graphs showing the sound decay for different frequencies in a room are in the 'Noise Decay Graphs' section. There are two sets of graphs, one for the case of a closed room without absorbers present, the second for the same room with the door open and a foam absorber placed behind the microphone. The RT60 values from each graph are given in Table I for the closed case, and Table II for the open case. The limit describes upper and lower decibel limit that the RT60 values was calculated from. Where no suitable limits were available this was calculated by hand by extending the first linear portion of the curve to a loss of $60dB$.

Frequency (Hz)	Limits (dB)	RT60 (s)
125	5-25	0.445
500	5-20	0.354
2000	5-20	0.236
8000	Hand	0.063

TABLE I: Decay time for closed room

Frequency (Hz)	Limits (dB)	RT60 (s)
125	5-20	0.406
500	5-20	0.265
2000	Hand	0.054
8000	Hand	0.046

TABLE II: Decay time for open absorbing room

By plotting the decay time against frequency we are able to analyse the speed of noise decay for the two different cases. A graph of time against frequency is given in figure 12.

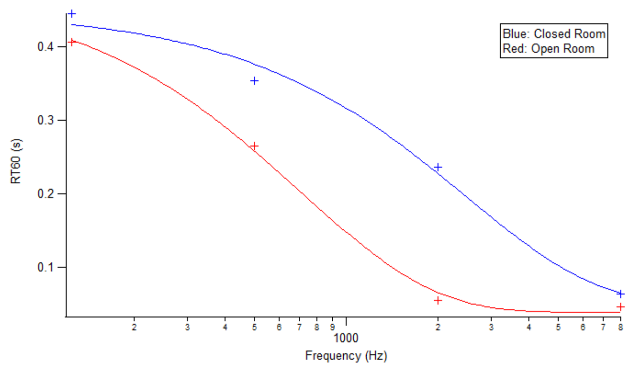

FIG. 12: Reverberation time versus frequency

It can be seen from the graph that lower frequencies of sound have a greater reverberation time than higher frequencies for the same configuration. It is also seen that for all frequencies the reverberation time is greater for the closed room than for the open room with absorbers. This is due to the increased reflection in the closed room, in the open room the absorber will stop some of the reflection and other waves will be lost through the open door.

Noise Decay Graphs

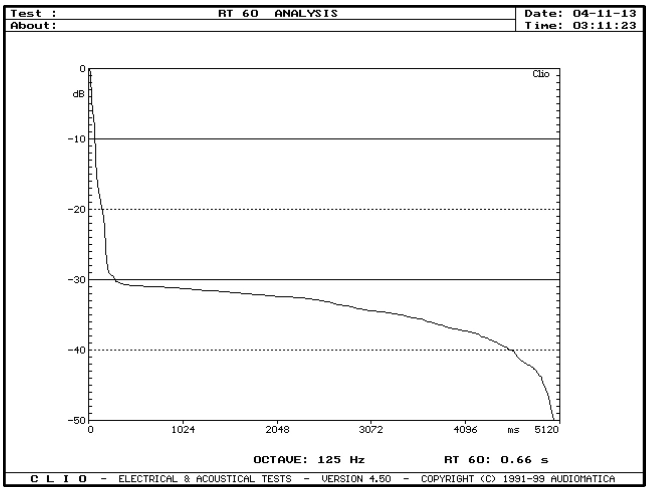

FIG. 13: Noise decay at 125Hz, door closed

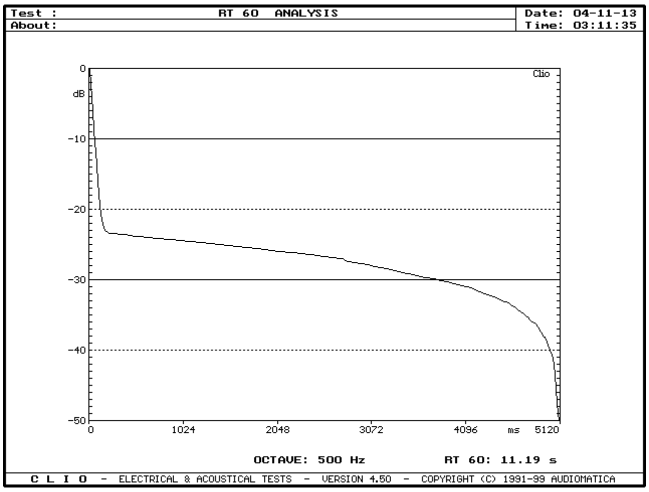

FIG. 14: Noise decay at 500Hz, door closed

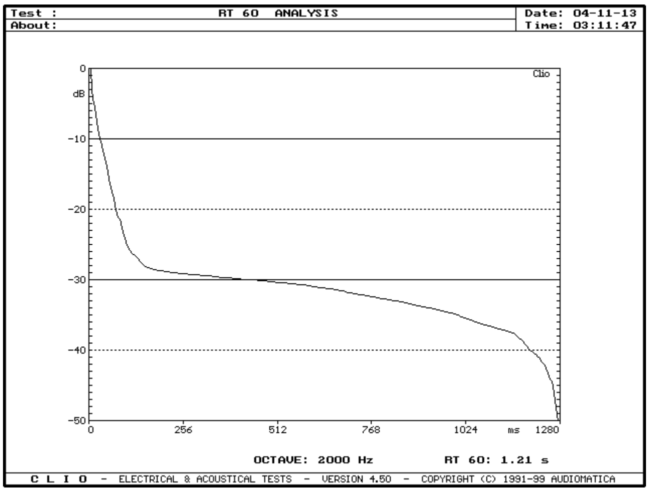

FIG. 15: Noise decay at 2000Hz, door closed

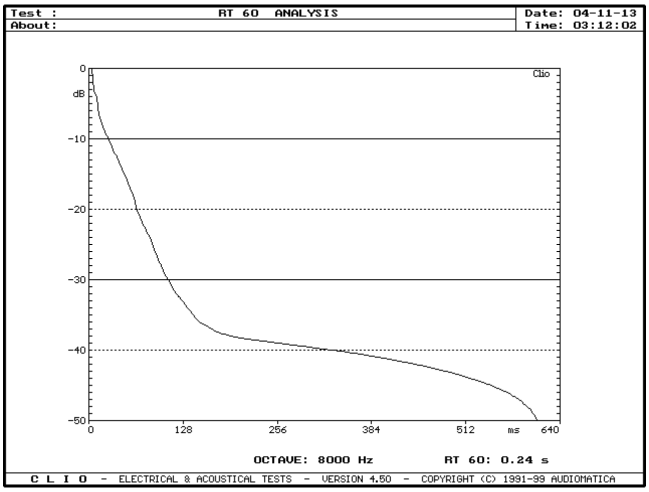

FIG. 16: Noise decay at 8000Hz, door closed

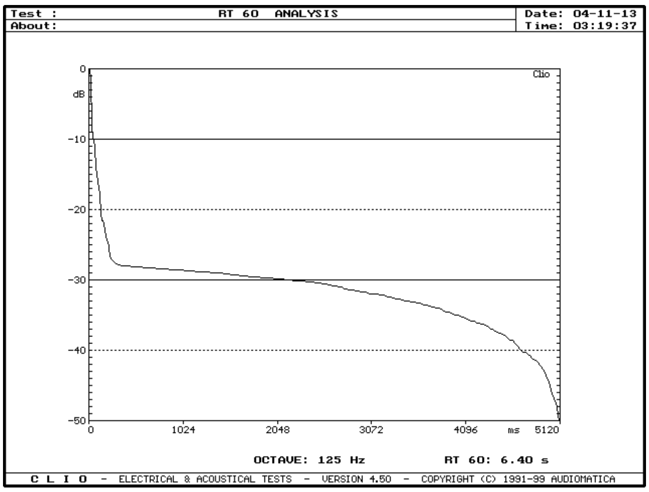

FIG. 17: Noise decay at 125Hz, door open and absorber present

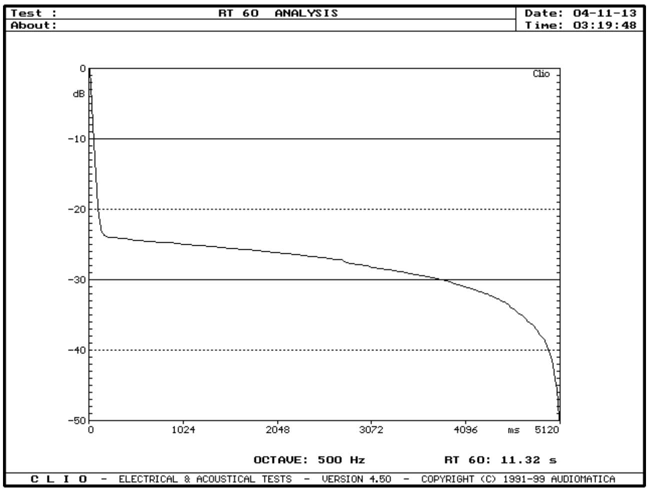

FIG. 18: Noise decay at 500Hz, door open and absorber present

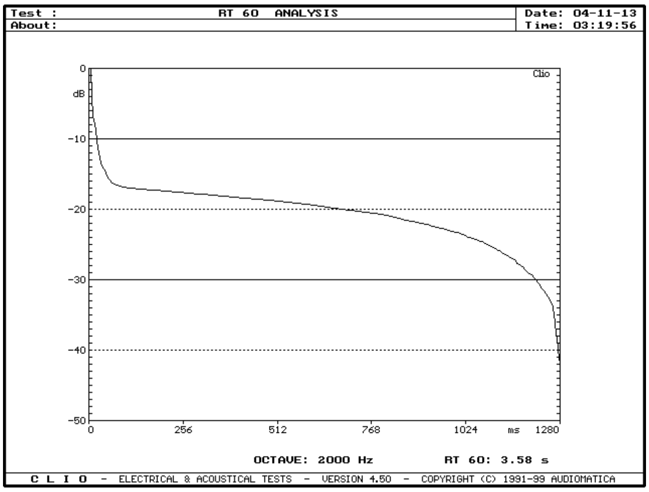

FIG. 19: Noise decay at 2000Hz, door open and absorber present

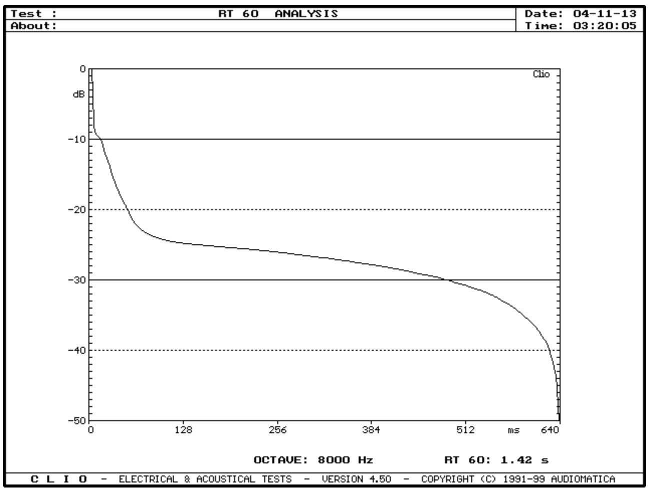

FIG. 20: Noise decay at 8000Hz, door open and absorber present

VI. CONCLUSION

We were able to analyse High and Low Pass filters and compare their operation with theory, and found a correlation with a difference of only 13 and 22% for Low and High pass filters respectively, and were able to measure the delay times for the signal response through the filter to be 0.02 for the High Pass Filter and 0.03 ms for the Low pass filter.

By measuring the time delay for an impulse from a loudspeaker to reach a microphone we determined the speed of sound to be $396ms^{-1}$, this value is 16% greater than the accepted value for the speed of sound at sea level. This could be due to more delaying factors in the microphone and the wire/circuit for measuring input in the measuring equipment which adds to the loopback delay time initially measured.

Upon measuring data for acoustic systems we found that the reverberation time of a room is dependent both on frequency and the configuration of the room. Presence of absorbing surfaces will decrease the reverberation time of all frequencies, as will having an open door to the room in which the experiment is taking place. Reverberation time is greater for sounds of a lower frequency than those of a higher frequency given the same settings.

VII. REFERENCES

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