

# Acoustic Measurements

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Section 1

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## I. ABSTRACT

## II. INTRODUCTION

## III. THEORETICAL BACKGROUND

## 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 input 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 1 and 2 respectively. The frequency response graphs have straight line extrapolations superimposed on them, the intercept of these two lines is the frequency breakpoint.

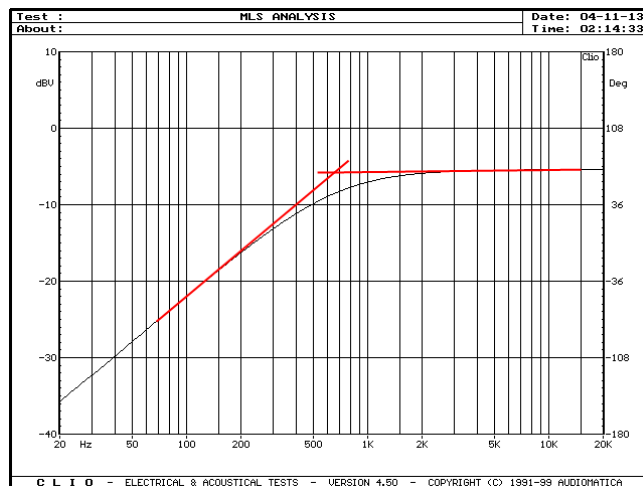


FIG. 1: Frequency response of a high pass filter

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 as follows

$$f_c = \frac{1}{2\pi RC} \quad (1)$$

Where R is the resistance of the component resistor and C is the capacitance of the component capacitor.

$$R = 15k\Omega$$

$$C = 20nF$$

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

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 2.

$$\tau = \frac{1}{2\pi f_c} \quad (2)$$

using the measured value for  $f_c$  from the graph gives

$$\tau_H = \frac{1}{2\pi f_c}$$

$$= \frac{1}{2\pi \times 650}$$

$$= 0.24ms$$

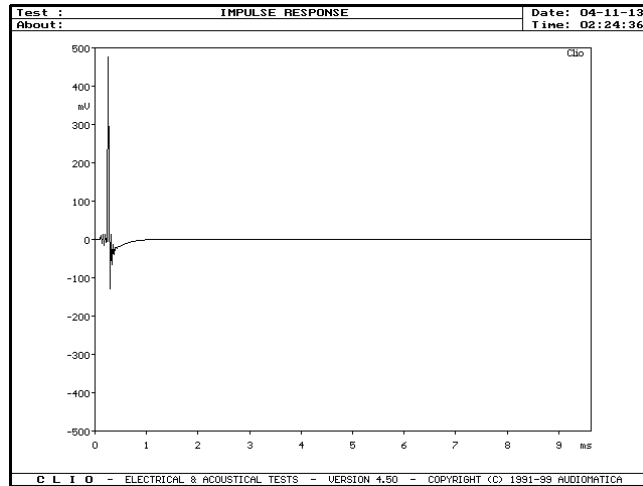


FIG. 2: Impulse response of a high pass filter

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.

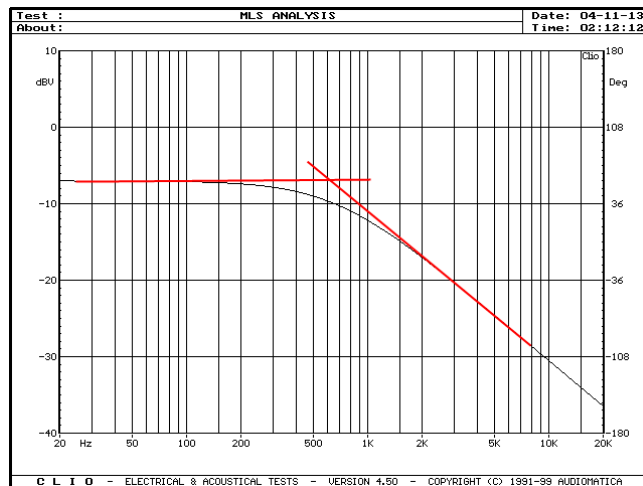


FIG. 3: Frequency response of a low pass filter

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 2 in order to calculate the time constant

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

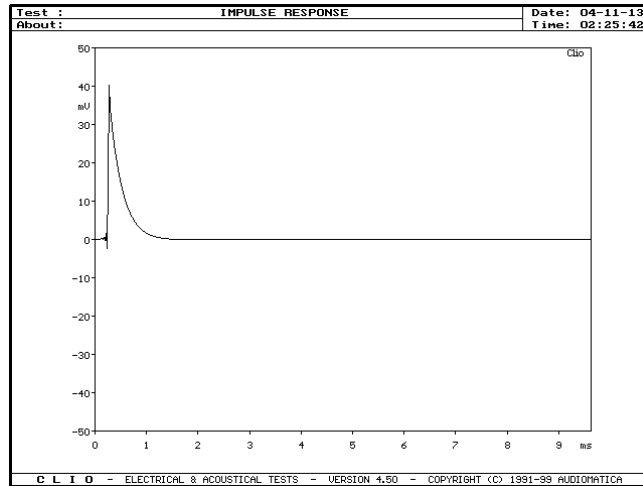


FIG. 4: Impulse response of a low pass filter

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. 3.

$$\tau = RC \quad (3)$$

Substituting values gives

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

The difference between the measured value for the high pass filter,  $\tau_H$ , and the calculated value is  $-20\%$ . The difference between the measured value for the low pass filter,  $\tau_L$ , and the calculated value is  $-10\%$ .

## Frequency Response of a Loudspeaker

The impulse and frequency response graphs for a microphone placed 0.5m from a loudspeaker, away from walls and obstructions, are given in figures 5 and 6 respectively.

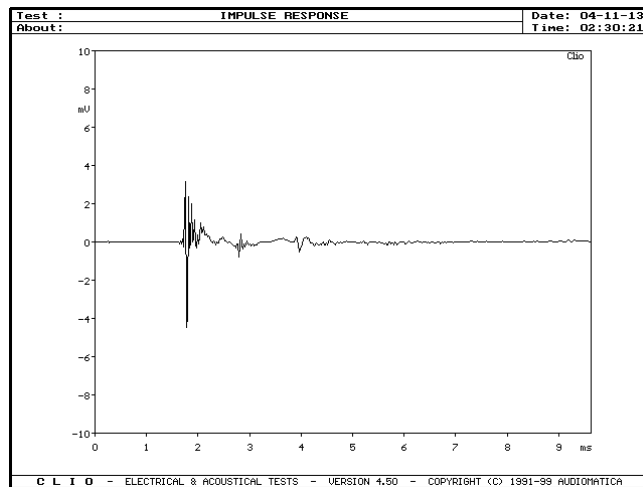


FIG. 5: Impulse response for a microphone adjacent to a loudspeaker, minimal reflection.

By reading the time until the first spike on the impulse response and the distance of the microphone from the speaker, we are able to 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

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

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

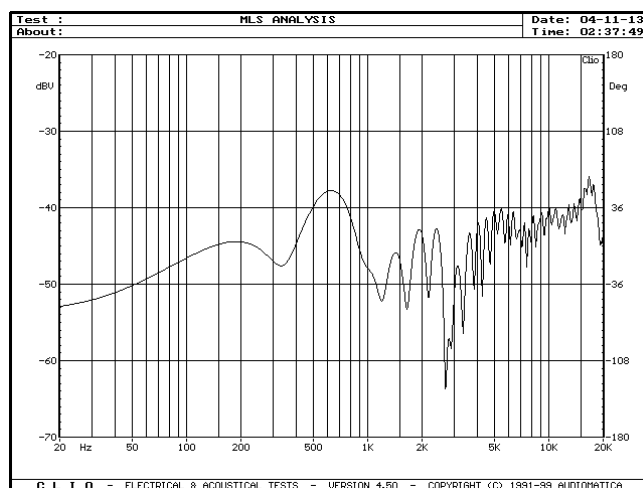


FIG. 6: Frequency response for a microphone adjacent to a loudspeaker, minimal reflection.

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 placed 0.5m from a loudspeaker are given in figures 7 and 8 respectively. In this case the speaker and microphone were placed close to a wall so that a strong reflection peak could be observed

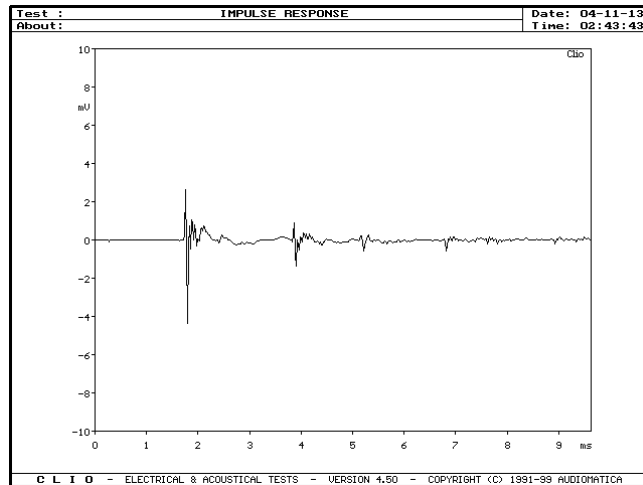


FIG. 7: Impulse response for a microphone adjacent to a loudspeaker, strong reflection

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

$$\begin{aligned} t_i &= 1.74ms \\ D_i &= 0.598m \\ t_r &= 3.85ms \\ D_r &= 1.324m \end{aligned}$$

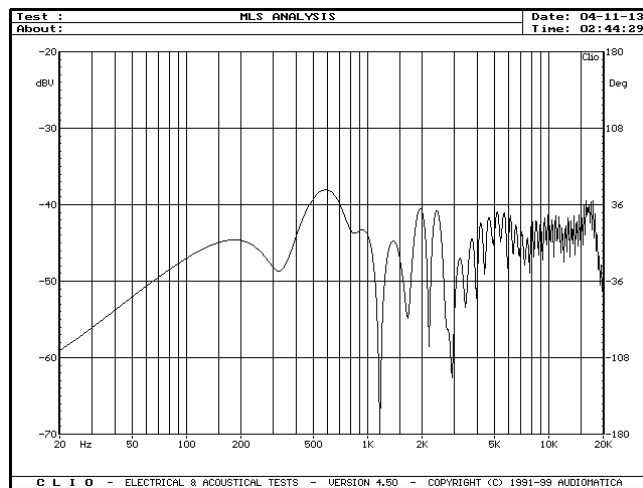


FIG. 8: Frequency response for a microphone adjacent to a loudspeaker, strong reflection

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 ??.

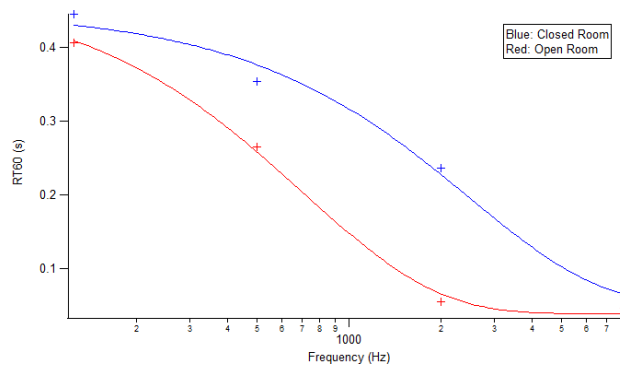


FIG. 9: Reverberation time versus frequency

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

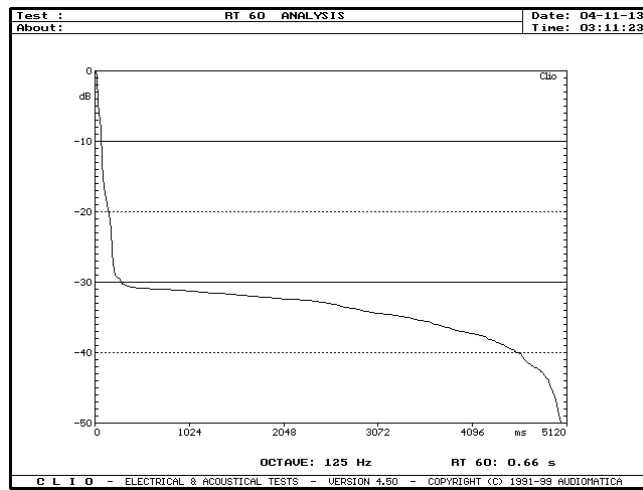


FIG. 10: Noise decay at 125Hz, door closed

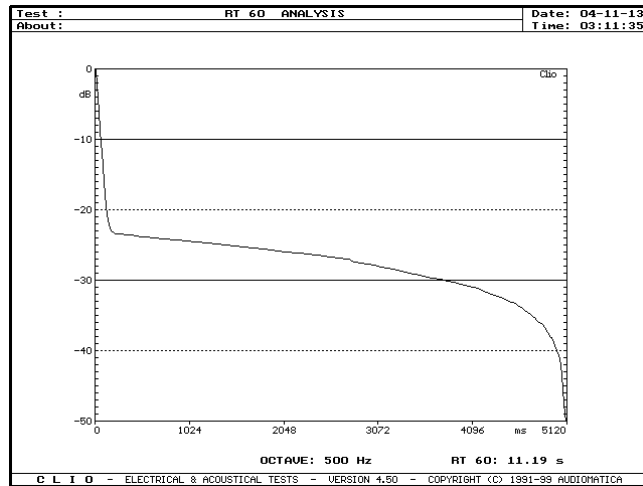


FIG. 11: Noise decay at 500Hz, door closed



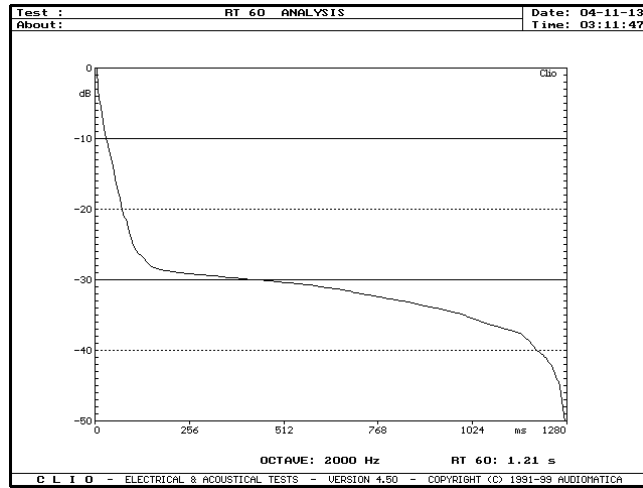


FIG. 12: Noise decay at 2000Hz, door closed

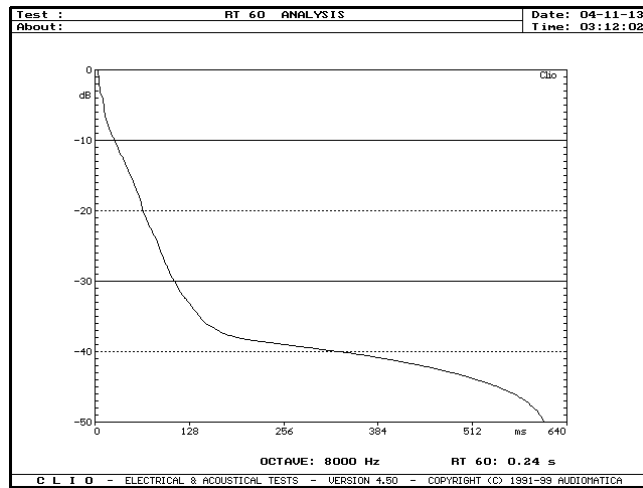


FIG. 13: Noise decay at 8000Hz, door closed

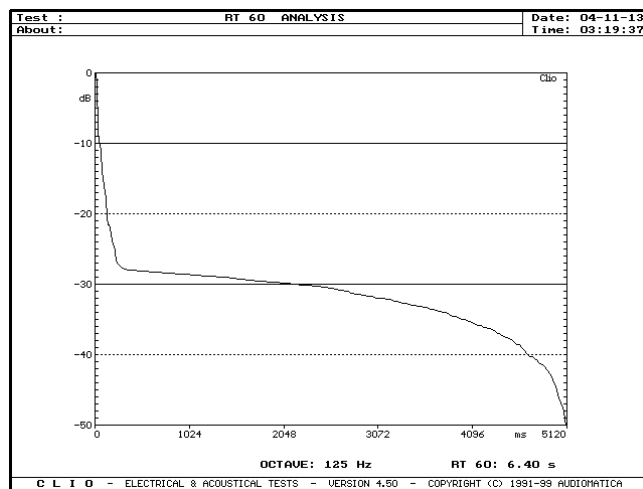


FIG. 14: Noise decay at 125Hz, door open and absorber present

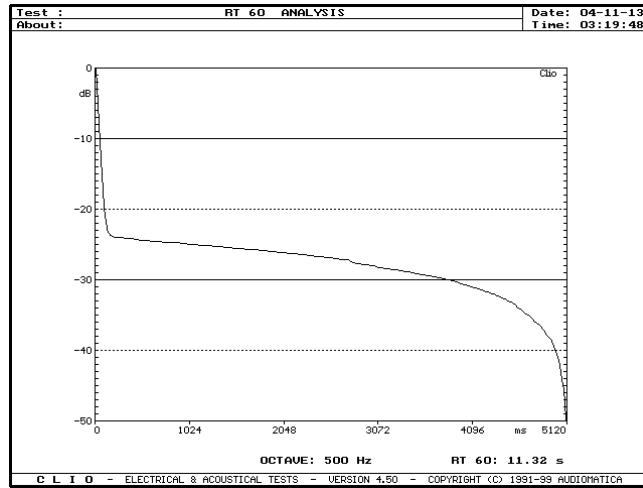


FIG. 15: Noise decay at 500Hz, door open and absorber present

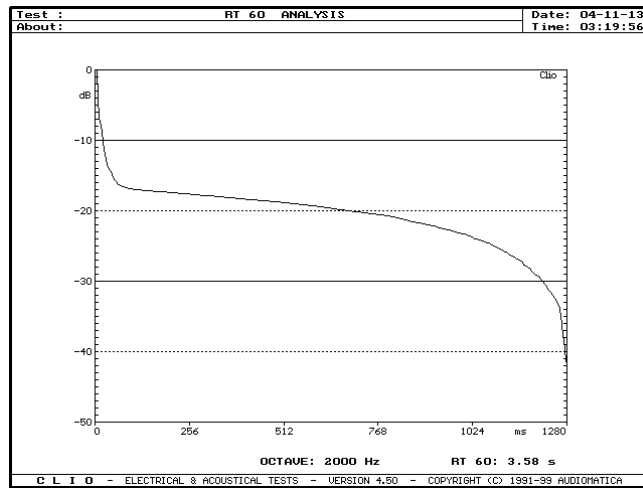


FIG. 16: Noise decay at 2000Hz, door open and absorber present

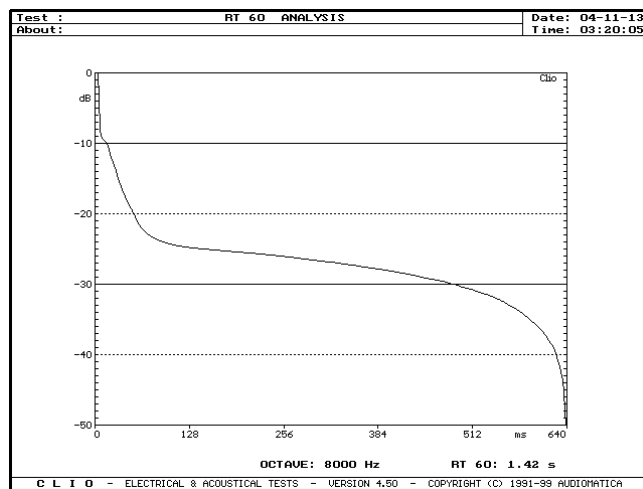


FIG. 17: Noise decay at 8000Hz, door open and absorber present

## VI. CONCLUSION

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.

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.

## VII. REFERENCES