A Critical Assessment of Sound Stimuli for Reverberation Time Measurement in Acoustic Performance Spaces

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1 INTRODUCTION

Reverberation time has been widely recognised as an important measurement of an acoustic space, this is particularly important in performance spaces where critical listening is required[3,11,12]. There are various methods to measure reverberation time in acoustic performance spaces according to recognised standards; the two main method categories are the Impulse Response and Interrupted Noise methods. This paper aims to compare and analyse reverberation time results derived from various sound source stimuli, in order to assess the suitability of all stimuli across various physical and acoustic characteristics.
Practical measurements were performed according to the guidelines set out in BS EN ISO 3382-1:2009 Measurement of room acoustic parameters: Part 1: Performance Spaces[4]. The reverberation times of three acoustic performance spaces (church, concert hall and lecture theatre) were measured, using three different types of sound stimuli (impulse, chirp and interrupted noise). The results are analysed in octave bands (125-4000Hz) and each stimulus will be critically assessed in each space and frequency band.
This paper will therefore present results, which could inform the choice of sound stimuli for reverberation time measurements based on the frequency range of interest and the characteristics of the space.

2 SOUND STIMULI TYPES

2.1 ISO Guidelines

The types of sound stimuli for reverberation time measurement are outlined in the official standards BS EN ISO 3382-1[4] where they are referred to as ‘Integrated Impulse Response’ and ‘Interrupted Noise’ methods. Both categories of stimuli are detailed in the standards, but with minimal associated guidelines and they are somewhat ambiguous in parts.
BS EN ISO 3382-1 does not state the type or size of the space to be measured in order to sufficiently excite, only stating that the initial sound source needs to be 35dB above the background noise of the space in a given frequency band[3]. The amount of energy to excite a church compared to a small recording studio would be significantly different and the size of a space does not dictate the background noise level.
The frequency characteristics of the stimuli sounds are also of interest due to the wide range of potential sources. Each stimulus source will have its own frequency characteristics, which will excite certain frequency bands more than others. For a single band frequency measurement, certain narrow band sources may suffice; however an important consideration of the stimulus sound source is that it has a broad enough band to cover the frequency range of interest[11].

2.2 Impulse Response

The impulse response of an acoustic system is its response to a Dirac Delta Function. The acoustic system can be a room, cavity, plate, beam or audio device[6]. The output of the system is measured in response to the input of the Dirac pulse. This is the theory of an impulse response; however in practice a Dirac pulse is impossible to recreate in acoustic situations, due to it requiring a pulse that has an infinitely high-energy spike at x=0 and is zero at all other points[2]. Dirac pulses may be used from loud speakers; however their short duration and finite amplitude will result in very small amounts of energy produced and will be dependent on the signal to noise ratio of the equipment used. The response of a speaker could not possibly be fast enough to respond to an infinitely narrow pulse also. In an acoustic test chamber, this may be possible; however in a concert hall for example where background noise is present, it would not be possible if accurate results were desired[1]. Some sound stimuli are used to try and closely replicate the ideal Dirac pulse; these include balloons, paper bags, starter pistols and other equipment that would produce high amplitude sound pulses. The issues with such sources are that the frequency characteristics are generally not uniformed and the source output is not repeatable, i.e. balloons cannot easily be inflated to a single identical size. The sound stimulus used in the integrated impulse response method will ideally be impulsive, broadband and low directivity and that the source produces enough sound energy in the frequency ranges in question. Starter pistols, balloons and paper bags have all been used, but with such a range of impulse sources available, each with different frequency characteristics, more specific guidelines are possibly required[10]. Despite the issues with such impulse sources, there is a distinct benefit, which is the portability of the equipment required, since no large loudspeaker or amplifier is required unlike alternative methods. This is why the method is still widely used; however the limitations and issues highlighted should still be considered.

Sine sweep integrated impulse response methods are not as portable and convenient as the aforementioned sources due to the requirement of a loudspeaker; however they can be identically repeated and provide a full range of frequencies if required. Short sine chirps or longer sine sweeps can be used and these provide an improved signal to noise ratio to that of the more traditional methods. Also due to there being no random fluctuations in this method, it therefore reduces the need for multiple measurements for averaging[6]. A disadvantage of sine sweep techniques is that specialist analysis software and methods are usually required, but these are generally accessible at all levels.

2.3 Interrupted Noise

The interrupted noise method uses a steady state level of random broadband noise, which is stopped once steady state is achieved and the subsequent sound pressure level decay is measured. The noise source most commonly used in room acoustic tests is pink noise, due to its presence of low frequency energy, as opposed to the lack of low frequency energy of a white noise signal. This method however lacks a smooth decay curve due to the random nature of the excitation and when below the background noise level, the curve is no longer representative of the actual decay. Due to the random nature of excitation, this method also requires multiple measurements of a space, due mainly to the effect of room modes resulting in inconsistent measurements[6]. It has been found that using the same microphone and loudspeaker positions repeatedly to measure the decay curve will result in random phases and amplitudes of room modes each time the excitation signal is stopped. The difference between the decaying modes of each measurement will result in varying decay curves to measure. Due to this phenomenon, a single decay curve measured using the interrupted noise method is not useful[6].

2.4 Reverberation Time Measurement and Room Characteristics

Practical measurements of reverberation time will somewhat be effected by room modes, where certain points in a room will have increased or decreased pressure at certain frequencies; thus being detected by the microphone if in these positions. It has been found that if more than three standing waves are excited, the fluctuations will average out and the intensity will diminish uniformly; however when only two or three room modes are excited, the intensity will fluctuate in diminishing and it will depend largely on the microphone position and the frequency content of the sound source[8]. The effect of room modes on reverberation time measurements is also frequency dependent, where higher frequencies are found to be more predictable than low frequencies. This is due to the decay of the sound at low frequencies being from only a few modes; thus resulting in more accurate higher frequency reverberation times[5]. The averaging of reverberation time results from multiple source and receiver positions is important to consider the effects of room modes in a space, especially if the space is thought to be small enough for the room mode distribution to be less uniformed and dense. Another consideration is if there are parallel surfaces in the room, as this will reinforce axial modes and therefore will contribute to the modal effect on the reverberation time measurements[9]. These characteristics of small rooms can increase the decay time of modal frequencies compared to non-modal frequencies and results in individual modal frequencies being represented, as opposed to the whole room in terms of decay time[5]. Practical studies have shown that reverberation time is less accurate in a small room at low frequencies. A particular study[7] measured various acoustic spaces, from irregular small shaped rooms to large diffuse rooms, using various stimuli to measure the reverberation time. The study found that in large diffuse rooms, there are no emphasized room modes in the frequency of interest; however in smaller more absorbent rooms, the results were less consistent at low frequencies between different stimuli and that more attention should be paid to positioning and sound source used[7].

In summary, small rectangular spaces are more likely to exhibit stronger axial room modes than larger and irregular shaped spaces and thus reverberation time measurements in smaller spaces can be effected by prominent room modes. BS ISO-3382-1:2009[4] does acknowledge the potential issues raised with measuring reverberation time at low frequencies, but does not provide guidance on the size of a space to be measured.

3 SPACES MEASURED

Three performance spaces were chosen for reverberation time measurements, with the aim for them to provide a variety of size, absorption, purpose and acoustic characteristics as per Table 3.1. Each space was measured using three different sound stimuli sources:

- Balloon
- Chirp
- Interrupted Noise

Location	Comments
The Royal Concert Hall, Nottingham,	• Seats 2315 + 186(choir) • Volume 17,510m^3 • RT 1.75 seconds (500/100Hz) • Floor – studded rubber on concrete, Walls – plaster on solid masonry, Ceiling – 100 mm sprayed concrete, plastered. • Balcony soffits of fibrous plaster • Stage – timber boarding. • Seating – upholstered tip-up with unperforated bases[3].
All Saints Church, Nottingham,	• Mainly stone and concrete, with some carpeting • 300 wooden seats across the width of the church and covering the main floor area • High arches, spire 175ft • Active tramline approx. 200m from the building
Lecture Theatre, Derby University, Derby	• 150, thick, soft furnished, raked seats • Sloping carpeted floor • Rectangular shape • Slatted wooden wall coverings • Absorption significant to be considered compared to the size of the space.

Table 3.1

4 METHODOLOGY

The measurements were carried out in the performance spaces stated, using the guidelines set out in BS-EN-ISO 3382-1:2009: Measurement of acoustic parameters. Part 1: Performance Spaces[4].

Summary of the guidelines followed[4]:
The T_{20} measurement is a preference that has been given to the 20 dB evaluation range, for the following reasons:

a) the subjective evaluation of reverberation is related to the early part of the decay;
b) for the estimation of the steady-state sound level in a room from its reverberation time, it is appropriate to use the early part of the decay: and
c) the signal-to-noise ratio is often a problem in field measurements, and it is often difficult or impossible to get a evaluation range of more than 20 dB. This requires a signal-to-noise level of at least 35 dB.

Source Positions

- Positioned where the natural sound sources in the room would typically be located.
- Minimum of two source positions shall be used.
- The height of the acoustic centre of the source shall be 1.5m above the floor.

Measurement Positions

- Microphone positions a minimum of 2m apart
- Microphone at least quarter of a wavelength i.e. normally 1m from a reflecting surface including floor
- Microphone positions should be representative to where the listeners would normally be located i.e. audience seats. Microphone height should be 1.2m above the floor, corresponding to the ear height of average listeners in typical chairs.
- Microphone positions to cover the seating area evenly and the results of the measurements may be averaged and the frequency range should cover 250-2000Hz in octave bands

All measured spaces followed an overall procedure of:

- Two sound source positions used in areas of the space where sound is usually generated i.e. stage, lectern, and choir area.
- Three identical microphone positions per space, source position and stimulus type.
- The microphone was placed left, centre and right of the typical sound stage of the space, in the area where the audience would usually be seated for a performance.

4.1 Equipment

The following equipment was used to measure the reverberation time of each space:

- NTi XL2 Sound Level Meter, SNo. A2A-03293-D1, FW2.32
  - Mic Type: NTi Audio M4260, S/N: 1774, User calibrated 2012-03-04
- SLM Measurement Setup
  - T_{20} ,1/3 Octave (results presented 1/1 octave)
  - Range: 30 - 130 dB Sound Calibrator 1251 (Cert No. 8185, Cal Date: 24/01/11, Cal Due Date: 01/12)
- NTi Minirator Pro audio generator
- Dodecahedron loudspeaker (omni-directional sound source)
- Balloons and pins

5 RESULTS AND ANALYSIS

5.1 Reverberation Time Measurement Results

In order to compare and analyse the stimuli used, Figures 5.1, 5.11 5.12 demonstrate the overall averaged T_{20} from all source and microphone positions used for each of the sound stimuli used.

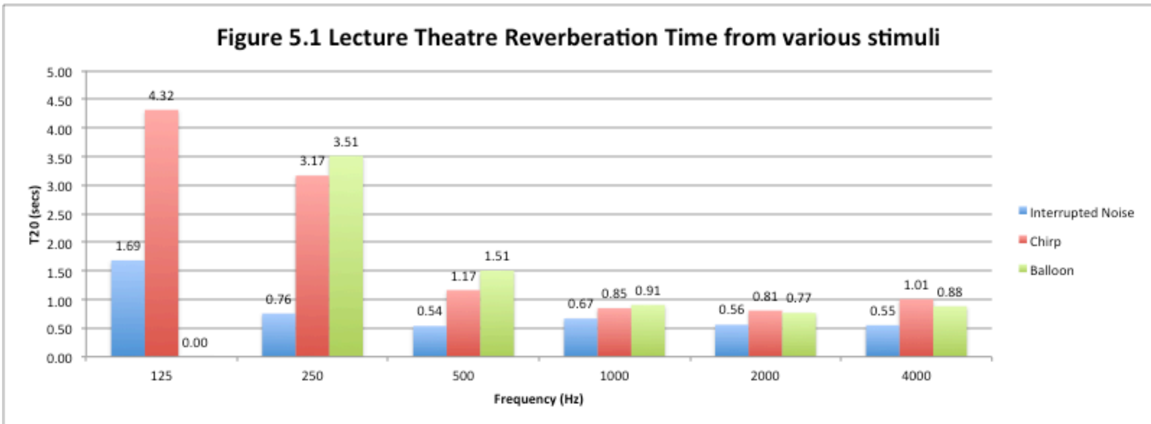

Figure 5.1 Lecture Theatre reverberation time measurements from various stimuli.

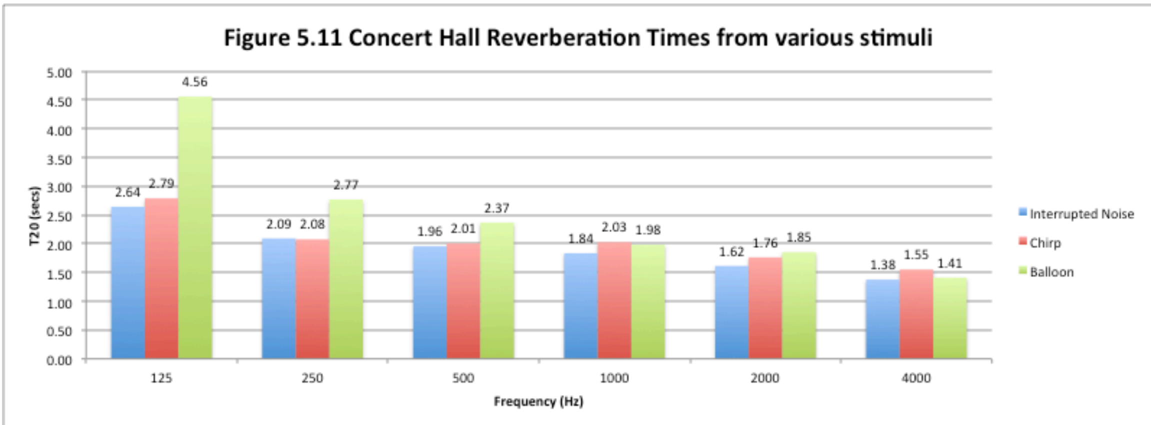

Figure 5.11 Concert Hall reverberation time measurements from various stimuli.

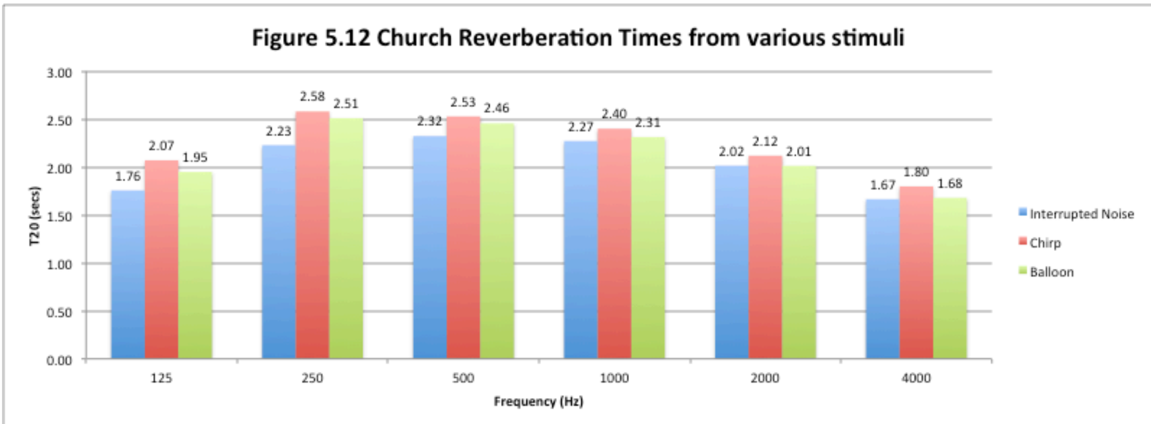

Figure 5.12 Church reverberation time measurements from various stimuli.

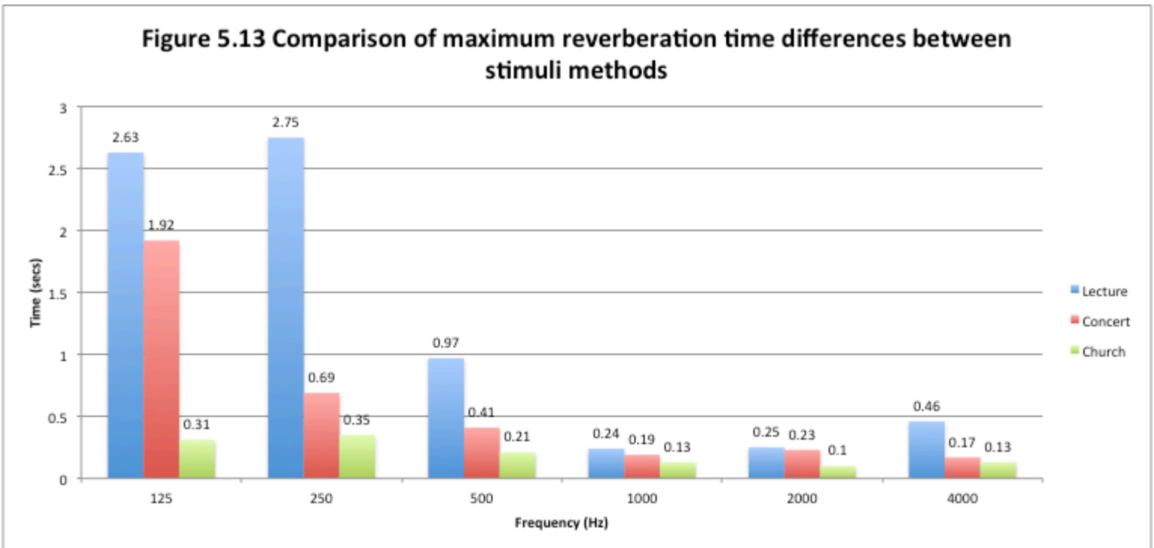

Figure 5.13 Comparison of maximum reverberation time differences of the stimuli methods used

5.2 Analysis of the Reverberation Time Measurements

Lecture Theatre

Figure 5.1 demonstrates the average T_{20} of the lecture theatre space measured. The low frequencies (125-500Hz) exhibit large differences between the measurements, this is most apparent between the Interrupted Noise when compared to the Chirp and Balloon methods. The balloon impulse response method resulted in no measurements at 125Hz and the Chirp method resulted in an implausible 4.32 average T_{20} measurement at 125Hz. At 250Hz the balloon and chirp methods produce higher than predicted T_{20} results. As the frequency band in question increases, there are generally less differences between the different sound stimuli methods used for measurement.

Concert Hall

Figure 5.11 demonstrates the average T_{20} of the concert hall space measured.
As with the Lecture Theatre, the low frequency range (125-500Hz) exhibits the largest range of results when comparing stimuli. The 125Hz band for the balloon stimulus has a 4.56 second T_{20} average, compared to the 2.64 and 2.79 for the interrupted noise and chirp respectively, this is a large difference and the balloon continues to exhibit higher average T_{20} times across 250-500Hz bands also. The 4.56-second measurement for the balloon is implausible if compared to an official reverberation time of the same Concert Hall[3], where the time is stated as 2.75 seconds, which approximately correlates with the interrupted noise and chirp method results.
The mid to high frequency bands (1-4kHz) demonstrate more parity of T_{20} averages for each method and the balloon is the closest method at 2000Hz to the results in Figure 5.14 and approximates to 1000Hz results also.

Church

Figure 5.12 demonstrates the average T_{20} of the church space measured. Compared to the other two spaces, the church did not exhibit the same extent of anomalies with low frequency reverberation time measurement. The results are a lot more consistent across the different methods used and there are no greatly implausible results evident. The balloon demonstrated implausible results in the concert hall and lecture theatre; however in the church the result evidenced some cohesion to the other two methods used. The chirp method produced consistently higher average T_{20} results against the other two methods as opposed to the balloon method in the other spaces measured.

Discussion of the average T_{20} results

Figure 5.13 demonstrates the maximum difference of average T_{20} measurements, between all stimuli methods, per frequency band and space measured. As previously stated, reverberation time measurements are more accurate at high frequencies; this supports the measurement results presented in Figure 5.13.

The lecture theatre exhibited the largest differences between results from the methods used. For each frequency band the differences were greatest in the lecture theatre, with as much as 2.75 seconds difference between the balloon and the interrupted noise method. It is thought that errors were the cause of large differences between methods used, in particular the frequency characteristic of each stimulus. It is thought that the balloon does not produce enough low frequency energy[10] and in the lecture theatre, which is a relatively small and absorptive room compared to the concert hall and church measured. Low frequency energy could be absorbed to inadequate levels if the low frequency energy is already lacking in the sound source. The interrupted noise method produces continued low frequency energy with the only limit being the response of the loudspeaker and microphone. The interrupted noise method data demonstrates more consistent results at low frequencies, compared to the other methods across all spaces measured. With the same limitations of the interrupted noise method, the chirp method does produce low frequency energy; however each frequency is represented at a single point in time during the sweep, this potentially would not excite the low frequency response of a small absorptive room enough to provide consistent and plausible results.

Examining the interrupted noise method in the 1000-4000Hz bands, the T_{20} differences in the lecture theatre were significantly reduced, with a maximum of 0.46 seconds difference at 4000Hz. 0.46 seconds is thought of as a significant difference in results and could cause errors in design and prediction if used. The church T_{20} measurements consistently produced the least differences between each method used. The low frequency bands of 125Hz and 250Hz produced the greatest differences in the church, 0.31 and 0.35 seconds respectively. 2000Hz measurements in the church presented a 0.1 second maximum difference between methods, with 1000Hz and 4000Hz bands resulting in a 0.13 second maximum difference; this is thought to be more acceptable.

The larger spaces of the concert hall and specifically the church produced the most consistent results across all methods. It is thought that the low frequency energy present in each stimulus is supported in the larger spaces, due to less absorption and more time to develop within each space. The church in particular has minimal absorption relative to the vast stoned surface space; this resulted in the low frequency measurements to be more consistent across all stimuli than the other spaces measured.

As discussed previously, room modes can also effect reverberation time measurements, specifically in smaller rectangular rooms[5,8]. The lecture theatre displays such characteristics; therefore the high differences for readings in the lower frequencies could be caused by reinforced room modes between the parallel surfaces. Room modes are not thought to be a problem within significantly large and irregular shaped spaces, such as the concert hall and church spaces measured.

6 CONCLUSION

The larger differences between stimuli were dominant in the low frequencies (125-500Hz) across all rooms measured, with the smaller, more absorbent room resulting in the most errors and differences across the types of stimuli used.

These results are in line with other studies, which also indicate that reverberation time is more difficult to measure at low frequencies and in smaller rooms, due to strong low frequency and low-density modal response in small rooms increasing the decay time at certain modal frequencies.

The interrupted noise method resulted displayed the least amount of errors and differences between measurements for two sources and across the frequencies measured. This method performed most consistently across all spaces and positions used, but the method did still display on average more differences between measurements at lower frequencies.

The impulse response method using the balloon demonstrated the most errors and inconsistencies from the stimuli used, again this was more evident at the lower frequencies; however the balloon was the least consistent in measurements across all stimuli used. It is thought that the balloon does

not generate enough low frequency energy to excite a room at these frequencies; thus resulting in errors and inconsistent results.
It is concluded that the choice of stimulus for low frequency measurements of reverberation time is important for accuracy and consistency of results. Above the 500Hz octave band has been demonstrated as less important in terms of stimuli used, with more consistency above this band across all stimuli used. It is recommended that the interrupted noise method be used, if low frequency reverberation time measurements are required, particularly in small rooms.

7 FUTURE WORK AND IMPROVEMENTS

To follow on from the results and conclusions of this paper, it is recommended that more measurements be performed using other stimuli, to gather more averages in order for more accurate analysis of the findings. One method not used in this paper due to the limitation of equipment available and the consistency of equipment for all measurements, was the reverse time integrated sweep response method. It is recommended that reverse time integration is tested and compared with the methods used in this paper. More measurement and source positions may also be considered for future work, allowing greater averaging of results and potentially increasing the accuracy. A wider range of acoustic spaces should be tested to provide further indication of possible correlation between acoustic characteristics and reverberation time accuracy.
Computational and formula based prediction of the spaces could be carried out to determine a predicted reverberation time in order to compare against actual practical measurements, this would allow greater determination of implausible results. Without any indication of actual reverberation time, it is difficult to determine the accuracy of the results; however no method would be able to precisely determine the actual reverberation time due to limitations in all practical and prediction methods. A modal frequency response measurement may also be considered for each space, this would allow for anomalies in low frequency reverberation time to be explained or used to inform any corrections at certain frequencies.

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