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Can We Save the Cost of Building Anechoic Chambers?

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Abstract

This work looks at using coherent averaging to measure transducer responses with high precision without using an anechoic chamber. The use of coherent averaging in building acoustics is familiar to those who use deterministic signals (e.g.. MLS) as the basis for their measurements. The primary concern is to achieve a sufficient signal to noise ratio so that results relate to the system being measured rather than unrelated noises. This technique requires conditions to be unchanging with time but coherent averaging can be used to advantage in situations where conditions are purposefully rendered time-varying. It is possible to select or reject contributions to a measurement by choosing some transmission paths to be time invariant and making others – ones we wish to remove from the measurement - varying. In previous work we have shown that by rotating a loudspeaker-microphone couple in a highly reverberant room we can measure the loudspeaker (or microphone) transfer function with the same accuracy achievable in an anechoic chamber by using this technique to suppress the reverberation in the room. In this more recent work we consider whether any particular deterministic signal – e.g. A log 'chirp' – is more advantageous for this application and whether the availability of the new soundfield-type microphones offers an improved way for making such measurements.

INTRODUCTION

Addressing the theme of the conference from the point of view of the Acoustics Testing Service (ATS) of the University of Auckland we consider that one of the largest risks to our continued existence (i.e. our sustainability) – and presumably that of similar testing services in other parts of the world - is the cost of our facilities. In the case of the ATS the maintenance and rental charges for our suite of reverberation and anechoic chambers comprise the largest items of the annual budget. The reverberation chambers are in regular demand for insulation, sound power and absorption measurements whereas in comparison the large anechoic room is little used commercially.

Building on work we have reported earlier [1] we have been exploring alternative techniques for obtaining the information traditionally measured in anechoic chambers. If these are satisfactory they will obviate the expense and resources required for building and maintaining these major items of acoustical equipment.

USES FOR ANECHOIC CHAMBERS

International and national standards show the uses which involve anechoic environments [e.g. 2, 3] for formal measurements. These are –

1. Transducer frequency responses and calibration
2. Transducer directionality
3. Sound power determinations.

In addition they are often used for:

1. Low noise emission measurements
2. Noise source identification in complex machinery or engines
3. Subjective experiments where highly controlled sound fields are required.

Finally, in our experience, the anechoic chamber can be a valuable teaching tool for students in acoustics and related courses. If we were to dispense with anechoic chambers alternatives for these uses would be desirable. Preferably these alternatives would only require other readily available spaces or environments.

MEANING OF ANECHOIC.

The word anechoic is quite clearly based on the word 'echo' to which the prefix 'an' meaning 'without' has been added. 'Echo' seems to have been adopted into the English language as early as the 15th century and originated from the Greek myth of the nymph (i.e. a minor divinity who is eventually mortal) Echo. Classical scholars are divided about whether Echo was named because she ultimately existed only through the sounds she could make (Note: echo derives from ekhe = sound or noise) or because – in an alternative version of the myth – she was fated only to be able to speak by repeating what was said to her

So 'anechoic' could be argued to mean either 'without any sound' or 'without audible repetition of sounds'.

Lay understanding of echo is an audible repeat of a sound separated in time from the original sound and therefore might support the latter meaning. However, typical anechoic chambers can be seen to address both meanings in that

1. Their reflection suppressing lining removes all audible repetitions, and
2. Their highly insulating wall construction removes all audible external sound.

ALTERNATIVES TO ANECHOIC CHAMBER MEASUREMENTS

1. Sound Power measurement in a reverberation chamber [4] or using an intensity probe [5] present more attractive alternatives to the time consuming series of measures over an enclosing surface as required for an anechoic determination of sound power.
2. Low noise measurements do not, in principle, require an anechoic chamber and any well-insulated space can substitute. Similarly, noise source identification can be carried out in non-anechoic environments by intensity measurement or using an acoustic camera [for example 6]
3. Sound fields for subjective experiments are arguably best controlled in an anechoic chamber but more and more we are realising that other factors (especially vision) strongly mediate reactions to sounds [7] so that results obtained in an anechoic chamber can be strongly biased or even artefactual. Thus more realistic environments for subjective assessments are to be preferred.

The remaining major use for anechoic chambers – that of transducer measurement and calibration – is what our present work addresses. The aim is to demonstrate that an alternative is viable which can be undertaken almost anywhere including highly reverberant rooms.

Whilst this might indicate that anechoic chambers are nor needed for objective sound measurements it must be admitted that continuing access to an anechoic chamber for students to experience this environment, the behaviour of sounds, and the attendant subjective effects provides a teaching tool of almost inestimable value!

TRANSDUCER RESPONSE MEASUREMENT

We propose that coherent averaging can be used to remove the reflected sound in an ordinary room by purposefully introducing time-variance in the room reverberant field that we want to discriminate against. The technique we propose leaves the direct sound unaffected by the variation.

Method

The traditional methods for measuring responses of microphones and loudspeakers require either an anechoic chamber for a quasi-steady state measurement - or windowing of a transient measurement (obtained by convolution of the results of a deterministic signal measurement e.g. MLS or chirp with the original source signal).

Since anechoic chambers are expensive and relatively rare, much use is made of the latter technique and measurements are therefore made in normal, reasonably reverberant rooms. Trace (II) in Figure 3(a) is a typical example of the impulse response of a loudspeaker obtained in proximity to a reflecting surface. The first and largest spike represents the direct sound from the loudspeaker. To obtain the loudspeaker response free from the distorting effects of room reflections we must window out this direct sound. The arrival time of the first reflection puts a limit to the width of the time window we can use. If this is smaller than the length of the impulse response of the loudspeaker we are not able to measure the response correctly, and the frequency resolution for our analysis of the response is limited.

These limitations may in principle be avoided by making measurements in an ordinary room but employing a coherent averaging technique where the repeated measurements have been made in such a way that the direct sound between source and transducer remains the same but the reflected sound is changed so as to be uncorrelated each time. For example in the case of the single reflecting surface we can use the method illustrated in Figure 1.

The loudspeaker and microphone are mounted on a turntable so that they are fixed relative to each other. The turntable is rotated whilst coherent averaging of MLS periods is carried out. The rotation varies the travel time of the unwanted reflection whilst keeping the direct path time-invariant.

The variation happening during a single MLS period has the

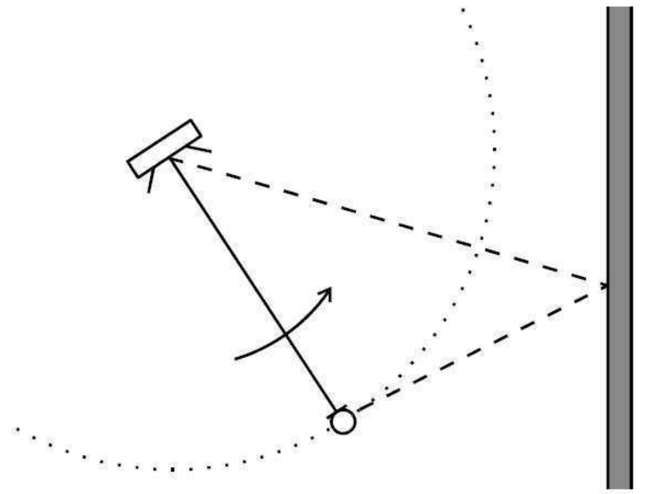

Figure 1. Measurement set-up with rotating transducers for reflection suppression by averaging

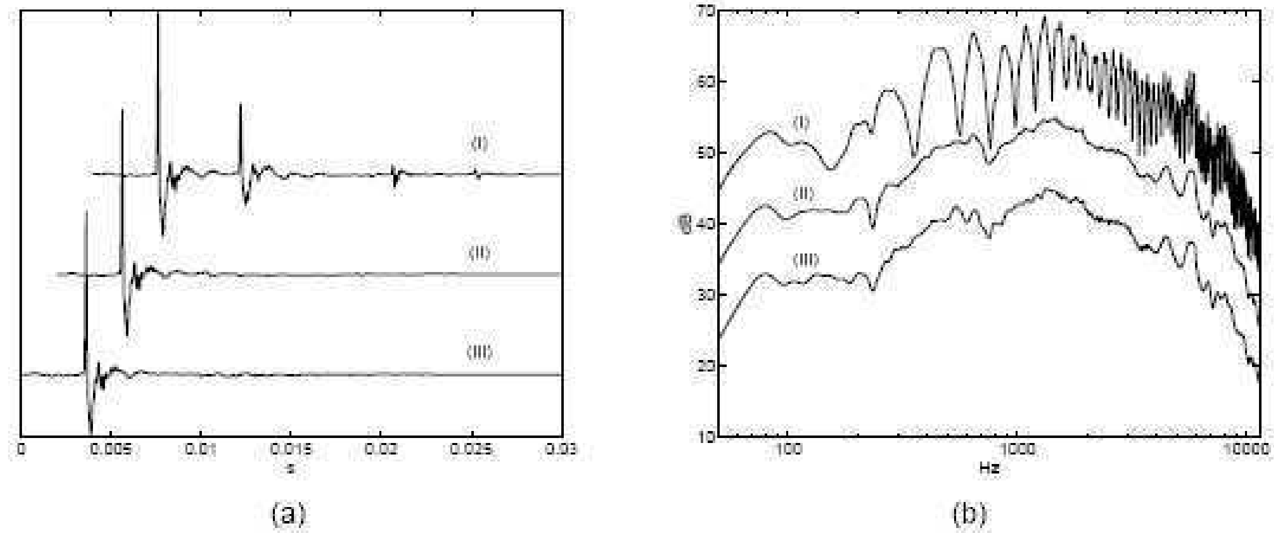

Figure 2. The responses of a loudspeaker found from MLS responses: (I) near a reflecting surface, (II) near a reflecting surface but with the loudspeaker and microphone rotating, and (III) in an anechoic chamber.

effect of transforming some of the reflected energy into a time-spread, frequency dependent noise-like component, which can be reduced by coherent averaging [8]. In addition, the responses from different periods will be different, and the reflected energy will behave more or less incoherently during averaging.

Therefore, when measuring in an ordinary room, if the plane of the rotation is skewed with respect to all significantly planar surfaces of the room, the contribution of the room reflections is made incoherent and a high direct-to-reverberant energy ratio can be built up by averaging a suitable number of periods of the signal.

Figure 2(a) and (b) shows how the true response of a loudspeaker is successfully extracted using this technique, the averaged response was found using 14 MLS periods of 5.7s each, during two full revolutions of the turntable. In principle this will also work with a single period if the length of the sequence is sufficiently long. The main requirement is that the path length variation of the major reflections are made sufficiently large, at least on the order of the wavelength for the relevant bandwidth.

Theory

For each location of the source and of the receiver, the impulse response is different. The impulse response is composed of:

- Direct sound
- Early reflections from the walls and objects in the room
- Reverberation i.e. where the reflection density is so high it is impossible to distinguish the different contributions.

Many measurements are made in various places in the room. By averaging them, we will build up the ratio DIRECT to REVERBERANT sound ratio (DRR).

In an anechoic room the impulse response, $h(k)$, is simply the direct sound component. In the "reverberant" room, the room produces an additional response $r(k)$ which is different for each measurement position (Figure 3).

If an average is made of N measurements, the pressure measured by the microphone becomes:

$$h_{av}(k) = h(k) + \frac{1}{N}\sum_{n=1}^{N} r_n(k) \approx h(k) \qquad (1)$$

Below the mixing time, t_{mix} $\sqrt{V}$ (V = the volume of the room), direct sound is summed coherently and the early reflections, will sum incoherently. Above the mixing time, each impulse response can be approximated as a noise sequence $x_n(k)$ with an exponential envelop.

$$r_n(k) \approx e^{-ak} x_n(k) \qquad (2)$$

The corresponding DRR in dB becomes:

$$DRR = 10\log\left(\frac{\left\langle \left|h(k)\right|^2 \right\rangle}{\left\langle \left| \frac{1}{N}\sum_{n=1}^{N} r_n(k) \right|^2 \right\rangle}\right) \qquad (3)$$

where <> indicates a time average.

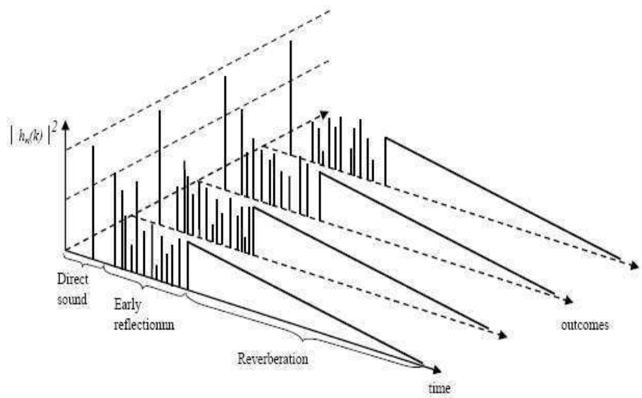

Figure 3: Examples of successive signals received by the microphone

Our task is now to predict the value of *N* that will be required to increase this to a value high enough, e.g. 10 dB, so that the measurement result can be considered to consist only of direct sound.

We can investigate this analytically for simple geometries (e.g. a rectangular room) by a standard Green's functions approach or, for more general geometries, by an image source model. But in all cases simplifications and approximations are required if we are to make predictions which can be tested against practical measurements. In the limit we might, for example, assume that the room satisfies the conditions of a Sabine space so we may regard the reverberant field (which includes the early reflections) as uniform throughout the volume. Then if the room has a total absorption, A, and an average surface absorption coefficient, α, a source with a sound power output, W, will create a reverberant sound with an equivalent intensity, I_{rev}, given by:

$$I_{rev} = 4W(1-\alpha)/A \qquad (4)$$

The direct sound from the source - if it has a directivity factor, Q, in the direction of the receiver a distance, r, away - will create a direct sound intensity, I_{dir}, given by

$$I_{dir} = QW/4\pi r^2 \qquad (5)$$

In this case the DRR reduces to:

$$DRR = 10\log\{QA/16\pi r^2(1-\alpha)\} \qquad (6)$$

Alternatively since we usually describe a room in terms of its reverberation time, T, we can use the fact that for a room of volume, V,

$$T = 0.16V(1-\alpha)/A \qquad (7)$$

to write:

$$DRR = 10\log\{QV/100\pi r^2 T\} \qquad (8)$$

This indicates that in order to have the direct sound component at least 10 dB above the reverberant sound the measurement process has to improve the DRR by an amount

$$\text{Gain required} = 10\log\{1000\pi r^2 T/QV\} \qquad (9)$$

If we assume that each time we repeat the signal the transducers have moved to a new position such that the reverberant sound components are uncorrelated, the number, *N*, of repeats

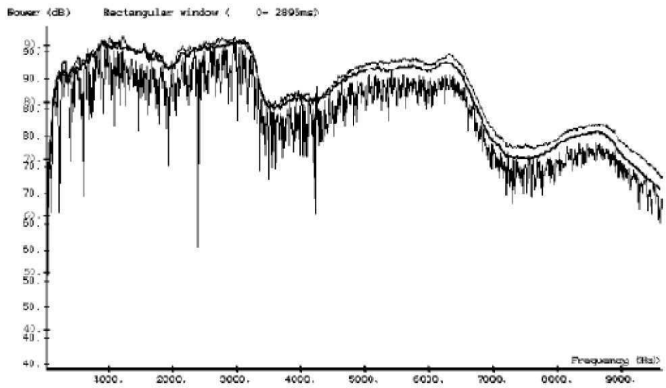

Figure 4. An example of how the proposed technique of coherent averaging an rotation of the source and microphone in a reverberant room can remove the reverberant sound component. Bottom trace is the stationary response, the middle trace is the rotated response and the top trace shows the anechoic chamber measurement of the response.

required – since the gain is 10logN – is:

$$N = 1000\pi r^2 T/QV \qquad (10)$$

In the case of an omnidirectional source measured at a microphone distance of 1m, in a room of 50m^3 and RT of 1s, this would imply that we need to repeat the signal a minimum of 65 times.

Using a source and microphone rotated together on a turntable the number of circles, S, ofdiameter, d, to be swept is given by:

$$S = N/P \text{ where } P = \pi(\sin^{-1}(\lambda/8d))^{-1} \qquad (11)$$

Practical Considerations

In practice the main difficulties are created by 1) the need to move the transducers throughout the room without changing their separation or orientation with respect to one another, and 2) the need to have sufficiently different positions that the reverberant components are uncorrelated.

Since we have no guarantee that in a particular measurement arrangement the unwanted components will present as incoherent we need to incorporate a check that the averaging process truly does produce a slower accumulation of unwanted compo nents compared with the wanted component. A "2 bin" approach can be used in which alternate sequence responses are accumulated in different bins which can then be compared both for estimates of the suppression of the incoherent signal in different parts of the spectrum. This then allows an estimate the measurement effort required to achieve a selected DRR given any room geometry and arbitrary transducer positions in the room.

These matters were addressed further in the conference presentation, but experience with using the technique has demonstrated that 1) rotation of the transducers is a feasible basis for the required movement and 2) that due to the larger correlation lengths in rooms at low frequencies it is at these frequencies that the room effect is more difficult to remove.

So far measurements have been made using MLS as the source signal but in principle any deterministic signal having an appropriate spectrum could be used. However, we have been investigating other time spread signals [e.g. 9, 10] which are more advantageous than others for system response measurements depending on the conditions of measurement (e.g. nonlinearity in the equipment, stationarity of conditions). Therefore consideration of the relative merits of these different signals for this application is part of our present work.

Recently affordable soundfield-type microphones have become available comprising multiple capsules able to decompose multi-path room sound fields. These might offer the possibility of reducing the measurement effort by reducing the number of sweeps required to achieve a desired DRR.

CONCLUSION

It has been argued that most of the measurements for which anechoic chambers have been used can be made in other environments and therefore there is a reduced need for building and maintaining anechoic chambers.

The need for an anechoic environment for measuring and calibrating transducers has for some time been obviated to some extent by the use of measuring impulse responses and then removing the reflections with a time window but this leads to a limitation on the resolution in the measurement and an associated low frequency limit.

This work has focussed on demonstrating the feasibility of making measurements of transducer responses in ordinary reverberant rooms without the need for a time windowing approach – and hence avoiding the limitations of such an approach

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