Using Coherent Averaging with time-varying conditions to remove room reverberation from transducer measurements.

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Abstract:
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.

The 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.

This presentation looks at using coherent averaging to measure transducer responses with high precision without using either an anechoic chamber or a time-windowing technique. Also we consider whether any particular deterministic signal – e.g. Farina's log sweep – is more advantageous for this application.

1. INTRODUCTION

The traditional aim in using coherent averaging when making response measurements has been to achieve sufficient signal so that results are dependable. This method is used, for example, in audiological measurements (for Evoked Response Audiometry and measurements of Stimulated Oto-acoustic Emissions) and building acoustics. Its application in field measurements in buildings permits successful measurements when background noise is high or signal power is limited. Coherent averaging can also allow us to be "environment friendly". For example, when measurements are needed in an already-occupied apartment building and neighbours are uncooperative or noise-sensitised, coherent averaging allows measurements to be made using signals radiated at sub-threshold levels and hence without disturbance.

The gain in S/N ratio when using coherent averaging results from the fact that in-phase coherent signals sum their amplitudes whilst incoherent sum their intensities. Thus, for a sound signal with sound pressure *P* a single repetition will

- upon coherent summing - produce a resultant sound pressure of 2 *P*. Accompanying noise having a sound pressure *N* - assuming it is not periodic and correlated with the signal - will produce a resultant sound pressure of $\sqrt{2}\,N$, thus improving the S/N ratio by 3 dB. Coherent averaging of *n* repetitions provides an improvement of $10\log n$ dB. When using this technique for measuring responses of linear systems it requires that the system remains invariant in time and that the unwanted (noise) sound does not contain components which are correlated with the signal.

In this presentation we extend consideration to a novel application for coherent averaging by using it to differentiate between direct and reflected transmission paths in rooms and permit anechoic measurements of transducers without the necessity for an anechoic chamber or time-windowing.

2. BUILDING MEASUREMENTS

When faced with an inadequate S/N ratio in a field measurement an alternative to bringing in a higher powered source or using higher signal levels (and potential disturbance to neighbouring occupants) is to repeat a deterministic signal a number of times and apply coherent averaging.

We have, for example, made insulation measurements into occupied dwellings without those occupants of the receiving spaces being aware of the test sounds which enter their dwelling [1]. In this case we used a Maximum Length Sequence (MLS) as our signal. In principle, any deterministic and repeatable excitation signal can be used but the phase-randomising property of MLS deconvolution makes this technique especially suited, as it provides transient noise immunity. Improved performance in the presence of non-stationary disturbances is also possible [2].

There are, however, limits to how much the S/N ratio (and hence dynamic range) can be improved by averaging. For MLS, non-linearities in any part of the measured system will impose what is effectively a noise floor which no additional averaging will improve upon [3]. Figure 1(a) shows this limit in a response measured in an auditorium, using a dodecahedron loudspeaker source. In this situation where the ambient noise is already low other sources may give a better dynamic range. Figure 1(b) shows a response measured in the same auditorium but this time using a starter pistol source. The form of the limit in fig 1(a) is particular to using MLS and results from distortion in the source. The implications of this are that for the best possible results using this technique it is necessary to drive the loudspeaker at very modest levels. Further illustrations are given in [4].

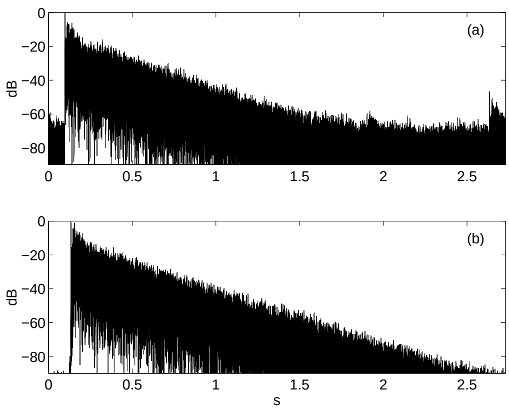


Figure 1. Impulse responses measured in a large hall: (a) With loudspeaker and MLS, and (b) with starter pistol

3. APPLICATION TO TRANSDUCER RESPONSE MEASUREMENT.

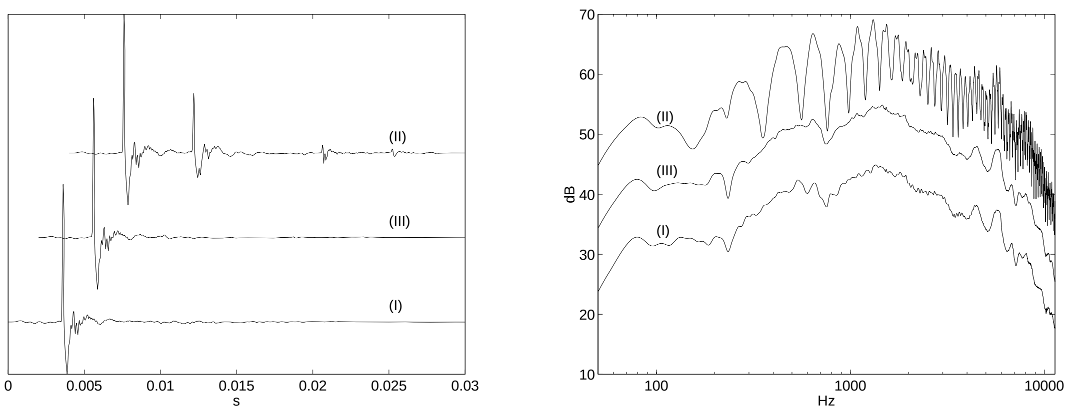
Possibilities for new uses for coherent averaging arise in those situations where we can purposefully introduce time-variance in any elements of a measured system that we would like to discriminate against. The measurement of transducer responses provides an example.

Transducer Responses

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 (which may be obtained in a variety of ways, e.g. MLS or time-delay spectrometry). 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 be avoided by making the measurement as shown in Figure 4. 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 effect of transforming some of the reflected energy into a time-spread, frequency dependent noise-like component, which can be reduced by coherent averaging [5]. 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



(a) (b)

Figure 1. The responses of a loudspeaker found from MLS responses:
(I) In an anechoic chamber, (II) near a reflecting surface, and (III) near a reflecting surface but with the loudspeaker and microphone rotating

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 3(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.

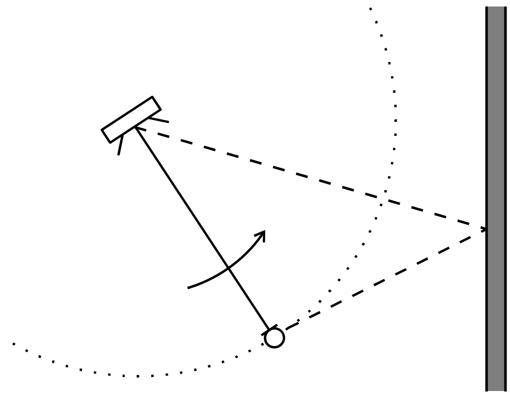


Figure 2. Measurement set-up with rotating transducers for reflection suppression by averaging

Measuring the Insulation of Building Partitions

This same technique of separating specific paths in a multi-path transmission process should, in principle, allow us to measure building component performance more exactly under conditions of high flanking transmission, and could be the basis of a tool for diagnosing and identifying construction defects.

Since reverberant sound can be suppressed with this approach, it might offer a useful alternative to measuring Sound Reduction Index by the intensity method when an intensity measurement is compromised by conditions which are too reverberant. The main challenge in developing suitable procedures is to find ways in which to render some transmission paths time-varying whilst leaving others time-invariant. Not only must they be effective in rendering the undesired transmissions incoherent over the whole of the relevant bandwidth, but they must also present as practicable in the field. We are presently considering various approaches, and results will be presented at the conference.

3. REFERENCES

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