Reverberation time measurements

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Contents

Chapter 1

What is reverberation time

Reverberation occurs in a room when sound is emitted in the room and then turned off. It is due to reflections from the walls, floor and ceiling. When the sound waves hit e.g. a wall it is partly absorbed and partly reflected and the sound level will diminish and eventually die out. The time it takes the sound to drop 60 dB is called the reverberation time. It is assumed that the sound die out exponentially, but this is sometimes not the case. Also the background noise is often high, which makes it impossible to measure over a 60 dB range. Therefore it is usually calculated, using linear regression, how long it takes the sound to drop 5, 25 and 35 dB and the the reverberation time constants are given by:

$$T_{20} = 3(t25 - t5)$$

$$T_{30} = 2(t35 - t5)$$

If T_{20} is very different from T_{30} the decay is not exponential. T_{30} is the preferred value, but T_{20} is often used if it is impossible to measure T_{30} due to a high background noise.
The reverberation time is a function of frequency. This means that the recorded sound is filtered by a filter bank and T_{20} and T_{30} is measured from the output from the filter bank. The filters in the filter bank are usually 1/1 or 1/3 octave filters. The lowest frequency of interest is seldom lower than 50 Hz and the highest is seldom higher than 5 kHz.

Chapter 2

The measurement

Two distinct methods can be used when measuring reverberation time:

1. Interrupted Noise
2. Impulse Response

2.1 Interrupted Noise

In this case a loudspeaker and a microphone is placed in the room. A white or pink noise signal generator with a constant output level is connected to the loudspeaker. The microphone is connected to a filter bank having an RMS detector on each output. The time constant of the detector must be somewhat shorter than the reverberation time of the room. When the signal generator is switched off the output of the detectors are recorded giving the decay curves of the room. Finally the procedure outlined in the previous section is followed.
Often a large power amplifier an loudspeaker is required, particular if background noise is present, making the method impractical is some situations.

2.2 Impulse Response

The Impulse Response method can be further divided into the following five methods:

1. Impulse excitation
2. Dual FFT
3. MLS
4. Exponential sweep and FFT
5. Heterodyne principle

The five methods reflect different common ways of obtaining the impulse response of a linear and time invariant system.
Once the impulse response is measured a common signal processing is used. The impulse response

is feed to a filter bank giving a filtered impulse response at each output. Each impulse response is then integrated going backwards from the end to the beginning giving the decay curves of the room. Finally the procedure outlined in the previous section is followed.
The impulse response can often be obtained without the use of a power amplifier and loudspeaker or using less heavy equipment making it ideal i many situations. Se the following sections.

2.2.1 Impulse excitation

In this situation no power amplifier and loudspeaker is used. The impulse response is obtained simply by generating a sound impulse in the room. The impulse must be short and have a flat spectrum. This can be difficult to achieve particularly at lower frequencies. The impulse is often obtained by puncturing a balloon, a gun shot or using an electrical spark. For less demanding measurements a hand clap may be sufficient.
The method is rather sensitive to background noise but is attractive since it only requires light weighted equipment.

2.2.2 Dual FFT

For this an the following three methods a power amplifier and a loudspeaker is required, but often simpler equipment than for the interrupted noise method can be used due to the much longer measuring time and signal averaging.The measurement set-up is shown in Figure 2.1 and Figure 2.2. The frequency response is given by:

$$H(\omega) = \frac{E[Y(\omega)X^*(\omega)]}{E[|X(\omega)|^2]}$$

where
$Y(\omega)$ is the FFT of Ch. 2 and $X(\omega)$ is the FFT of Ch. 1. $E[]$ denotes ensemble averaging.
The impulse response is then found as the IFFT of H. Due to the averaging the background noise and distotion in the loudspeaker can be suppressed.

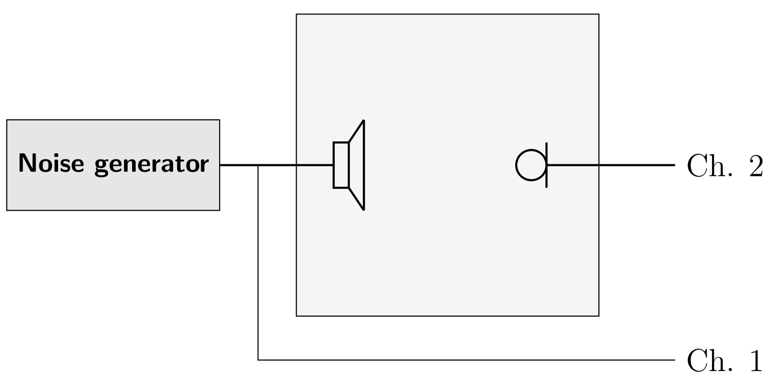

Figure 2.1:

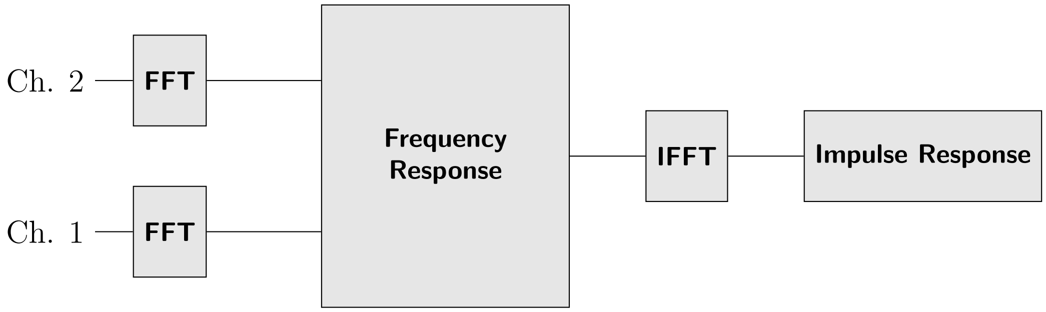

Figure 2.2:

2.2.3 Exponential sweep and FFT

The process outlined in the Dual FFT section can be done without averaging if the signal generated is a deterministic signal with an approximately flat frequency response. Such a signal could be a sine sweep. An exponential sweep is often used in order to give a better reduction of the background noise at low frequencies. The necessary sweep time is often relatively long (e.g. one minute) since there is no averaging involved. The consequence is that the FFT size becomes large (e.g 2^{22}) which can be a problem if the signal processing is to be done using simple hardware.
The method only requires one channel, since the generator signal is deterministic and the FFT need only be calculated once. The advantages is the same as for dual channel measurements regarding suppression of distortion and background noise.

2.2.4 MLS

In this situation only the output from the microphone is measured, see Figure 2.3. The generator output is a MLS signal as described in [1]. The output from the microphone is a periodic signal and in order to reduce the background noise every period is averaged together and followed by the signal processing also described in [1].
The method is somewhat sensitive to distortion in the loudspeaker, but if averaging is done over a large number of periods the background noise can be reduced even if the generator level is low. One of the advantages of this method is the simplicity of the signal processing and the low memory requirement.

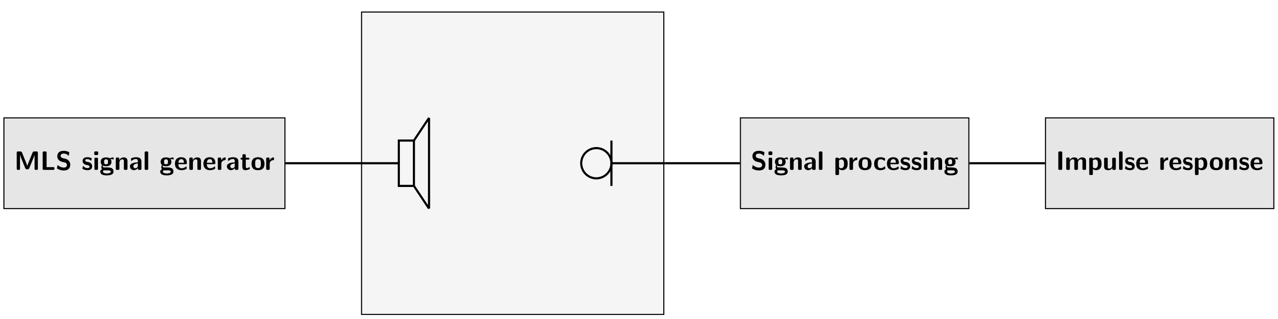

Figure 2.3:

2.2.5 Heterodyne principle

A sine sweep, linear or exponential, is used for excitation. The technique is described in [2]. The advantage compared to the method described in section 2.2.3 is that the signal processing can begin when the sweep begins making it possible to use virtually any sweep length. Most of the signal processing is thus "hidden" behind the sweep an only the final FFT and sweep compensation are visible. There is a running data reduction making the method well suited for measurement equipment with a limited memory size and low processing capabilities.

Bibliography

[1] Impulse response measurements using MLS by Jens Hee, http://jenshee.dk/signalprocessing/mls.pdf

[2] Sine and Sine Sweep measurements by Jens Hee, http://jenshee.dk/signalprocessing/sweepmeas.pdf