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MULTI-CHANNEL REVERBERATION TIME MEASUREMENT WITH SWEEPS

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Abstract

This work presents aspects about simultaneous reverberation times measurements at a number of points inside rooms. The reverberation time values are determined from impulse responses measured by an indirect technique, using sweep sine excitation signals. The first part of the paper deals with the sweep generation, considering the room behaviour and the electroacoustical sound source response. Doing so, a high and frequency-independent signal-to-noise ratio can be achieved in a broad frequency band, while at the same time keeping constant time amplitude. A unique convolution signal reference for all the channels can be determined from the designed sweep signal. The impulse response from each channel can be filtered by standard fractional octave bands and then backward integrated to yield filtered energy decay curves. The filtered impulse responses have different limits after the convolution. Because of that, the integration limits have to be considered carefully, avoiding the well know bias errors from the impulse response noise energy. The paper also considers a method for averaging the filtered impulse responses of different room locations after the backward integration, reducing decay curve oscillations due to lack of diffusion, mainly at the lower frequencies. Some experimental results are presented for a six-microphone system in a reverberation room for sound power measurements. The measurement chain is described, including the sound source and the devices used. Establishing a multi-channel system for reverberation time measurements is easily feasible nowadays, with many options of high quality AD/DA converters. Such a system dramatically reduces the measurement time and could increase the precision of various acoustic test types that require spatial average reverberation times.

INTRODUCTION

The task of reverberation time measurement in rooms and the evaluation of other acoustical parameters can be fulfilled fast and accurate by using transfer function methods. The reverberation times RT are estimated with the well know RIR (room impulse response) backward integration technique which results in the sound energy decay curve. In order to minimise systematic errors caused by the noise floor in the RT results, the transfer function method should yield the maximum decay range possible. The sweep technique provides many options for the selection of excitation signals to maximise the decay range, as show in [1].

Multi-channel RT measurement systems are a very efficient tool to reduce test execution time, allowing for a number of positions inside a room to be measured simultaneously. This paper presents some aspects of a multi-channel system used by the Inmetro acoustic laboratory for room measurements. The system is composed of an AD/DA frontend for PC use, targeted mainly to multi track audio recording. It has 8 input and output channels with low noise and distortion audio AD/DA converters. A software called Monkey Forest controls the frontend, performs the sweep measurements and processes the RIR for room acoustical parameters calculations. The Inmetro system is used in field applications and in-laboratory applications [2,3]. Accessories are available for signal conditioning and amplification, allowing the use of different instrumentation, like measurement microphones, artificial head, two-way omnidirectional source and artificial speaker.

The following sections present details of the Inmetro system for multi-channel RT measurement. Particularly, some solutions applied to support a reference sound source calibration procedure in our reverberation room are pointed out. The description includes the sweep design, considering the loudspeaker response and the predicted RT. The reference spectrum's group delay establishes the limits of the RIR after filtering it into 1/3-octave bands. Some results are presented, showing that very high decay ranges were obtained.

SWEEP SINE MEASUREMENT

Figure 1 presents the steps involved in a RIR measurement with sweeps. The sweep is divided into two channels to feed a two-way omnidirectional sound source, composed of a dodecahedron and a subwoofer. After captured by a microphone, the sweep response is Fourier-transformed and multiplied by a reference spectrum, resulting in the room transfer function RTF. Finally, the RIR is obtained by the inverse Fourier transform of the RTF. For a number of microphones distributed inside a room, all these steps can be done simultaneously.

Sweep Design

Although the swept sine is an old and well know signal for acoustical transfer function measurements, there are some design possibilities, as shown in the reference [1], that can improve substantially the measurement performance. The sweeps are easy to generate, being a sine with variable instantaneous frequency. The sweep crest factor can be kept similar to that of a sine (3 dB), resulting in good performance of power amplifiers and loudspeakers. By sweeping the frequencies from low to high, the response can fit both the longest low frequency RT and the shortest high frequency RT. Both the sweep duration and the frequency range can be extended as needed by defining the sweep limits. The sweep's magnitude spectrum depends of the sweep rate. For instance, linear sweeps own a white spectrum, while log sweeps possess a pink one. The instantaneous sweep rate can be controlled to optimize the signal to noise ratio, considering the source's frequency response, the background noise and the expected reverberation time. The harmonic distortion from the loudspeaker can be distinguished and segregated from the RIR, this way substantially increasing the measurement signal-to-noise ratio in comparison to measurements with pseudo-random noise.

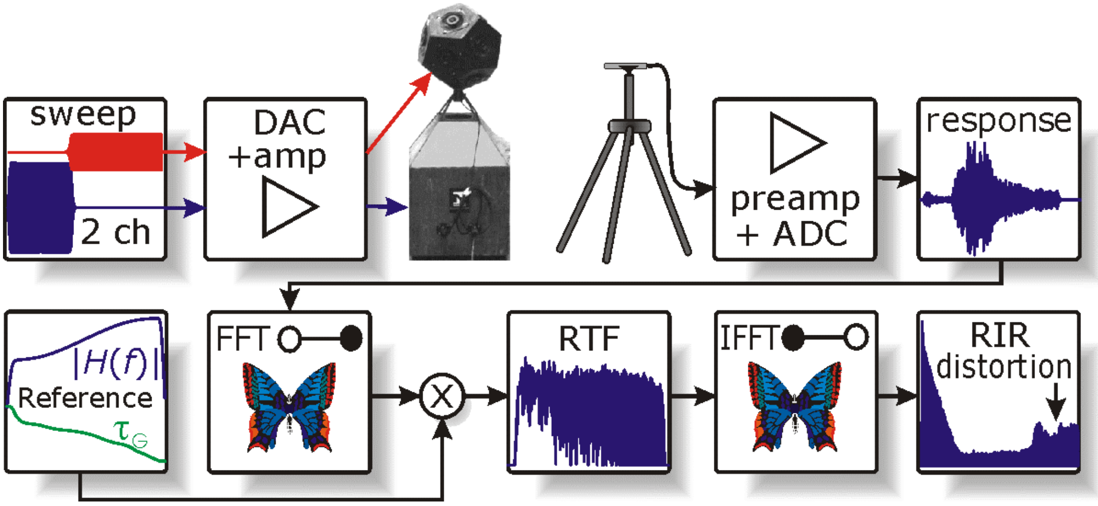

Figure 1 – Multi-channel Room impulse response measurement with sweeps and circular deconvolution.

The sweep used in this work was designed in the frequency domain, based on a desired excitation spectrum, see Figure 2. The dodecahedron and the subwoofer power spectra, both measured in diffuse field, were inverted to equalize them. Additionally, an extra emphasis at low frequencies (30 dB at 100 Hz) was applied to improve the signal to background noise ratio. The excitation was divided in two channels to substitute external crossover circuits. Once the desired excitation spectrum $|H(f)|$ has been defiened, the group delay $\tau_G(f)$ was calculated by the equation (1) relating the two variables by a constant equal to the ratio between the sweep length and the spectral energy. The sweep complex spectrum composed of the desired amplitude and the phase, calculated from the group delay by integration, was then transformed into a time signal by IFFT.

$$\frac{\partial \tau_G(f)}{\partial f} = \frac{\tau_G(f_{END}) - \tau_G(f_{START})}{\sum_{Ch=1}^{2} \sum_{f=0}^{f_S/2} |H(f)|^2} \cdot \sum_{Ch=1}^{2} |H(f)|^2 \qquad (1)$$

The sweep generated with this method had a nearly constant time envelope despite of the arbitrary spectrum defined as target. Because the subwoofer can bear more power than the dodecahedron, its output gain was set 8 dB higher.

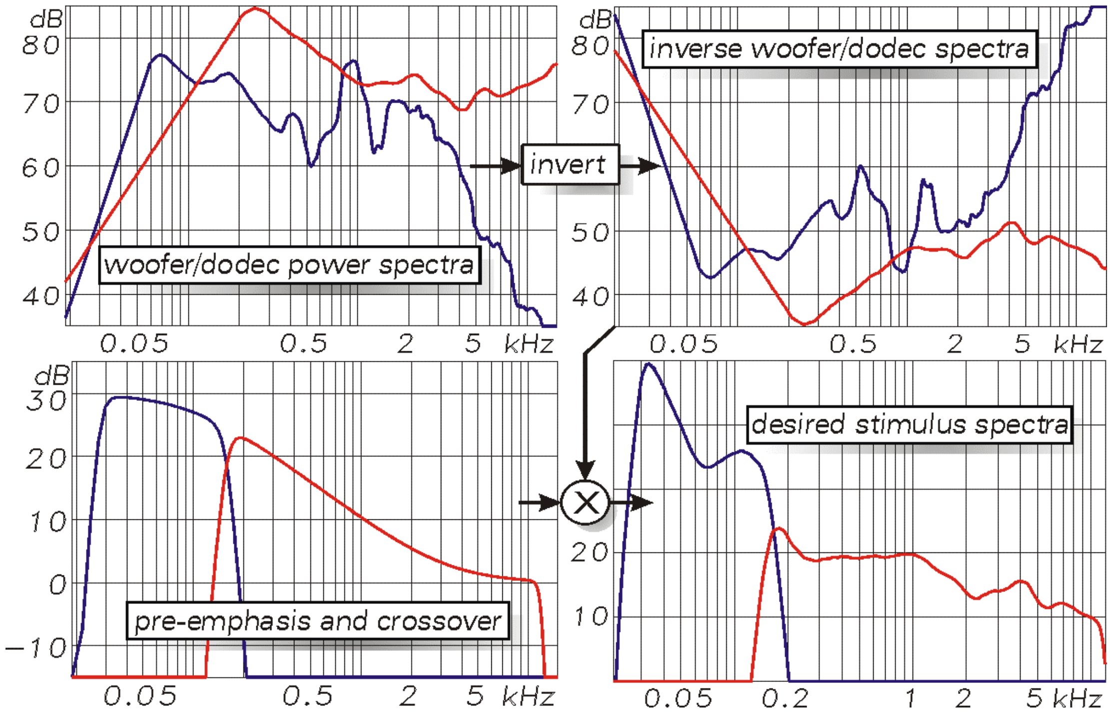

Figure 2 – Steps to derive the desired excitation signal spectra: Inversion, application of pre-emphasis and crossover band-pass functions.

Room Impulse Response

The reference spectrum for the deconvolution was created within the design procedure. Its modulus consists of the inverse pre-emphasis curve. To recover the RIRs from the sweep responses captured by the microphones, the group delay of the reference spectrum has the opposite course of the one of the excitation signal. This means that any frequency line of the sweep response will be shifted left by a time period equal to the group delay for that frequency. In other words, the reference spectrum's group delay determines the frequency dependent limits of the RIR after the deconvolution. Depending on the deconvolution mode, circular or linear, the area behind the group delay limit can contain harmonic distortion or no signal at all. In both cases, this region should be discarded. Otherwise, the RT values could present serious systematic errors from incorrect back integration. Figure 3 shows the RIR limits governed by the reference spectrum's group delay in the case of circular convolution and how this affects the filtered RIR for different frequency bands.

MULTI-CHANNEL REVERBERATION TIME MEASUREMENT

The multi-channel system was used to measure RTs for reference sound source (RSS) calibration in reverberation room, according the ISO 6926:1999. The ISO procedure requires RT measurements at 6 microphone and 3 source positions, yielding a total of 18 spatial combinations. The RT is taken from T_{10} or T_{15}. RT values in 1/3-octave bands from 100 Hz to 10kHz are required. The system has been set up with 6 microphones inside the reverberation chamber. The 3 source positions were set manually. A single 24 seconds sweep was enough to obtain RIRs with very high dynamic range.

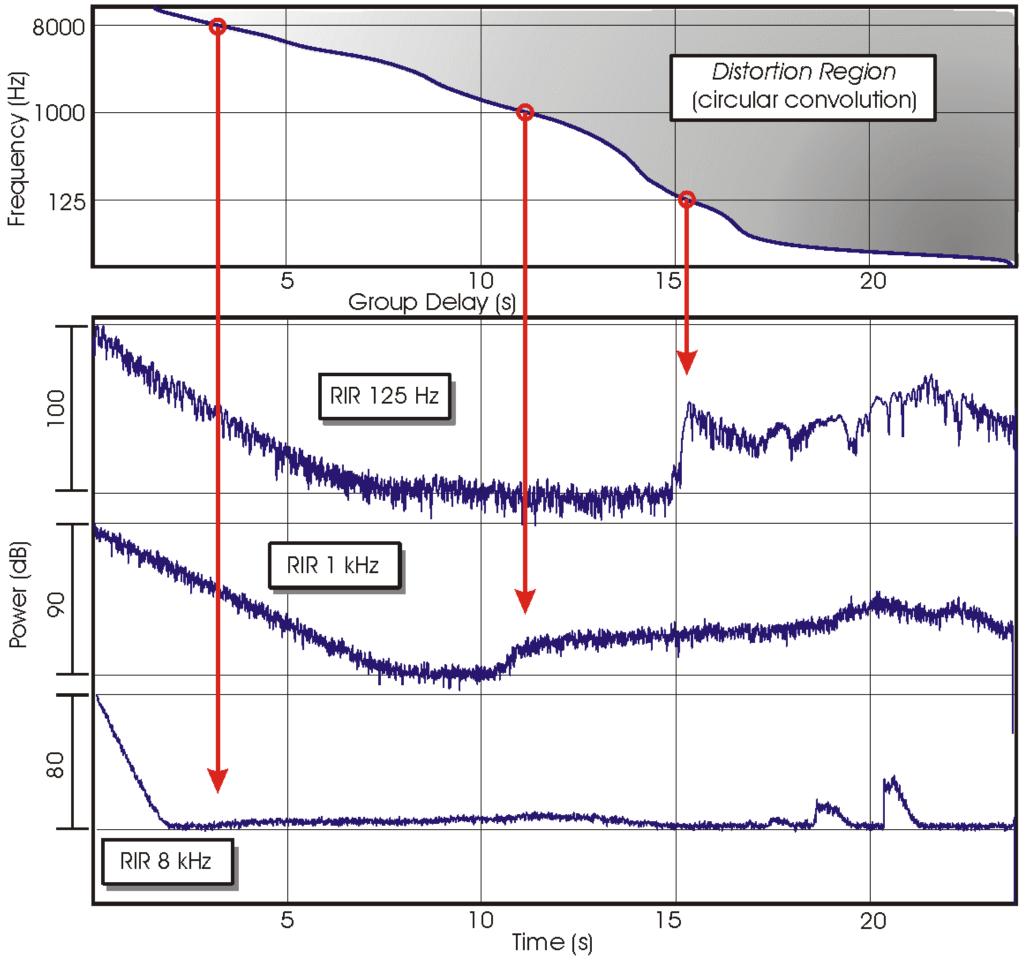

Figure 3 – Reference group delay, upper part, limiting filtered RIR after circular convolution.

Some preliminary tests were conducted to adjust the multi-channel system. First, the RT has been measured by the traditional interrupted noise technique with the help of a sound analyzer Norsonic RTA-840. The differences found between the RT results from a measurement with the new system and from RTA-840 (3 averages per position) were less than 6% in the 125 Hz band for T_{20}. The RT derived from the former technique served as reference for the sweep length selection. For instance, the expected RT was around 4 s for the 125 Hz, 5 s for 1kHz and 1.5 s for 8 kHz. The reference spectrum group delay and the achievable decay range (also referred as S/N) also have to be considered in the sweep length selection. In our case, the measured decay range was far more than 60 dB with a single sweep in all frequency bands, meaning that the reverberant decay dropped into the noise floor at a time above the RT.

The plots shown in Figure 3 were obtained for a sweep having 2^{20} (order 20) values, resulting in a length of 23.7 s (sampled at 44.1 kHz). The reference spectrum's group delay at the top indicates the effective duration of the RIR after filtering. The RIR results after filtering for the 125, 1000 and 8000 Hz 1/3 octave bands are shown. For the 125 Hz band, the RIR fits into an interval of 15 s. This is enough to accommodate the initial propagation gap between the source and microphone, the reverberation decay and still a strip of the noise floor. The decay range was around 100 dB. After 15 s, harmonic distortion occurs because the circular deconvolution [1]. The limit for the 1 kHz band was 11 s and the decay range was around 90 dB. For the 8 kHz band, the limit was 3.5 s and the decay range was 80 dB. The differences between the distortion behavior in the 3 bands are remarkable.

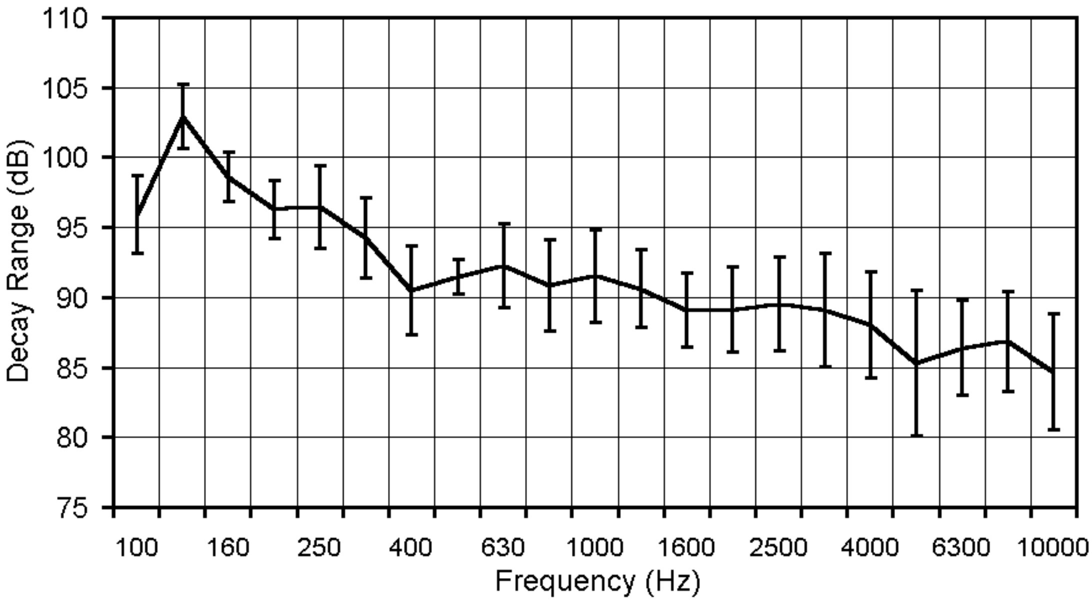

Figure 4 – RIR mean decay range and the deviation in 1/3-octave bands for 18 spatial combination of source and microphone positions.

The filtered RIR obtained with the system and a single sweep has satisfactory decay range for all the source and microphone combinations. Figure 4 shows the mean decay range and the variation of the values for the 18 source and microphone positions. Even in the worst case, the decay range was still 80 dB, much higher than the 20 dB decay range necessary for the T_{15} evaluation. This difference gives an enormous safety margin against systematic errors in the backward integration process to calculate the reverberation times [4].

Decay Average

The multi-channel RT analysis can be improved in precision and time execution if the integrated impulse response IIR are averaged before the RT calculation. A proposed averaging method is currently under investigations. Until now, the comparison between the mean RT value and the RT value from decay average shows very good agreement. The proposed method, however, is believed to approximate the real RT value with fewer measurements than the standard averaging of the final RT values.

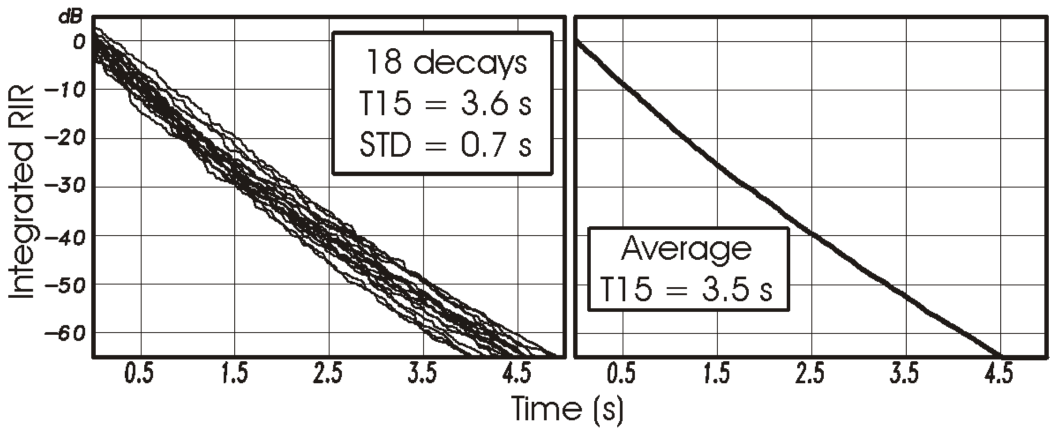

Figure 5 – 18 decays in 125 Hz 1/3-octave band, left, and the decay average, right.

The effect of IIR averaging can be seen in the Figure 5. The left plot shows 18 decays in the 125 Hz band. The curves have nearly parallel paths, but present typical oscillations because of the low modal density at low frequencies, and also some bending. The mean T_{15} value is 3.6 s with a standard deviation of 0.7 s. In the right plot of Figure, all the 18 curves are averaged by the following proposed method. In a first step, the RIR start of each channel is found and the curves are shifted in time to synchronize the decay start point. Then the energies of each IIR (so the backward integration value at zero time) are normalized. Finally, the IIR are averaged. The normalization process guarantees that each integrated impulse response has the same weight in the averaging process, regardless of microphone sensitivity differences. The decay average results in a curve free of oscillations, but with the same bending tendency. The T_{15} in this case is 3.5 s.

CONCLUSIONS

A reverberation time measurement system was presented. The system operates with 8 channels simultaneously and so reduces the execution time. By using a single shot of a specially designed sweep, the impulse responses of a reverberation chamber were measured with a very high decay range. It was shown that the time limits between the room impulse response and the harmonic distortion region are governed by the group delay reference spectrum. A technique for decay averaging to achieved reduced curve oscillations caused from lack of room diffusion has also been presented.

The reverberation time values from the Inmetro reverberant chamber have being measured many times until the present. The system stability is verified by the repetitions. Some measurements were also made using shorter sweeps (order 19), with similar emphasis characteristics. The values found were very close to those obtained with the longer sweep. The difference for the shorter sweep is that the distortion products almost smear into the reverberant tail in some frequency bands. Figure 6 shows the T_{15} measured and compared to the reference spectrum group delay for the two sweeps.

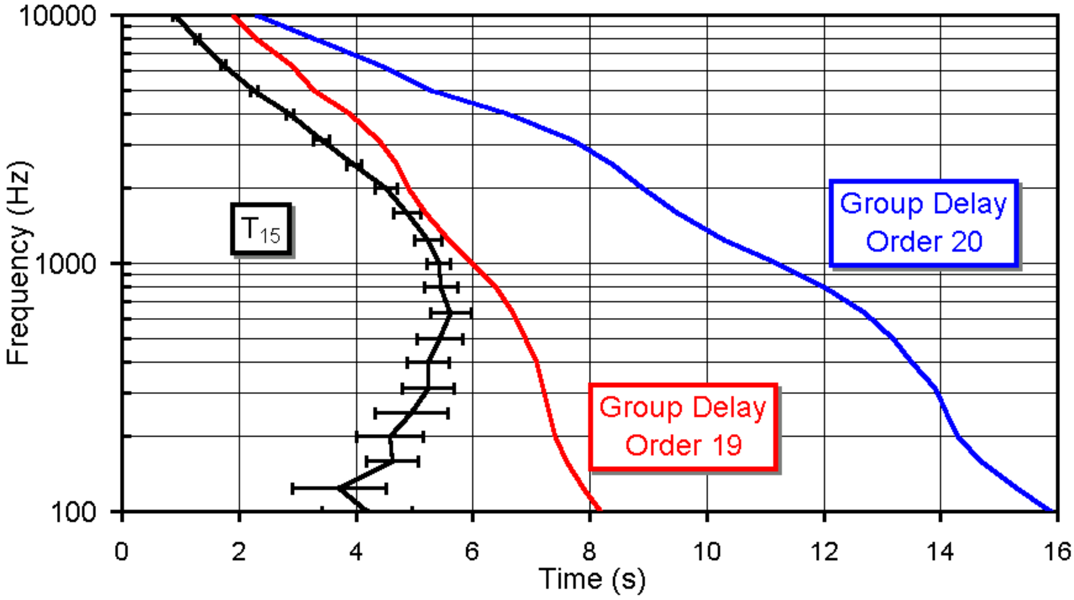

Figure 6 – Mean value and variation of RT from 18 decays and reference group delay for two different length sweeps.

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