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Short Reverberation time measurements using MLS

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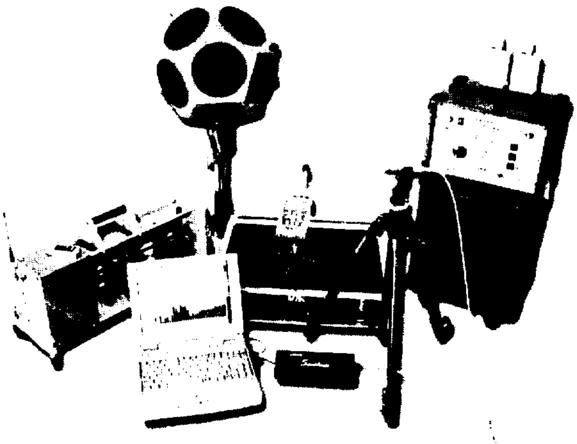

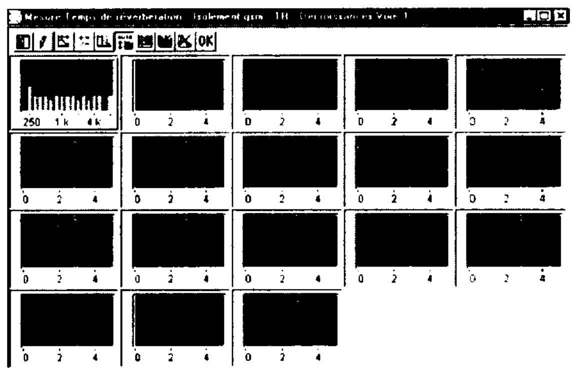

Measuring reverberation time has been discussed for years, with different types of sources and different types of algorithm for any kind of rooms, concert, industrial or general purpose. For short reverberation times, the problem is especially interesting due to an increased difficulty of having a reliable and repeatable measurement and to the sensitivity of the analysis techniques applicable to short signals.
This paper presents a method, which allows a reliable estimation of reverberation time for small spaces, in a software environment, and using a commercial product (Symphonie analyser), designed for in-situ measurements.
The recent implementation of the MLS (Maximum Length Sequence) technique is not only a general improvement for building acoustics but also a new feature for room acoustics.
In order to take a full benefit from the MLS technique, a new filtering method has been implemented in order to process efficiently short signals as explained in this paper.
The MLS technique is also very useful for other applications such as ground absorption measurement or evaluation of the acoustical performance of cellular phones.

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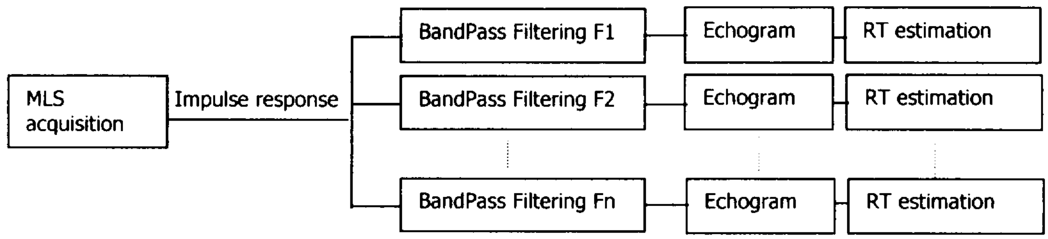
1. ANALYSIS METHOD

1.1. *Architecture*

One method for obtaining a precise measurement of reverberation time is to apply the definition (decay of energy by 60dB) to echograms of the impulse response filtered in frequency bands. A general schematic of the tools available for arriving at this result is shown below. Each of these operations is available as tools developed in software, based on a Symphonie hardware acquisition platform.



1.2. *Impulse response acquisition*

Impulse measurement techniques (pistol shot, handclap) for obtaining the impulse response of a room suffer from a low signal to noise ratio, and very poor repeatability. In addition, it is very difficult to excite the room sufficiently in each frequency band to be able to reliably calculate the corresponding reverberation time.

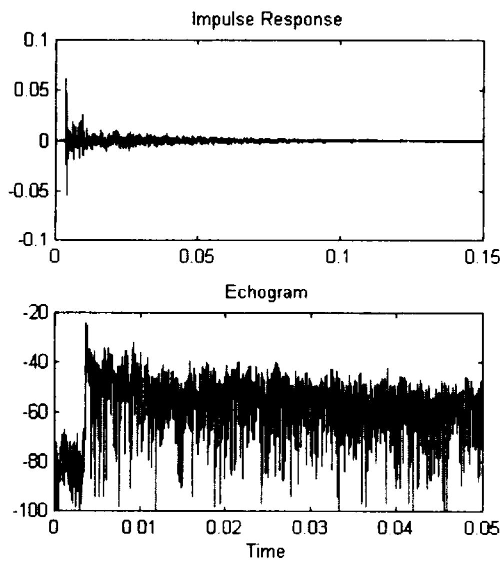
For this technique, it is preferred to use an MLS-type measurement, where the principle is based upon the emission of maximum-length sequences and cross correlation between the emission and the received response.

These signals have the advantage of generating a significant energy at all frequencies and immunity to noise and distortion.

This measurement method, now accepted as a standard, is well described in references [1] and [2]. In addition, this digital method has the great advantage of a fast transformation, developed by Hadamard, which allows the efficient calculation of the cross-correlation function between signals [3].

The implementation of this method on the acquisition platform offers the following characteristics:

- ✓ Generated sequences of orders 8 to 18,
- ✓ Pass-band up to 25.6kHz,
- ✓ Dual channel acquisition, and averaging of up to 2048 sequences.



The wide range of available orders allows the identification of varying responses and adaptability to the measurement. An additional possibility is the calculation of two impulse responses from the same excitation, which allows the application of several methods of determining binaural transfer functions based on the use of an artificial head. An example of a classical method and its wide-band echogram (time zoom) is presented here.

The identified volume corresponds to a room of 3.5 x 3 x 2.5 - 25 m^3.
13^{th} order was used (response of 160ms). The number of averages is 32, giving a measurement time of 5.120s.
Despite the low number of parameters used, we obtain a signal to noise ratio of the order of 40dB, with a noise level in the room of around 40dB (equipment noise).

1.3. *Filtered echogram computation*

To calculate the reverberation time, we need the time signal in each frequency band (partial octave) for which we need a result. We therefore need the best possible representation of the signal in both time and frequency domain, for post-calculation of the energy.

For this, two classical techniques are commonly used: FFT analysis/synthesis in partial octave bands and digital filtering.

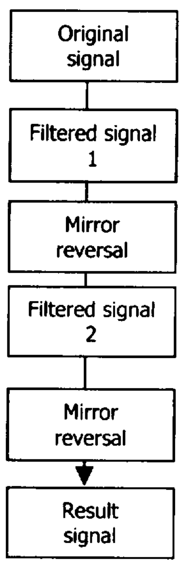
Intuitively, one can achieve mediocre results from an FFT analysis, this method being best suited to the analysis of stationary signals and very sensitive to transient analysis.

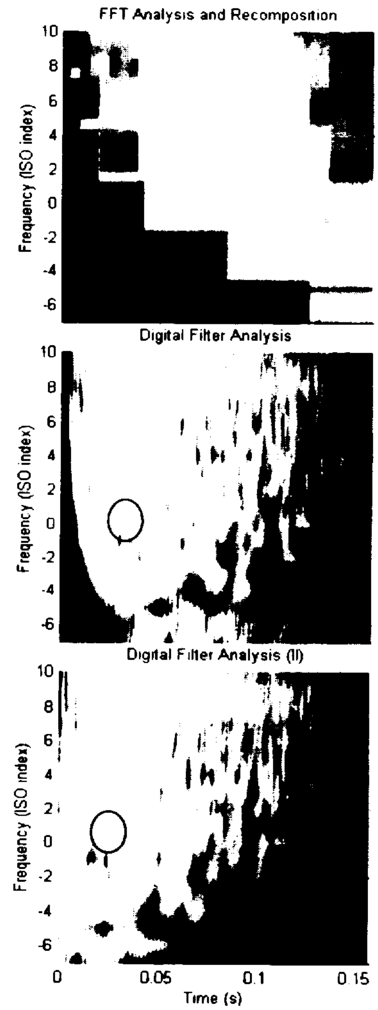
In addition, we must retain a good time resolution as the resulting calculation is based on the decay of the direct wave. The principle of block processing (size of FFT) and the need to avoid using a large overlap (keeping the time resolution) appears to limit this method.

This is confirmed by processing the same signals with the two techniques to obtain a spectral evolution in third octaves, in bands 50 Hz to 10 kHz, and with a time resolution of 1 ms. The « blocky » appearance obtained with FFT synthesis (top picture on following page) completely eliminates the method because of inadequate time resolution, especially at low frequencies.

The digital filtering method (middle figure on the following page) appears more suited to our calculation.
Offering a time resolution identical to the signal source, it needs a large calculation overhead, since the signal must pass through as many filters as required at the same time.
To overcome this inconvenience, a classical improvement consists of using only one bank of filters and to take advantage of digital processing, by defining the filter shapes as a function of their centre frequency, independently of the sampling frequency.
In this way, by dividing the sampling frequency by 2 in each filtering stage (decimation), we reduce the number of points by the same factor and thereby the calculation load.
Even if the calculation overhead is overcome, the middle picture shows also one of the other major problems of digital filtering: the propagation time of the energy leads to a time delay in the output, which is accentuated by the bandwidth of the filter.

Also, at low frequencies, this very visible phenomenon leads to a significant offset between the moment of detection of the direct waves at low frequencies, and therefore inaccuracies in the estimation of the decay slopes.
To overcome this problem, we therefore propose an intelligent filter in « mirror image », which compensates for the time delays in the filters.

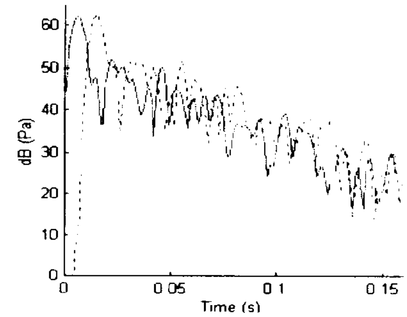




This algorithm consists of filtering the signal twice: once in the direct sense, then again in reverse, whereupon the signal is filtered again.

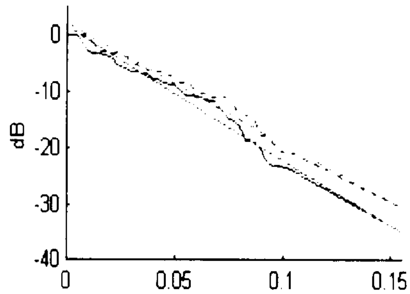
The final result is achieved after another reversal of the output signal. The operation consists of two filtering operations instead of one, which might seem slow, but in fact is quite simple: we can apply a filter of half the order of the traditional method, in a way that its double application gives the same characteristics at the output. The filters used are 4th order Butterworth, designed to meet the requirements of ISO Class 0. Only the two reversals weigh down the processing, but the improvement gained in the result (bottom figure) largely justifies this treatment. With a standard PC, the result shown is achieved in around six seconds.

To finish with the performance of the filtering algorithms built-in to the 01dB software, here are two time histories extracted from the 1kHz bands using the two previous filtering methods.



As can be seen, the filtering method used (solid line) compensates the delay introduced by the classical method (dotted line) that is about 10 ms.
By looking at the two curves carefully, we can see that there is more than just a time shift: during the decay, the details appear more precise (see the zones circled on the time-frequency maps for channel 0) and are better localised. An analysis of a unique wave can therefore be better synchronised across all the frequency bands.

For the estimation of short RT, this improvement can have significant consequences in such a way that the estimations of energy decay will also be more accurate. The figure here compares the two decays corresponding to the two filtering methods obtained by classical backwards energy integration according to Schröeder [4]. A simple linear regression shows the decay slope.



We can then take the estimation of RT60: 290 ms against 260 ms with the corrected method of filtering, a significant difference of around 10%.

2. APPLICATIONS

As demonstrated MLS method suits to measurement of short reverberation time. But this method can also be used to other applications.

- ✓ **Room criteria:** The calculation of many criteria is also possible, such as EDT, clarity, and definition as well as the classic STI and RASTI.
- ✓ **High insulation:** High insulated places like Music halls require an accurate method to make the measurement of the insulation. For the same reason than short RT measurement, this method fulfils the requirement.
- ✓ **Simulation of the response of a system**: MLS method gives a very good estimation the response of a system. It is possible to make a convolution of this response with another signal to emulate the response of the system. For example, it is possible to measure the response of a cellular phone. Then you can make a convolution with an audio signal "emitted" to determine the audio signal "received" by the receptor. You will then know the sound quality of this product.

3. CONCLUSION

The measurement of short reverberation times requires an adaptation of classical methods to guarantee an optimal accuracy. By using Symphonie and its associated software suite, the tools are available for the measurement of impulse responses using the MLS method and at the same time for optimised analyses of these responses.

The « mirror image » filtering allows the conservation of the direct wave synchronisation after filtering, and assures the accuracy required for the calculation of different criteria, such as the reverberation time.

Now that this method is validated and implemented, we can see that it would be useful for many other applications, either in building acoustics and industry.

4. REFERENCES

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