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건축음향을 위한 LMS 측정방법

MLS for Building Acoustics Measurements

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1. Summary

The paper describes how a Maximum Length Sequence (MLS) and correlation technique in the form of Hadamard transformation may be applied for the measurement of airborne sound reduction. It is shown that in most cases, the values obtained by the MLS technique will be equal to the expected values obtained by the classical method. However, due to the correlation process involved, the MLS method will be much less sensitive to disturbances from extraneous noise. The described method has been implemented in an RTA acoustic analyser.

2. Introduction

Present standardised techniques related to building acoustics for the measurement of airborne sound reduction and reverberation time prescribe the use of broadband noise for the excitation. The level reduction is obtained by calculating the difference of the mean sound level in the source- and the receiving-rooms in response to the continuous noise excitation. Reverberation is calculated from the decay of the sound level just after switching a steady noise signal off in the receiving-room.

The method is described in the international standard ISO 140. The measurements should be made with instrumentation fulfilling the requirements for sound level meters (IEC 651/IEC 804) and with frequency selective filters, normally in 1/3 octave bands (IEC 225/IEC 1260).

This paper describes how a Maximum Length Sequence and correlation technique may favourably be used to obtain the same measurement values. In most cases, the values obtained by the MLS technique will be equal to the expected values obtained by the classical method. However, due to the correlation process involved in the MLS method, the measurement will be much less sensitive to disturbances from extraneous noise. Even clicks, pops, footsteps, etc. will all be transferred into benign noise distributed evenly over the entire measurement-time.

3. Relations to the impulse response

For a sound reduction measurement, the mean levels in the source- and the receiver-room are measured using a loudspeaker with required directional specifications placed in the source-room. The loudspeaker is fed from a noise generator covering at least the frequency band for the measurement. As the system is considered to be linear, the setup may be modelled as a white noise source followed by a frequency selective measurement channel.

Due to the stochastic character of the noise, the mean level over a certain measurement time T will have a standard deviation given by the square root of the inverse of the observation time multiplied by the bandwidth of the measurement.

Reverberation time is measured using a similar setup. The noise is on until a stationary value is obtained. Then the source is switched off and the decay in the observed level is recorded. Again the stochastic character of the excitation signal will give different decay curves. An averaged decay may be obtained by averaging more measurements (ensemble average).

By the application of stochastic signal theory Schroeder has shown that if the noise is switched off at the time t=0, the expected decay, L(t), may be calculated from the impulse response, r(t), of the system comprising of the chain between the noise generator to the level detector for the decay:

$$L(t) = 10\log\left(\int_t^{\infty} r^2(\tau)d\tau\right)$$

The integrand is a squared function and thus positive. The integral will therefore have its maximum value for t=0 and fall monotonously as t is increased. Due to the theory, the value L(0) corresponds to the expected noise level just before the noise was switched off. The equation may therefore be used for estimating the excitation level as well as the reverberation decay.

The stated results hold as long as the transfer channel is linear and time-invariant (time-independent). We will later discuss how these requirements hold in a normal measurement in building acoustics.

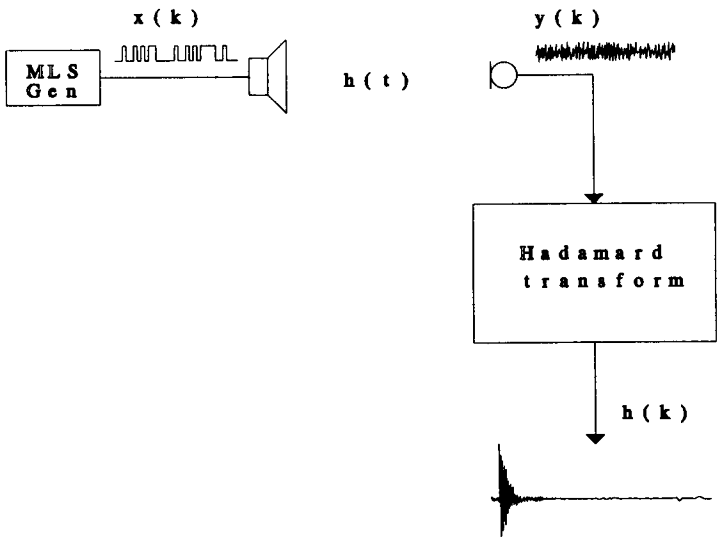
The impulse response may be measured in different ways. An obvious application would be to apply an acoustic impulse and simply record the output from the microphone. However, it is very difficult to obtain such an impulse in a controlled and reproducible way. Signal theory shows that the impulse response may be obtained from a correlation calculation between the input- and the output-signal.

4. The M-sequence as noise source

The international standard ISO 140 prescribes bandpass-filtered broad band noise as an excitation signal. Such a signal has a continuum of frequencies between the band-limits. This is an attractive signal because all room-resonances will be excited. The M-sequence will have a certain length. Therefore, if replayed sequentially, it will have a line spectrum with a line separation equal to the inverse of the sequence length. For example, a sequence of 8 seconds length will have a line spectrum where the distance between the lines is 0.125 Hz.

The acoustic response of a room may be described by a number of resonances, the lowest determined from the dimensions of the room. The width of each resonance or the Q-value, is related to the reverberation time at that particular frequency. It may be shown that if the sequence length is longer than the reverberation time, more than two spectral lines falls within the bandwidth of each elementary resonance. It can therefore be concluded that an M-sequence of this length will excite every resonance. As a substitute for a true noise signal an M-sequence may therefore be used to test room-responses as long as the length is longer than half the reverberation time and preferably longer than the reverberation time. This has also been verified by a number of practical measurements where M-sequences have been used and where no deviations from values obtained by true noise are reported.

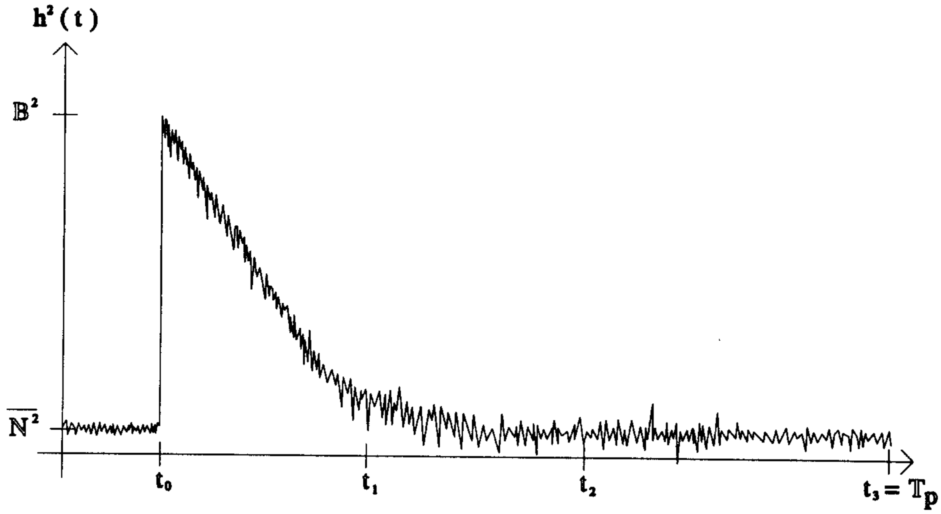
The M-sequence is a deterministic signal in contrast to a real noise signal which is stochastic. This difference is not likely to cause any problems for this application. As the result may be shown to have a correct expectation, it is in general positive that these results show no stochastic uncertainty.



5. Measurement of the impulse response

Statistic signal theory shows that if white noise is used as an input signal to a linear, time-invariant system, the cross-correlation between the output and the input will be equal to the impulse response of the system. An M-sequence may be considered to be a discrete-time approximation to white noise.

A number of papers have shown that the cross-correlation and thus the impulse response may be computed in a very effective way by the **Fast Hadamard Transform (FHT)** if the white noise is approximated by an M-sequence. The maximum length sequence of order N consists of (2^N -1) samples. Each sample has the value +1 or -1. The values seem to appear randomly. A benefit of the FHT is that it requires, like the more familiar **Fast Fourier Transform**, only **n $\log_2(n)$** operations. Since the M-sequence is represented by +1 and -1, the FHT consists of additions and subtractions only.



6. Level measurements

If the response to the noise excitation is measured in the conventional way, it is impossible to distinguish between the response and the extraneous noise. However, if the measured signal is Hadamard transformed, almost all signal energy is collected in the first part of the record. If we then only integrate over those parts of the measured impulse response where the signal energy dominates, and skip the part where the background noise dominates, we will get a much better estimate of the signal. If we apply the same power for excitation within a frequency band, the enhancement in the signal to noise ratio can be shown to be approximately given by:

$$\Delta_{S/N} = 10\log\left(\frac{T_{meas}}{T_{rev}}\right) + 3dB$$

where $\mathbf{T_{rev}}$ is the reverberation time and $\mathbf{T_{meas}}$ is the total measurement time consisting of a whole number of M-sequence periods. Signal-to-noise enhancement may therefore be achieved by investing in prolonged measurement time.

The method allows us to a certain extent to separate the signal and the noise. After the measurement it is therefore possible to indicate the signal-to-noise ratio obtained and thereby also compensate for the noise.

The noise reduction also applies to inherent noise in the measurement instrumentation. The user should therefore be aware that the measurement values thus obtained may be limited by cross talk in the signal cables and the instrumentation setup.

7. Reverberation time measurements

For reverberation time measurements the signal-to-noise ratio may be enhanced in a way similar to that of the level measurement. Again the noise will be attenuated by the square root of the number of M-sequence periods applied. A detailed analysis shows that the enhancement in the signal-to-noise ratio will be given by the following equation:

$$\Delta_{S/N} = 10\log\left(\frac{T_{meas}}{T_{rev}}\right) + 11.4dB$$

where again $\mathbf{T_{rev}}$ and $\mathbf{T_{meas}}$ are the reverberation time and the total measurement time, respectively. For example, if one minute is used for the measurement of a reverberation time of around one second, the enhancement will be 29 dB or corresponding to an increase in the excitation power from 1 to 1000 watt.

8. Time invariance

The M-sequence may be described as a series of pulses of equal amplitude but appearing in positive or negative direction in a random manner. The measured signal will be the response to these pulses. The effect of the Hadamard transformation is to transfer the responses at different times and in different direction so they will add together simultaneously. It is therefore important that the system does not change during the measurement. The requirement to time-invariance is therefore very important and should always be considered when MLS techniques are applied.
In building acoustics it is common to move the microphone during a level measurement to obtain spatial averaging. This cannot be used when correlation or MLS techniques are used. If this is done, responses with different phases are averaged together and the mean value will be too low.

Elecroacoustic components, such as microphones, loudspeakers, power amplifiers and measurement channels are normally so stable that few problems due to time-invariance are likely to occur. However, the acoustical transmission may cause problems. The transmission of sound is sensitive to change in the environmental conditions as temperature, humidity and movement of the air by the wind. The temperature and humidity in a room are normally changing slowly. Time-invariance due to temperature and humidity is therefore more pronounced if the measurement is performed over a long period of time. Few problems related to change in atmospheric pressure, temperature and humidity are reported in the literature, but the phenomenon should always be considered when the MLS method is applied.

Time-invariance due to wind normally shows short-time fluctuation and is always a problem for the MLS technique. As the errors are caused by averaging impulse responses where the phase is not stable, the effect is most pronounced at higher frequencies. The wind will normally influence the sound transmission by altering the speed of the sound. The phase-error will therefore increase with the distance the sound has been transmitted. In a room the last part of the reverberation decay will be more sensitive to this effect and the recorded decay will decrease too fast in the last part of the decay.

Outdoor measurements with MLS should only be applied with the utmost care. Normally, the method cannot be applied to measurements of sound transmission over large distances. The sensitivity to wind will increase with the frequency and with the distance between the sound source and the receiver.

9. Linearity

The theory requires the system to be linear in order to be valid. The effect of unlinearities has been discussed in a number of papers (Ref). An analysis reveals that if the system includes unlinear terms, the impulse response will not decrease asymptotically to zero as it should for a linear system. The effect will therefore appear in the impulse response as irregular noise. However, unlike the case with "regular" noise, this error will not decrease by further averaging.

Analysis as well as experiments shows that the MLS method is not very sensitive to unlinearities and that the sensitivity will decrease by applying longer sequences. For room and building acoustic measurements, the sequence is normally long. Minor unlinearities of 1-3% will therefore not influence a level measurement if the integration period is limited to about half the reverberation time. The unlinearities may limit the dynamic range for reverberation curves. However, as the errors show up as noise in the reverberation curve, normal detection of background level errors will indicate this type of problems.

10. Practical realisation

The described MLS method to obtain level measurements and reverberation time measurements has been implemented in the digital real time analyser Norsonic RTA840. Most of the signal flow remains unmodified. The samples from the A/D-converter are stored. Synchronous averaging may be applied, after which the stored response is Hadamard transformed to obtain the broadband impulse response for the system to be investigated. The impulse response replaces the normal values from the AD-converter and further processing is based on the normal signal processing blocks. The impulse response is sent twice to the filter in order to prevent initial transients from corrupting the results.

In MLS mode, most of the signal processing blocks are retained. The normal procedure for testing the characteristics of the instrument is therefore appropriate. This may include tests for filter selectivity or level detector linearity.

The instrument operates with a sequence length of 2^{17}-1. The sampling frequency is selectable in binary steps from 64 kHz and downwards. For a sampling frequency of 16 kHz (measurements up to 5 kHz), the sequence length corresponds to 8.2 seconds. This allows recording of the impulse-response for a similar period of time.

The way the MLS method has been implemented in the analyser makes the use of it very similar to the use of the instrument in its ordinary operating mode. The user will not need a deep knowledge of the theory of M-sequences nor the Hadamard transformation. However, a certain experience in how unlinearities and extraneous noise show up in the results is always valuable to secure a reliable measurement.

The number of averages to obtain a reliable result may be estimated by a measurement of the background noise and the measured level with excitation.

The instrument has two channels opening up for measurements of two levels or two decays simultaneously.

As the broadband impulse response is stored in the instrument, it may be replayed in opposite direction. In this way the virtual reverberation times of the fractional octave filters are reduced significantly (1/10 at 50 Hz).

The instrument has a signal output feeding the M-sequence excitation signal. However, as this is a known signal, this signal may as well be generated by a stand-alone device. This will eliminate the need for cables between the source and the receiving room. An accurate frequency to control the sampling is essential, but this can be obtained by other means. The starting time of the sequence is normally required to find the beginning of the impulse-response. However, for a normal impulse-response for a room, the start point within required accuracy is easy to regenerate based on typical characteristics of the decay.

11. Conclusion

Experience with the described implementation, as well as a number of papers, has shown that the MLS method gives results similar to the standardized method for measurement of airborne sound reduction and reverberation time. The method has proved to give reliable results even in situations where the traditional method suffers from disturbance by extraneous noise. We therefore recommend that the MLS method should be allowed as a viable extension within the present formulation of the standard provided that it is implemented in a compatible way. Future standards should cover the described method more directly.

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