An improved MLS measurement system for acquiring room impulse responses

Olesen, Søren Krarup; Plogsties, Jan; Minnaar, Pauli; Christensen, Flemming; Møller, Henrik

Published in:
Proceedings of NORSIG 2000, IEEE Nordic Signal Processing Symposium

Publication date:
2000

AN IMPROVED MLS MEASUREMENT SYSTEM FOR ACQUIRING ROOM IMPULSE RESPONSES

Søren Krarup Olesen, Jan Plogsties, Pauli Minnaar, Flemming Christensen, Henrik Møller

Department of Acoustics, Aalborg University,
Fredrik Bajers Vej 7 B4, DK-9220, Aalborg, Denmark
acoustics@acoustics.auc.dk

ABSTRACT

Maximum-length sequences (MLSs) have proven very useful for the measurement of acoustical transfer functions. However, measurements are affected by background noise, nonlinearities and time variance in the system under test. These errors can cause a distorted impulse response, add spurious impulses to the measurement or reduce the effective signal-to-noise ratio. The origins of errors made in an MLS measurement system are described with focus on room impulse response (RIR) measurements. Practical solutions are given so that room impulse responses can be acquired with high immunity against nonlinear distortion and noise. Based upon this knowledge a PC-based measurements system was designed and implemented.

1. INTRODUCTION

A RIR can be defined as the time domain transfer function in a room from the electric loudspeaker terminals to the microphone terminals. It characterises the effect of the sound emitted by the source reaching the receiver point on a direct path and after having been reflected and diffused by the room boundaries. Measurements of RIRs are mostly used in architectural and building acoustics, since they give access to the acoustical properties of a room. A second interest for highly accurate RIR measurements arises from its application in auralization, where recorded signals are convolved with a RIR. Measurements made for auralization are usually done using an artificial head as receiver, i.e. there are two receiving microphones and a pair of RIRs referred to as a binaural room impulse response (BRIR). BRIR measurements were the motivation for this study.

2. THE MLS MEASUREMENT TECHNIQUE

Binary maximum-length sequences have certain properties that make them attractive as excitation signals for measurements of acoustical impulse responses. Several publications [1], [2] give detailed descriptions of the theory and the implementation of MLS measurement systems.

2.1. Maximum-length sequences

An MLS is a periodic sequence of binary numbers (±1). The magnitude of an MLS is the same at all frequencies except at DC. Using binary sequences as measurement signals ensures the highest possible excitation level and therefore high noise immunity. An MLS is created by reading from a linear shift register, where defined register outputs are summed (modulo 2) and fed back to the shift register input. MLSs are pseudorandom noise sequences known as m-sequences in coding theory. They are generated by the primitive polynomials over Galois fields of 2^m elements. The coefficients of the primitive polynomials determine, which registers are combined in the feedback structure. The length of an MLS or rather its period is $L=2^m-1$ where m is the order of the sequence. There exists several primitive polynomials and thus several sequences of each order, while they are different in the sense of being cyclically distinct, i.e. they cannot be matched by linear rotation. The calculation of the primitive polynomials over Galois fields is not trivial especially for high orders. Different polynomials for different orders have been calculated by Chabaud and Lercier [3].

2.2. Obtaining the impulse response

The linear transfer function of a system is calculated by a division in frequency domain,

$$\mathrm{H}(z) = \frac{\mathrm{Y}(z)}{\mathrm{X}(z)}. \tag{1}$$

In this way the excitation signal, X, is deconvolved from the system output, Y, to obtain the transfer function of the system, H. The division with X may cause problems if the magnitude response of the signal displays very low values at some frequencies. Hence, it is desirable to use an excitation signal that excites all frequencies equally much, such as an MLS. A sequence order m has to be chosen in advance depending on the expected length of the impulse response of the measured system. The length of the sequence L should be chosen sufficiently long to avoid time aliasing.

3. SOURCES OF ERRORS

There are several hurdles for the assessment of highly accurate measurements in practise; noise, nonlinear distortion and time-variance. A good description of the "pathology" of MLS measurement systems is given in [4]. While measurement noise affects the measurement result additively, the latter two sources of errors arise from the measured system violating the assumption of being linear and time-invariant.

3.1. Background noise

The acoustic background noise during a RIR measurement affects the measured response directly. Theoretically, an MLS provides the maximum excitation level for a given amplitude (low crest factor) and the highest obtainable SNR. The acoustical excitation level during the measurement is limited by the equipment used. However, care should be taken not to introduce nonlinear distortion (see next section). Interfering noise during the measurement adds noise distributed equally over the measurements time to the resulting impulse response. The common practice to reduce the affect of noise is to average repeated measurements. This averaging is preferably carried out on the system output, Y, before deconvolution (*pre-averaging*). The SNR improves by approximately 3 dB if the number of pre-averaged measurements is doubled.

3.2. Nonlinearities

If the system to be measured shows nonlinear behaviour, the measured impulse response will be falsified. This effect has been reported in many studies and has been well described and analysed by Dunn and Hawksford [5]. An example is shown in Figure 1; a RIR has been measured in a standard listening room using an 18th order sequence.

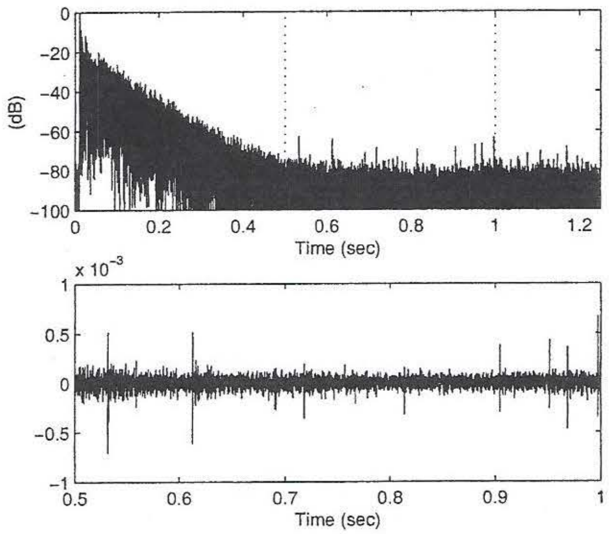


Figure 1. Top: Logarithmic plot of the absolute values of the room impulse response measured with a certain 18th order MLS. Bottom: Zoom on the impulse response from 0.5-1 s.

A closer look at the late part of the response reveals irregularities, which are impulsive and non-uniformly spread and cannot be interpreted as random background noise. These are caused by nonlinearities in the system, such as the electrodynamic driver of the loudspeaker. The position of the pulses cannot be predicted [6], unless the nature of the system's nonlinearity is known. However, for a certain system the appearance of these artifacts is directly associated with the particular MLS used, i.e. they occur at the same position in a repeated measurement, and the error due to nonlinearity cannot be reduced by simple averaging. If a longer sequence is used the error is distributed over the longer measurement period. Therefore, truncation of the impulse response reduces the error due to nonlinearity [5]. Figure 2 shows the same RIR as in Figure 1 but measured with a shorter (16th order) MLS, and the transient errors are more dense than those of Figure 1.

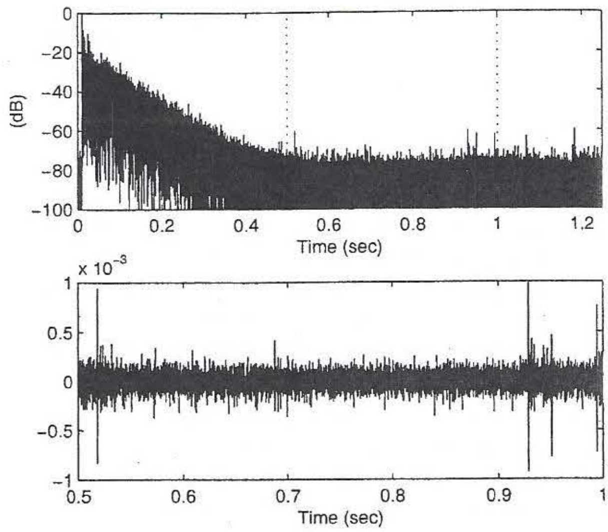


Figure 2. Impulse response measured with a shorter (16th order) MLS.

The amplitude of the observed artifacts is within the dynamic range to be considered. If the RIR is directly auralized, the artifacts are heard as clicks. Furthermore, it is shown by Vanderkooy [4] that the calculation of the reverberation time from the Schroeder plots of such perturbed impulse responses can lead to inaccurate results. Figure 3 depicts a measurement of the same channel using another, cyclically distinct sequence of 18th order. The same artifacts are observed, while the spurious impulses appear at different positions. Hence, if the two measured responses shown in Figures 1 and 3 are averaged, these noise peaks are reduced by 6 dB.

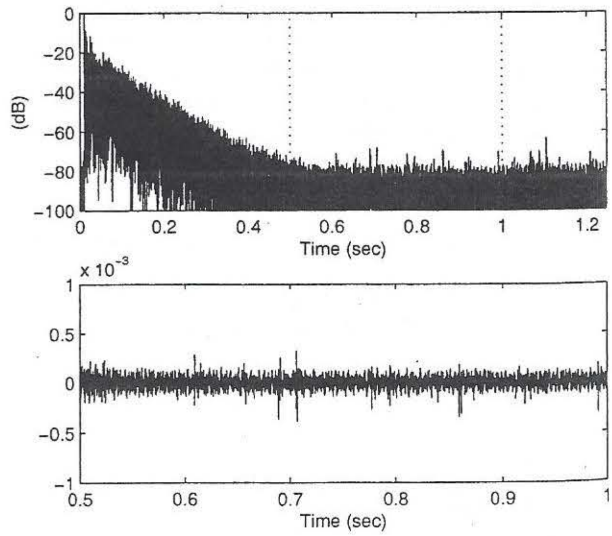
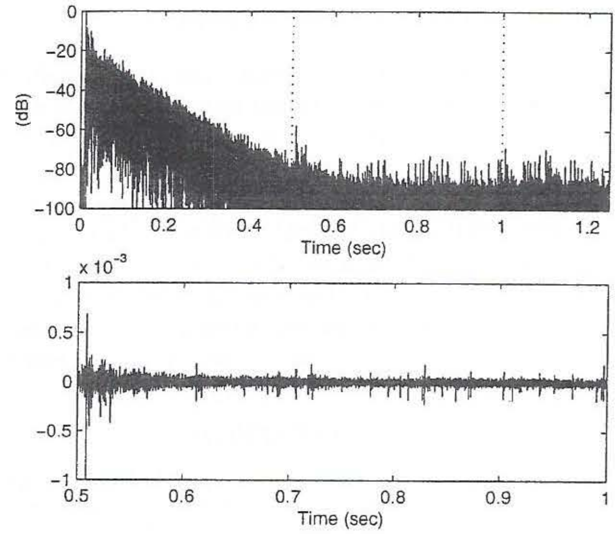
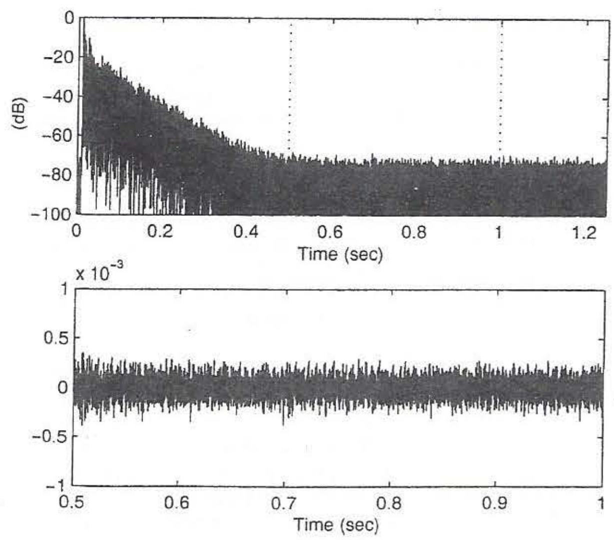


Figure 3. Top: Logarithmic plot of the absolute values of the impulse response measured with a different (compared to Figure 1) 18th order MLS. Bottom: Zoom on impulse response from 0.5-1 s.

The proposed solution is to perform repeated measurements using different MLSs and average the responses H of these measurements (*post-averaging*). Each time the number of post-averages is doubled, the noise due to nonlinearities is reduced by approximately 6 dB. Evidently post-averaging (like pre-averaging) also enhances the SNR with respect to background noise. Rife [7] pointed out that a measurement can be made that exhibit complete immunity against even-order distortion if the Volterra theory of nonlinear systems is applied. An MLS measurement is performed followed by a measurement using the same sequence with reversed polarity. Subtracting the resulting impulse responses of the two measurements from each other yields a response that is unaffected by even-order distortion. Care should be taken that the system behaves as linear as possible during the measurement, e.g. by avoiding too high excitation levels. On the other hand the excitation levels should be kept high enough to reduce the influence of background noise. An optimal gain value can be found as illustrated in Figures 4a-4c. Figure 4c shows that the errors introduced are approximately 80 dB below the peak value of the RIR.





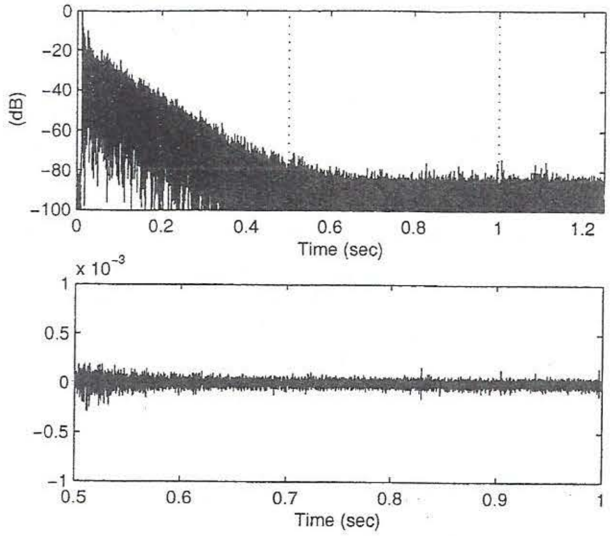


Figure 4a-c. RIR measurements with 18th order sequences (4 pre-averages, 4 post-averages) for different gain settings. 4a) High gain leading to nonlinearities. 4b) Low gain causing high background noise level. 4c) Proper gain setting yields high overall SNR.

3.3. Time variance

Another type of error that affects the impulse response measurement is time variance; well described by Svensson and Nielsen [8] who studied *interperiodic* and *intraperiodic* time-variance effects. Interperiodic time variance are effects that occur between two measurements, like slow temperature changes, and intraperiodic time variance refers to effects that occur within an MLS period, such as clock jitter and wind. Both types of time variance introduce an error with a +12 dB per octave frequency characteristic. Since this error is localised around the impulse response, it cannot be reduced by averaging or truncation. Also a noise-like error with a +6 dB per octave characteristic is introduced by intraperiodic time variance. This component is spread in time and can therefore be reduced by both averaging and truncation of the impulse response. Since the +6 dB per octave characteristic represents energy which has been removed from the system response, the presence of such noise reveals that the measured impulse response is distorted.

Time variance during MLS measurements can be a significant source of errors in reverberant rooms [8]. Especially the errors caused by intraperiodic time variance can dominate the apparent noise at high frequencies. In order to examine this, the background noise in the room can be compared to the apparent noise in the measurement by inspecting the initial part of the impulse response. Averaging should be used only at frequencies where the background noise dominates the apparent noise, since averaging decreases the random noise at the expense of a more distorted system response.

4. THE IMPLEMENTED SYSTEM

A system for measuring BRIRs has been implemented, and for that purpose two synchronised channels were required. The system software runs on a PC with two soundcards installed (16 bits, 48 kHz). The exact timing of record and playback cannot be controlled with sample accuracy using standard soundcard drivers. Thus, in order to ensure complete synchronisation between input and output, the MLS is fed back directly to the soundcards left channels. The MLS is played by one soundcard and the stereo system's output is sampled by the right channels of the two soundcards (see Figure 5).

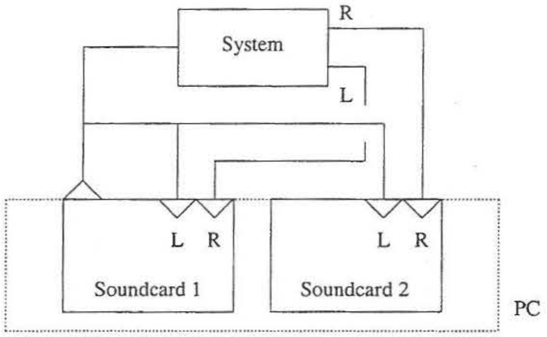


Figure 5. The soundcard setup.

On the input side, the left and right channels of the soundcards are always synchronous, so the two responses are easily obtained by deconvolving the right input, Y, with the left input, X. In advantage, the converters involved are effectively taken out of the measured system. A discrete Fourier transform of the two signals was performed using a non-power-of-two algorithm [9]. The order of the MLS can be chosen from 12 to 20 corresponding to a sequence lengths of approximately 0.1 s to 20 s depending on the sampling frequency, and also the number of pre-averages and post-averages can be set. An example of a complete measurement sequence with 2 pre-averages and 2 post-averages is outlined in Figure 6a. Figure 6b shows the recorded system output signal. The first block consists of 4 repetitions; sequence A followed by 4 repetitions of sequence B.

In general, one sequence is required in the beginning and one in the end of every pre-averaging block. The first sequence ensures that the system is in a stationary state of excitation [1]. The last sequence is appended to guarantee that the system is still excited, when the system output is selected for deconvolution. The next block of sequences (type B) follows after a pause of length L to let the system response decay before exciting it with a new sequence. The overall system response is then calculated as follows: The system input sequences X_{AI} and X_{AII} are pre-averaged to form X_A, and also the two output signals Y_{AI} and Y_{AII} starting at exactly the same point in time are pre-averaged to form Y_A. Deconvolution with X_A yields H_A, and similar for H_B. Post-averaging H_A and H_B forms the final result H. In this case, the gain in suppression of noise due to nonlinearity and the background SNR should both be improved by approximately 6 dB.

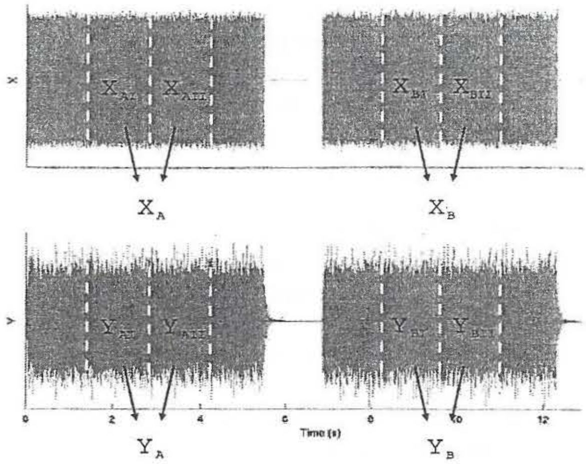


Figure 6a-b. 6a) System input signal, X for 2 pre-averages and 2 post-averages. 6b) System output signal, Y.

5. CONCLUSION

The MLS measurement technique has proven useful for measurements of room impulse responses. Errors due to acoustic background noise can be efficiently reduced by averaging the output from a system driven by MLSs. By applying different (cyclically distinct) MLSs to the system, errors caused by nonlinearities can be reduced as well. These errors are often more severe than background noise errors. A PC based measurement system employing two low-cost soundcards was developed. It was shown that high quality 2-channel room impulse response measurements can be made utilizing this system.

6. REFERENCES

[1] J. Borish, J. B. Angell, "An efficient algorithm for measuring the impulse response using pseudorandom noise", *J. Audio Eng. Soc.*, vol. 31(7), pp. 478-488, July/August 1983.

[2] D. D. Rife, J. Vanderkooy, "Transfer-function measurements with maximum-length sequences", *J. Audio Eng. Soc.*, vol. 37(6), pp. 419-444, June 1989.

[3] F. Chabaud, R. Lercier, "Tables of polynomials over Galois fields", *http://www.dmi.ens.fr/~chabaud/Poly/GF.html*, Département d'Informatique, École Normale Supérieure, Paris, France.

[4] J. Vanderkooy, "Aspects of MLS measuring systems", *J. Audio Eng. Soc.*, vol. 42(4), pp. 219-231, April 1994.

[5] C. Dunn, M. O. Hawksford, "Distortion immunity of MLS-derived impulse response measurements", *J. Audio Eng. Soc.*, vol. 41(5), pp. 314-335, May 1993.

[6] M. Wright, "Comments on: 'Aspects of MLS measuring systems'", *J. Audio Eng. Soc.*, vol. 43(1/2), p. 48, January/February 1995.

[7] D. D. Rife, "Comments on 'Distortion immunity of MLS-derived impulse response measurements'", *J. Audio Eng. Soc.*, vol. 42(6), pp. 490-491, June 1994.

[8] U. P. Svensson, J. L. Nielsen, "Errors in MLS measurements caused by time variance in acoustic systems", *J. Audio Eng. Soc.*, vol. 47(11), pp. 907-927, November 1999.

[9] FFTW, a Fast Fourier Transform development library at *http://www.fftw.org*