MLS IMPULSE RESPONSE MEASUREMENTS FOR UNDERWATER BOTTOM PROFILING

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This work describes the use of the MLS (Maximum Length Sequence) signal for the measurement of impulse responses, in alternative to the traditional techniques based on the use of impulsive sources.

The mathematical theory for the generation of the MLS signal and for the deconvolution of the system's impulse response are first described. Then a software implementation of the measurement method, based on the creation of plug-ins for a shareware waveform editor on a low cost PC is demonstrated.

Some preliminary experiments conducted under controlled conditions show that the proposed technique produces images of the bottom profile with higher signal-to-noise ratio and better spatial resolution than those obtainable with the impulsive technique.

Furthermore, the new technique seems superior also for some practical aspects connected with the very nature of the MLS signals, which makes it almost inaudible and makes the sound source barely localizable: this is very important for military applications.

1. INTRODUCTION

Impulsive techniques are used from the beginning of the Underwater Acoustic science, to evaluate the distance of objects and to plot the profile of the sea bottom.

Although the use of impulses has many advantages, it causes also some problems: to obtain high signal-to-noise ratios very large impulse amplitudes are needed. These can be generated efficiently only through resonating transducers, which have of course only limited bandwidth.

When wide-band pulses are required, usually a high power sparker is employed. Anyway also this kind of source has some well known defects: it requires a large equipment, it is expensive, and the pulses produced are not very repeatable.

In this work the use of an alternative approach is presented: a small wide-band piezoelectric transducer is used as an underwater loudspeaker, being fed with continuous MLS (maximum length sequence) pseudo random signal [1,2,3]. A second transducer (a wide band hydrophone) is used as the receiving sensor, located near the sound source. The couple of source and receiver is moved on a straight line at constant depth, scanning the bottom profile.

A personal computer samples the signal coming from the hydrophone, and through the well known Hadamard transform it deconvolves a series of impulse responses in real time. A standard visualization tool is then used to map the sequence of impulse responses (with logarithmic amplitude) in pseudo-color plots, which enables the direct visualization of the sea bottom profile.

Through a proper synthetic aperture focalization algorithm, the transducer directivity can be greatly increased. Furthermore, the vertical resolution can be improved employing a modified version of the MLS signal, preconvoluted with the inverse impulse response of the transducers, in such a way that the emitted signal deconvolves to an almost perfect Dirac's Delta function., as suggested by Mommertz [4].

The paper presents the theory behind the process stated above, and the first experimental results obtained in a real, highly reverberant environment. The comparison with experiments conducted employing the same experimental setup but with impulsive signals, shows that the novel technique greatly reduces the noise contamination of the signals, making it possible to work with lower power and without any averaging.

2. THEORY

The MLS signal is well known since at least two decades [1,2,3]: it is a binary sequence, in which each value can be simply 0 or 1, obtained by a shift register as the one shown in fig. 1. The obtained signal is periodic, with period of length L given by:

$$L = 2^{N-1} \tag{1}$$

in which N is the number of slots in the shift register, also called the order of the MLS sequence. Thus an order N=16 means a sequence with a period of 65535 samples. In this work, a software MLS generator is employed: it can produce sequences up to order 21, which correspond to 2,097,151 samples, or 47.55 s at a sampling rate of 44.1 kHz.

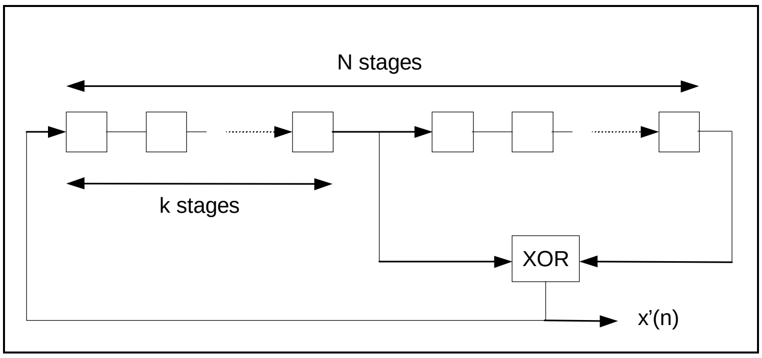


Fig. 1 – Shift register for the creation of the MLS signal

Another very important point is the position of the tap inside the shift register: it is possible to generate MLS sequences also with multiple taps, and the position of the taps influences the behavior of the sequence, particularly when it is used for the excitation of system which are not perfectly linear. The work of Vanderkooy [5] suggests that some sequences are better than others, reporting a list of known taps positions which works reliably for various orders of the MLS sequence. All these are included in the software module, and the user can choose between the various available sequences for any order.

After having sampled the response of the system to the excitation signal, this response has to be processed, for extracting the system's IR. Thanks to the favourable mathematical properties of the MLS excitation signal, the deconvolution of the IR can be made with the well known Fast Hadamard Transform (FHT), as originally suggested by Alrutz [1], and clearly explained by Ando [2] and Chu [3]. The Ando formulation was employed here. The process is very fast, because the transformation happens "in place", and requires only addition and subtractions. The computations are done in floating point math. Here a brief description of the method is presented.

When the periodic MLS signal m(j) [j=0..L-1] is applied at the input of a linear system, characterized by an impulse response h(i) [i=0..P], in which P<L, the measured output signal y(k) can be interpreted as the convolution of the excitation signal with the system's impulse response:

$$y(k) = \sum_{j=0}^{L-1} h(j) \cdot m(k-j) \tag{2}$$

Let we transform eq. 2 in matrix notation, defining a matrix M so that:

$$M(i,j) = m[(i+j-2) \bmod L] \tag{3}$$

Thus, eq. 2 becomes simply:

$$\{y\} = [M] \cdot \{h\} \tag{4}$$

To obtain the impulse response h, we find the inverse matrix of M, named $\tilde{M}$:

$$\tilde{M}(i,j) = M(i,j) - 1 \tag{5}$$

Thanks to the mathematical properties of the MLS sequence, it can be shown that the product of this inverse matrix with the original one produces a slightly modified unit matrix:

$$\tilde{M} \cdot M = (L+1) \cdot I \tag{6}$$

in which I is the identity matrix. Thus, for extracting h from y, it is required to compute:

$$h = \frac{1}{L+1} \cdot \tilde{M} \cdot y \tag{7}$$

The above process is effective, but computationally quite heavy. There is a trick for obtaining the same result with a very little number of mathematical operations. Let we introduce a larger square matrix U, of order L+1:

$$U = \begin{bmatrix} 1 & K & 1 \\ M & M & \\ 1 & & \end{bmatrix} = \begin{bmatrix} 0 & K & 0 \\ M & \tilde{M} & \\ 0 & & \end{bmatrix} + \begin{bmatrix} 1 & K & 1 \\ M & 1 & M \\ 1 & \Lambda & 1 \end{bmatrix} \tag{8}$$

This makes eq. 7 to rewrite as

$$\begin{Bmatrix} 0 \\ h \end{Bmatrix} = \frac{1}{L+1} \cdot \left(U \cdot \begin{Bmatrix} 0 \\ y \end{Bmatrix} - \begin{Bmatrix} \bar{y} \\ \vdots \\ \bar{y} \end{Bmatrix} \right) \tag{9}$$

in which $\bar{y}$ is simply the sum of the L measured samples y(k).

The matrix U can be expressed in terms of an Hadamard-type matrix H, thanks to a pair of permutation matrixes P and P^T:

$$U = p^T \cdot H \cdot p \tag{10}$$

The multiplication of the Hadamard matrix H with the measured vector y can be done very efficiently through the Fast Hadamard Transform (analogous to the Fast Fourier Transform) algorithm. Thus, the steps for deconvolving the impulse response from the measured signal y are:

1. generate the permutation vector p from one period of the MLS sequence with the well-known technique of Cohn and Lempel [6]
2. reorder the elements in y according to the permutation vector p
3. Add a zero element at the beginning of the permuted vector, for making it of length L+1; let we call y' this modified vector
4. apply the Fast Hadamard Transform to y': the result is a "shuffled" version of h, called h', in which the first element is always zero
5. throw away the zero at the beginning of h', and reorder backward the elements following the inverse permutation described by p.
6. After subtracting the DC offset $\bar{y}$ and adjusting the scale factor dividing by (L+1), the wanted impulse response is obtained.

It must be noted that the generation of the permutation vector p can be done once, and stored for every future use of the same MLS signal. Thus the only heavy computation is the Fast Hadamard Transform, which requires only $L \cdot \log_2(L)$ summations: on modern PCs, this task can be easily done in real time, even for very long MLS signals. This means that the continuous acquisition of the signal can be done simultaneously with its processing, and even with the visualization of the subsequently deconvolved impulse responses, making use simply of the computer's CPU and of a standard sound board (nowadays incorporated in any notebook computer).

3. SOFTWARE IMPLEMENTATION

Both the generation of the MLS signal and the deconvolution of the impulse responses have been implemented in specially made software plug-ins, which are added to the shareware program CoolEdit by David Johnston [7]. The first plug-in is simply a generator of MLS signals; after the signal is generated, an instance of CoolEdit is maintained in playback (in loop mode), continuously emitting the excitation signal.

A second instance of CoolEdit is then opened, and the signal going to the input(s) of the sound board are sampled; once chosen the MLS sequence to be deconvolved (which has obviously to be the same used for excitation), the deconvolver plug-in transforms the sampled signal in a sequence of impulse responses. It is possible to extract only a portion of each impulse response, for reducing the memory consumption and maintaining only the required information.

After the measurement is done, a separate visualization programs makes it easy to display the sequence of impulse responses as a traditional sonar graphical plot; a new version of the measurement software is under development, capable of real-time display of the sonar plot during the measurement.

4. EXPERIMENTS

A preliminary experiment was undertaken for checking the feasibility of the new technique. The experiment was conducted in a large pool, in which a car was placed on the bottom (this was the object to visualize). A rudimental mechanical scanning equipment was attached over the pool, moving at a speed of 0.05 m/s.

A small piezoelectric loudspeaker and a waterproof microphone were attached to the movable platform, and connected directly with the input and output connectors of the audio board of a notebook PC, capable of 16-bits full-duplex operation.

The measurement was performed twice; the first time the continuous MLS signal was employed, at a sampling rate of 44.1 kHz and with a period L equal to 32767 samples (0.75 s long). The system’s response was deconvolved to a series of impulse responses, storing only the first 2048 points of each of them. The extra length of the sequence was required because the pool was quite reverberant, and the use of a shorter sequence caused serious time aliasing.

The second measurement was made generating a series of periodic pulses, with a repetition rate of 32767 samples. Also in this case, only the first 2048 samples of each impulse response were stored.

The same settings were employed for processing the two measurements and for transforming them in sonar images: each impulse response was first logarithmically transformed, then each sample was mapped to a 256-levels grayscale. The result was assembled in a matrix of pixels, in which each vertical line (from left to right) represents an impulse response, with the zero of time at the top of the image.

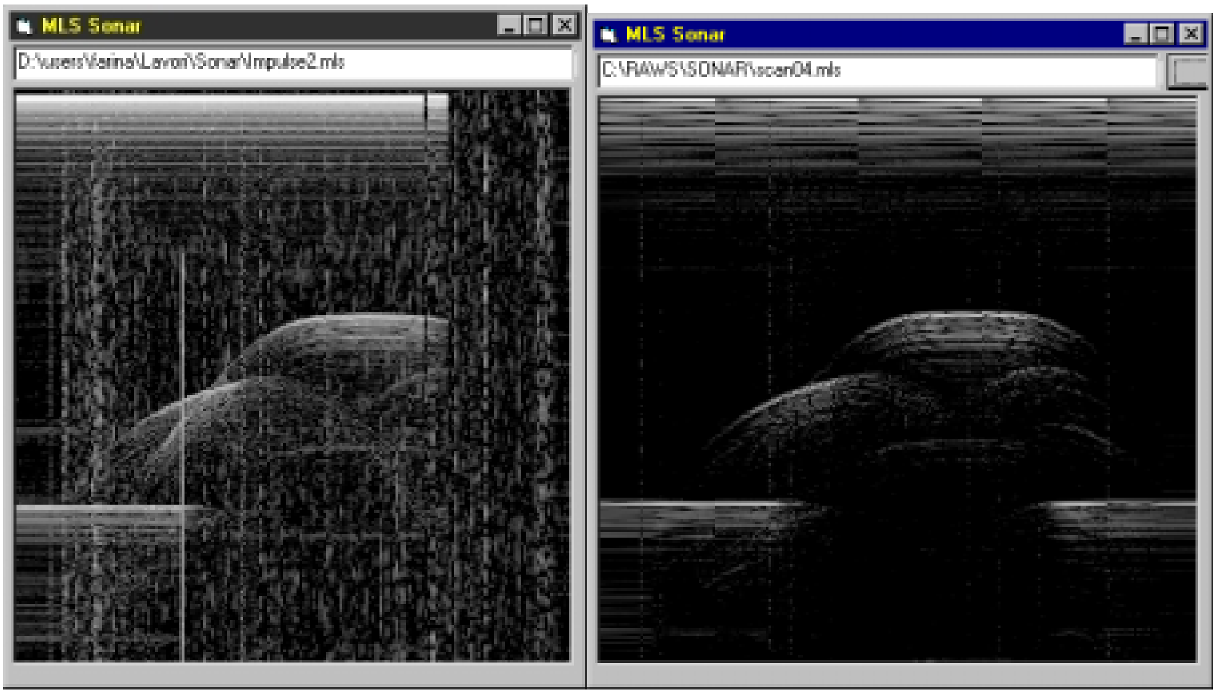
Figures 2 and 3 show the images obtained with the two different techniques.

It is obvious from them that the signal-to-noise ratio of the MLS technique is largely better than with impulses: this is easily explained with the fact that the continuous MLS signal radiates an amount of energy which is very large compared with the energy of a single pulse.

Nevertheless some artifact is evident also with the MLS signal: these are due to missed samples from the sound board, which was not able to sustain such an high bi-directional data flow. Due to the presence of these missed samples, any attempt to improve the resolution of the images making use of the synthetic aperture focalization was unsuccessful.

In the prosecution of the work the notebook PC will be substituted with a new one, equipped with a more powerful CPU and with an hardware buffered sound chip: this should completely remove the above problem.

Furthermore, it was not attempted to linearize the response of the transducers creating a proper inverse filter pre-convoluted with the excitation signal, as suggested by Mommertz [4], although the software for performing this task was already developed [7].



Figg. 2,3 – Images obtained with the impulsive technique (left) and with the MLS (right)

5. CONCLUSIONS

The first results of this research show that the MLS signal is very appealing for underwater measurements, due to his high immunity to the external noise and to the exceptionally easy and fast processing required for the deconvolution of the impulse responses. The software implementation on a notebook PC resulted in a powerful, low-cost system.

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