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PRINCIPLES AND PRACTICE IN MAXIMUM LENGTH SEQUENCE ANALYSIS

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1. INTRODUCTION

The paper gives an overview of the principles of Maximum Length Sequences applied to acoustic testing and its implementation in the MLSSA (Maximum Length Sequence System Analyser) system - a commercially available Audio test instrument. Some of the applications and practicalities of the system in use are then covered. By necessity much of the work in this paper has already been covered by Doug Rife at DRA Laboratories (1), the designer and manufacturer of the system.

2. MAXIMUM LENGTH SEQUENCES.

The idea behind MLS dates back at least 2 decades and was shown to be a practicable measurement method some time ago, but it was quite recently that the first commercially available instrument was developed by DRA Laboratories.

A binary maximum length sequence is a periodic two level pseudo random sequence of length $2^N - 1$ where N is an integer. These can be generated by digital shift registers.

The sequence can be thought of as a series of weighted delta functions (values of 1 or -1). These are physically unrealisable and so are convolved with a unit pulse function (a zero order hold function). The resultant boxcar waveform is then realisable using a 1 bit D/A converter(see figure 1).

MAXIMUM LENGTH SEQUENCE ANALYSIS

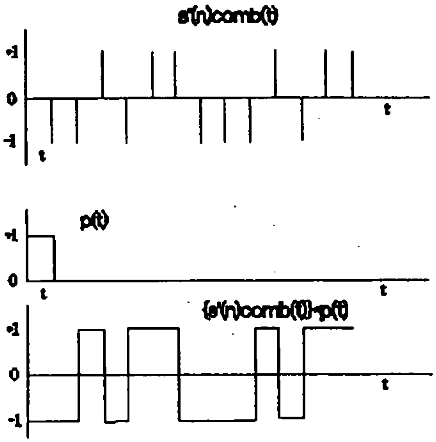


Fig. 1

After application to the system under test a circular cross correlation is performed to derive the Periodic Impulse Response of the system under test. For a perfectly linear noiseless system this PIR is identical to one obtained by periodic pulse excitation. There is now however a vast improvement in noise and distortion immunity. Note that because the MLS sequence is non random and entirely repeatable it is not necessary to sample the sequence sent out as with Dual Channel FFT analysers.

The measured PIR can in turn be considered identical to the impulse response provided time aliasing effects are eliminated. Time aliasing occurs if the impulse response of the system has not fully died away before the MLS finishes. Then the remaining part of the response is effectively added back at the start of the impulse response causing inaccuracies. Thus the MLS period should be greater than the impulse response of the system under test. In fact it was found that the minimum measurement times is two full MLS periods with the period being long enough to render time aliasing effects negligible. The first period allows the system to settle while the second is used for the actual measurement.

Some of the features of this method are as follows:

1- DC Offset.
The MLS essentially rejects dc. Most systems are ac coupled anyway but any residual dc offsets are severely attenuated in the cross correlation.

2- Truncation effects.
The MLS is periodic, so providing exactly one MLS period is used there are no truncation effects so no data windows are required.

3- Crest factor.
Theoretically = 0dB thus measurements have a high energy content giving high signal to noise ratios. Passing the MLS signal through an average linear system gives the signal a near Gaussian Probability Density Function. This equates to a crest factor of approximately 11dB. This can be improved by the use of the Pre averaging and pre emphasis of the signal.

4- Transient noise immunity.
It is shown that transient noise is transformed into low level benign noise spread across the PIR thus transient noise immunity is very high.

5- Non Linearity in MLS measurements.
Is shown to cause quasi-stationary fixed pattern noise to appear in the PIR. The effect of this is minimised by using an MLS period much longer than the chosen analysis window. Then most of this noise is lost in the unused tail of the PIR.

Pre averaging is simply averaging the raw sampled data of a number of MLS periods. This reduces any random noise in the measurement. 16 averages gives a 12dB improvement in Signal to noise.

Pre emphasis involves shaping the MLS sequence. The MLS has an energy characteristic similar to white noise so on a wideband measurement there can be little energy at low frequencies. Unfortunately most disturbing background noise occurs at low frequencies (traffic rumble, Air handling equipment etc.). By filtering the MLS on output and "unfiltering' it when it comes back an improvement can be achieved. It is found that a simple pink noise filter gives a near ideal characteristic.

So the PIR obtained using the MLS and circular cross correlation is essentially the complete description of the system under test. This data can be stored and further and exhaustive analysis of it can take place at a later date.

3. MLSSA. IMPLEMENTATION OF MLS

DRA Laboratories produced the first commercial MLS instrument called MLSSA. In order to maximise flexibility and minimise cost it was designed around the IBM-PC format. The system consists of an IBM standard size expansion card and appropriate controlling software.

The card houses the MLS generator, A 12 bit A to D converter and an 8 pole programmable anti-aliasing filter. The 12 bit resolution is improved to an effective 16 bits by the use of analogue dither. A number of anti-aliasing filter alignments are available. The Chebychev for example gives maximum roll off at the expense of passbard ripple and phase distortion while the Bessel gives very little phase distortion at the expense of a drooping frequency characteristic.

All calculations are performed by the host computer (a math coprocessor is a necessity) DSP hardware is not used. The advantage here is that as PC technology improves, MLSSA's performance improves with it. On a 80386 20 MHz system the cross correlation algorithm completes in 3.5 seconds for a 65535 point MLS.

Menu driven software is provided to do a wide range of measurements. Low level software is also provided for user programming of the system for special purposes. The software includes a 32kpt FFT routine for measurement in the frequency domain. The following table gives an idea of MLSSA's capabilities.

Bandwidth	Sample Rate	MLS Length	Impulse length	FFT size	Frequency resolution
1kHz	4kHz	64kpt(16.38s)	32kpt(8.195s)	32kpt	0.12Hz
10kHz	30.1kHz	64kpt(2.179s)	32kpt(1.090s)	32kpt	0.92Hz
40kHz	117.6kHz	64kpt(557ms)	32kpt(278ms)	32kpt	3.59Hz

Fig 2 shows the main measurements available from MLSSA.

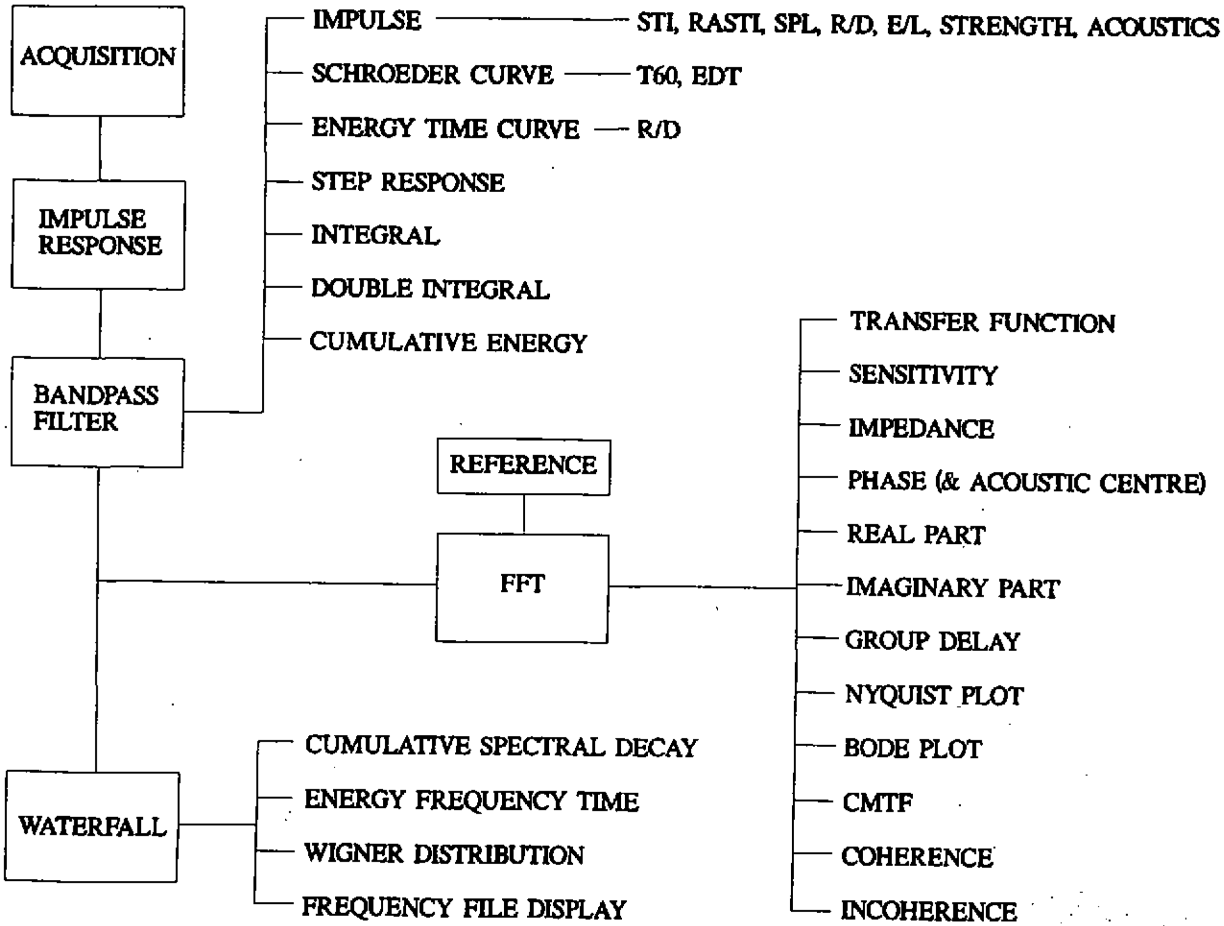
MLSSA can also operate in scope mode where the cross correlation is disabled. Here it becomes a digital storage scope/single channel FFT analyser.

4. APPLICATIONS

MLSSA can be applied to many linear two port systems ranging from auditoria through loudspeakers to electrical circuits. You can even set up tape machines with it. To illustrate the operation of the system in use two common testing scenarios will be considered.

4.1 'Anechoic' loudspeaker measurement

The aim here is to get an anechoic measurement of a loudspeaker without actually using an anechoic chamber. This can be done by selecting just the initial 'reflection free' part of the impulse response and applying the FFT to that part only. First various measurement parameters are set up on the system:


ACQUISITION
IMPULSE RESPONSE
BANDPASS FILTER
IMPULSE
STI, RASTI, SPL, R/D, E/L, STRENGTH, ACOUSTICS
SCHROEDER CURVE
T60, EDT
ENERGY TIME CURVE
R/D
STEP RESPONSE
INTEGRAL
DOUBLE INTEGRAL
CUMULATIVE ENERGY
REFERENCE
FFT
TRANSFER FUNCTION
SENSITIVITY
IMPEDANCE
PHASE (& ACOUSTIC CENTRE)
REAL PART
IMAGINARY PART
GROUP DELAY
NYQUIST PLOT
BODE PLOT
CMTF
COHERENCE
INCOHERENCE
WATERFALL
CUMULATIVE SPECTRAL DECAY
ENERGY FREQUENCY TIME
WIGNER DISTRIBUTION
FREQUENCY FILE DISPLAY

Measurement Bandwidth and Anti-Aliasing filter type. Once these are set MLSSA automatically sets an appropriate sampling rate at 3 or 4 times the required bandwidth.

MLS period. Normally set to the maximum 65535 points to guard against time aliasing effects.

Acquisition Length. Although we have selected a 65535 point MLS we can choose how much of this we process into an impulse response. For this particular measurement we only need a short impulse length as we are only interested in the response of the loudspeaker itself. We might choose to process say 1024 points only. Note that once this is done the remaining unused points are lost. The benefits are faster processing and improved rejection of any noise due to system non-linearities.

Output level. Normally adjusted to be loud enough for good signal to noise ratio without overdriving the system.

Once these are set the measurement is triggered from the keyboard. The MLS is passed into the loudspeaker via a power amplifier, collected via a microphone (& preamp) and returned to MLSSA. MLSSA will first adjust the input gain to make best use of the A/D converter. It will then acquire the full MLS and perform the cross correlation. The impulse response (of pre selected length) is then displayed on the monitor. At this stage the impulse response can be saved to disc as all subsequent analysis is post-processing and can be done at any time. Figure z shows a typical impulse.

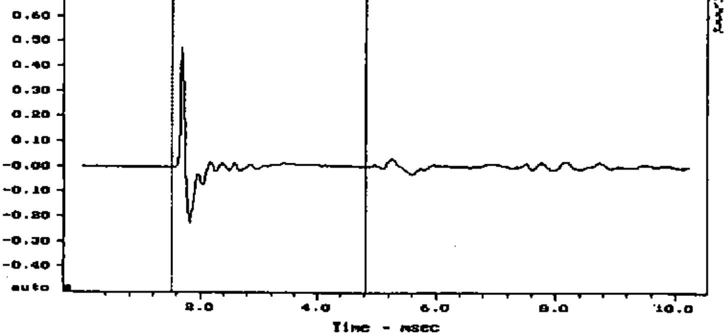


Fig 3. Windowing the impulse response

To get the anechoic frequency response the two cursors are positioned to select the portion of the impulse that contains only the direct sound and no reflections. The FFT is then applied to this section only. The problem here is that as the analysis window is shortened the true frequency resolution is worsened. For example a 10ms section of impulse can only give a frequency resolution of 100Hz no matter how big an FFT you apply to that section. So in this case you might decide that the resulting frequency response can only be used above say 500Hz if any degree of accuracy is to be maintained. Thus like all measurement systems the lowest frequency that an anechoic measurement can be obtained at is determined by the arrival time of the first reflection ie. the size of the room.

To improve the accuracy of the measurement the output of the power amplifier can be measured and its frequency response stored as a reference. When the loudspeaker response is subsequently measured the response of the power amplifier is removed as are the characteristics of the anti-aliasing filter. Pre-averaging can also be employed to improve signal to noise ratio.

Figure x shows that many curves can be obtained from this single measurement, all after the event. Phase would be of interest to a loudspeaker designer, software allows minimum, excess, wrapped and unwrapped phase to be displayed. An additional routine allows the elimination of any phase delay in the measurement (eg the 'time of flight' between loudspeaker and microphone). Even the 3D waterfall plots are derived from this single measurement.

4.2 Room measurement

The measurement parameters would be set up. This time the maximum MLS length is more of a necessity (depending how reverberant the room is of course). Normally the maximum 32768 points of impulse would be displayed so that the full decay of the room could be seen. Pre averaging is especially useful here for improving low frequency noise eg. air conditioning noise, traffic rumble etc. A loudspeaker is still required to excite the room and it should be chosen to cover the frequency band of interest. MLSSA would find it difficult to recover the 63Hz band RT if the loudspeaker rolls of at 100Hz. The measurement would be triggered as before.

Once the long impulse is displayed (a 10kHz bandwidth measurement would give a 1 second impulse length) much data can be obtained from it. A number of things can be calculated direct from the impulse. These include Early/Late ratio, Reverb/Direct ratio, Source strength, SPL, STI & RASTI (note that for best accuracy some shaping of the MLS is recommended for these). There is an Acoustics page that gives an octave band analysis of many of these parameters plus EDT and Reverb time over several decay ranges.

When in the time domain measurements can be band filtered using a software programmable filter. Thus the full range of octave band Schroeder decay curves can be generated or third octave even. It is when doing these filtered decay curves that signal to noise problems are most likely to be encountered, usually at low frequency. MLSSA includes a noise compensation routine here which assesses the background noise in the initial 'time of flight' gap then applies a correction to the decay curve. To further improve signal to noise use pre-averaging or lower the measurement bandwidth so that relatively more energy is present at low frequencies.

These examples hopefully give an indication of the operation of MLSSA in practice. There are of course many other applications of MLSSA but it is not possible to discuss all of them here.

5. CONCLUSION

The MLSSA system is an effective implementation of the MLS measurement method. The method shows good immunity to noise and non-linearity. The software supplied caters for a wide range of measurements and applications and is simple to operate due to the menu format. System upgrading is a simple case of copy a disc with new software.

6. ACKNOWLEDGEMENT

We would like to acknowledge the work of Doug Rife of DRA laboratories for his work in designing and manufacturing MLSSA.

7. REFERENCES

(1)Douglas D Rife and John Vanderkooy, "Transfer-Function Measurement with Maximum Length Sequences," JAES Vol 37, N0. 6 1989 June

DRA Laboratories, MLSSA Handbook