When using ML sequences for LTI system's measurements you should **keep in mind, that the MLS method is accurate only for LTI systems** meaning their transfer function does not change over the measurement interval. Therefore, you should not attempt to make room measurements while the room acoustics are changing. This might occur if enough people are moving around a room during a measurement session. The opposite is also true – if your sound card (or PC) is unable to keep the In/Out timing accurate, the MLS measurement system will not work.

Basic MLS/FFT specification

The program will generate 6 different MLS sequences and use associated FFTs as shown below:

MLS order	MLS length	FFT length
13	8191 points	8192 bins
14	16383	16384
15	32767	32768
16	65535	65536
17	131071	131072
18	262143	262144

Smoothing: 1/3, 1/6, 1/12, 1/24 and 1/48 octave smoothing
Sampling: 11025, 12000, 22050, 24000, 44100, 48000, 96000 (192000 where applicable).

The MLS signal is periodic with period 2^N -1 where N is the order of the sequence. The MLS generator is clocked by the sample rate clock. Therefore, the period of a MLS, in seconds, is equal to the sampling interval times the sequence period in points. For example, for a 48 kHz sampling rate (20.8 microseconds sampling interval) a 262144-point MLS will have a period of 5.452 seconds.

The MLS can be regarded as a train of pulses of equal amplitude but appearing in positive or negative direction in pseudo-random manner. Such signal is fed into the loudspeaker, so that he microphone will pick up the response to these pulses. The application of Hadamard transform, causes the individual responses at different times to be re-arranged, so they appear simultaneous and in one direction. It is now easy to understand, why the LTI requirement for the system – it must not change during the measurement.

True Frequency Resolution

The system transfer function is computed by applying an FFT to the impulse response or a portion of it as in the case of windowed, anechoic loudspeaker measurements. The displayed frequency resolution is given as the sample rate divided by the FFT size (i.e. 48000/262144=0.183Hz). Displayed frequency resolution is just the FFT bin separation. However, the true frequency resolution of a measurement is **at best equal to 1/T** where T is the length of the impulse response segment used in the FFT calculation (i.e. the width of a rectangular time window). **Thus, it is possible to have a very fine displayed resolution but a rather coarse true resolution.** For example, you could apply a 32768-point FFT to a 512-point segment of impulse response. In that case, the large FFT only acts to interpolate the true frequency resolution to form a much finer displayed resolution.

Data-tapering window shapes would yield even worse resolution than a rectangular window. In the frequency domain, SoundEasy indicates the true frequency resolution by **a gray-shading the invalid section of the graphics display**. The data falling in the grayed frequency range should be regarded as invalid.

For example, if the time window T = 4 milliseconds the lowest accurate frequency is 250 Hz. The true frequency resolution at all higher frequencies will, in this case, also be 250 Hz. Therefore, the next correctly measured and calculated data point will be at 500Hz, and the next at 750Hz, and so on…..
Please note, that between 250Hz and 750Hz there is only ONE valid data point !.

To obtain better true resolution you must measure an extended impulse response and use an equally long time window and FFT size to compute the transfer function – go outdoor, close-mike or anechoic.