

Time and frequency domain sampling principles

Time domain

For a vector of N samples in time domain, the RMS calculation is:

$$x_{RMS} = \sqrt{\frac{1}{N} \sum_{i=1}^N x_i^2}$$

Frequency domain

If Y is the FFT of x, where Y is a complex number. It contains N samples as well:

$$Y = FFT(x)$$

The RMS of Y (Parseval's theorem):

$$y_{RMS} = \frac{1}{\sqrt{N}} \sqrt{\frac{1}{N} \sum_{i=1}^N y_i^2}$$

Considering P1 as the normalized spectrum of N samples, and P2 the normalized reduced spectrum (of N/2 samples):

$$P1_i = \frac{|Y_i|}{N}$$

$$P2_1 = P1_1$$

$$P2_i = 2P1_i \quad \text{for } i \in (2: \frac{N}{2})$$

Where we have already taken modulus of each of Y's components. Then the corrected RMS of the normalized reduced spectrum equates to:

$$P2'_{RMS} = \frac{\sqrt{N}}{2} \sqrt{\frac{1}{N/2} \sum_{i=1}^{N/2} P2_i^2} = \frac{\sqrt{N}}{2} P2_{RMS}$$

Conversion of RMS values to dB scale

Microphone values are referenced as peak values:

$$peak = RMS\sqrt{2}$$

The largest word size by the microphone is $BITLENGTH = 24$, so the maximum value it can read is $2^{BITLENGTH}$:

$$FULL\ SCALE(dB_{FS}) = 20 \log_{10}(2^{BITLENGTH})$$

And the conversion of the peak values to FS (full scale) of the microphone:

$$P2_{RMS}(dB_{FS}) = 20 \log_{10}(peak)$$

Hence, the SPL values, mapped to the sensor's range are:

$$P2_{RMS}(dB_{SPL}) = FULL\ SCALE(dB_{SPL}) - [FULL\ SCALE(dB_{FS}) - 20 \log_{10}(peak)]$$

Where $FULLSCALE(dBSPL) = 120$ according to the manufacturer's data-sheet.

