A-weighting filter for 44.1 and 48 kHz sampling

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Definition

The A-weighting filter used for sound level measurements is described in the IEC 61672:2003 standard. In this standard only the analog specifications are given. Using a digital filter these specifications have to be transformed for each sampling frequency to be used. The following IIR filter coefficients can be used for 44.1 kHz and 48 kHz sampling.

	44.1 kHz	48 kHz
a1	-1.31861375911	-1.34730722798
a2	0.32059452332	0.34905752979
b0	0.95616638497	0.96525096525
b1	-1.31960414122	-1.34730163086
b2	0.36343775625	0.38205066561
a1	-1.88558607420	-1.89387049481
a2	0.88709946900	0.89515976917
b0	0.94317138580	0.94696969696
b1	-1.88634277160	-1.89393939393
b2	0.94317138580	0.94696969696
a1	-1.31859445445	-1.34730722798
a2	0.32058831623	0.34905752979
b0	0.69736775447	0.64666542810
b1	-0.42552769920	-0.38362237137
b2	-0.27184005527	-0.26304305672

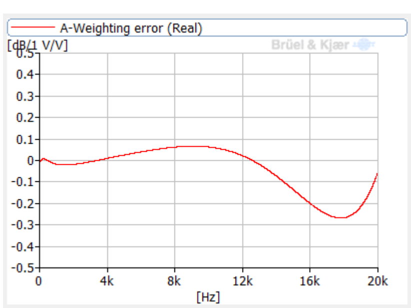

Figure 1: 44.1 kHz sampling
Error curve 0 - 20 kHz

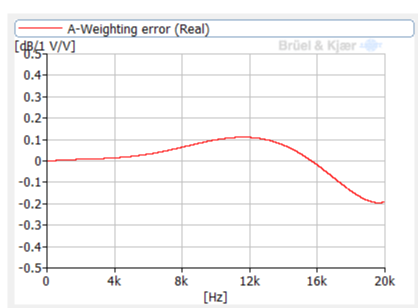

Figure 2: 48 kHz sampling
Error curve 0 - 20 kHz