Qualcomm Technologies International, Ltd.

User EQ

Application Note

80-CT445-1 Rev. AE

October 26, 2017

Revision history

Revision	Date	Description
1	DEC 2013	Original publication of this document. Alternative document number CS-00309844-DC.
2	DEC 2013	Minor editorial corrections
3	APR 2015	Updated for ADK 4
4	SEP 2016	Updated to conform to QTI standards; No technical content was changed in this document revision
5	OCT 2017	Added to the Content Management System. Removed reference to ADK 4.0 to make generic. Updated DRN to Agile numbering.

Contents

Tables

Figures

1 User EQ - overview

User equalization is provided in the Sink and Subwoofer applications available in ADK/ROM projects. The user EQ uses technology that allows filter coefficients to be controlled from the VM in real time and over GAIA.

The coefficient calculation increases the number of filter types available, see Table 1-1.

Table 1-1 Available filter types

1st Order Filters	2nd Order Filters
Low pass	Low pass
High pass	High pass
All pass	All pass
Low shelf	Low shelf
High shelf	High shelf
Tilt	Tilt
-	Parametric EQ

The filters implemented follow their analog prototypes. The analog prototypes are transformed into their digital counterparts together with appropriate warping of the frequency and q where necessary to compensate for the frequency response distortion which occurs with various digital filters. See Table 1-2.

Table 1-2 Filter analog prototypes

Filter type	1st order	2nd order
Low pass	$\frac{w}{s+w}$	$\frac{w^2}{s^2 + {}^{w}/_{q}s + w^2}$
High pass	$\frac{s}{s+w}$	$\frac{s^2}{s^2 + {}^{w}/_{q}s + w^2}$
All pass	$\frac{s-w}{s+w}$	$\frac{s^2 - {}^{w}/_{q}s + w^2}{s^2 + {}^{w}/_{q}s + w^2}$

Table 1-2 Filter analog prototypes (cont.)

Filter type	1st order	2nd order
Shelf	$g\frac{s + w_1}{s + w_2}$ $w_1 = w^*10^{dB/40}$ $w_2 = w/10^{dB/40}$ g is varied to change between low shelf, high shelf and tilt filters	$g\frac{s^2 + {}^{w_1}/_{q}s + w_1^2}{s^2 + {}^{w_2}/_{q}s + w_2^2}$ $w_1 = w^*10^{dB/80}$ $w_2 = w/10^{dB/80}$ g is varied to change between low shelf, high shelf and tilt filters
Parametric EQ	-	$\frac{s^2 + {}^{w}/_{q_1}s + w^2}{s^2 + {}^{w}/_{q_2}s + w^2}$ $q_1 = q/10^{dB/40}$ $q_2 = q^*10^{dB/40}$

2 User EQ filter calculation library

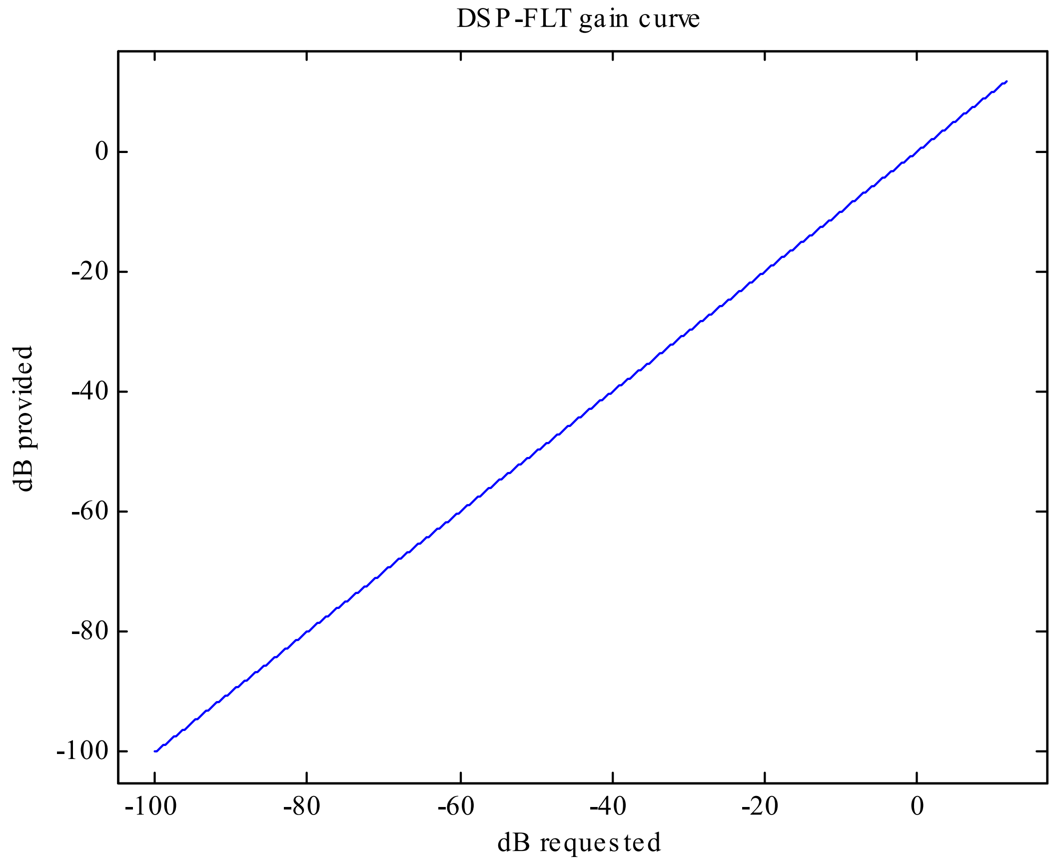
2.1 Master gain

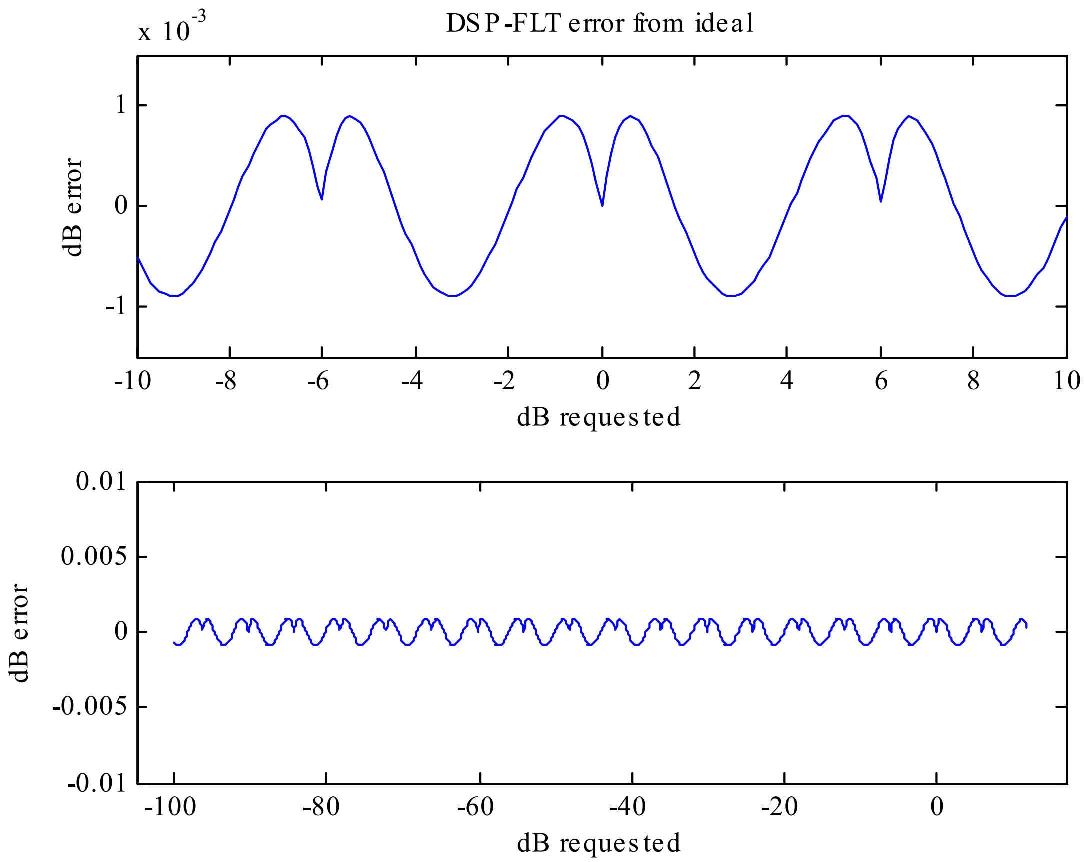
The master gain uses an exponent and shift gain element that provides accurate gain tracking over its entire range. The variation in gain is as a result of the truncated Taylor series used to calculate the gain coefficient.

Table 2-1 shows the master gain parameter limits.

Table 2-1 Master gain parameter limits

Limit	Gain
Low limit	-36 dB
High limit	+12 dB


DSP-FLT gain curve
dB provided
dB requested
0
-20
-40
-60
-80
-100
-100
-80
-60
-40
-20
0



2.2 First order low pass

The first order low pass filter is a digital implementation of the following analog prototype.

$$\frac{w}{s+w}$$

Table 2-2 shows the low pass parameter limits.

Table 2-2 First order low pass parameter limits

Limit	Filter Frequency
Low limit	0.333 Hz
High limit	20 kHz

If the requested filter is above 0.453515 of the sample frequency (fc = 20 kHz @ Fs = 44.1 kHz), then the filter is substituted with bypass coefficients.

Figure 2-1 shows the filter response calculated at octave frequencies from 40 Hz to 10 kHz with a sample rate of 48 kHz.

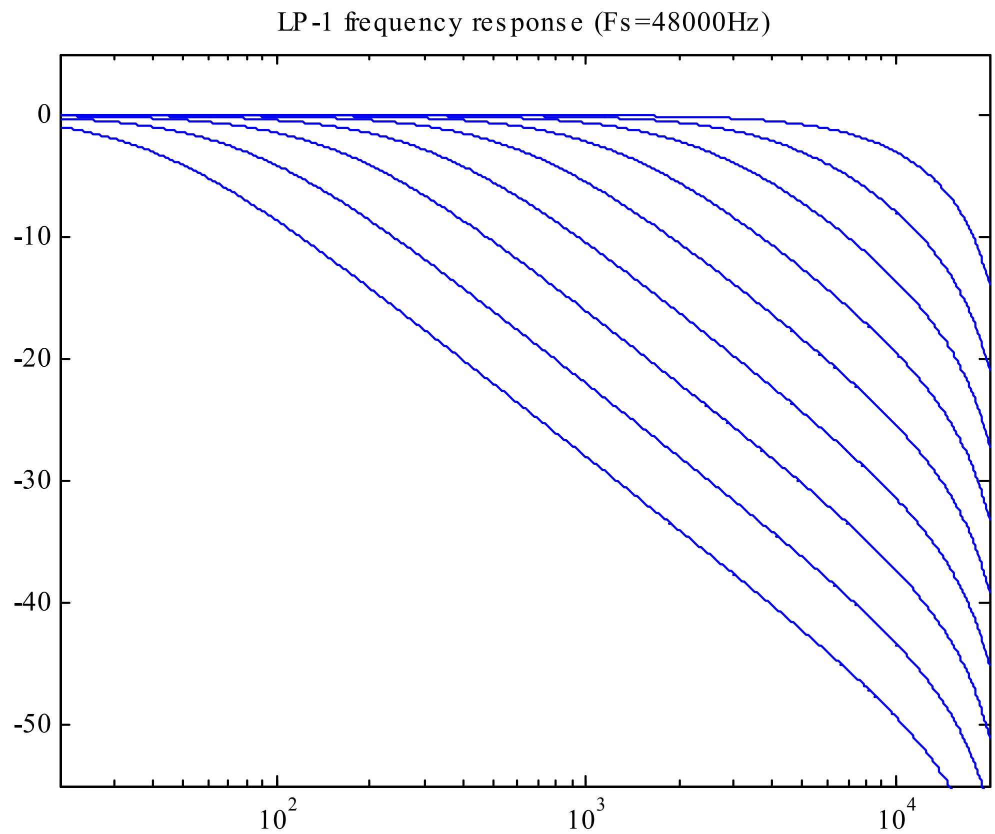


Figure 2-1 First order low pass filter response

2.3 First order high pass

The first order high pass filter is a digital implementation of the following analog prototype.

$$\frac{s}{s + w}$$

Table 2-3 shows the 1st order high pass filter parameters limits.

Table 2-3 First order high pass parameter limits

Limit	Filter Frequency
Low limit	0.333 Hz
High limit	20 kHz

If the requested filter is above 0.453515 of the sample frequency (fc = 20 kHz @ Fs = 44.1 kHz), then the filter is substituted with bypass coefficients.

Figure 2-2 shows the filter response calculated at octave frequencies from 40 Hz to 10 kHz with a sample rate of 48 kHz.

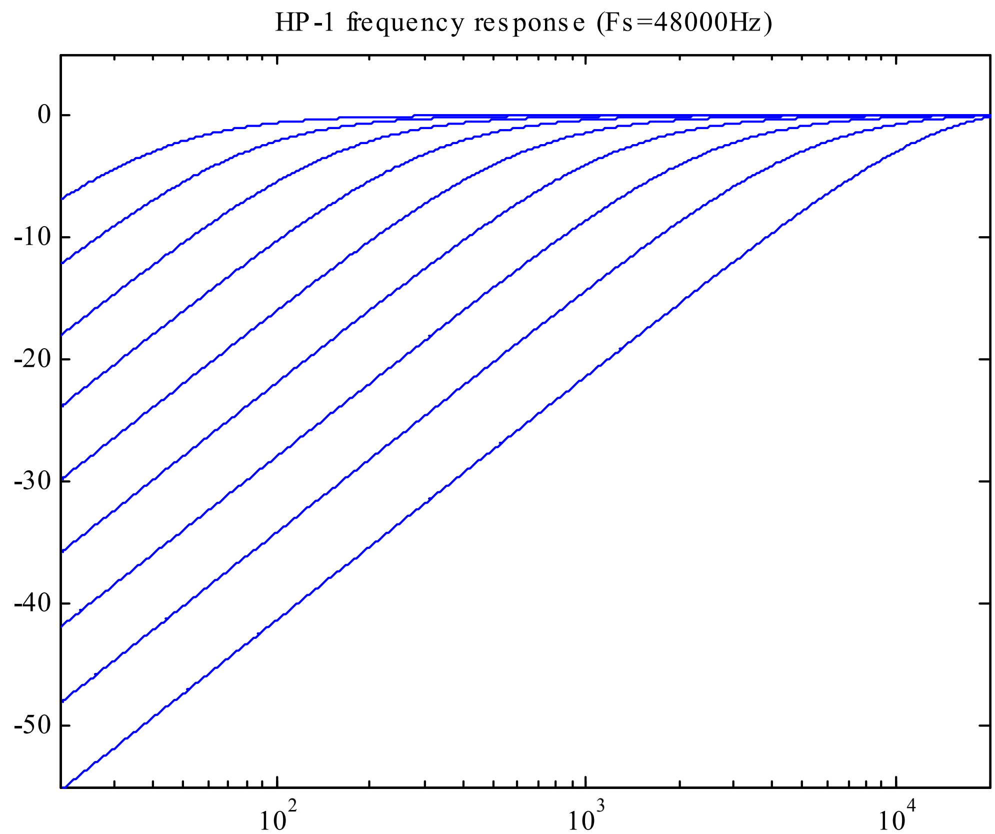


Figure 2-2 First order high pass filter response

2.4 First order all pass

The first order high pass filter is a digital implementation of the following analog prototype.

$$\frac{s-w}{s+w}$$

Table 2-4 shows the 1st order all pass parameter limits.

Table 2-4 First order all pass parameter limits

Limit	Filter Frequency
Low limit	0.333 Hz
High limit	20 kHz

If the requested filter is above 0.453515 of the sample frequency (fc = 20 kHz @ Fs = 44.1 kHz), then the filter is substituted with bypass coefficients.

Figure 2-3 shows the filter response calculated at octave frequencies from 40 Hz to 10 kHz with a sample rate of 48 kHz. The top graph shows the frequency response. The bottom graph shows the phase response in degrees.

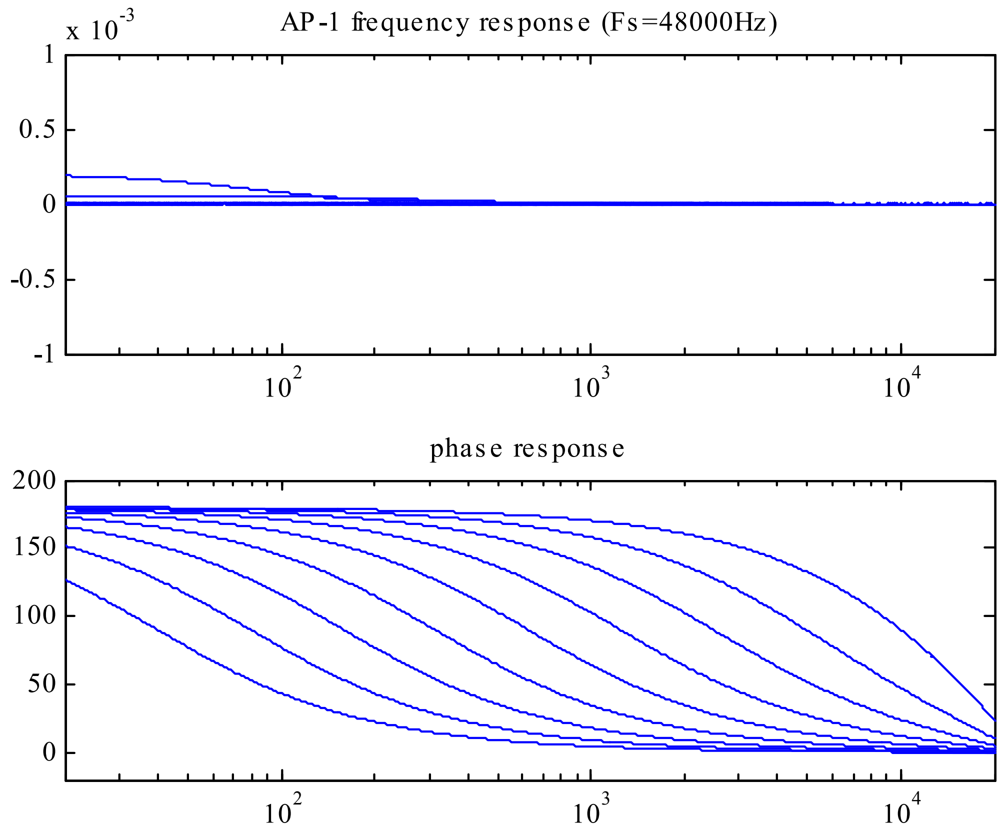


Figure 2-3 First order all pass filter response

2.5 Second order low pass

The second order low pass filter is a digital implementation of the following analog prototype.

$$\frac{w^2}{s^2 + {}^{w}/_{q}s + w^2}$$

Table 2-5 Second order low pass parameter limits

Limit	Filter Frequency	Filter Q
Low limit	40 Hz	0.25
High limit	20 kHz	2.0

If the requested filter is above 0.453515 of the sample frequency (fc = 20 kHz @ Fs = 44.1 kHz), then the filter is substituted with bypass coefficients.

Figure 2-4 shows the filter response calculated at octave frequencies from 40 Hz to 10 kHz with a sample rate of 48 kHz and a Q of 0.707107.

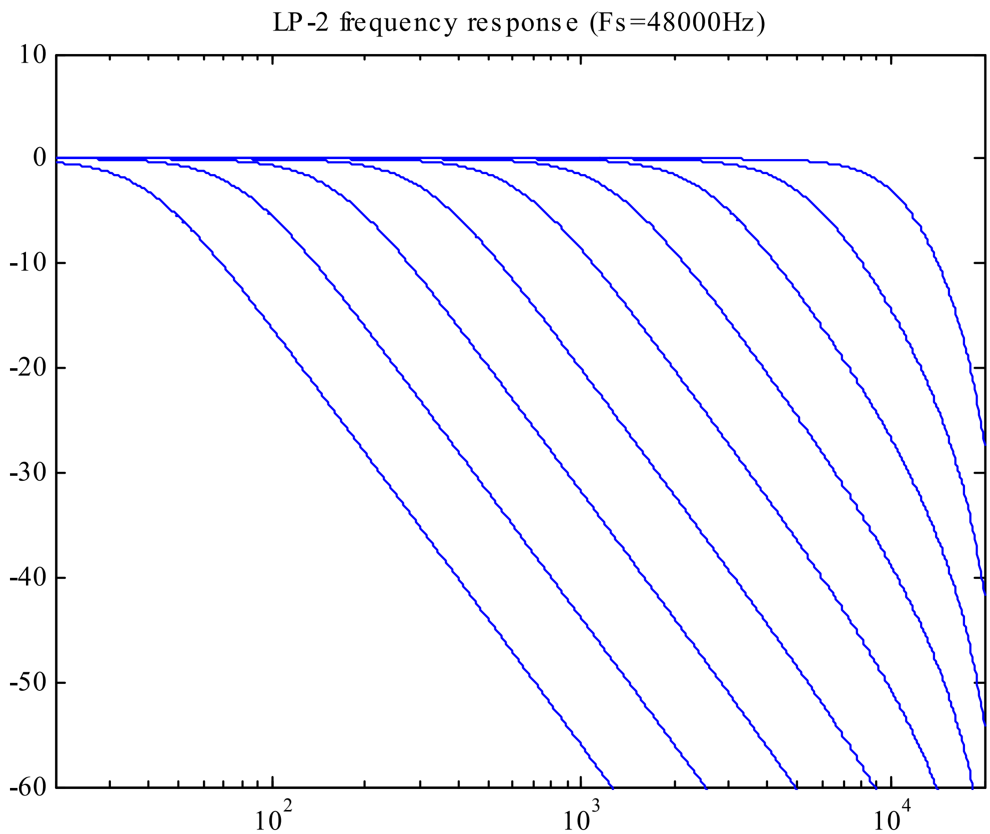


Figure 2-4 Second order low pass filter response

Figure 2-5 shows the filters calculated at 640 Hz, with Qs from 0.5 to 2.0

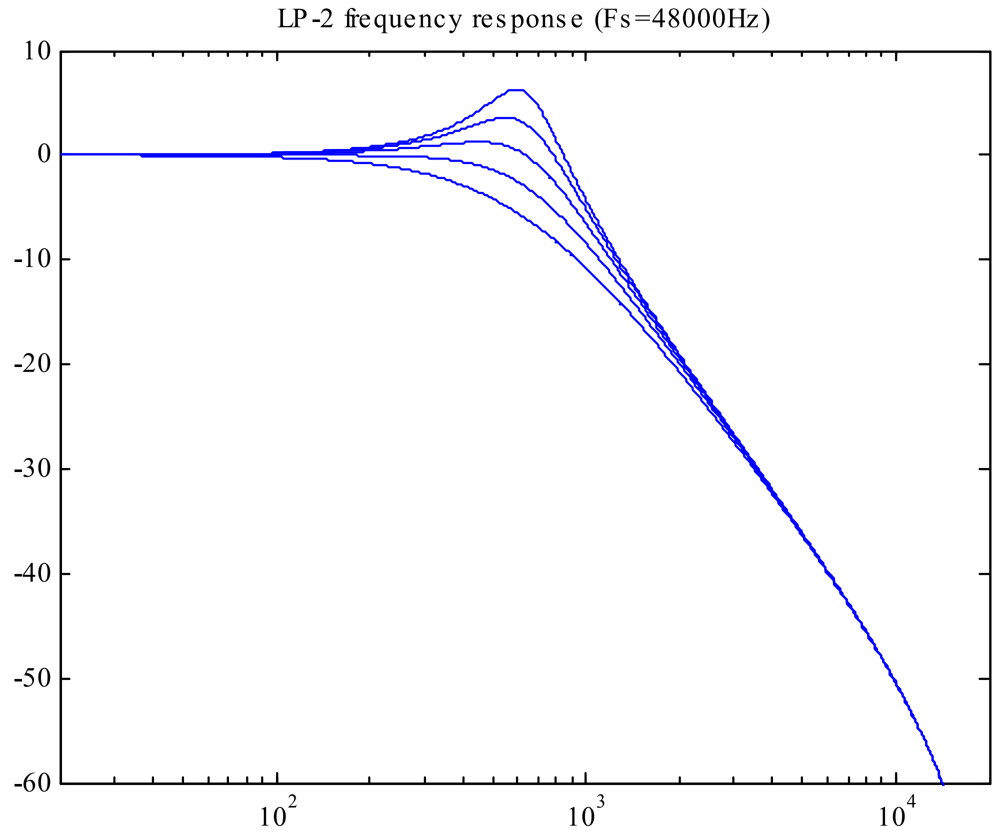


Figure 2-5 Second order low pass filter response at 640 Hz

2.6 Second order high pass

The second order high pass filter is a digital implementation of the following analog prototype.

$$\frac{s^2}{s^2 + {}^{w}/_{q}s + w^2}$$

Table 2-6 shows the 2nd order high pass parameter limits.

Table 2-6 Second order high pass parameter limits

Limit	Filter frequency	Filter Q
Low limit	40 Hz	0.25
High limit	20 kHz	2.0

If the requested filter is above 0.453515 of the sample frequency (fc = 20 kHz @ Fs = 44.1 kHz), then the filter is substituted with bypass coefficients.

Figure 2-6 shows the filter response calculated at octave frequencies from 40 Hz to 10 kHz with a sample rate of 48 kHz and a Q of 0.707107.

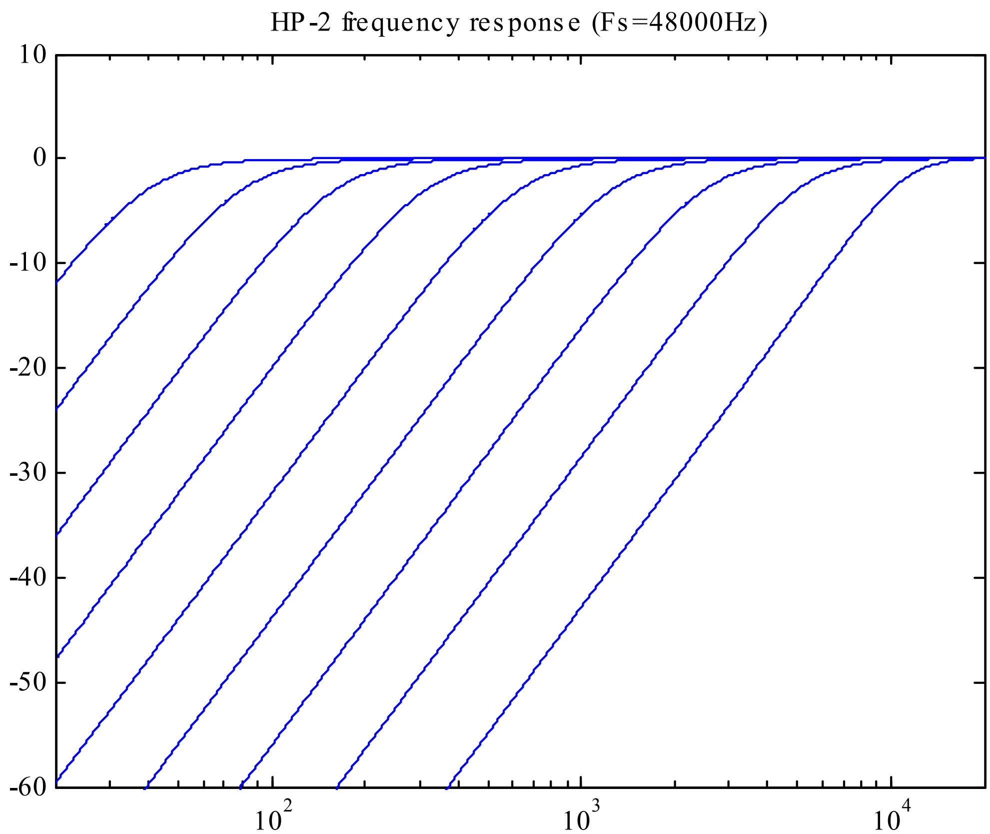


Figure 2-6 Second order high pass filter response

Figure 2-7 shows the filters calculated at 640 Hz, with Qs from 0.5 to 2.0

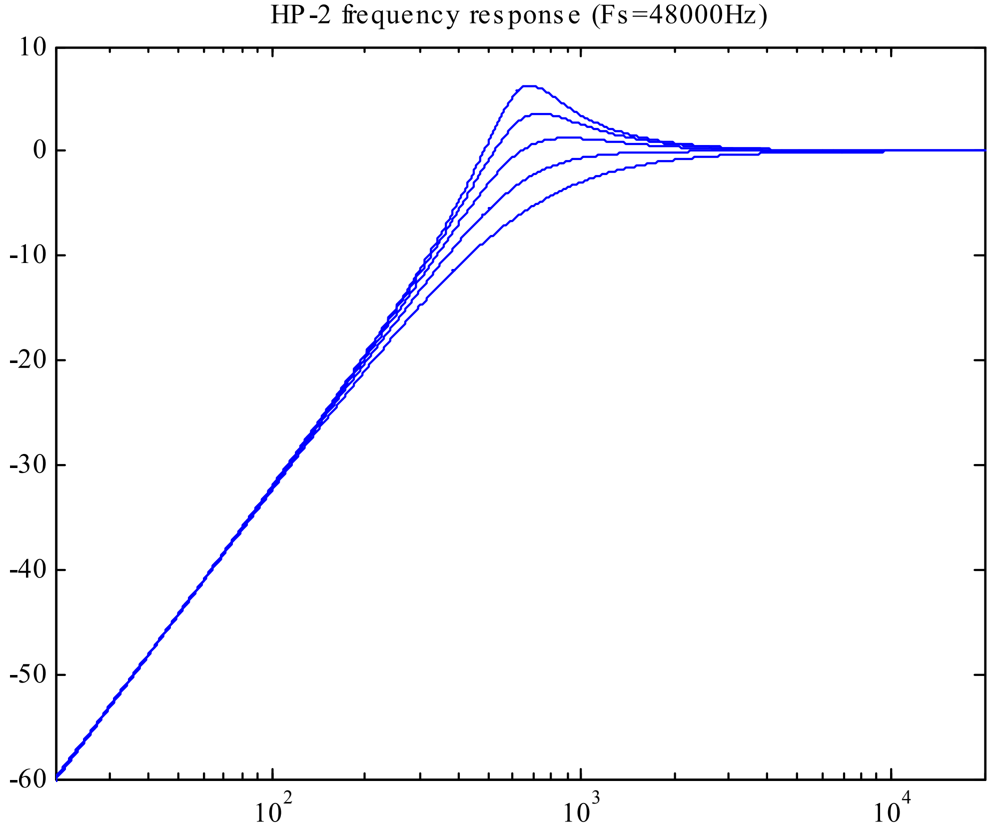


Figure 2-7 Second order high pass filter response at 640 Hz

2.7 Second order all pass

The second order all pass filter is a digital implementation of the following analog prototype.

$$\frac{s^2 - {}^{w}\!/_{q}s + w^2}{s^2 + {}^{w}\!/_{q}s + w^2}$$

Table 2-7 shows the 2nd order all pass parameter limits.

Table 2-7 Second order all pass parameter limits

Limit	Filter Frequency	Filter Q
Low limit	40 Hz	0.25
High limit	20 kHz	2.0

If the requested filter is above 0.453515 of the sample frequency (fc = 20 kHz @ Fs = 44.1 kHz), then the filter is substituted with bypass coefficients.

Figure 2-8 shows the filter response calculated at octave frequencies from 40 Hz to 10 kHz with a sample rate of 48 kHz and a Q of 0.707107.

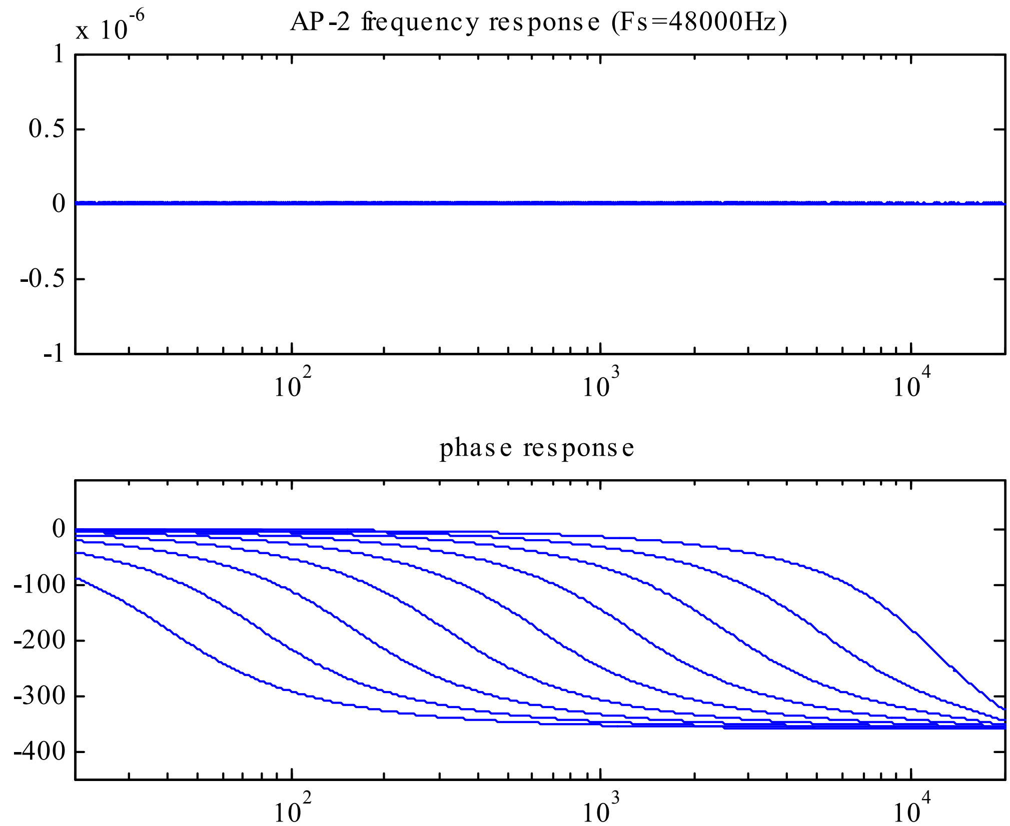


Figure 2-8 Second order all pass filter response

Figure 2-9 shows the filters calculated at 640 Hz, with Qs from 0.5 to 2.0

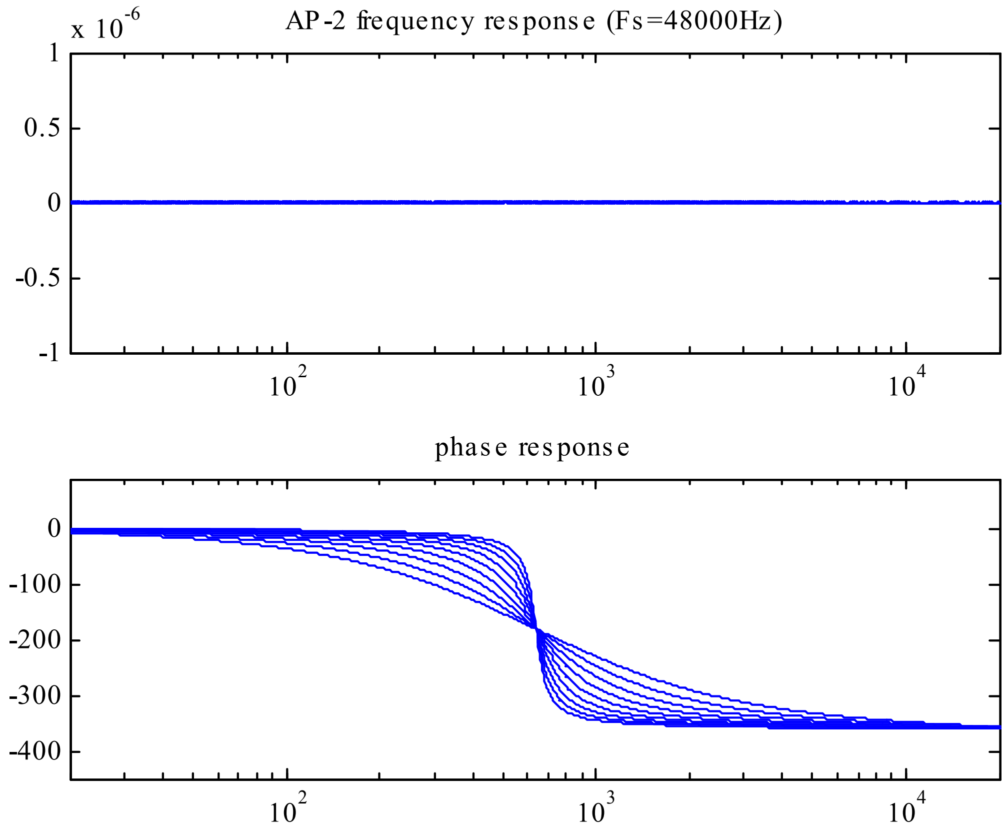


Figure 2-9 Second Order All Pass Filter Response at 640 Hz

2.8 First order low shelf

The first order low shelf filter is a digital implementation of the following analog prototype.

$$\frac{s + w_1}{s + w_2}$$

Where:

$$w_1 = w * 10^{dB/40}$$

$$w_2 = w / 10^{dB/40}$$

Table 2-8 shows the 1st order low shelf parameter limits.

Table 2-8 First order low shelf parameter limits

Limit	Filter Frequency	Filter Gain
Low limit	20 Hz	-12 dB
High limit	20 kHz	+12 dB

If the requested filter is above 0.453515 of the sample frequency (fc = 20 kHz @ Fs = 44.1 kHz), then the filter is substituted with bypass coefficients.

Figure 2-10 shows the filter response calculated at octave frequencies from 40 Hz to 10 kHz with a sample rate of 48 kHz and gains of +12 dB and -12 dB.

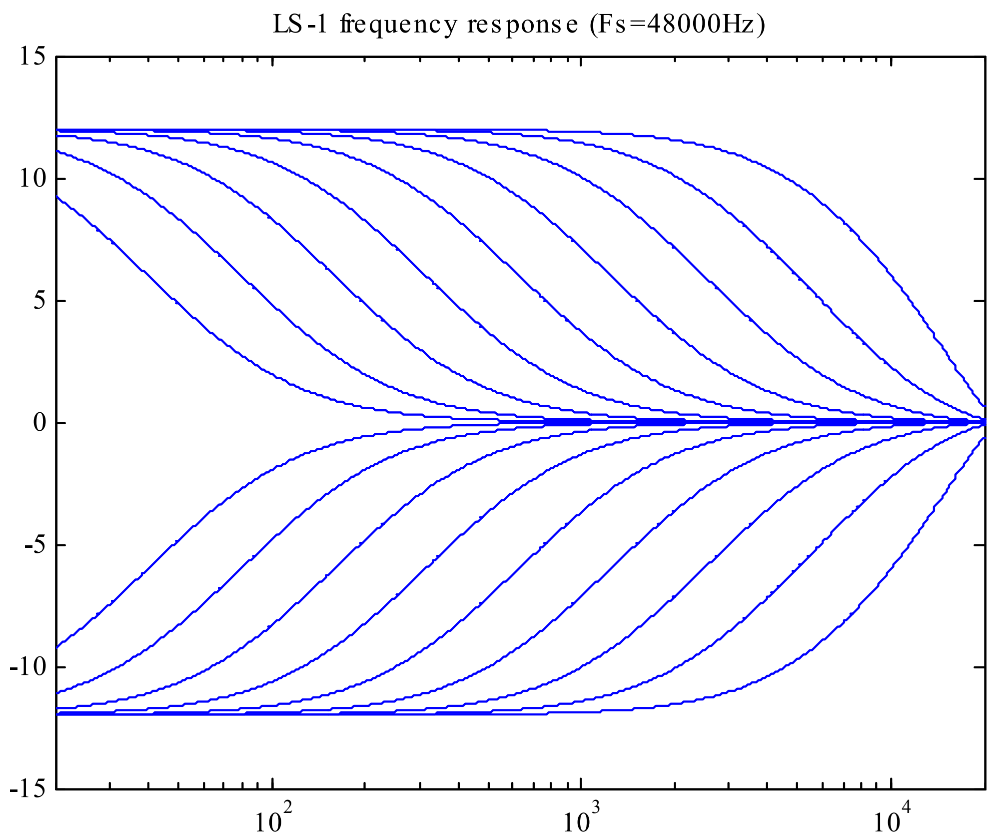


Figure 2-10 First order low shelf filter response

Figure 2-11 shows the filter response calculated at 640 Hz with gains from +12 dB to -12 dB in 1 dB steps.

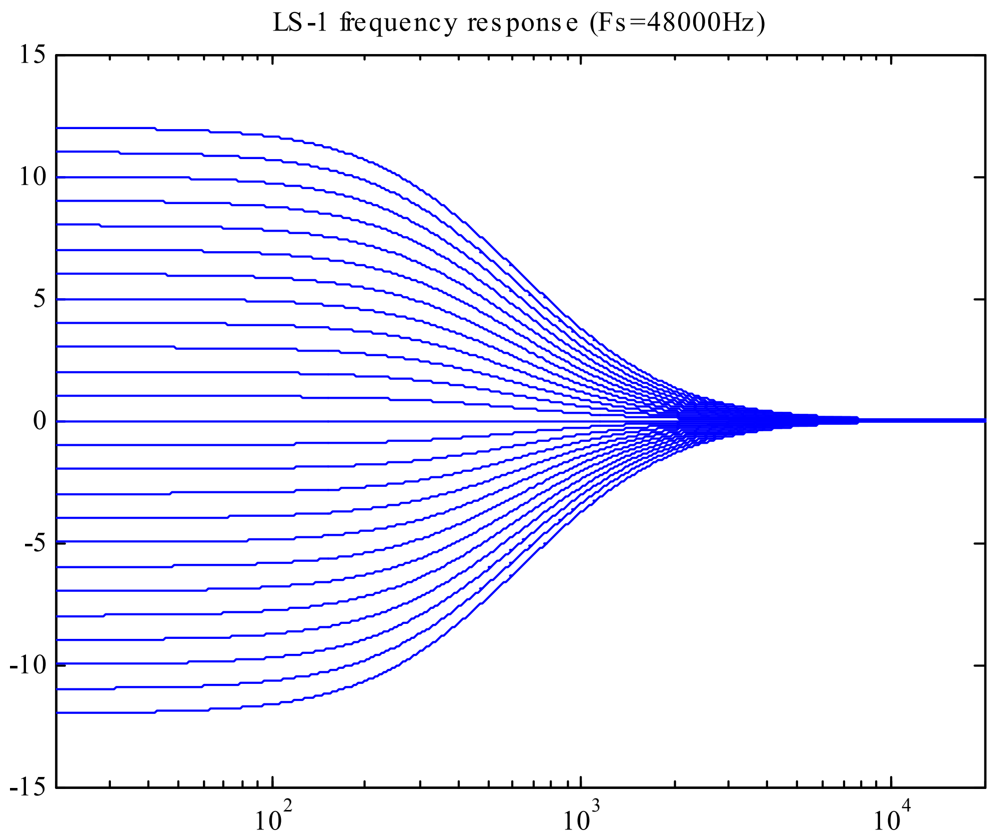


Figure 2-11 First order low shelf filter response at 640 Hz

2.9 First order high shelf

The first order high shelf filter is a digital implementation of the following analog prototype.

$$10^{dB/20} * \frac{s + w_1}{s + w_2}$$

Where:

$$w_1 = w * 10^{dB/40}$$

$$w_2 = w / 10^{dB/40}$$

Table 2-9 shows the 1st order low shelf parameter limits.

Table 2-9 First order low shelf parameter limits

Limit	Filter Frequency	Filter Gain
Low limit	20 Hz	-12 dB
High limit	20 kHz	+12 dB

If the requested filter is above 0.453515 of the sample frequency (fc = 20 kHz @ Fs = 44.1 kHz), then the filter is substituted with bypass coefficients.

Figure 2-12 shows the filter response calculated at octave frequencies from 40 Hz to 10 kHz with a sample rate of 48 kHz and gains of +12 dB and -12 dB.

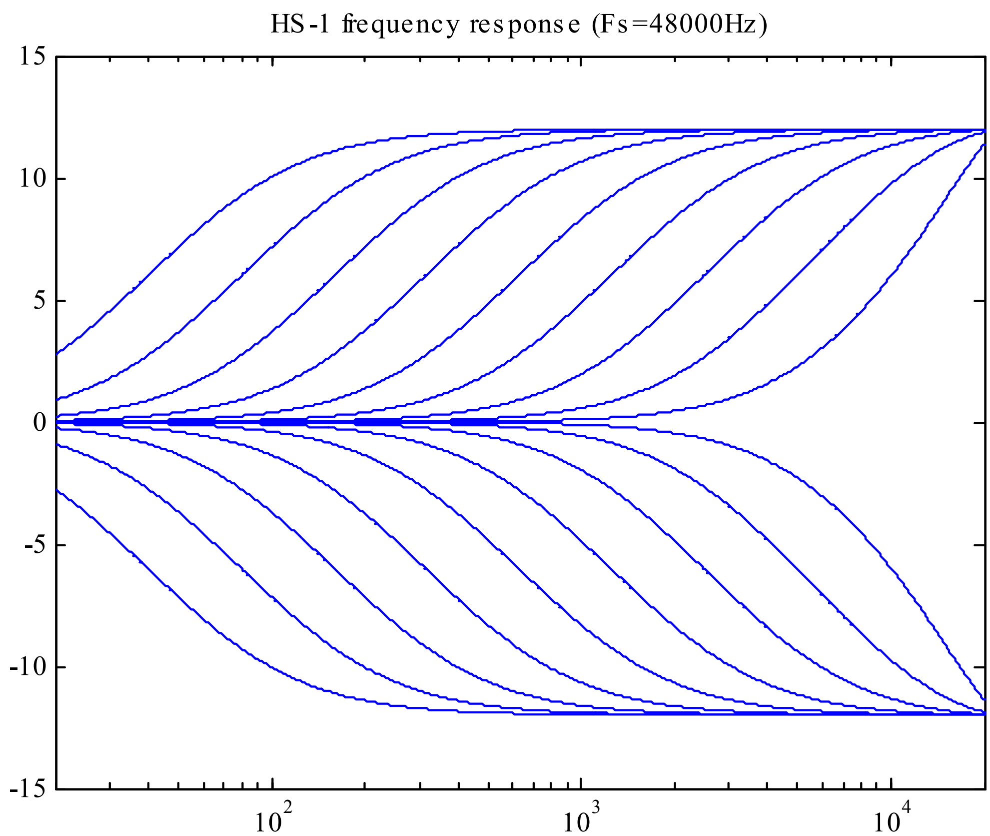


Figure 2-12 First order low shelf filter response

Figure 2-13 shows the filter response calculated at 640 Hz with gains from +12 dB to -12 dB in 1 dB steps.

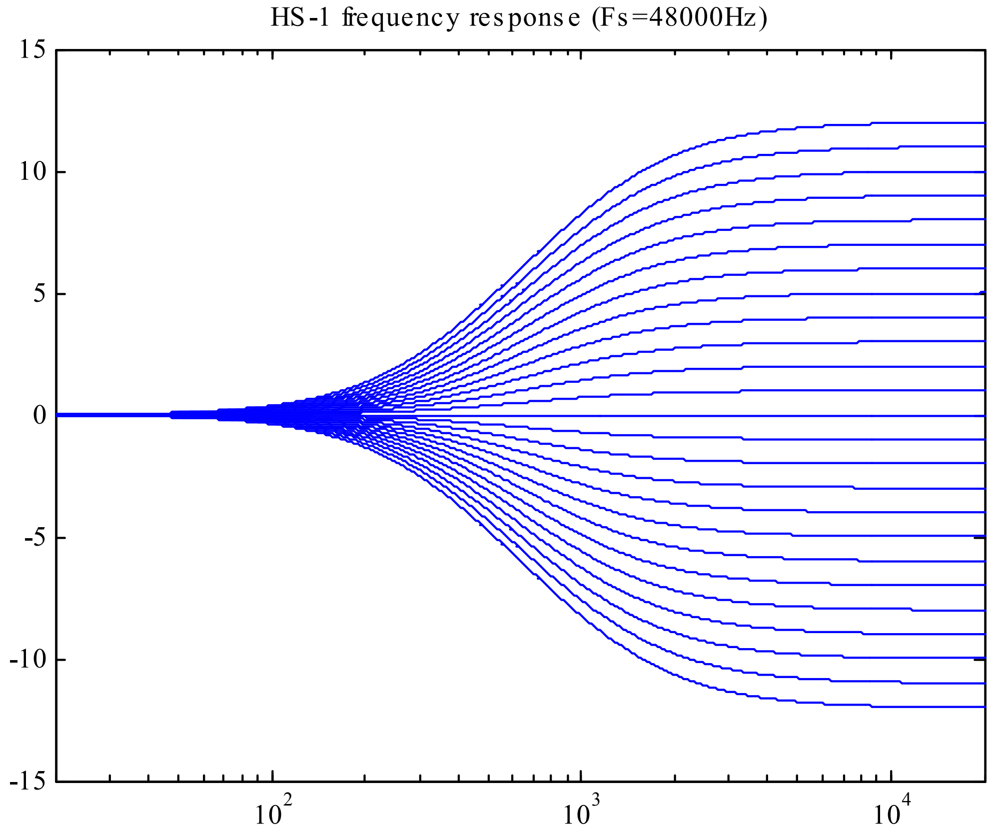


Figure 2-13 First order low shelf filter response at 640 Hz

2.10 First order tilt

The first order tilt filter is a digital implementation of the following analog prototype.

$$10^{dB/40} * \frac{s + w_1}{s + w_2}$$

Where:

$$w_1 = w * 10^{dB/40}$$

$$w_2 = w / 10^{dB/40}$$

Shows 1st order tilt parameter limits.

Table 2-10 First order tilt parameter limits

Limit	Filter Frequency	Filter Gain
Low limit	20 Hz	-12 dB
High limit	20 kHz	+12 dB

If the requested filter is above 0.453515 of the sample frequency (fc = 20 kHz @ Fs = 44.1 kHz), then the filter is substituted with bypass coefficients.

Figure 2-14 shows the filter response calculated at octave frequencies from 40 Hz to 10 kHz with a sample rate of 48 kHz and gains of +12 dB and -12 dB.

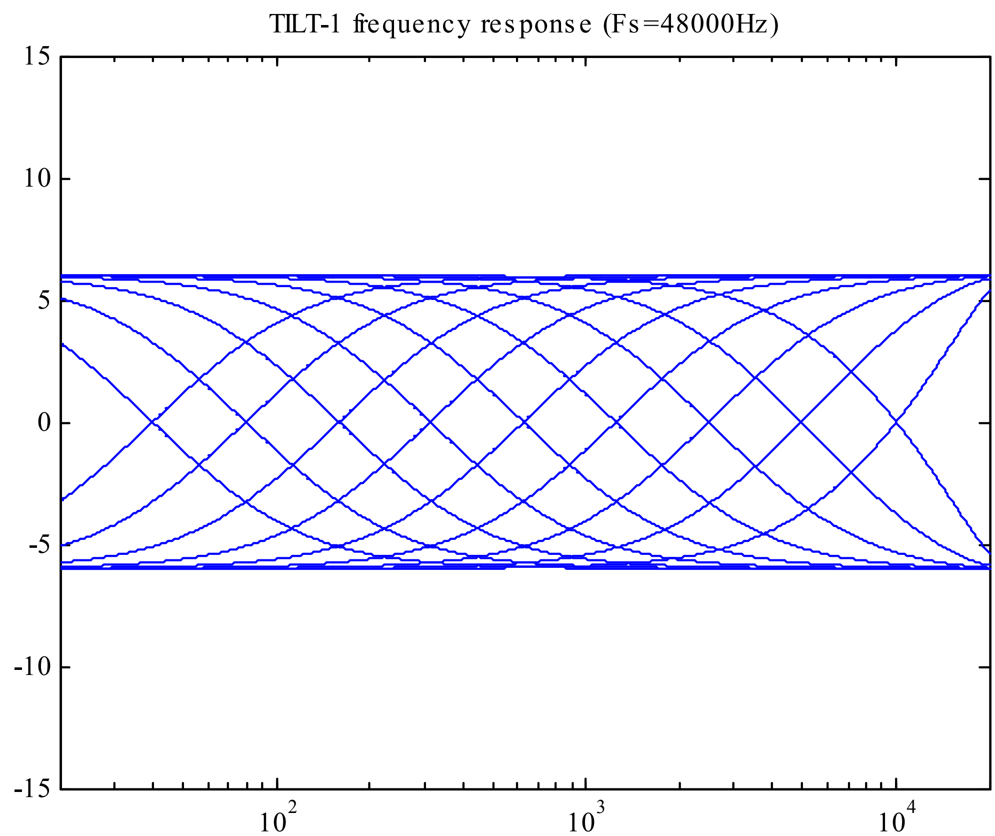


Figure 2-14 First order tilt filter response

Figure 2-15 shows the filter response calculated at 640 Hz with gains from +12 dB to -12 dB in 1 dB steps.

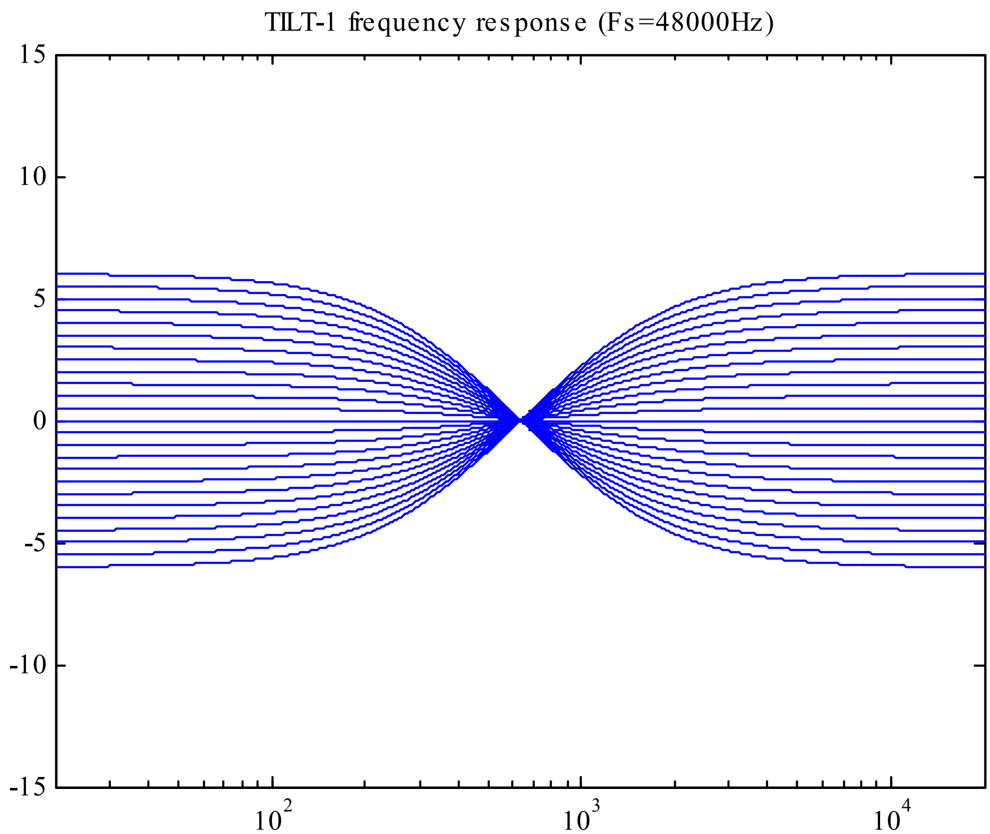


Figure 2-15 First order tilt filter response at 640 Hz

2.11 Second order low shelf

The second order low shelf filter is a digital implementation of the following analog prototype.

$$\frac{s^2 + {}^{w_1}\!/_{q}\, s + w_1^{\,2}}{s^2 + {}^{w_2}\!/_{q}\, s + w_2^{\,2}}$$

Where:

$$w_1 = w * 10^{dB/80}$$

$$w_2 = w / 10^{dB/80}$$

Table 2-11 shows the 2nd order low shelf parameter limits.

Table 2-11 Second order low shelf parameter limits

Limit	Filter Frequency	Filter Gain	Filter Q
Low limit	40 Hz	-12 dB	0.25
High limit	20 kHz	+12 dB	2.0

If the requested filter is above 0.453515 of the sample frequency (fc = 20 kHz @ Fs = 44.1 kHz), then the filter is substituted with bypass coefficients.

Figure 2-16 shows the filter response calculated at octave frequencies from 40 Hz to 10 kHz with a sample rate of 48 kHz, gains of +12 dB & -12 dB and a Q of 0.707107.

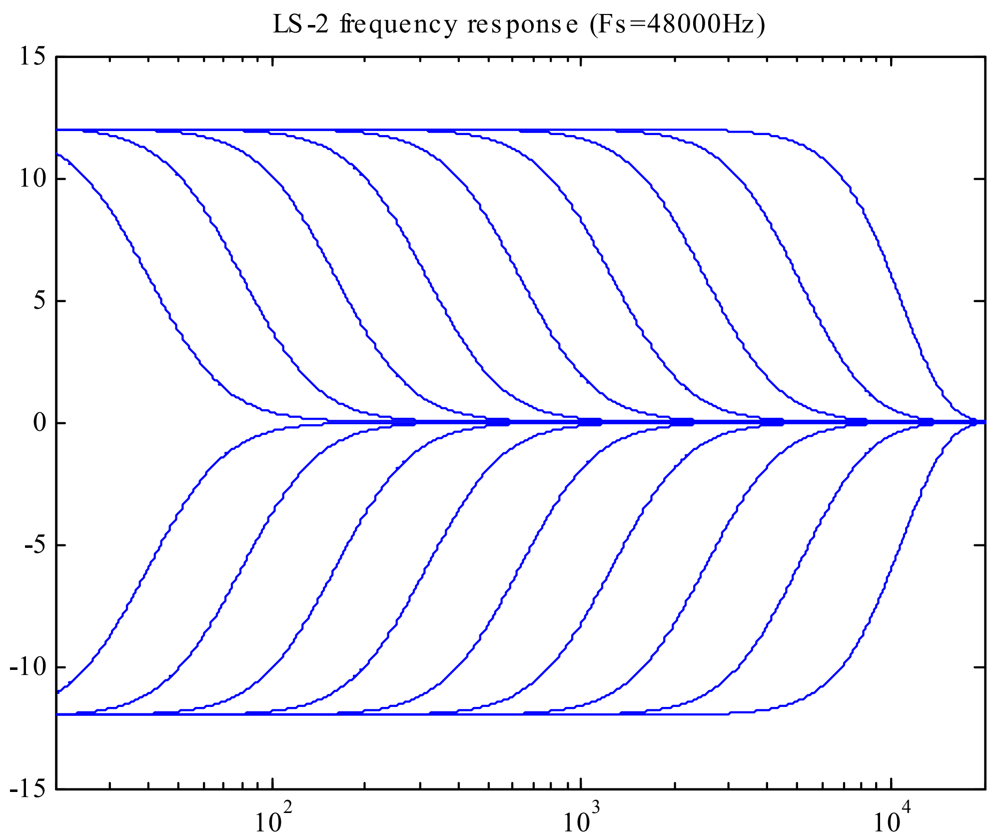


Figure 2-16 Second order low shelf filter response

Figure 2-17 shows the filter response calculated at 640 Hz and Q of 0.707107 with gains from +12 dB to -12 dB in 1 dB steps.

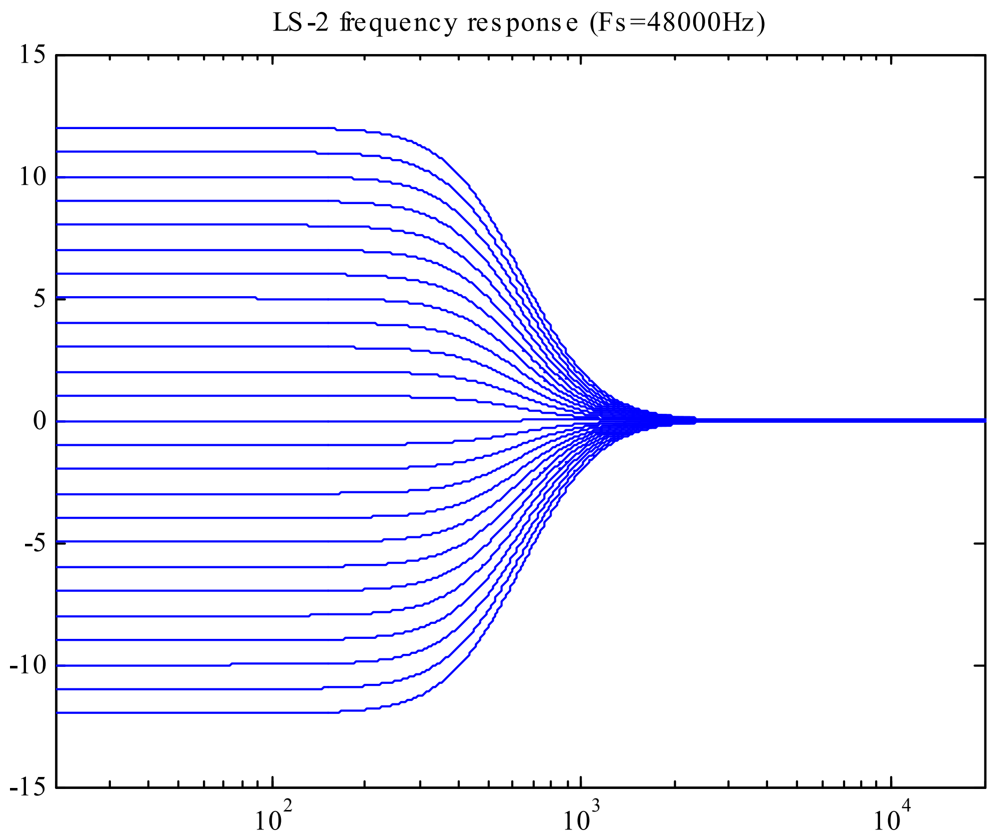


Figure 2-17 Second order low shelf filter response at 640 Hz (Q of 0.707107)

Figure 2-18 shows the filter response calculated at 640 Hz and +12 dB with Qs from 0.5 to 2.0.

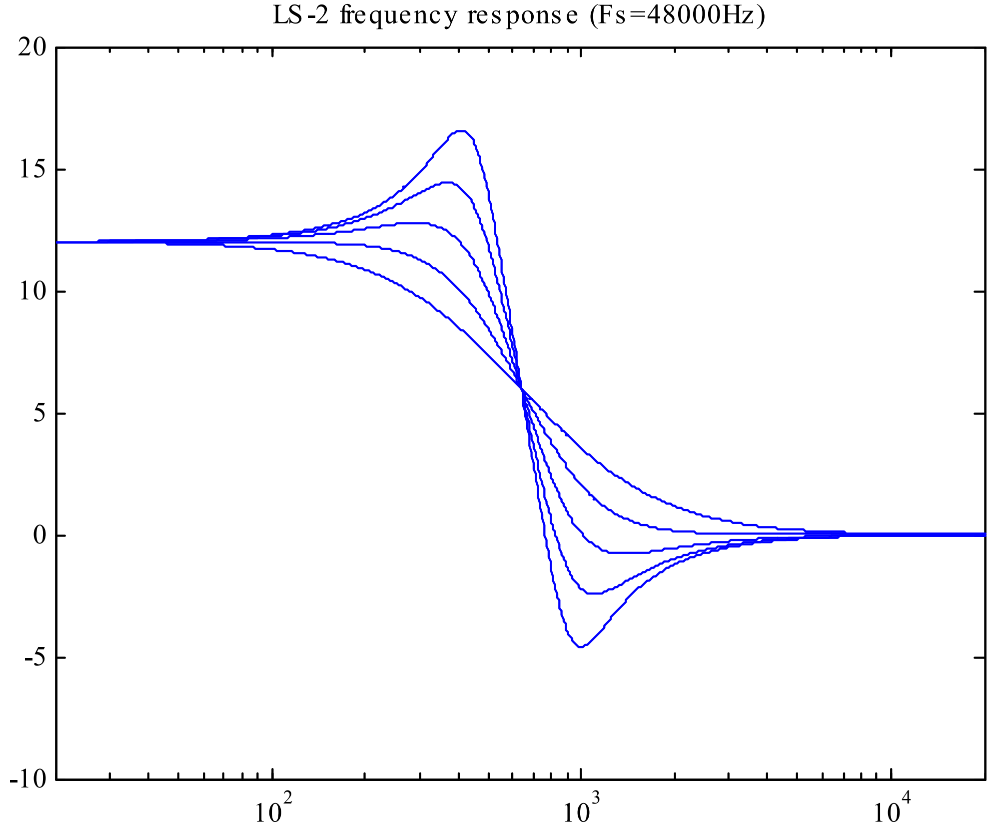


Figure 2-18 Second order low shelf filter response at 640 Hz (Qs from 0.5 to 2.0)

2.12 Second order high shelf

The second order high shelf filter is a digital implementation of the following analog prototype.

$$10^{dB/20} * \frac{s^2 + {}^{w_1}\!/_{q}\, s + w_1^2}{s^2 + {}^{w_2}\!/_{q}\, s + w_2^2}$$

Where:

$$w_1 = w * 10^{dB/80}$$

$$w_2 = w / 10^{dB/80}$$

Table 2-12 shows the 2nd order high shelf parameter limits.

Table 2-12 Second order high shelf parameter limits

Limit	Filter Frequency	Filter Gain	Filter Q
Low limit	40 Hz	-12 dB	0.25
High limit	20 kHz	+12 dB	2.0

If the requested filter is above 0.453515 of the sample frequency (fc = 20 kHz @ Fs = 44.1 kHz), then the filter is substituted with bypass coefficients.

Figure 2-19 shows the filter response calculated at octave frequencies from 40 Hz to 10 kHz with a sample rate of 48 kHz, gains of +12 dB & -12 dB and a Q of 0.707107.

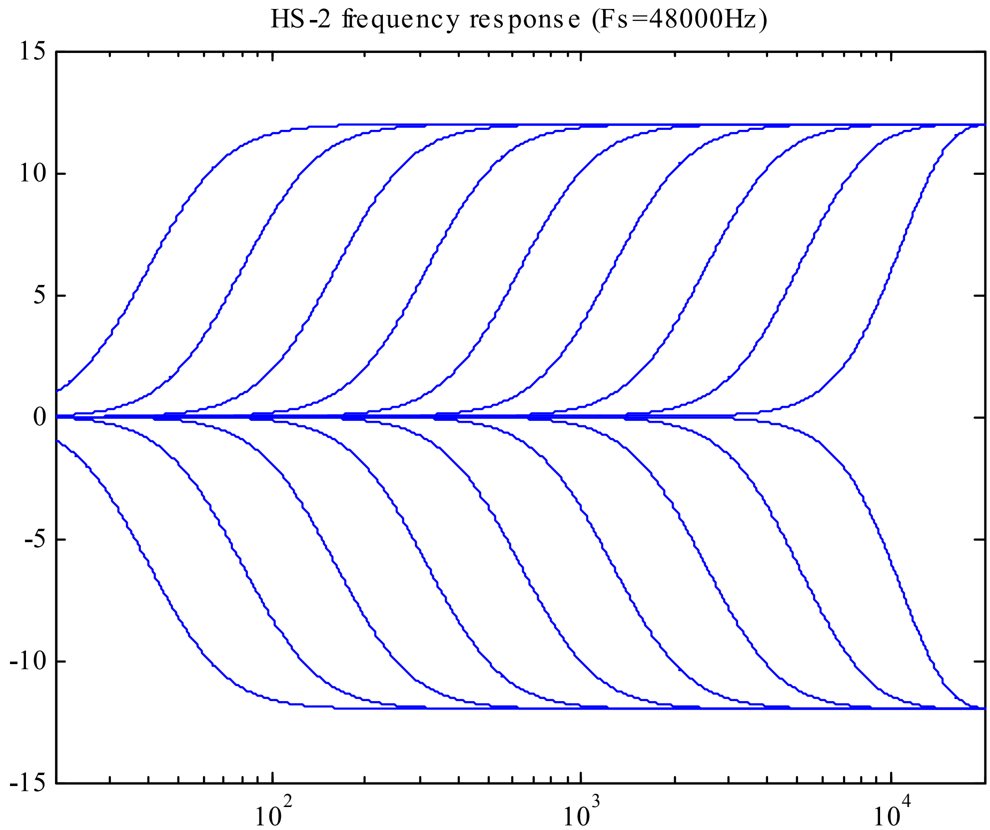


Figure 2-19 Second order high shelf filter response

Figure 2-20 shows the filter response calculated at 640 Hz and Q of 0.707107 with gains from +12 dB to -12 dB in 1 dB steps.

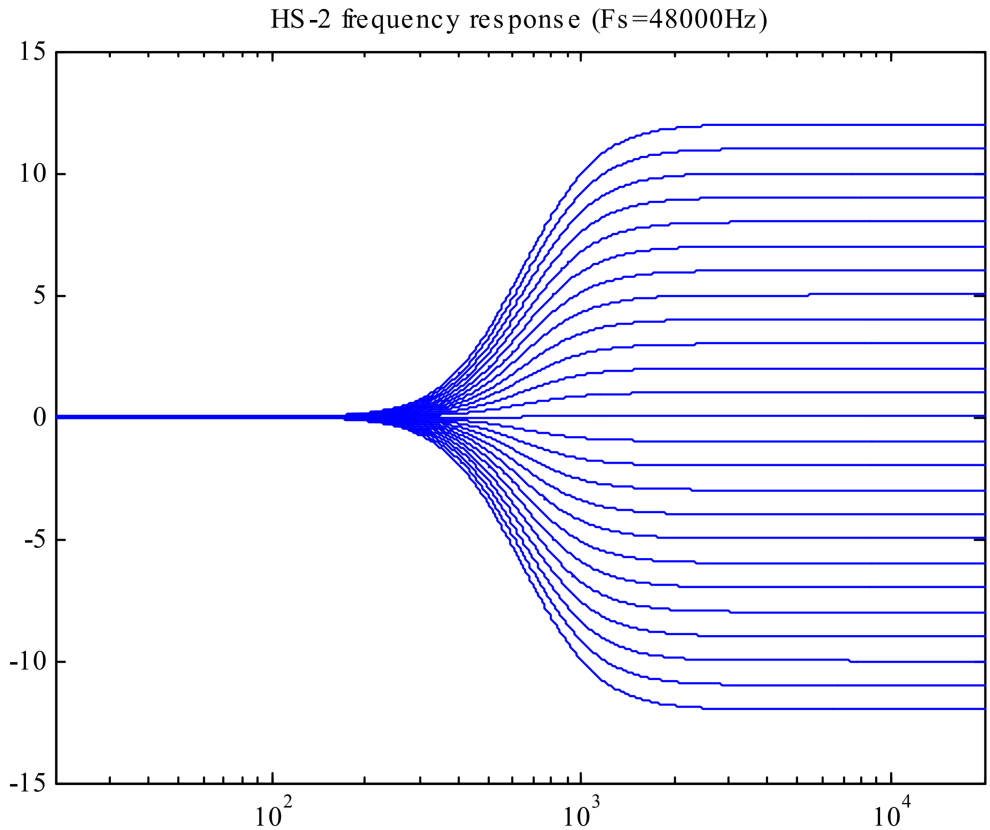


Figure 2-20 Second order high shelf filter response at 640 Hz (Q of 0.707107)

Figure 2-21 shows the filter response calculated at 640 Hz and +12 dB with Qs from 0.5 to 2.0.

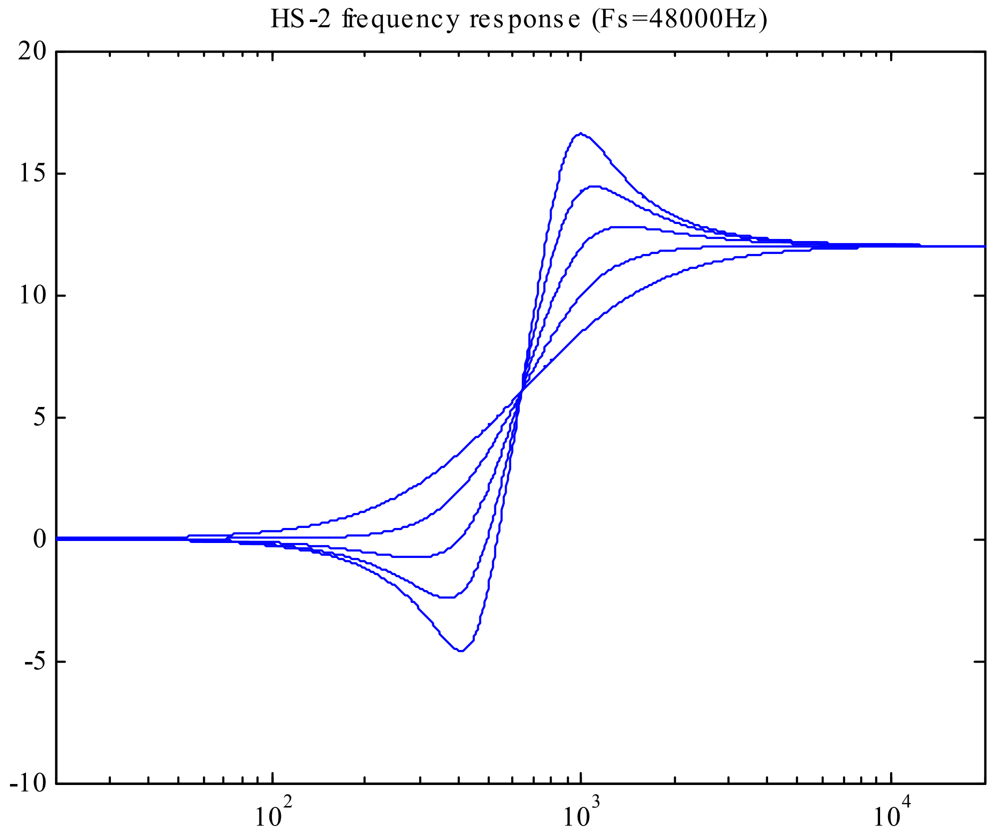


Figure 2-21 Second order high shelf filter response at 640 Hz (Q from 0.5 to 2.0)

2.13 Second order tilt

The second order tilt filter is a digital implementation of the following analog prototype.

$$10^{dB/40} * \frac{s^2 + {w_1}/{q}\, s + w_1^2}{s^2 + {w_2}/{q}\, s + w_2^2}$$

Where:

$$w_1 = w * 10^{dB/80}$$

$$w_2 = w / 10^{dB/80}$$

Table 2-13 shows the 2nd order tilt parameter limits.

Table 2-13 Second order tilt parameter limits

Limit	Filter Frequency	Filter Gain	Filter Q
Low limit	40 Hz	-12 dB	0.25
High limit	20 kHz	+12 dB	2.0

If the requested filter is above 0.453515 of the sample frequency (fc = 20 kHz @ Fs = 44.1 kHz), then the filter is substituted with bypass coefficients.

Figure 2-22 shows the filter response calculated at octave frequencies from 40 Hz to 10 kHz with a sample rate of 48 kHz, gains of +12 dB & -12 dB and a Q of 0.707107.

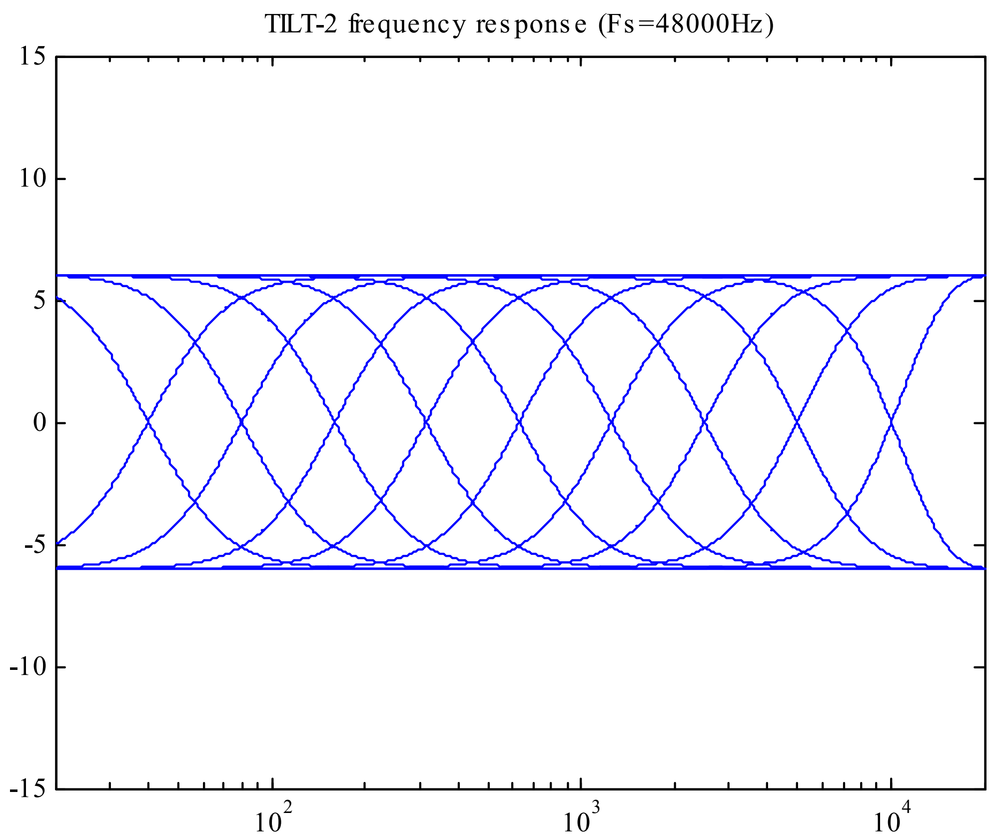


Figure 2-22 Second order tilt filter response

Figure 2-23 shows the filter response calculated at 640 Hz and Q of 0.707107 with gains from +12 dB to -12 dB in 1 dB steps.

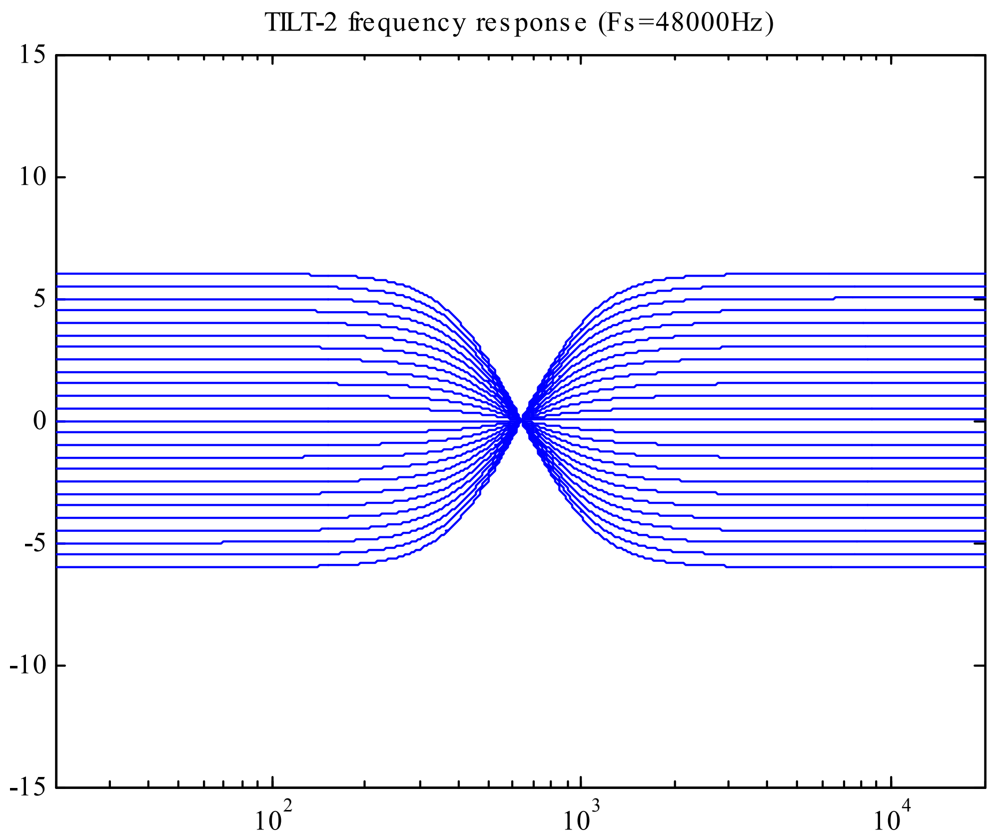


Figure 2-23 Second order tilt filter response at 640 Hz (Q of 0.707107)

Figure 2-24 shows the filter response calculated at 640 Hz and +12 dB with Qs from 0.5 to 2.0.

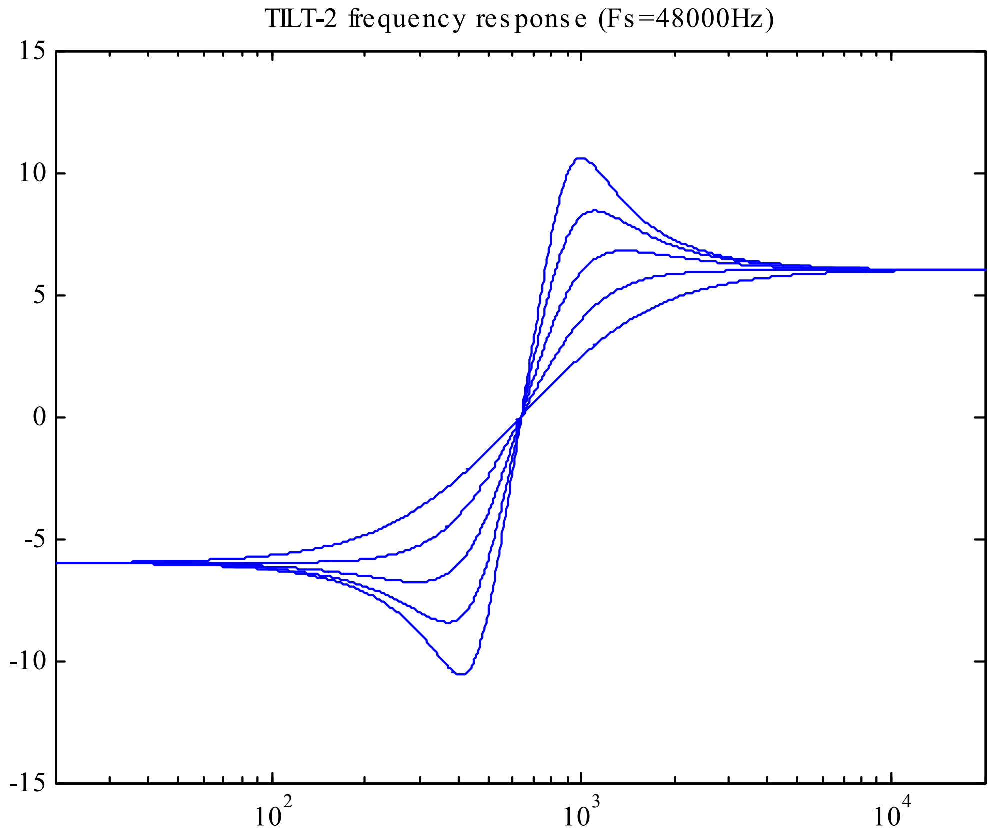


Figure 2-24 Second order tilt filter response at 640 Hz (Q from 0.5 to 2.0)

2.14 Parametric EQ

The parametric EQ filter is a digital implementation of the following analog prototype.

$$\frac{s^2 + {}^{w}\!/_{q_1} s + w^2}{s^2 + {}^{w}\!/_{q_2} s + w^2}$$

Where:

$$q_1 = q/10^{dB/40}$$

$$q_2 = q*10^{dB/40}$$

Table 2-14 shows the parametric EQ parameter limits.

Table 2-14 Parametric EQ parameter limits

Limit	Filter Frequency	Filter Gain	Filter Q
Low limit	20 Hz	-36 dB	0.25
High limit	20 kHz	+12 dB	8.0

If the requested filter is above 0.453515 of the sample frequency (fc = 20 kHz @ Fs = 44.1 kHz), then the filter is substituted with bypass coefficients.

Figure 2-25 shows the filter response calculated at octave frequencies from 20 Hz to 20 kHz with a sample rate of 48 kHz, gains of +12 dB & -12 dB and a Q of 0.707107.

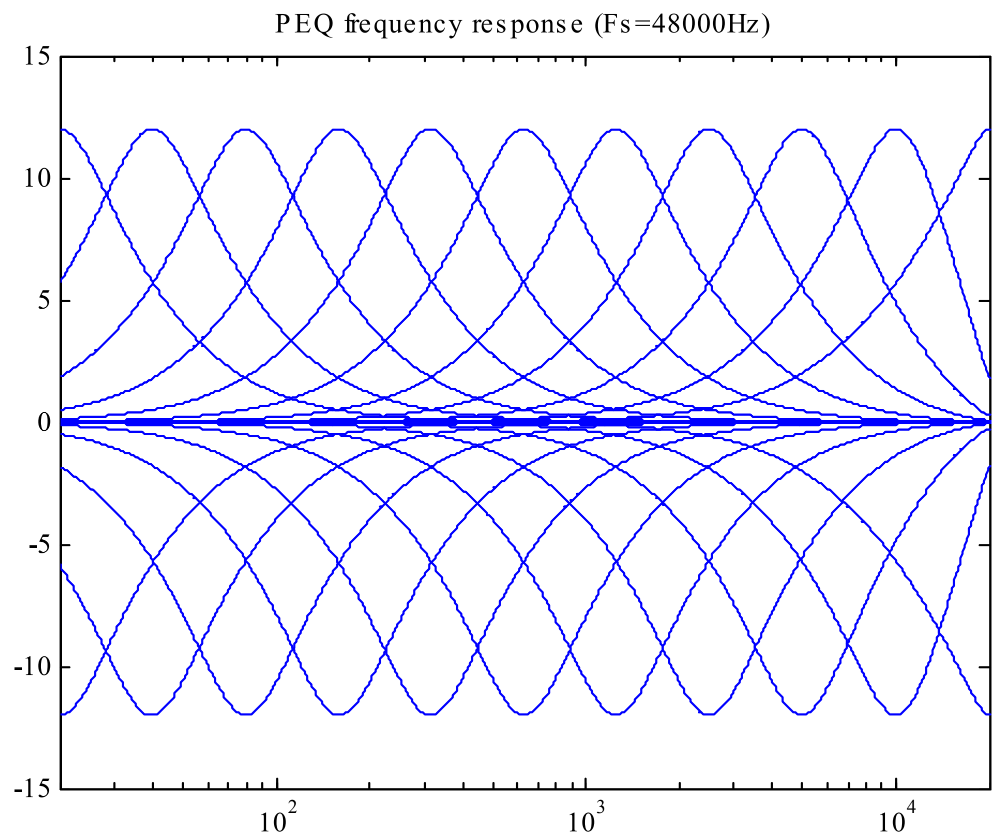


Figure 2-25 Parametric EQ filter response

Figure 2-26 shows the filter response calculated at 640 Hz and Q of 0.707107 with gains from +12 dB to -12 dB in 1 dB steps.

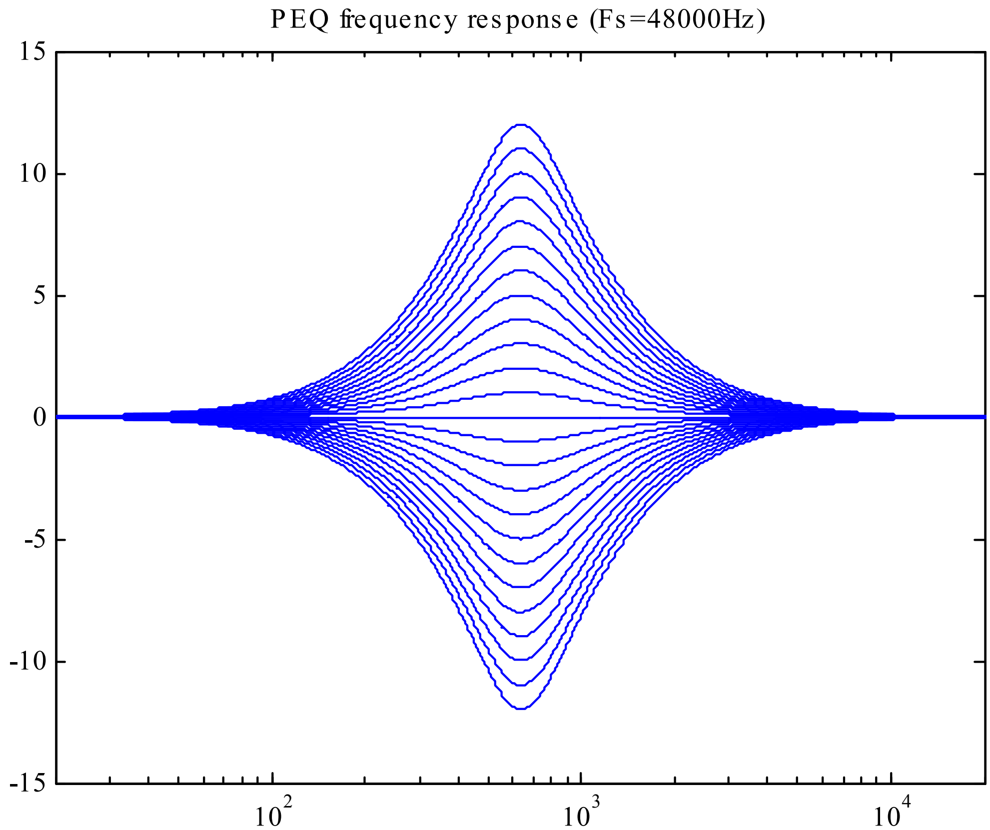


Figure 2-26 Parametric EQ filter response at 640 Hz (Q of 0.707107)

Figure 2-27 shows the filter response calculated at 640 Hz and +12 dB & -12 dB, with Qs from 0.25 to 8.0.

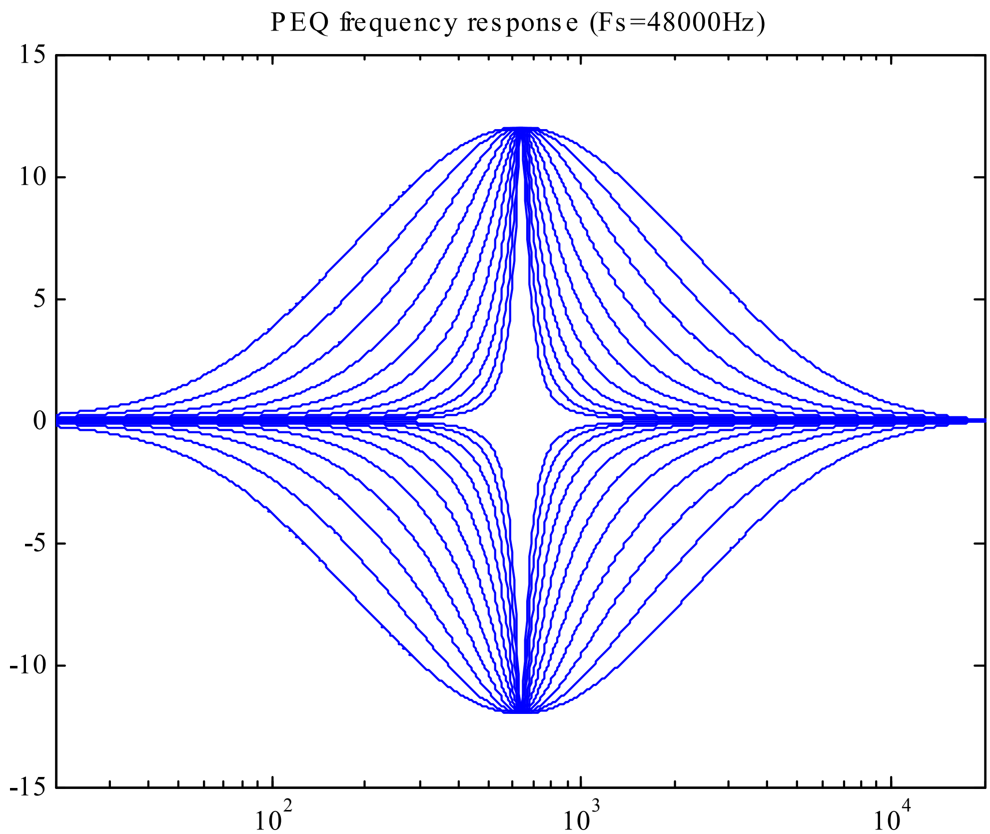


Figure 2-27 Parametric EQ filter response at 640 Hz (Q from 0.5 to 2.0)

A Simulation of coefficient calculation

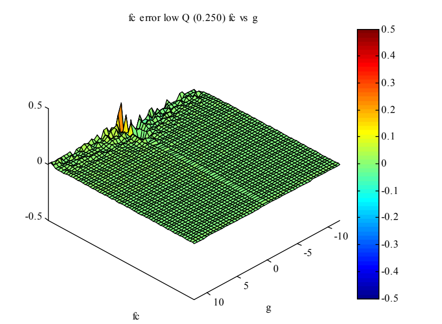
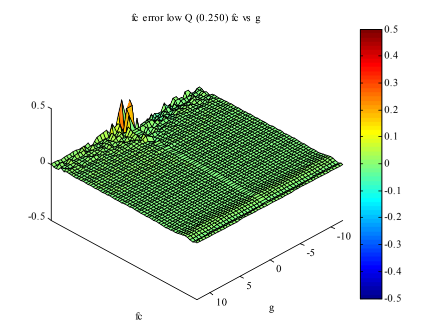
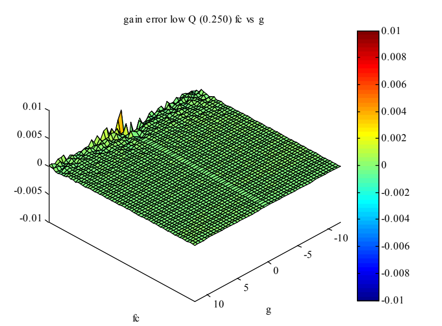
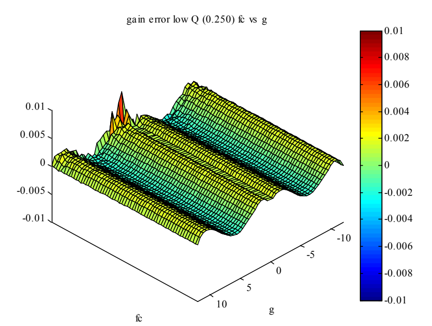
To simulate the coefficient calculation within the DSP using floating point arithmetic, a test program was written in C, which would calculate the coefficients for a parametric filter.

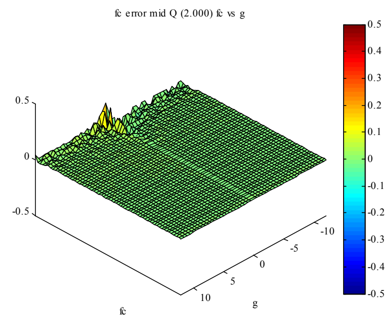
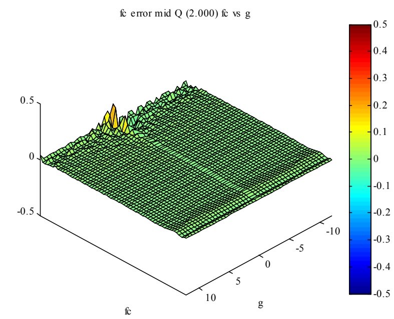
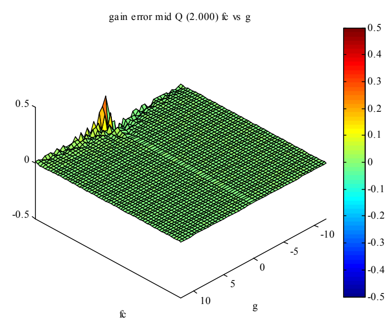
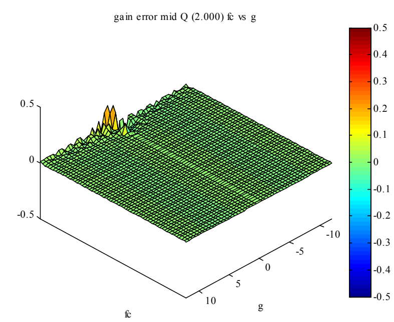
The simulations were performed assuming a 24-bit implementation of a DF-I biquad filter with the coefficients scaled to allow up to 12 dB of gain.

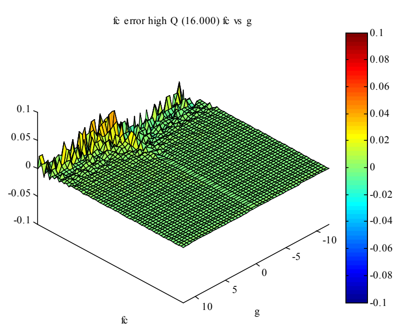
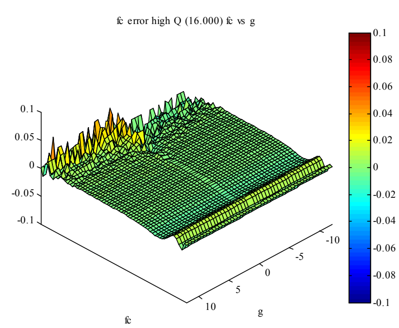
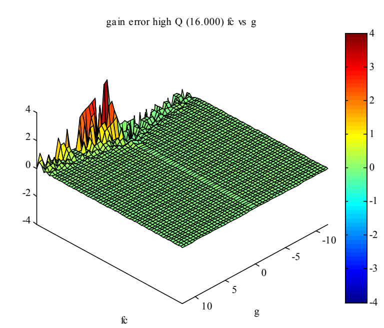
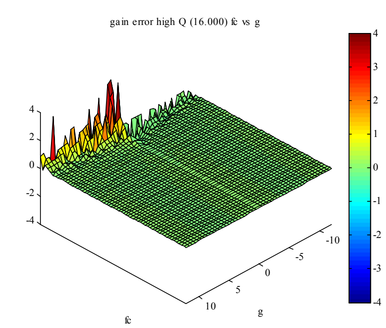
The graphs in this appendix show the error in center frequency as the frequency and gain of the filter are modified. The first graph shows the error when the filter Q is set to its lowest setting, 0.25.

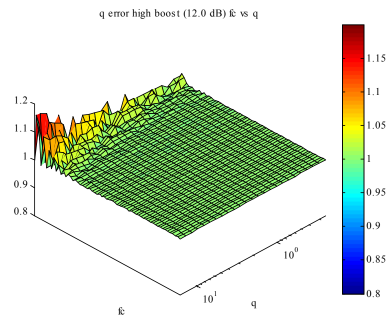
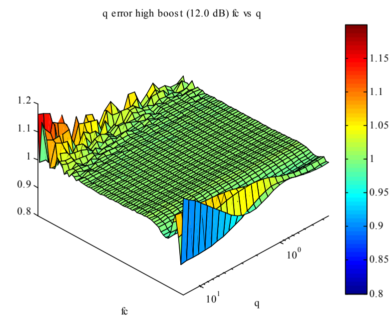
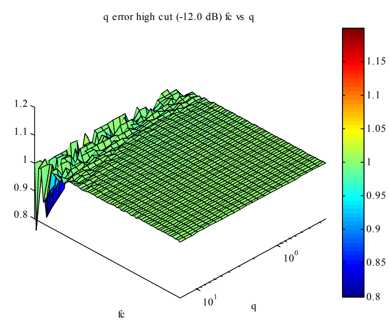
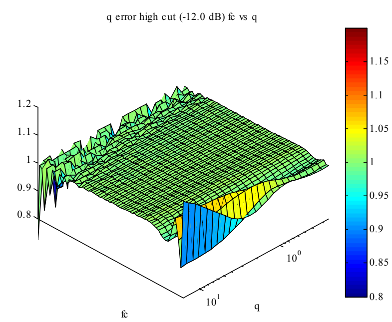
NOTE The first column shows coefficients calculated in C using double precision arithmetic and standard library functions.

The second column shows coefficients calculated using the floating point library simulation.

	24-bit coefficients calculated using double precision maths in C	**24-bit coefficients using floating point library simulation**
Center frequency error with low Q filters		
Gain error with low Q filters		

	24-bit coefficients calculated using double precision maths in C	24-bit coefficients using floating point library simulation
Center frequency error with mid Q filters		
Gain error with mid Q filters		

	24 bit coefficients calculated using double precision maths in C	24 bit coefficients using floating point library simulation
Center frequency error with high Q filters		
Gain error with high Q filters		

	24-bit coefficients calculated using double precision maths in C	**24-bit coefficients using floating point library simulation**
Q error with high boost filters		
Q error with high cut filters		

The tables in this appendix show two main sources of errors in the responses of the filters:

1. The pole zero space of the direct form I filters means that low frequency filters (close to 20 Hz) suffer from distortions in the frequency responses. This is worst for high Q filters.
2. Replacing the complex maths functions in the C library with approximations reduces the filter gain accuracy and also the frequency accuracy, but to an amount, which is often better than errors, introduced due to the coefficient resolution.

The situation where the coefficient calculation routines deviate most significantly from the ideal is with high Q filters at high frequency. This is an area where digital filters traditionally have difficulty due to frequency warping, but may require more investigation to see where the error is being introduced.

Terms and definitions

Term	Definition
ADK	Audio Development Kit
Bluetooth	Set of technologies providing audio and data transfer over short-range radio connections
DSP	Digital Signal Processor
EQ	Equalizer
IC	Integrated Circuit
QTIL	Qualcomm Technologies International, Ltd.