Application Report

April 2010

Bi-quad filters realization on AIC Codecs

HPA Software

ABSTRACT

This application report provides information regarding filter equations and co-efficient format representation that can be used to realize digital filters on the AIC3xxx miniDSP platform. It also explains ways to update the filter coefficients on the fly using a host processor.

Contents

Figures

Tables

1 Standard Bi-quad filter equations

1.1 All Pass (Phase shift) filters

Filter parameters

BW = Bandwidth in Hz
F_c = Center frequency in Hz
F_s=Sample frequency in Hz

Error Checking

$$0 \le BW \le \frac{F_s}{2}$$

$$0 \le F_c \le \frac{F_s}{2}$$

$$0 \le F_s \le 192K$$

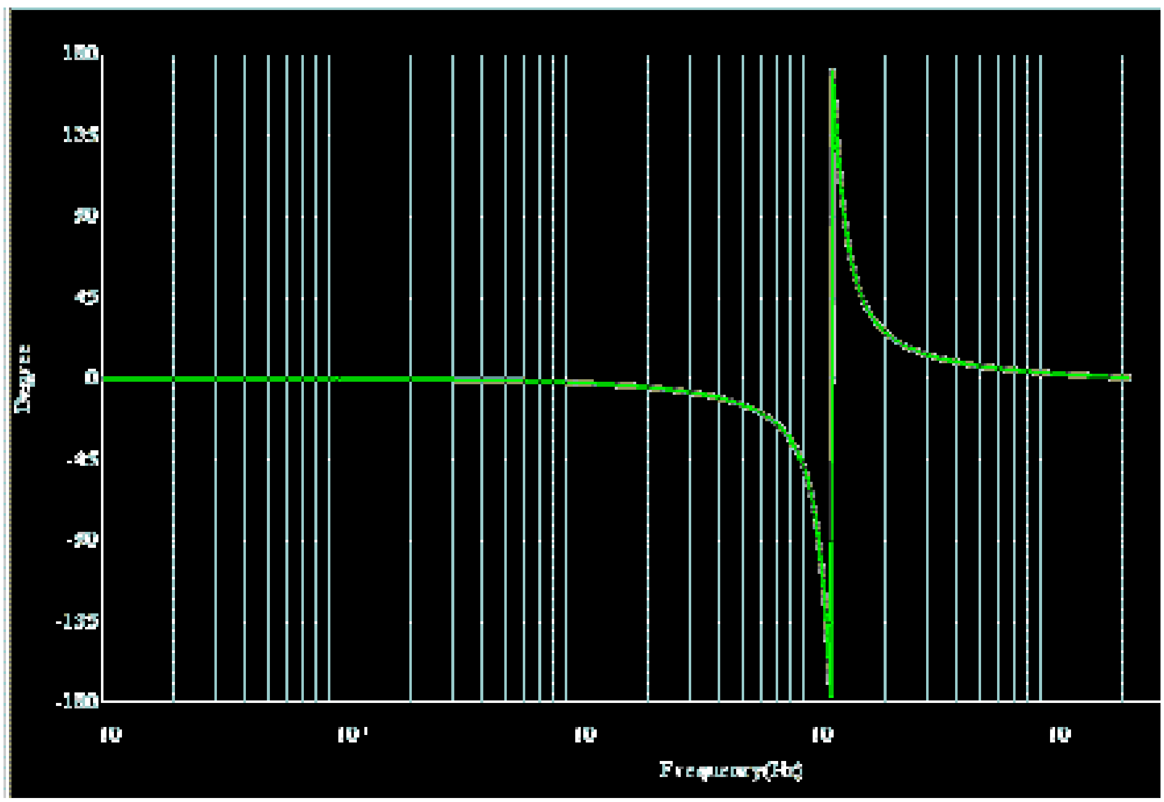

Figure 1. Phase shift filter phase response (F_c=1200Hz, BW=300Hz)

Equation

$$a = \frac{1 - \tan\left(\pi \times \frac{BW}{F_s}\right)}{1 + \tan\left(\pi \times \frac{BW}{F_s}\right)}$$

$$d = -\cos\left(2 \times \pi \times \frac{F_c}{F_s}\right)$$

Filter Coefficients

$$b0 = a$$
$$b1 = d \times (1 + a)$$
$$a1 = b1$$
$$b2 = 1$$
$$a2 = a$$
$$B = [b0\ b1\ b2]$$
$$A = [1\ a1\ a2]$$

1.2 Equalization filters

Filter Parameters

BW = Filter Bandwidth in Hz

F_c = Filter Center Frequency in Hz

F_s = Sample rate in Hz

G = Filter Gain in dB

Error Checking

$$0 \le BW \le \frac{F_s}{2}$$
$$0 \le F_c \le \frac{F_s}{2}$$
$$0 \le F_s \le 192K$$
$$-140 \le G \le 48$$

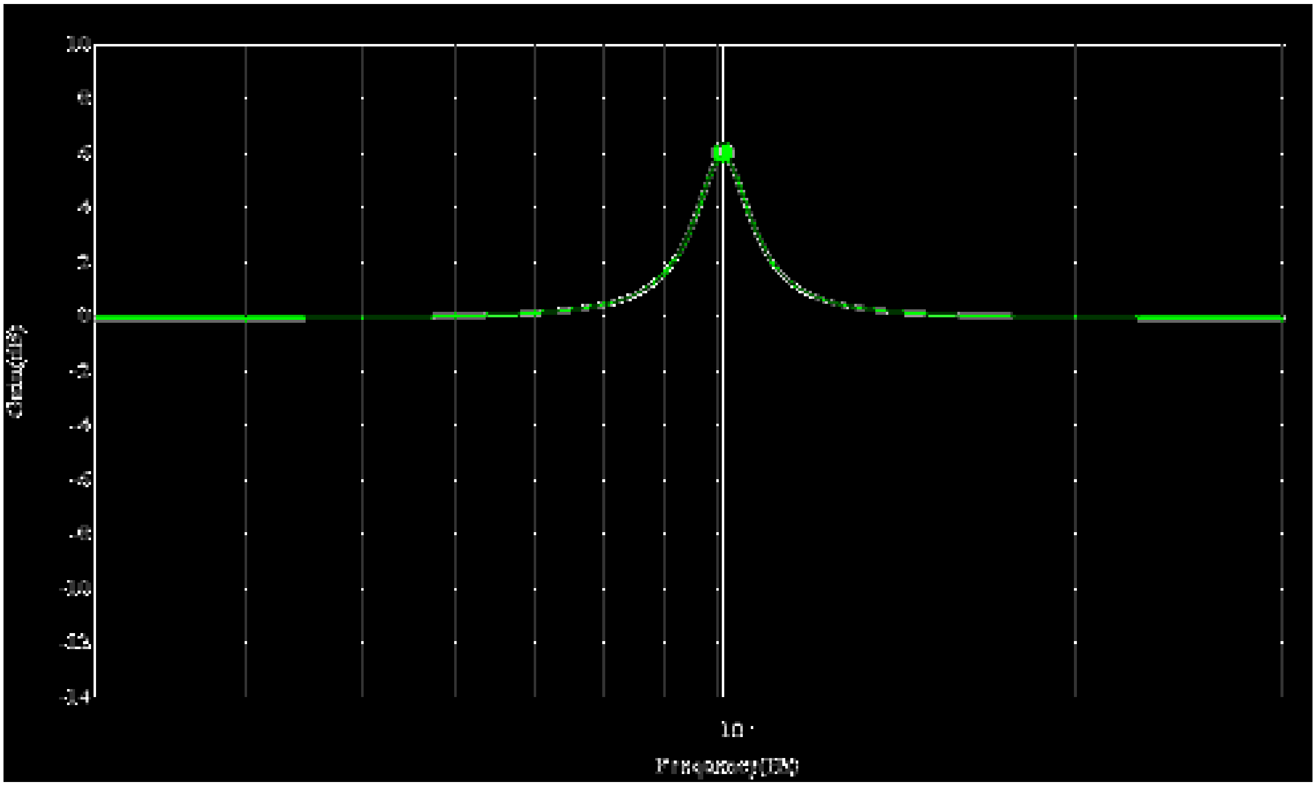

Figure 2. EQ filter with F_c=1kHz, BW=100Hz,G=6dB

Equation

$$A = 10^{\frac{G}{20}}$$

$$if\,(A < 1)$$

$$a = \frac{\left(\tan\left(\pi \times \frac{BW}{F_s}\right) - A\right)}{\left(\tan\left(\pi \times \frac{BW}{F_s}\right) + A\right)}$$

$$else$$

$$a = \frac{\left(\tan\left(\pi \times \frac{BW}{F_s}\right) - 1\right)}{\left(\tan\left(\pi \times \frac{BW}{F_s}\right) + 1\right)}$$

$$H = A - 1$$

$$d = -\cos\left(2 \times \pi \times \frac{F_c}{F_s}\right)$$

Filter Coefficients

$$b0 = 1 + (1 + a) \times \frac{H}{2}$$

$$b1 = d \times (1 + a)$$

$$b2 = \left(-a - (1 + a) \times \frac{H}{2}\right)$$

$$a1 = b1$$

$$a2 = -a$$

$$B = [b0\ b1\ b2]$$

$$A = [1\ a1\ a2]$$

1.3 Notch Filters

Filter Parameters

BW = Notch Bandwidth in Hz

F_c = Notch Center frequency in Hz

F_s=Sample rate in Hz

Error Checking

$$0 \le BW \le \frac{F_s}{2}$$

$$0 \le F_c \le \frac{F_s}{2}$$

$$0 \le F_s \le 192K$$

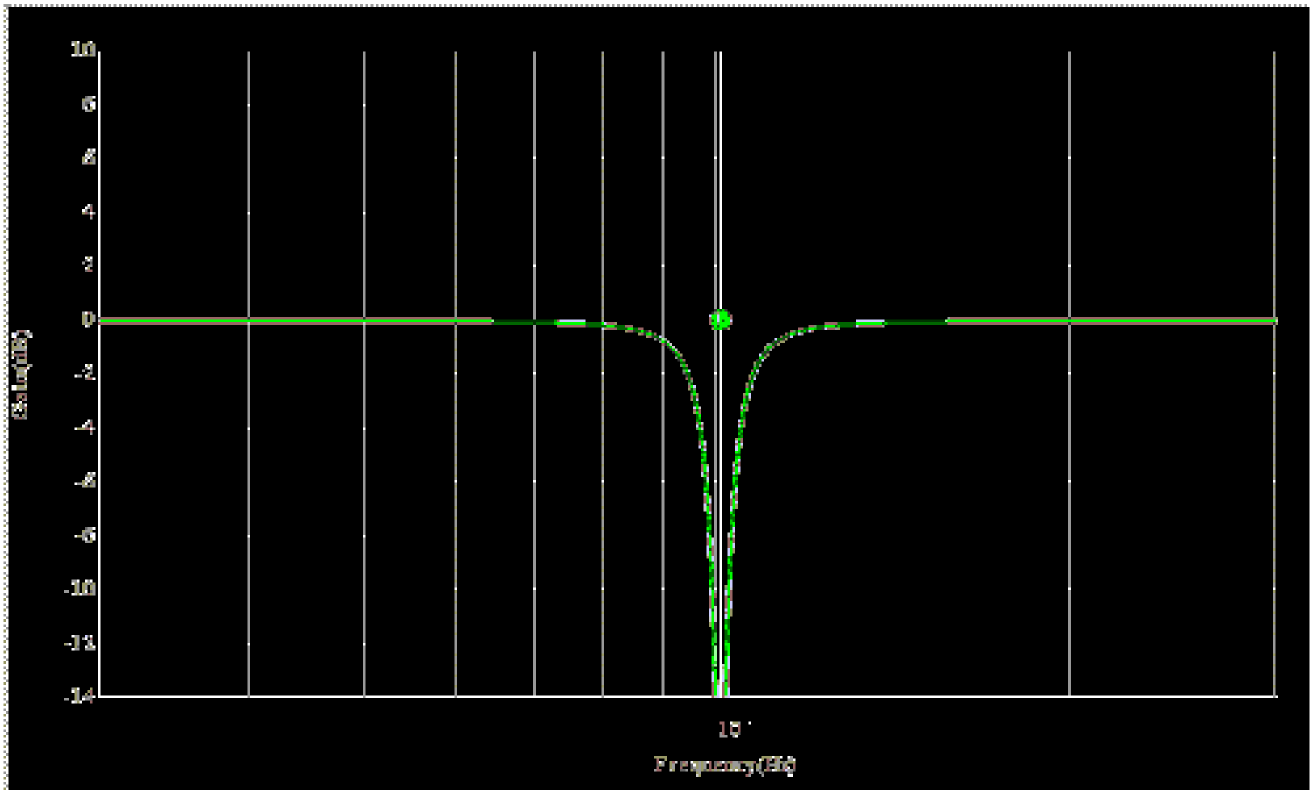

Figure 3. Notch filter magnitude response with F_c=1kHz,BW=100Hz

Equation

$$a = \frac{1 - \tan\left(\pi \times \frac{BW}{F_s}\right)}{1 + \tan\left(\pi \times \frac{BW}{F_s}\right)}$$

$$d = -\cos\left(2 \times \pi \times \frac{F_c}{F_s}\right)$$

$$b0 = a$$

$$b1 = d \times (1 + a)$$

$$a1 = b1$$

$$b2 = 1$$

$$a2 = a$$

Filter Coefficients

$$B = [b0\ b1\ b2]$$

$$A = [1\ a1\ a2]$$

$$B = 0.5 \times (B + A)$$

1.4 Treble Shelf

Filter Parameters

F_c = Treble Shelf Corner frequency in Hz

F_s = Sample rate in Hz

G = Treble Shelf Gain in dB

Error Checking

$$0 \le F_c \le \frac{F_s}{2}$$

$$0 \le F_s \le 192K$$

$$-24 \le G \le 24$$

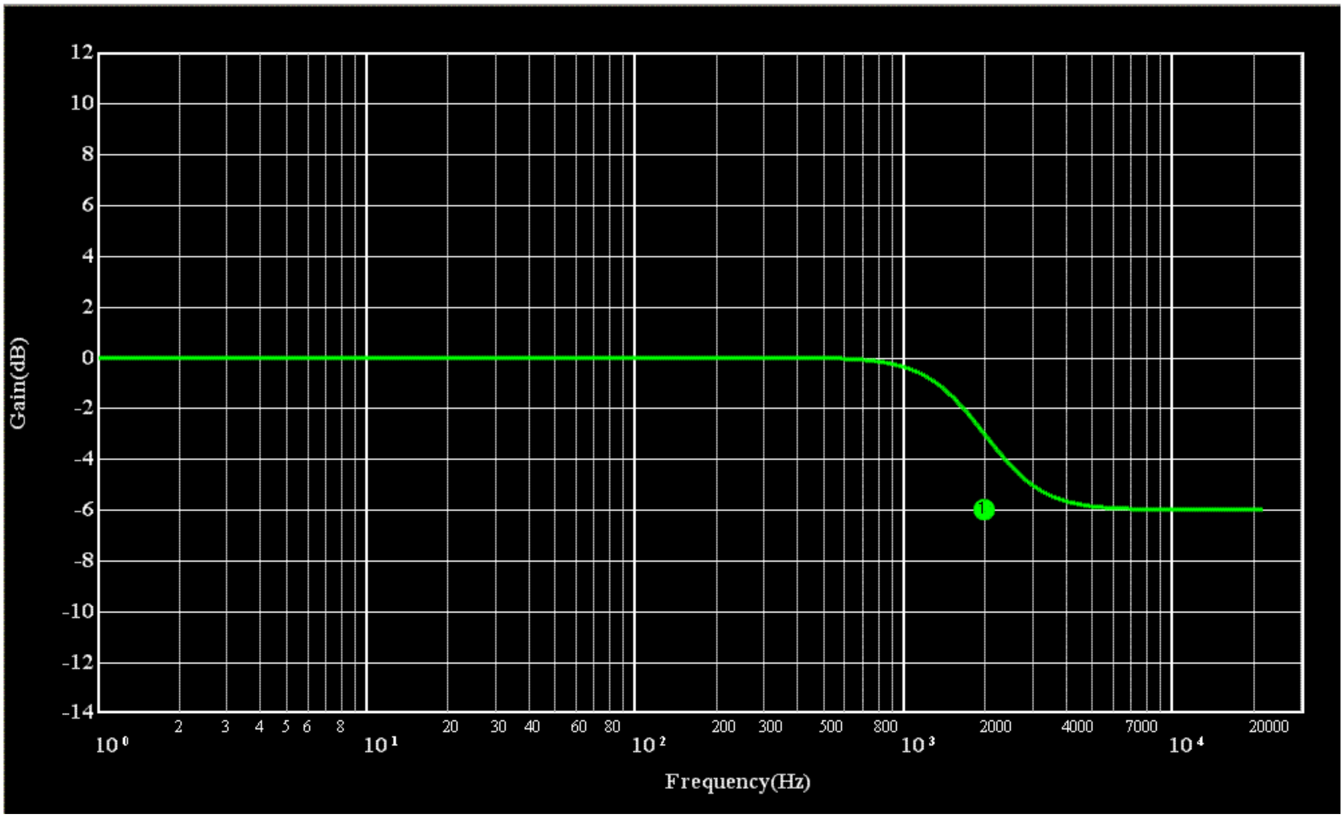

Figure 4. Treble shelf filter magnitude response (F_c=2kHz, G=-6dB)

Equation

$$g = 10^{\frac{G}{20}}$$

$$s = \frac{sqrt(2)}{2}$$

$$\rho = \frac{\pi}{2}$$

$$\varphi = \frac{F_c}{F_s} \times \pi$$

$$A = g$$

$$G = 20 \times \log 10(A)$$

```
If G > -6 & G < 6
```
$$F = sqrt(A)$$
```
elseif A > 1
```
$$F = \frac{A}{sqrt(2)}$$
```
else
```
$$F = A \times sqrt(2)$$
```
end
```

$$gd = \sqrt[4]{\frac{(F^2 - 1)}{A^2 - F^2}}$$

$$gn = sqrt(A) \times gd$$

$$a = \tan\left(\pi \times \left(\frac{F_c}{F_s} - \frac{1}{4}\right)\right)$$

$$b0 = \frac{gn^2 \times a^2 + 2 \times s \times gn - 2 \times gn^2 \times a + 1 - 2 \times s \times gn \times a^2 + a^2 + gn^2 + 2 \times a}{1 + gd^2 + 2 \times s \times gd - 2 \times s \times gd \times a^2 + gd^2 \times a^2 - 2 \times gd^2 \times a + a^2 + 2 \times a}$$

$$b1 = \frac{2 - 2 \times gn^2 \times a^2 + 4 \times gn^2 \times a + 4 \times a - 2 \times gn^2 + 2 \times a^2}{1 + gd^2 + 2 \times s \times gd - 2 \times s \times gd \times a^2 + gd^2 \times a^2 - 2 \times gd^2 \times a + a^2 + 2 \times a}$$

$$b2 = \frac{1 + 2 \times s \times gn \times a^2 - 2 \times s \times gn + 2 \times a + a^2 + gn^2 - 2 \times gn^2 \times a + gn^2 \times a^2}{1 + gd^2 + 2 \times s \times gd - 2 \times s \times gd \times a^2 + gd^2 \times a^2 - 2 \times gd^2 \times a + a^2 + 2 \times a}$$

$$a0 = 1$$

$$a1 = \frac{2 - 2 \times gd^2 \times a^2 + 4 \times gd^2 \times a + 2 \times a^2 - 2 \times gd^2 + 4 \times a}{1 + gd^2 + 2 \times s \times gd - 2 \times s \times gd \times a^2 + gd^2 \times a^2 - 2 \times gd^2 \times a + a^2 + 2 \times a}$$

$$a2 = \frac{1 - 2 \times gd^2 \times a + 2 \times a + gd^2 - 2 \times s \times gd + a^2 + gd^2 \times a^2 + 2 \times s \times gd \times a^2}{1 + gd^2 + 2 \times s \times gd - 2 \times s \times gd \times a^2 + gd^2 \times a^2 - 2 \times gd^2 \times a + a^2 + 2 \times a}$$

Filter Coefficients

$$B = [b0\ b1\ b2]$$
$$A = [a0\ a1\ a2]$$

1.5 Bass Shelf Filters

Filter Parameters

F_c = Bass Shelf Corner frequency in Hz

F_s = Sample rate in Hz

G = Bass Shelf Gain in dB

Error Checking

$$0 \le F_c \le \frac{F_s}{2}$$

$$0 \le F_s \le 192K$$

$$-24 \le G \le 24$$

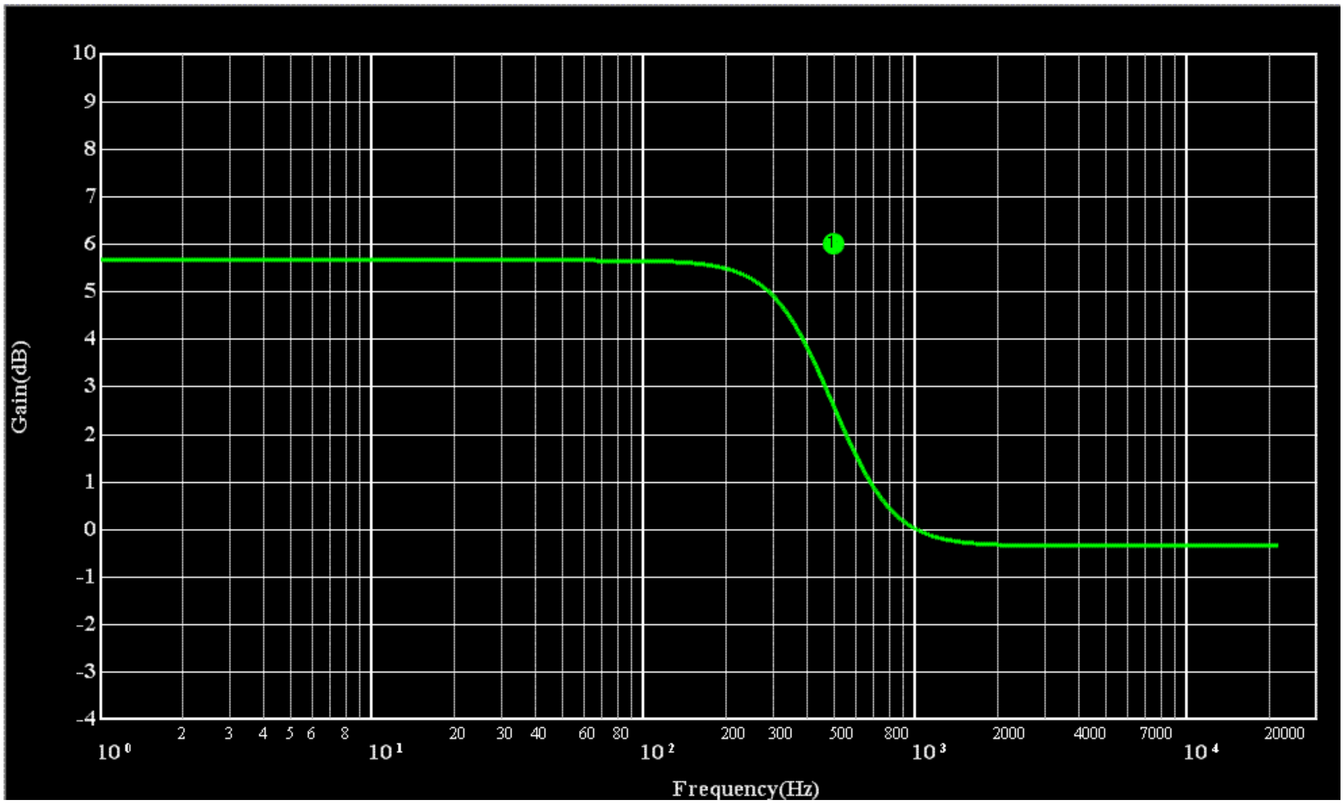

Figure 5. Bass shelf filter magnitude response(F_c=500Hz,G=6dB)

Equation

$$g = 10^{\frac{G}{20}}$$

$$s = \frac{sqrt(2)}{2}$$

$$\rho = \frac{\pi}{2}$$

$$\varphi = \frac{F_c}{F_s} \times \pi$$

$$A = g$$

$$G = 20 \times \log 10(A)$$

```
If G > -6 & G < 6
```

$$F = sqrt(A)$$

```
elseif A > 1
```

$$F = \frac{A}{sqrt(2)}$$

```
else
```

$$F = A \times sqrt(2)$$

```
end
```

$$gd = \sqrt[4]{\frac{(F^2 - 1)}{A^2 - F^2}}$$

$$gn = sqrt(A) \times gd$$

$$a = \tan\left(\pi \times \left(\frac{F_c}{F_s} - \frac{1}{4}\right)\right)$$

$$b0 = \frac{-\left(-1 - gn^2 \times a^2 - a^2 - 2 \times gn^2 \times a - gn^2 - 2 \times s \times gn + 2 \times s \times gn \times a^2 + 2 \times a\right)}{2 \times s \times gd + 1 - 2 \times s \times gd \times a^2 + gd^2 \times a^2 + 2 \times gd^2 \times a + a^2 + gd^2 - 2 \times a}$$

$$b1 = \frac{-\left(2 - 4 \times a - 4 \times gn^2 \times a - 2 \times gn^2 \times a^2 - 2 \times gn^2 + 2 \times a^2\right)}{2 \times s \times gd + 1 - 2 \times s \times gd \times a^2 + gd^2 \times a^2 + 2 \times gd^2 \times a + a^2 + gd^2 - 2 \times a}$$

$$b2 = \frac{1 + 2 \times s \times gn \times a^2 - 2 \times a + gn^2 - 2 \times s \times gn + 2 \times gn^2 \times a + a^2 + gn^2 \times a^2}{2 \times s \times gd + 1 - 2 \times s \times gd \times a^2 + gd^2 \times a^2 + 2 \times gd^2 \times a + a^2 + gd^2 - 2 \times a}$$

$$a0 = 1$$

$$a1 = \frac{-2 + 2 \times gd^2 \times a^2 + 4 \times gd^2 \times a - 2 \times a^2 + 2 \times gd^2 + 4 \times a}{2 \times s \times gd + 1 - 2 \times s \times gd \times a^2 + gd^2 \times a^2 + 2 \times gd^2 \times a + a^2 + gd^2 - 2 \times a}$$

$$a2 = \frac{gd^2 \times a^2 - 2 \times a + 1 + 2 \times gd^2 \times a - 2 \times s \times gd + a^2 + 2 \times s \times gd \times a^2 + gd^2}{2 \times s \times gd + 1 - 2 \times s \times gd \times a^2 + gd^2 \times a^2 - 2 \times gd^2 \times a + a^2 + gd^2 - 2 \times a}$$

Filter Coefficients

$$B = [b0\ b1\ b2]$$
$$A = [a0\ a1\ a2]$$

1.6 Second order Linkwitz Riley

Filter Parameters

F_s = Sample rate in Hz

F_c = Cut frequency in Hz

HL=LR Filter type (high, low)

Error Checking

$$0 \le F_c \le \frac{F_s}{2}$$
$$0 \le F_s \le 192K$$
$$HL = (high, low)$$

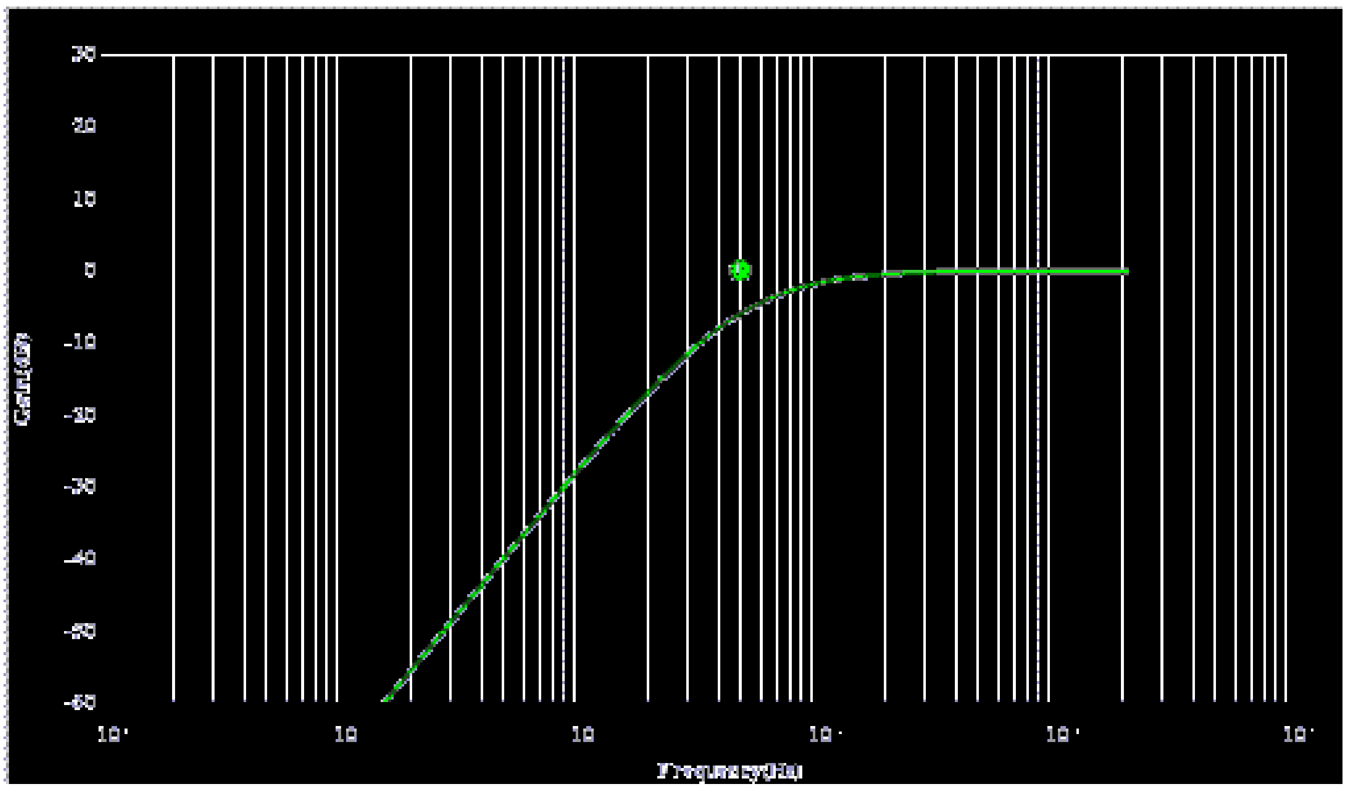

Figure 6. Second order Linkwitz Riley (F_c=500Hz, HL=high)

Equation

$$
\begin{aligned}
&wc = 2\times\pi\times F_c\\
&if\ \ HL(1:3) == 'low'\\
&\quad Ba = \begin{bmatrix}0 & 0 & wc^2\end{bmatrix}\\
&\quad Aa = \begin{bmatrix}1 & 2\times wc & wc^2\end{bmatrix}\\
&else\\
&\quad Ba = \begin{bmatrix}1 & 0 & 0\end{bmatrix}\\
&\quad Aa = \begin{bmatrix}1 & 2\times wc & wc^2\end{bmatrix}\\
&end\\
&k = 2\times\pi\times\frac{F_c}{\tan\left(\pi\times\frac{F_c}{F_s}\right)}\\
&B = \left[Ba(1)\times k^2 + Ba(3) + Ba(2)\times k, -2\times Ba(1)\times k^2 + 2\times Ba(3), -Ba(2)\times k + Ba(1)\times k^2 + Ba(3)\right]\\
&A = \left[Aa(1)\times k^2 + Aa(3) + Aa(2)\times k, -2\times Aa(1)\times k^2 + 2\times Aa(3), -Aa(2)\times k + Aa(1)\times k^2 + Aa(3)\right]
\end{aligned}
$$

Filter Coefficients

$$B = \frac{B}{A(1)}$$

$$A = \frac{A}{A(1)}$$

1.7 Second Order Variable Q Filter

Filter Parameters

F_s = Sample rate in Hz

F_c = Cut frequency in Hz

HL=LR Filter type (high, low)

Q=Filter Q $\left(s^2 + \frac{wc}{Q}\times s + wc^2\right)$

Error Checking

$$0 \le F_c \le \frac{F_s}{2}$$

$$0 \le F_s \le 192K$$

$$HL = (high, low)$$

$$0 \le Q \le 100$$

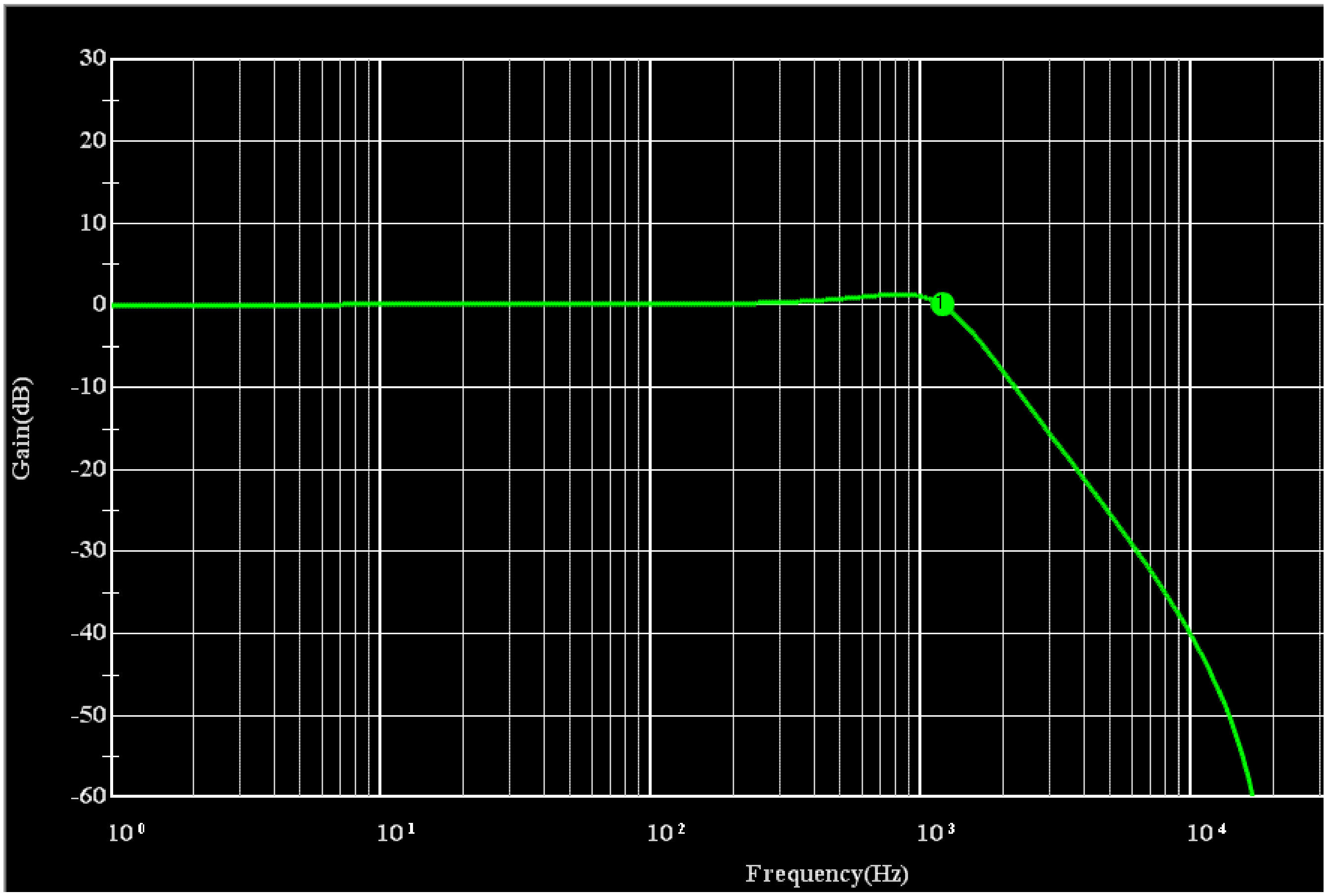

Figure 7. **Second order variable Q filter(F_c=1200Hz,Q=1,HL=low)**

Equation

$$
\begin{aligned}
& wc = 2 \times \pi \times F_c \\
& if\ HL(1:3) == 'low' \\
& \quad Ba = \begin{bmatrix} 0 & 0 & wc^2 \end{bmatrix} \\
& \quad Aa = \begin{bmatrix} 1 & \frac{wc}{Q} & wc^2 \end{bmatrix} \\
& else \\
& \quad Ba = \begin{bmatrix} 1 & 0 & 0 \end{bmatrix} \\
& \quad Aa = \begin{bmatrix} 1 & \frac{wc}{Q} & wc^2 \end{bmatrix} \\
& end \\
& k = 2 \times \pi \times \frac{F_c}{\tan\left(\pi \times \frac{F_c}{F_s}\right)} \\
& B = \left[Ba(1) \times k^2 + Ba(3) + Ba(2) \times k, -2 \times Ba(1) \times k^2 + 2 \times Ba(3), -Ba(2) \times k + Ba(1) \times k^2 + Ba(3)\right] \\
& A = \left[Aa(1) \times k^2 + Aa(3) + Aa(2) \times k, -2 \times Aa(1) \times k^2 + 2 \times Aa(3), -Aa(2) \times k + Aa(1) \times k^2 + Aa(3)\right]
\end{aligned}
$$

Filter Coefficients

$$B = \frac{B}{A(1)}$$

$$A = \frac{A}{A(1)}$$

1.8 Second order Butterworth Filter from Variable Q

Second order Butterworth filter can be realized by using variable Q filter with Q = 0.707

1.9 Second order Bessel Filter from Variable Q

Second order Bessel filter can be realized by using variable Q filter with Q = 0.5

1.10 First Order Butterworth Filters

Filter Parameters

F_s = Sample rate in Hz

F_c = Cut frequency in Hz

HL=LR Filter type (high, low)

Error Checking

$$0 \le F_c \le \frac{F_s}{2}$$
$$0 \le F_s \le 192K$$
$$HL = (high, low)$$

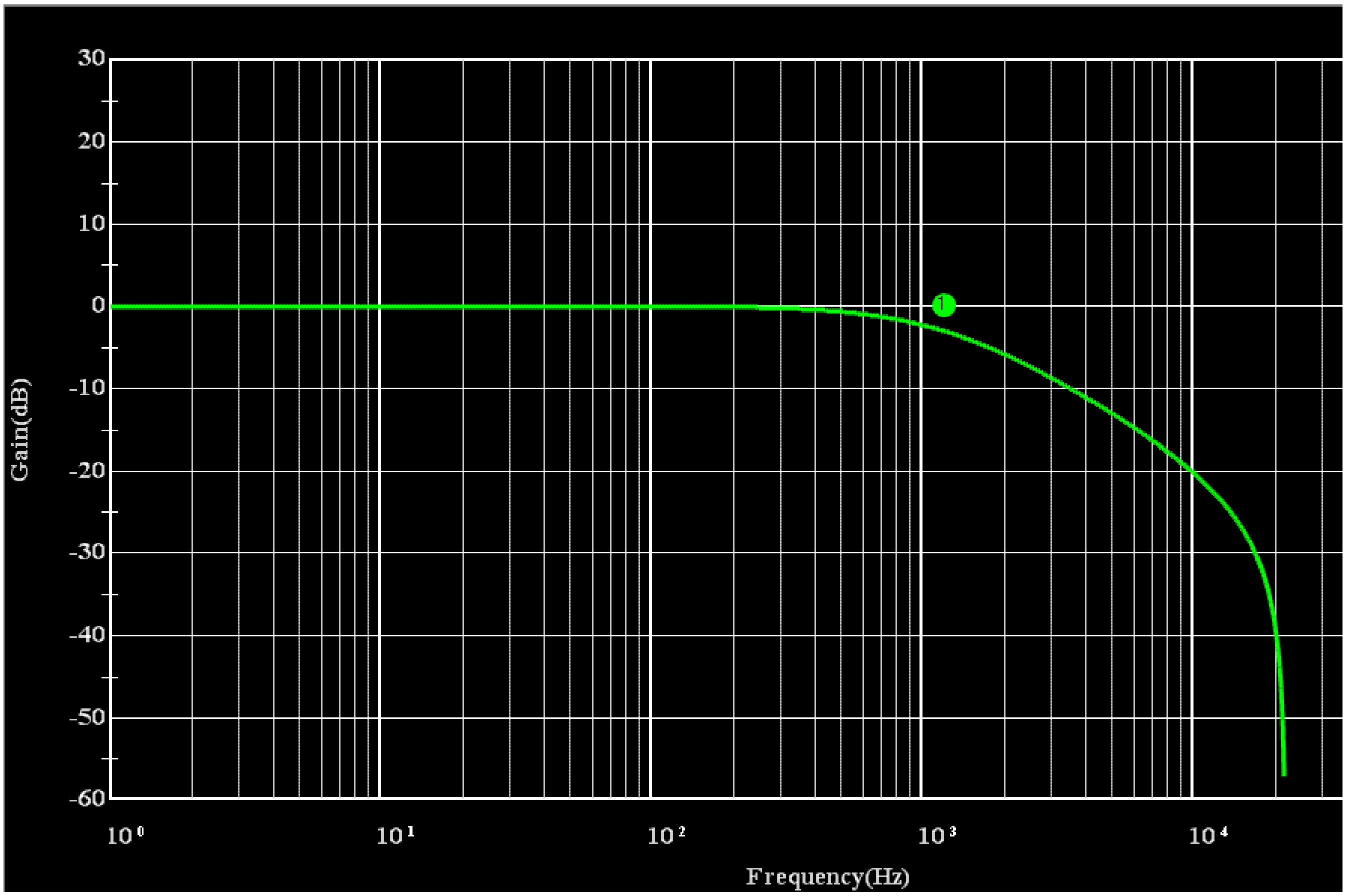

Figure 8. First order Butterworth(F_c=1200Hz, HL=low)

Equation

$$k = \frac{2 \times \pi \times F_c}{\tan\left(\frac{\pi \times F_c}{F_s}\right)}$$

$$Wc = 2 \times \pi \times F_c$$

$$If\ \ HL(1:3) =' low'$$

$$b0 = \frac{Wc}{k + Wc}$$

$$b1 = \frac{Wc}{k + Wc}$$

$$else$$

$$b0 = \frac{k}{k + Wc}$$

$$b1 = -\frac{k}{k + Wc}$$

$$end$$

$$a1 = \frac{Wc - k}{k + Wc}$$

Filter Coefficients

$$B = [b0\ \ b1\ \ 0]$$

$$A = [1\ \ a1\ \ 0]$$

1.11 Second order Chebychev

Filter Parameters

F_s=Sample rate in Hz

rip=Ripple specification in dB

typ=Filter type (high,low,stop)

```
if typ(1:3) == 'sto'
        Fc=Stop band Input Lower and upper frequencies [f1,f2]
Else
        Fc=Cutoff frequency in Hz
        If Scale peak to 0dB
                      Nrm=1
        If Scale PB to 0dB
                      Nrm = -1
```

$$\begin{aligned} &if \quad nrm \cong 1 \\ &\quad rip = rip \times -1 \\ &end \end{aligned}$$

```
End
```

Error Checking

$$0 \le F_c \le \frac{F_s}{2}$$

$$0 \le F_s \le 192K$$

$$0 \le rip \le 10$$

$$Nrm = 1, -1$$

$$HL = (high, low, stop)$$

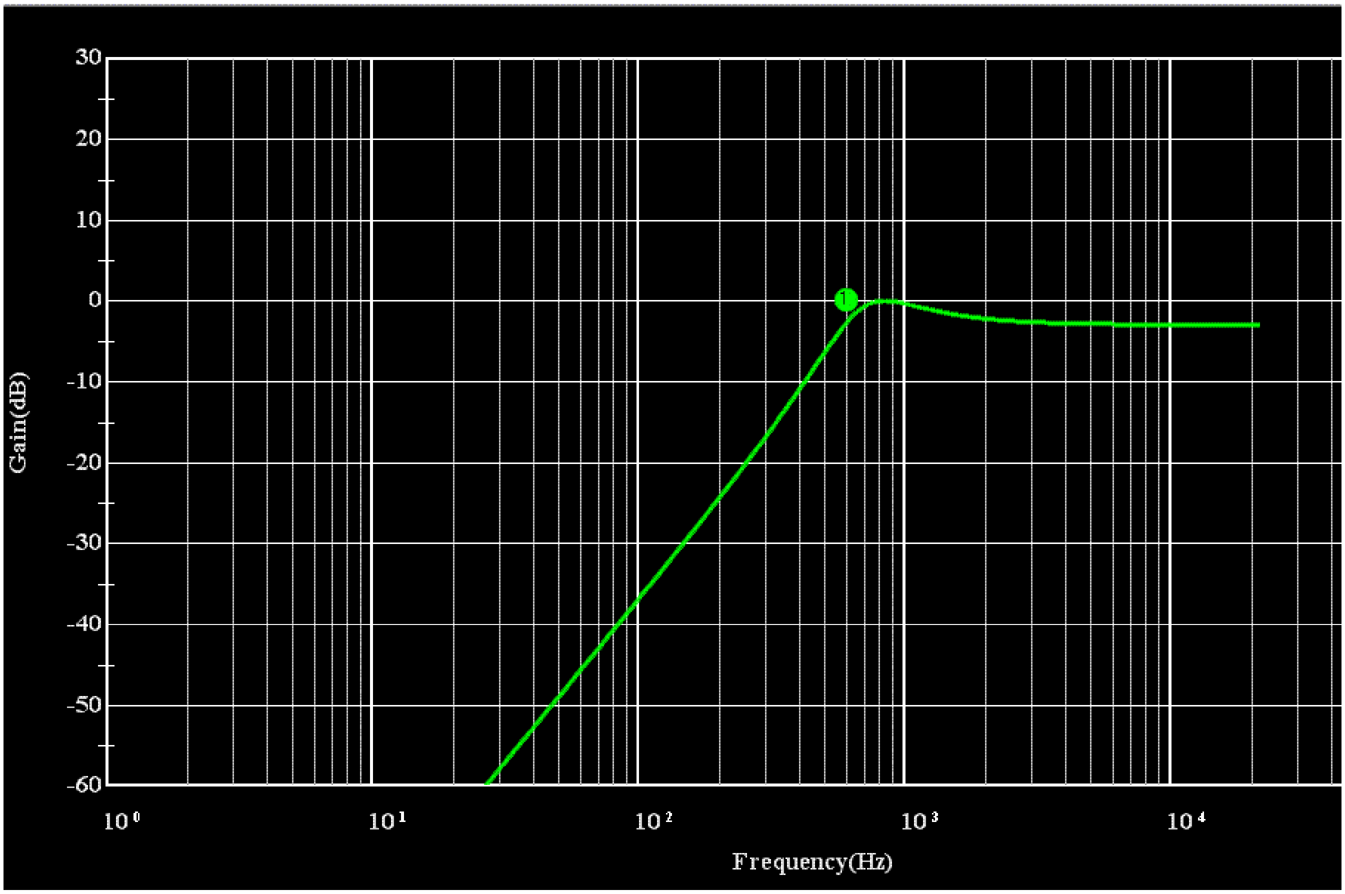

Figure 9. Second order Chebychev (F_c=600Hz, typ=high, rip=3dB)

Equation

```
If typ(1:3)=='sto'
        Call cheby1(ord,rip, (2×Fc)/Fs ,HL)
else
        Call soCHBI(Fc,Fs,rip,HL)
End
```

$$\frac{2\times F_c}{F_s}$$

```
Function soCHBI(Fc,Fs,rip,HL)

       If sign(rip)>0
                     Sf=1
       Else
                     Sf=0
       End
```

$R = |rip|$

$If\ \ R == 0$

$B = [1\ 0\ 0]$

$A = [1\ 0\ 0]$

$else$

$wc = 2 \times \pi \times F_c$

$\varepsilon = \sqrt{10^{\frac{R}{10}} - 1}$

$\alpha = \dfrac{a\sinh\left(\frac{1}{\varepsilon}\right)}{2}$

$\beta 1 = 3 \times \dfrac{\pi}{4}$

$\beta 2 = 5 \times \dfrac{\pi}{4}$

$s1 = \sinh(\alpha) \times \cos(\beta 1) + \cosh(\alpha) \times \sin(\beta 1) \times i$

$s2 = \sinh(\alpha) \times \cos(\beta 2) + \cosh(\alpha) \times \sin(\beta 2) \times i$

$a = real(s1 + s2)$

$b = real(s1 \times s2)$

$c = b$

$if\ \ sf$

$c = \dfrac{c}{\sqrt{1 + \varepsilon^2}}$

end

$if\ \ HL(1:3) == 'low'$

$Ba = \left[0\ \ 0\ \ c \times wc^2\right]$

$Aa = \left[1\ \ wc \times a\ \ b \times wc^2\right]$

Else

$Ba = \left[\frac{c}{b}\ \ 0\ \ 0\right]$

$Aa = \left[1\ \ wc \times \frac{a}{b}\ \ \frac{wc^2}{b}\right]$

End

$$k = 2 \times \pi \times \frac{F_c}{\tan\left(\pi \times \frac{F_c}{F_s}\right)}$$

$$B = \left[Ba(1) \times k^2 + Ba(3) + Ba(2) \times k, -2 \times Ba(1) \times k^2 + 2 \times Ba(3), -Ba(2) \times k + Ba(1) \times k^2 + Ba(3)\right]$$

$$A = \left[Aa(1) \times k^2 + Aa(3) + Aa(2) \times k, -2 \times Aa(1) \times k^2 + 2 \times Aa(3), -Aa(2) \times k + Aa(1) \times k^2 + Aa(3)\right]$$

$$B = \frac{B}{A(1)}$$

$$A = \frac{A}{A(1)}$$

```
End
```

2 Number representation format for filter coefficients

AIC codecs comes in two flavors. The “standard” version of the MiniDSP uses a 16 bit coefficient and a 24 bit data word for miniDSP_A and 28 bit data word for miniDSP_D while the “enhanced” version of the MiniDSP uses a 24 bit coefficient and a 32 bit data word. The AIC3254 and AIC3204 are enhanced devices. The TSC2117, AIC36, AC3110, AIC3111, and AIC3120 are standard devices.

All of the AIC codec devices use a 3.x data format (3.21 for the standard devices and 3.29 for the enhanced devices). This permits only two magnitude bits of headroom for signals that are greater then 1. To reduce the chance of clipping the signal in the AIC devices, the overall gain of the filter is moderated by scaling the numerator value based upon the value of the b0 term.

The coefficient size in the AIC codec family is 16 bit 1.15 format for the standard devices and a 24 bit 1.23 format for the enhanced AIC devices. With these formats the MiniDSP coefficients are able to represent a maximum positive gain of $1\text{-}2^{-15}$ for a 16 bit coefficient and $1\text{-}2^{-23}$ for a 24 bit coefficient. When filter coefficients are computed for an AIC codec, the gain of the filter response is scaled to permit the values to be represented in a 1.23 or a 1.15 format.

Once we have computed the filter equations from above we then must perform a couple of steps prior to loading the coefficients into the codec.

In the AIC codecs use a specific biquad implementation to accommodate the 1.15 and 1.23 coefficient data format.

$$y(n) = b_0 \times x(n) + 2 \times b_1 \times x(n-1) + b_2 \times x(n-2) + 2a_1 \times y(n-1) + a_2 \times y(n-2)$$

1. For format we first must scale the b terms by the b0 value. The b0, b1, and b2 terms are multiplied by a scaling value to limit the overall gain of the filter.
   - If b0 is greater than 1, then the default value for the scaling value is 1/b0, otherwise it is 1
   - The scaling factor is then applied to the b0, b1, and b2 terms of the filter.
   - This default value is computed and displayed in a user modifiable field when the filter coefficients are computed.
   - The user is permitted to change this value to a smaller value. However if they attempt it set it to a larger value than the default scaling value then the value will snap to the default scaling value.
2. Then both the numerator and denominator coefficients are scaled by a constant value
   - If the coefficients are being computed for a “standard” device then

     The b0, b1, b2, a1, and a2 terms are multiplied by 2^{15} and rounded to integer

   - If the coefficients are being computed for an “enhanced” device then

     The b0, b1, b2, a1, and a2 terms are multiplied by 2^{23} and rounded to integer.
3. Then we scale the b1 and a1 terms by 0.5.

Appendix A illustrates the generation of coefficients using the above procedure.

2.1 Filter coefficient normalization

Filter coefficient normalization is performed to limit the size of the coefficient values, as described above, and to limit the maximum gain of the filter to avoid clipping.

There are two places where clipping can occur.

1. Internal signal levels and clipping

   The miniDSP is able to internally represent a data value using a 3.29 or 3.21 format. This permits signal levels as large as 12 dB to be represented. However, the intermediate gains inside of a filter can be higher than the signal levels that are visible at the output of a component. To account for this, filter gains are typically scaled to reserve part or all of this 12 dB as headroom for internal computations.

2. Output signal levels and clipping

   The maximum signal level that can be output without clipping by the I2S output or the DAC interpolator is a data value represented as a by a 1.29 or 1.21 value (0.99 and -1.0). This corresponds to a signal level of 0 dB.

Coefficient scaling can be used to avoid clipping during internal operations and when the signal is sent to the I^2S or interpolator outputs.

For example, we wish to use an EQ filter that has a gain of 15 dB at 500 Hz. This filter response is shown as the blue curve in Figure 10. To avoid overflow for our internal representation, we normalized the filter coefficients so that the signal that is output does not exceed 5 dB. In this example, the filter gain is scaled by 10 dB (multiplying the coefficients by 0.316227766). The scaled filter response is shown by the red curve in Figure 10.

Similarly to avoid clipping when the signal was output, the filter gain should be scaled so that it does not exceed 0 dB. In this example, the filter gain is scaled by 15 dB (multiplying the coefficients by 0.177827941).. This is shown by the green curve in Figure 10.

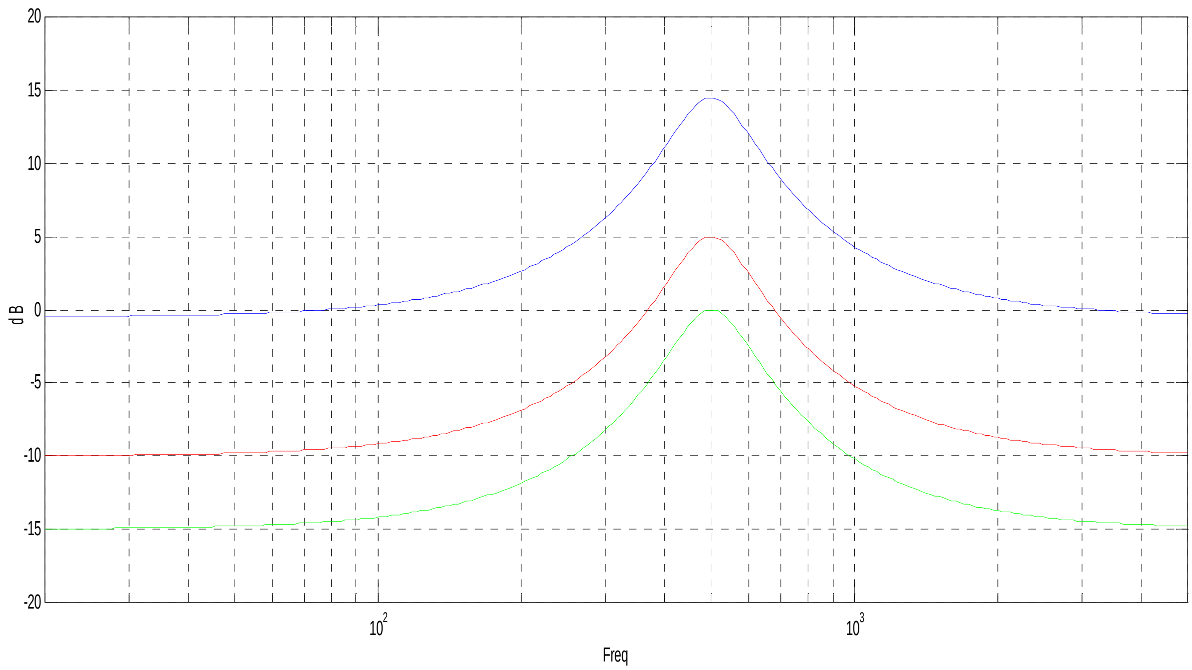

Figure 10. Normalizing filter response

3 Updating the codec filter coefficients

A host controller needs to provide the filter coefficient values over I2C to the AIC codecs. On the controller, the coefficient values need to be pre computed and stored in a table. The host controller will read the values from the table and download the coefficient values in a sequence that starts at the current biquad filter setting to the desired new value when an update is requested. The change of gains must be in increments/decrements of ¼ dB to avoid pops and clicks. For instance, if an EQ filter at 200Hz needs to be operated between ranges of -8dB to 8dB, the host controller needs to maintain a table of b0, b1, b2, a1 and a2 terms through -8db, -7.75dB, -7.5dB… 7dB, 7.25dB, 7.5dB, 7.75dB, 8dB.

Table 1. EQ Filter coefficients table in host controller for "enhanced" codecs

	b0	b1	b0	a1	a2
-8dB	0x7F5119	0x812FC5	0x7E69BC	0x7ED03B	0x82452A
-7.75dB					
-7.5dB					
-7.25dB					
…					
0 dB	0x7FFFFF	0	0	0	0
…					
7.5dB					
7.75dB					
8dB	0x7F6664	0x81C7F3	0x7D23F5	0x7F7E70	0x80E89C

Appendix A. Biquad coefficients computation example

This appendix illustrates the computation biquad coefficients and converting them into format required to load in AIC codecs.

1. Filter Specification: EQ filter with F_c = 5000 Hz, Gain = 6dB and Q=2.87 (BW=1742Hz) on AIC3254.

   Applying the equations in section 1.2 we get the biquad coefficients as

   $b0 = 1.03381744095486$

   $b1 = -1.85413395878212$

   $b2 = 0.898225719031722$

   $a1 = 1.85413395878212$

   $a2 = -0.932043159986584$

   Before writing these coefficients to codec memory, couple of normalization steps must be performed based on their values according to section 2.

2. Since b0 > 1, we need to scale the numerator coefficients by scale factor $\frac{1}{b0}$

   $b0 = 0.992464542388916015625$

   $\mathrm{b1} = -1.77996826171875$

   $\mathrm{b2} = 0.862296581268310546875$

3. Scale the b1 and a1 by 0.5.

   $b0 = 0.992464542388916015625$

   $\mathrm{b1} = -0.889984130859375$

   $\mathrm{b2} = 0.862296581268310546875$

   $\mathrm{a1} = 0.92706697939106$

   $\mathrm{a2} = -0.932043159986584$

4. Since the coefficients are to be computed for ‘enhanced’ device, we need to scale it by 2^{23} and rounded to nearest integer. Hence the final coefficient hexadecimal values that has to be loaded onto the AIC3254 are

   $b0 = \mathrm{0x7F0914}$

   $\mathrm{b1} = \mathrm{0x8E1500}$

   $\mathrm{b2} = \mathrm{0x6E5FBC}$

   $\mathrm{a1} = \mathrm{0x76AA20}$

   $\mathrm{a2} = \mathrm{0x88B2D1}$