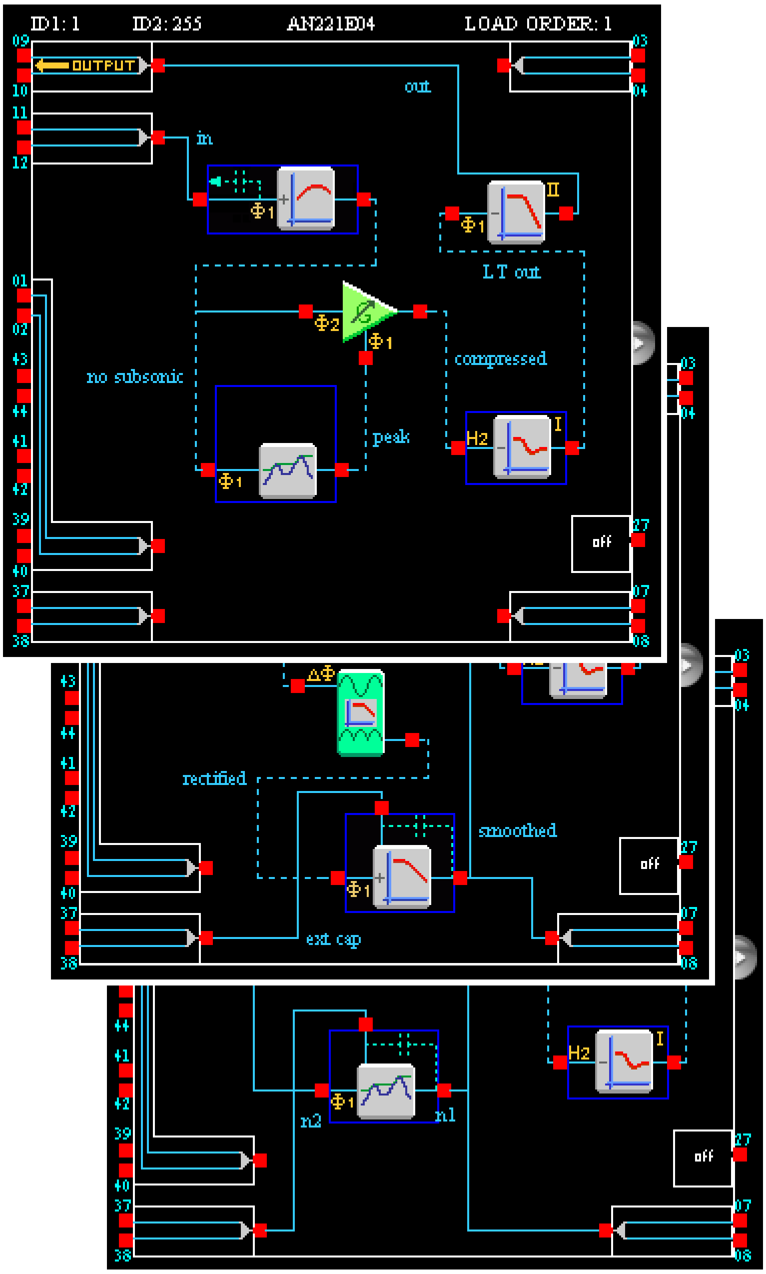
Reference Design: Subwoofer Signal Conditioner

Rev: 1.0.2

Date: 26th March 2004



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1 Purpose

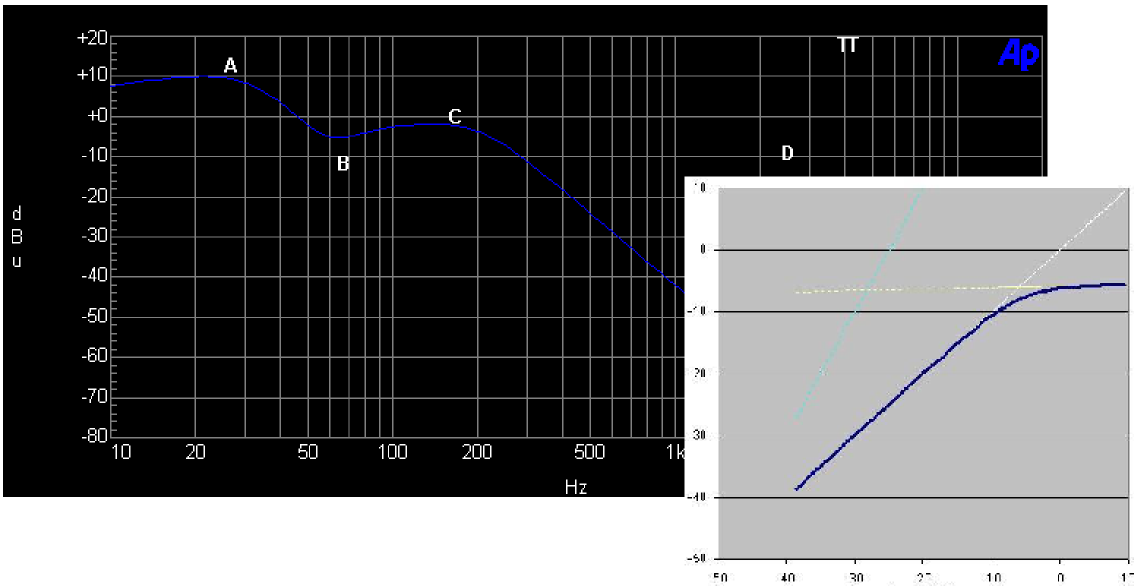
This document describes a reference design for implementing low frequency filtering and compression techniques using Anadigm®'s AN221E04 FPAA device.

The functions supported by this Subwoofer Signal Conditioning circuit are:

- **3rd order low-pass filter** with dynamically programmable cut-off frequency
- **Windows-based software interface** to perform adjustment of the above
- **Sub-sonic filter** with user-selectable corner frequency
- **Amplitude compression circuit** with user-specified compression characteristic, either peak-based with asymmetric attack/decay or rectifier-based.
- **Linkwitz Transform equalizer** circuit with user-programmable pole, zero and Q-factors

This entire circuit is supported by a single AN221E04 device.

This document also discusses design variations which offer different options to the user, and provides some bench characterizations of the design. The design itself is included as an AnadigmDesigner®2 design file as part of this Starter Kit.



The objective of the design is to provide a system which offers independent control of:

- **Linkwitz Transform** low frequency characteristic:
  - Frequency and Q settings of point '**A**'.
  - Frequency and Q settings of point '**B**'.
- **Corner frequency setting** of low pass filter '**C**'.
- Arbitrary **Compression characteristic** '**D**'.

2 Setup

2.1 Boards and interface

2.1.1 Inputs and outputs

The AN221K04 development board should be connected to the PC via the serial interface & cable provided. Depending on the type of signalling involved, the user may wish to make some custom modifications to the board. This section recommends various options.

It is recommended that all signalling into and out of the evaluation board be done differentially.

The reader is referred to the AN221K04 Evaluation Board User Manual, (Anadigm Document Number UM30900-U010) for full details.

The above user manual recommends various input and output interface circuits for ease of connection to audio sources and active speakers.

For formal performance tests all measurement instrumentation should connect directly to the FPAA header pins without any additional circuitry in the signal path.

It is assumed that a subsonic filter is required to remove very low frequency components and DC offsets. This being the case, interfacing to the FPAA is extremely simple.

If a differential input signal is available, the designer should apply the input signal through a pair of identical capacitors connected in series with the signal to input terminal I2 as shown in Figure 1. These capacitors form part of the input filter (see section 3.1).

With this arrangement, the circuit automatically adjusts for common-mode level differences between the signal source and the FPAA. No DC level shift is necessary.

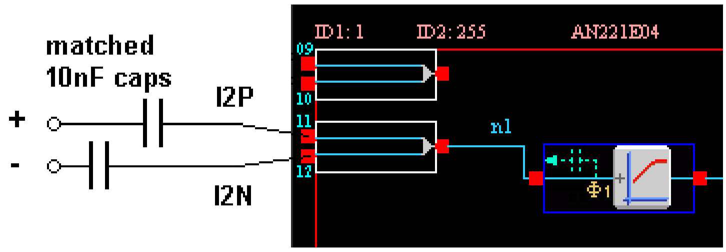


Figure 1

If the input signal is single-ended, then an alternative input arrangement is shown in Figure 2. If using this arrangement, the gain of the input filter stage should be doubled (see section 3.1).

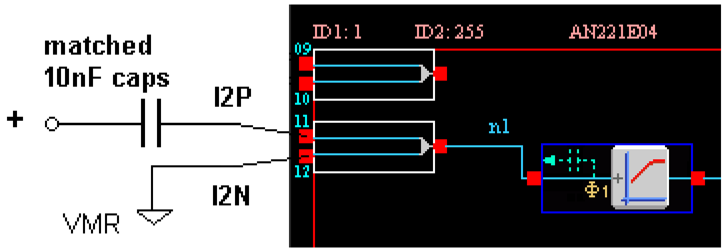


Figure 2

Figure 3 shows a differential-to-single-ended converter for a single-channel audio output, with 2V->0V common mode level shift.

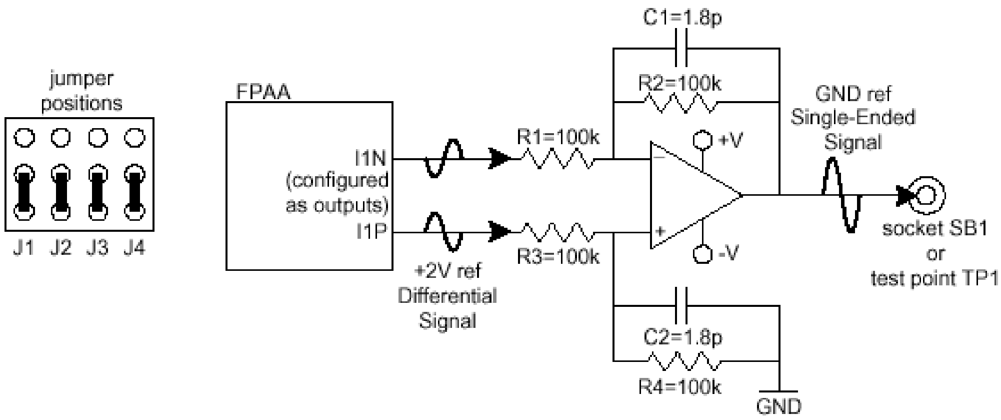


Figure 3

Alternatively, a single-ended output can be taken directly from the I1P pin, though the user is reminded of the 2V DC reference used for this signal, so a DC level shift or decoupling capacitor may be required.

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For more complete information on the AN221K04 evaluation board, the user is referred to the user manual:

AN221K04 Evaluation Board User Manual (click to download).
www.anadigm.com/support/literature library/Doc.No UM30900-U010

2.2 Optional Software installation

For convenience, the Starter Kit includes a simple Windows application called ***subCond.exe***.

This is not a mandatory component, but it does allow for a more rapid exploration of the circuit options discussed in this document, either for demonstration or evaluation purposes.

See Section 5 for more details.

Having installed the Starter Kit, no further installation is necessary. Simply double-click on this file or invoke it from the Windows Start menu.

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The executable **subCond.exe** *is installed automatically as part of the Starter Kit. Its default location is*

<Subwoofer Conditioner install dir>\subCond Application

or it can be invoked from the Start menu:

Start->Anadigm->Starter Kits->Subwoofer Conditioner->subCond

This software will only operate with an AN221K04 evaluation kit, connected to the PC's *serial port* using standard serial connector.

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Disclaimer:

Anadigm does not make any warranty or representation as to the functionality or otherwise of the "subCond" application and all warranties implied or express are excluded to the extent permitted by applicable law. Anadigm does not provide product support for this software.

3 Circuit description

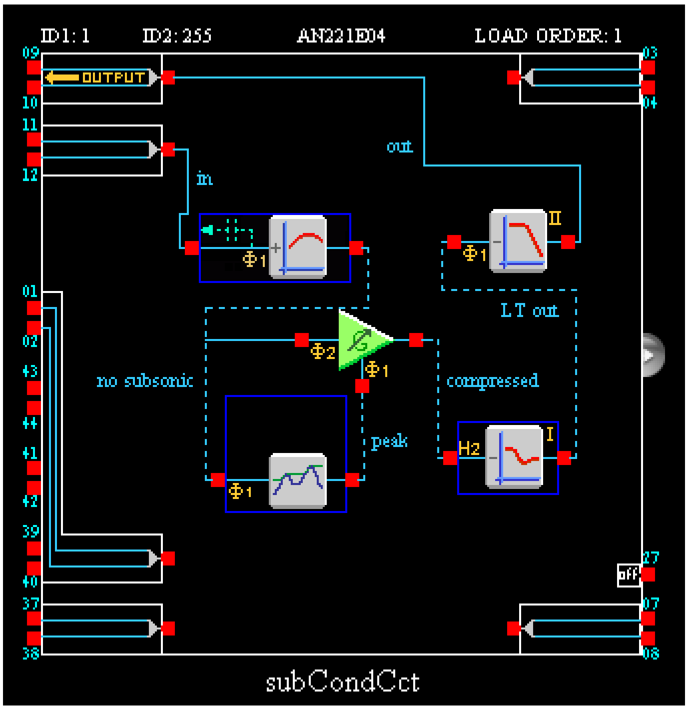

Figure 4 – Circuit of *Subwoofer Conditioner Peak.ad2*

The circuit of Figure 4 contains four main functional elements:

- **Sub-sonic filter** with user-selectable corner frequency
- **Amplitude compression circuit** with user-specified compression characteristic, either peak-based with asymmetric attack/decay or rectifier-based.
- **Linkwitz Transform equalizer** circuit with user-programmable pole, zero and Q-factors
- **Low-pass filter** with dynamically programmable cut-off frequency

The design for this Subwoofer Signal Conditioner uses a combination of Standard library CAMs and some prototype Customer CAMs.

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*If requested, any **Customer CAMs** required for this application will have been automatically added to the AnadigmDesigner®2 installation during the installation of this Starter Kit.*

For further information on Customer CAMs, their availability, reference kits, documentation and other technical support matters, contact your Sales Representative or visit the Anadigm Technical Support pages under

www.anadigm.com

3.1 Sub-Sonic Filter

The sub-sonic filter is an optional item, depending on the nature of the input signal presented to the Subwoofer Conditioner. This design assumes that no suppression of very low frequency and DC signals exists and that it must be supported by the design.

A Customer CAM has been developed to support this function. The CAM's name is ***xFilterLowFreqBilinearHP*** is assigned the instance name ***subSonic*** in this design. It is shown in Figure 5.

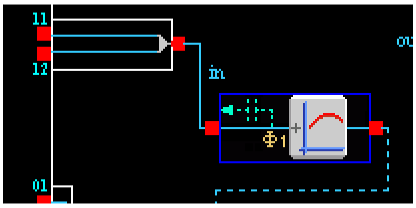


Figure 5

The CAM requires external capacitors to complete the sub-sonic filter function. See Section 2.1.1 for details on external components.

If the recommended **10nF** capacitors are used, this CAM will deliver a 1st order subsonic filter with –3dB corner frequency of **11.6Hz.**

This CAM also performs two other functions:

- **Gain Prescaling:** if the differential input signal level is other than 0dBu nominal (with **maximum** peak level of 8.75dBu – see Section 3.2) then this CAM should be used to pre-scale the gain accordingly. This is necessary for the optimal operation of the compressor.

  Default gain values are **1.0** for differential input and **2.0** for single-ended input.

- **Low-Pass Filter**: the CAM has an option to add a low-pass roll-off at higher frequencies than the HP filter corner. This feature is used in this design to add an order of magnitude to the sub-woofer lowpass filter.

  This filter is used in conjunction with the 2nd order output low-pass filter to create an overall 3rd order (18dB/8ve) low-pass function. See also Section 3.4.

CAM Parameters

Parameter:	Value:	Limits:	Realized
External Cap Value [nF]	10	0 To 100000	10.0
Gain	1	0.5 to 10	1.00
LP Corner Freq [kHz]	0.2	0.0550 To 5.00	0.200
=> HP Corner Freq [kHz]	0.0116	0.0116 To 0.0116	0.0116

Figure 6 – *subSonic* CAM settings

3.2 Audio Compressor

The audio peak-level compressor is shown in Figure 7. It comprises two key elements: a ***GainVoltageControlled*** CAM (instance name ***VGC***) and a ***PeakDetect*** CAM.

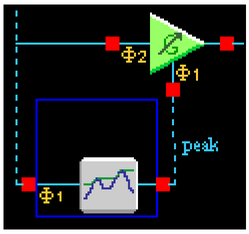
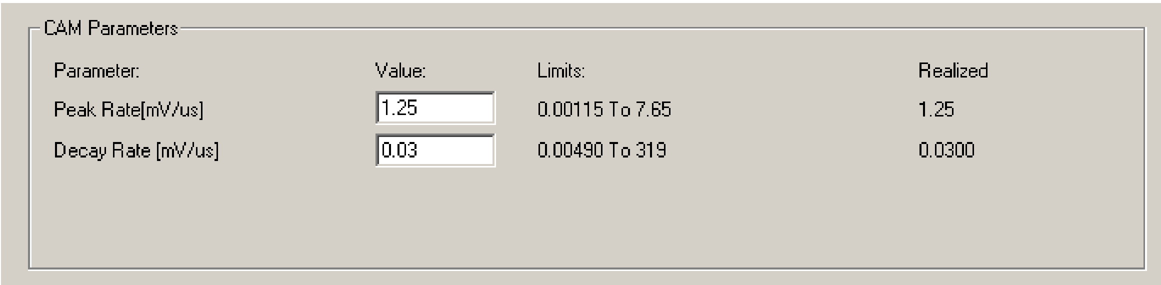


Figure 7

PeakDetect is a Customer CAM which delivers a voltage equal to (in this case) the positive peak level of its input. It has independently controllable attack and decay rates which combine a fast response with a longer 'hold' time. These are set to 1.25mV/µs and 0.03mV/µs respectively.

For alternative amplitude detection techniques, see Sections 4.1.1 and 4.1.2.



CAM Parameters

Parameter:	Value:	Limits:	Realized
Peak Rate[mV/us]	1.25	0.00115 To 7.65	1.25
Decay Rate [mV/us]	0.03	0.00490 To 319	0.0300

Figure 8 – *PeakDetect* settings

The ***GainVoltageControlled*** CAM is a gain stage whose forward gain is dependent on the input control voltage - in this case the output of the peak detector, giving amplitude-dependent gain.

The actual characteristic of this dependency is arbitrary, and something that the user can set. This is done by using a look-up-table that forms part of the ***GainVoltageControlled*** CAM. The gain for each of 256 individual control voltage values can be set independently. This is best done by loading in a set of values from a comma-separated-values (.csv) file.

This Starter Kit includes an excel spreadsheet called

Audio Peak Level Compressor.xls

contained in the directory

<Subwoofer Conditioner install dir>\Compressor Tools & Data

*Which can be used to generate **.csv** data files suitable for loading into the* **GainVoltageControlled** *CAM. The directory also includes three ready-made .**csv files**t for experimentation purposes.*

As referred to in the note above, the kit contains three ready-made .csv files. These can be directly loaded into the CAM called ***VCG***: In its ***Edit Parameters*** dialog box, select "***Lookup Table***", then "***Load…***".

The three files have the following characteristics:

unity_gain.csv:	No compression: gain of 1.0 for all signal amplitudes.
compress_m6_c40.csv:	Nearly flat-characteristic compression for all signals above **–6dBu**. (See below)
compress_m15_c40.csv:	Nearly flat-characteristic compression for all signals above –**15dBu**. (See below)

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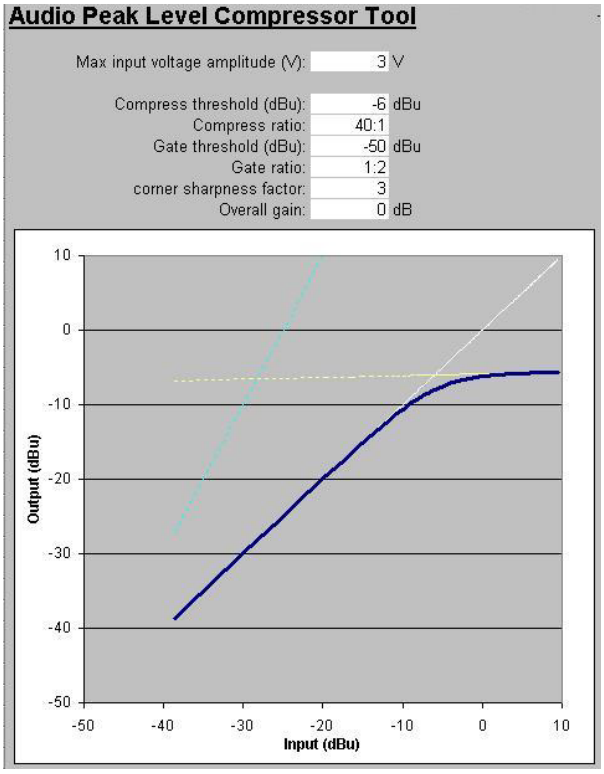
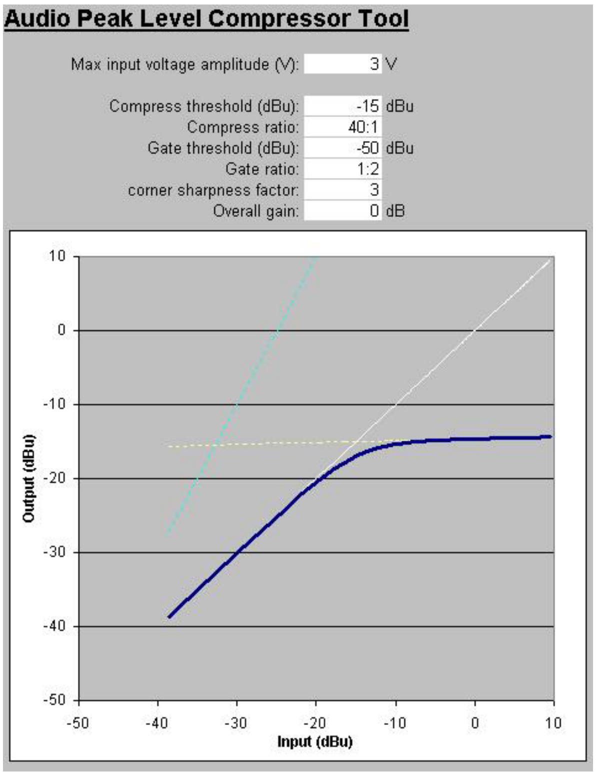
*The **GainVoltageControlled** CAM has an operational voltage range limit of +/-3V (diff) on its **Control** input. This means that for practical purposes, the peak amplitude limit on its **Input** signal (which is equal to the **Control**) must also be +/-3V, which = **8.75dBu**.*

For signals larger than this, the compressor will no longer operate, it will simply attenuate at the level demanded for an 8.75dBu signal.

If compression of larger input signals is required, gain pre-scaling should be done using the Sub-Sonic Filter (Section 3.1) and the compression characteristic should be adjusted accordingly (see remainder of this section).



compress_m6_c40.csv



compress_m15_c40.csv

Measurement results of these characteristics can be seen in Sections 6.5 and 6.6, and an FFT showing harmonic content in Section 6.7.

The FFT shows spur-free dynamic range relative to the compressed fundamental of –40dB. This is a classic result of the peak detect mechanism, where the attack/decay tracking can give a marked 'sawtooth' characteristic. This causes harmonic modulation of the signal (see note below).

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Note on PeakDetect Harmonic Effects

The linearity (40dB SFDR) of a Subwoofer Conditioner based on a standard peak detector such as this should be sufficient for many applications. If greater linearity is required, alternatives are available or under development.

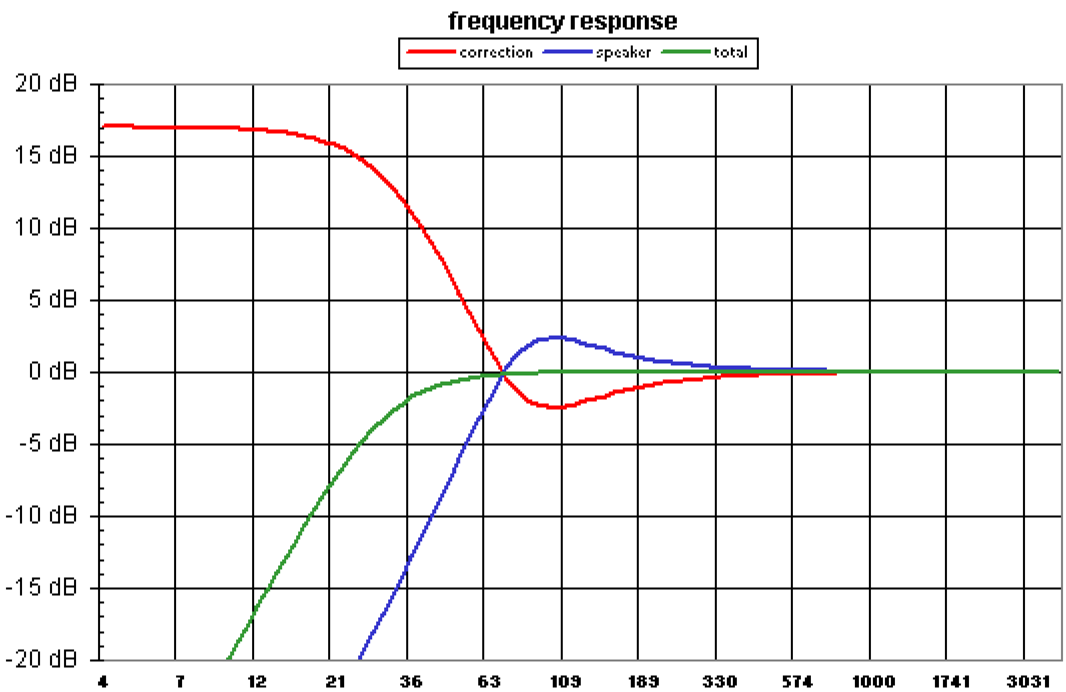
*A **rectifier-based compressor** is presented in Section 4.1.2 and an **extended low frequency peak detect** is presented in Section 4.1.1.*

☞

Using the spreadsheet **Audio Peak Level Compressor.xls** *other compression characteristics can be created with different thresholds, flat compression, or even negative compression ratios (output increasingly attenuates with increasing input levels) etc.*

3.3 Linkwitz Transform Equalizer

This circuit is a second-order system of poles and zeros that delivers a combined ability to suppress resonance from speaker cabinets at particular frequencies, and to boost the gain of very low frequency signals with independent control over both. This allows for equalization of the low frequency response of a completed speaker system (see Figure 9).



source: True Audio

Figure 9

In our Subwoofer Signal Conditioner, the Linkwitz Transform function is provided by the Customer CAM ***FilterSubwooferCompensation*** (instance name ***LwXform***) shown in Figure 10.

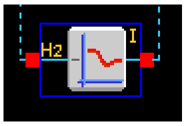


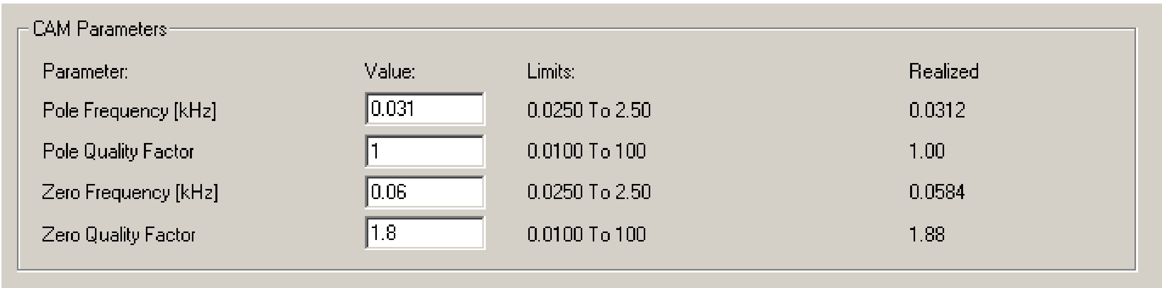
Figure 10

This CAM allows for independent control of

- **pole** frequency and Q
- **zero** frequency and Q

within the constraints of the internal component settings (Figure 11).

Details of the operation of the CAM are provided within its online CAM Documentation. Using the parameters above, the designer can set the frequency & depth of the notch, the frequency and height of the pole, and the resulting DC gain.



CAM Parameters

Parameter:	Value:	Limits:	Realized
Pole Frequency [kHz]	0.031	0.0250 To 2.50	0.0312
Pole Quality Factor	1	0.0100 To 100	1.00
Zero Frequency [kHz]	0.06	0.0250 To 2.50	0.0584
Zero Quality Factor	1.8	0.0100 To 100	1.88

Figure 11

The are many 3rd party references to the Linkwitz Transform on-line, examples included here for convenience:

ESP – The Linkwitz Transform Circuit

True Audio – Designing a Linkwitz Transform

3.4 Low Pass Filter

This CAM is a standard ***FilterBiquad*** whose parameters are:

- ***Corner Frequency***: **200Hz** nominal. This must be set to the same value as the *LP Corner Frequency* of the sub-sonic filter ***subSonic***. (Section 3.1).
- ***Gain*** **1.0**
- ***Q Factor:*** **1.0**

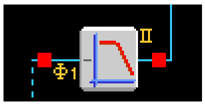
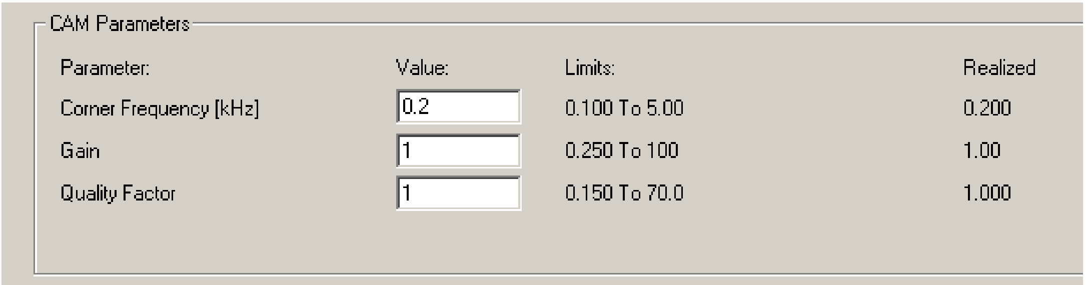


Figure 12

This corner frequency is the value that is ‘swept’ in the accompanying Windows-based software tool ***subCond.exe*** (see Section 5).

When building the facility for any such dynamic programming, it is important that the *Corner Frequency* of ***subFilter*** and the *LP Corner Frequency* of ***subSonic*** track each other to maintain the overall 3rd order Butterworth characteristic of the low-pass filter.



CAM Parameters

Parameter:	Value:	Limits:	Realized
Corner Frequency [kHz]	0.2	0.100 To 5.00	0.200
Gain	1	0.250 To 100	1.00
Quality Factor	1	0.150 To 70.0	1.000

Figure 13

4 Circuit Variations

4.1 *Compressor Options*

4.1.1 Long Time-Constant Peak Amplitude Detection

Another variant of the compressor deploys a extended Peak Detector CAM. This CAM uses external capacitors to achieve a significantly extended time-constant for the peak decay (and attack) rates.

This design is included in the Starter Kit as ***Subwoofer Conditioner LTC Peak.ad2*** (Figure 14). See note below.

☞

Note: *The circuit file* ***Subwoofer Conditioner LTC Peak.ad2*** *is only compatible with AnadigmDesigner2 version 2.4.1.2 and above. Anadigm respectfully requests that the reader awaits the imminent release of this version of the tool and the* ***PeakDetectExt*** *CAM to view and access this design.*

The reader is invited to monitor Anadigm's website Customer Support pages for updates.

However, this circuit configuration has been built into the experimentation software tool ***subCond.exe*** *should the reader wish to evaluate it. (See Section 5).*

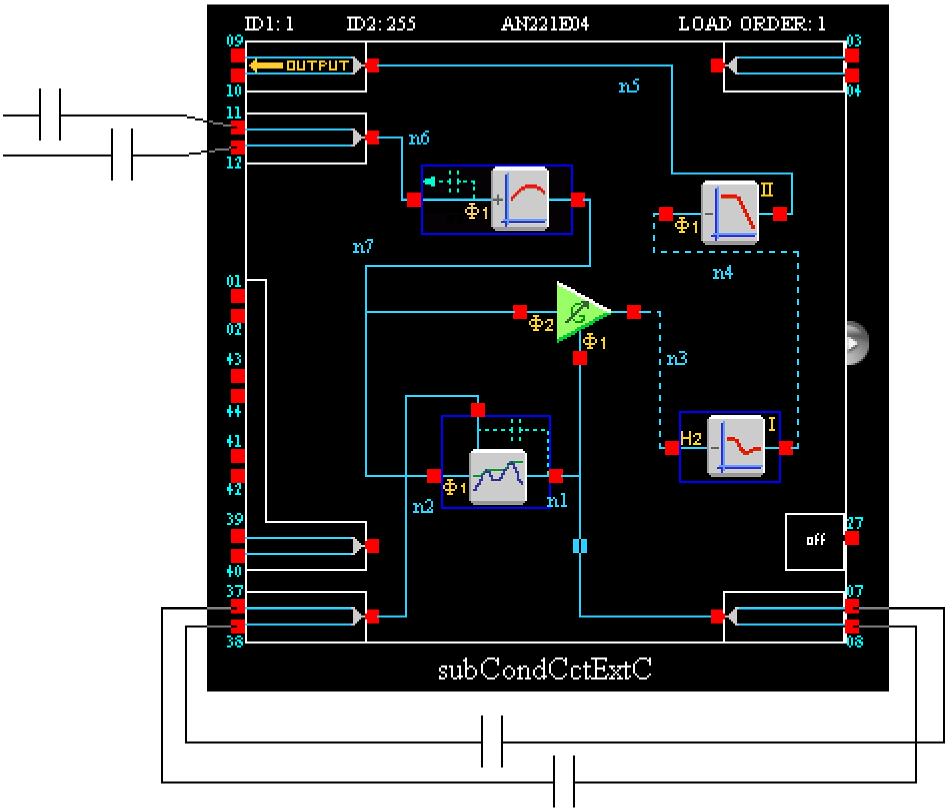


Figure 14 - Circuit of *Subwoofer Conditioner LTC Peak.ad2*

The long time-constant peak detector is a Customer CAM called ***PeakDetectExt*** (see note above). This CAM uses an external capacitor pair to introduce extended time-constants on attack and decay rates in the peak detector.

The design requires that two **100nF** capacitors be added: one connected between **O2P and I3N**, the other connected between **O2N and I3P**. In this design the resulting linear attack and decay rates are **20V/s** and **0.4V/s** respectively. This gives a much 'smoother' representation of the signal strength giving a much improved THD characteristic (see Section 6.8).

4.1.2 Rectifier-Based Compressor

The Starter Kit also includes an alternative design for the Subwoofer Signal Conditioner called ***Subwoofer Conditioner Rect.ad2*** (Figure 15). In this circuit, the peak detector in the compressor circuit is replaced by a rectifier + very long time-constant filter.

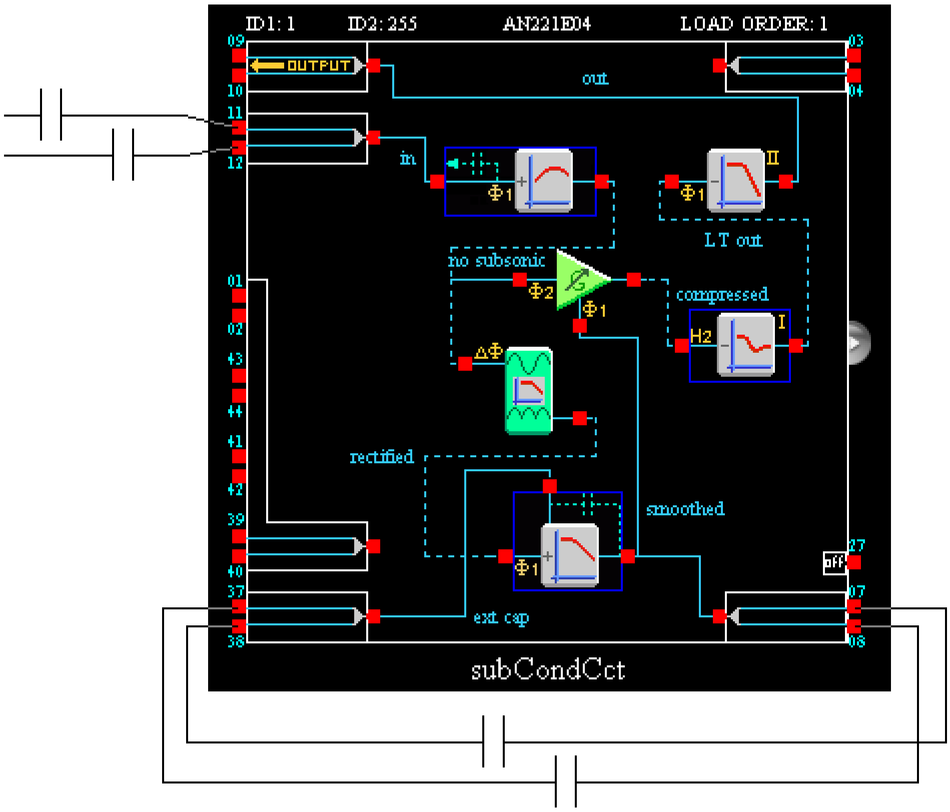


Figure 15 – Circuit of *Subwoofer Conditioner Rect.ad2*

The rectifier is a standard ***RectifierFilter*** CAM, and the long time-constant filter is a Customer CAM called ***FilterLowFreqBilinear*** . This CAM uses an external capacitor pair to introduce a long time-constant low-pass filter.

The design requires that two **1nF** capacitors be added: one connected between **O2P and I3N**, the other connected between **O2N and I3P** In this design, the resulting –3dB corner frequency is **4.9Hz**. This gives a much 'smoother' representation of the signal strength giving a much improved THD characteristic (see Section 6.9). For full details on the use of the ***FilterLowFreqBilinear***, the reader is referred to the CAM online Documentation.

By replacing the ***PeakDetect*** CAM, the signal being monitored is now the *average absolute voltage* rather than peak level, and the attack/decay rates are symmetrical.

4.1.3 RMS-Based Amplitude Detection

A further option for signal strength monitoring for the compressor is to use an RMS detector rather than voltage averaging (rectify + filter). This option requires additional resources and would require more than one FPAA device to implement.

This option is available now, and the reader is referred to Anadigm's website Customer Support pages for details.

5 The subCond Evaluation Assistant Tool

This tool is not an essential part of the Starter Kit, but it is included to make exploration and evaluation of the circuits described in this document more convenient.

The tool contains configurations and parameter controls for the two Peak Detector based compressor designs described in Sections 3 and 4.1.1.

Install and run the subCond.exe program as described in Section 2.2.

When subCond.exe is invoked, the dialog shown in Figure 16 appears.

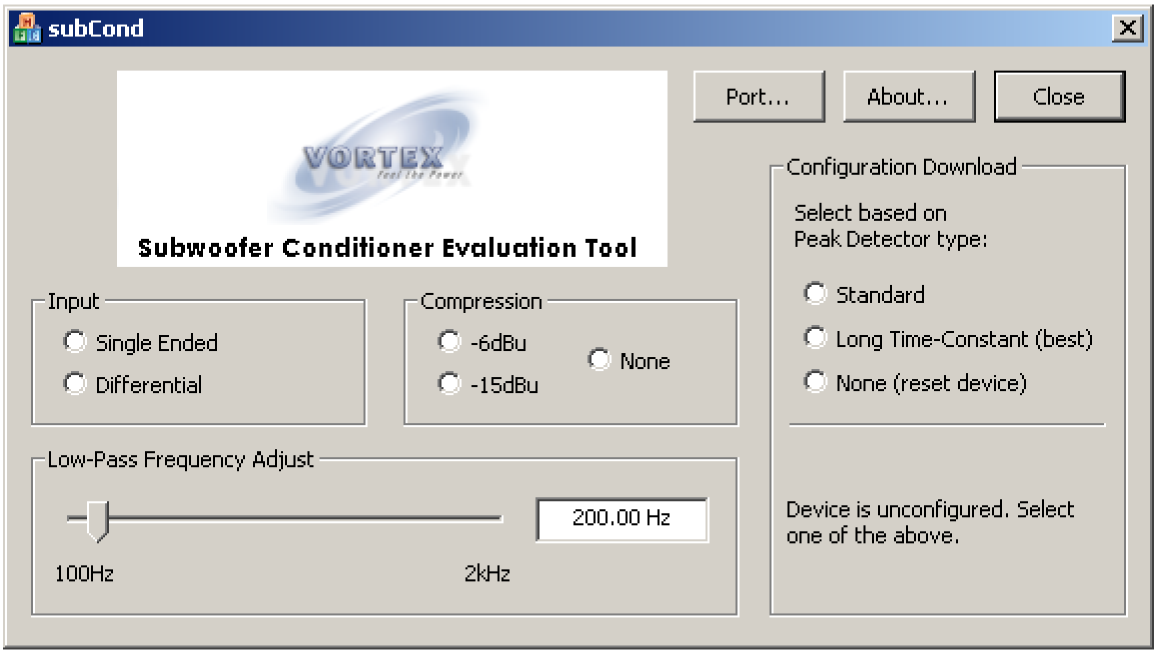


Figure 16

Press "***Port…***" to select the serial port to which the evaluation board is connected.

First, a configuration should be loaded. This is done by selecting one of the ***Configuration Download*** options. The tool contains two circuit configurations:

Standard: This is the circuit described in Section 3 and contained in the file ***Subwoofer Conditioner Peak.ad2.***

(Ensure that the input and output connections are made as described in Section 2.1.1).

Long Time-Constant: This circuit is described in Section 4.1.1 (see Note in that section) and contained in the file ***Subwoofer Conditioner LTC Peak.ad2.***

(Ensure that the input and output connections are made as described in Section 2.1.1, and that the external capacitors needed by the peak detector are added as described in Section 4.1.1).

There are three main controls:

Input: Use this to select whether input is single-ended or differential. For the former, it pre-scales the gain by 2.0

Compression: This control selects the compression threshold. The options are those described in Section 3.2.

Low-Pass Frequency Adjust: Slider control or 'Edit' box to modify the low-pass cut-off frequencies of the sub-sonic filter (section 3.1) and the low-pass filter (section 3.4) which combine to give an overall 3rd order lowpass characteristic.

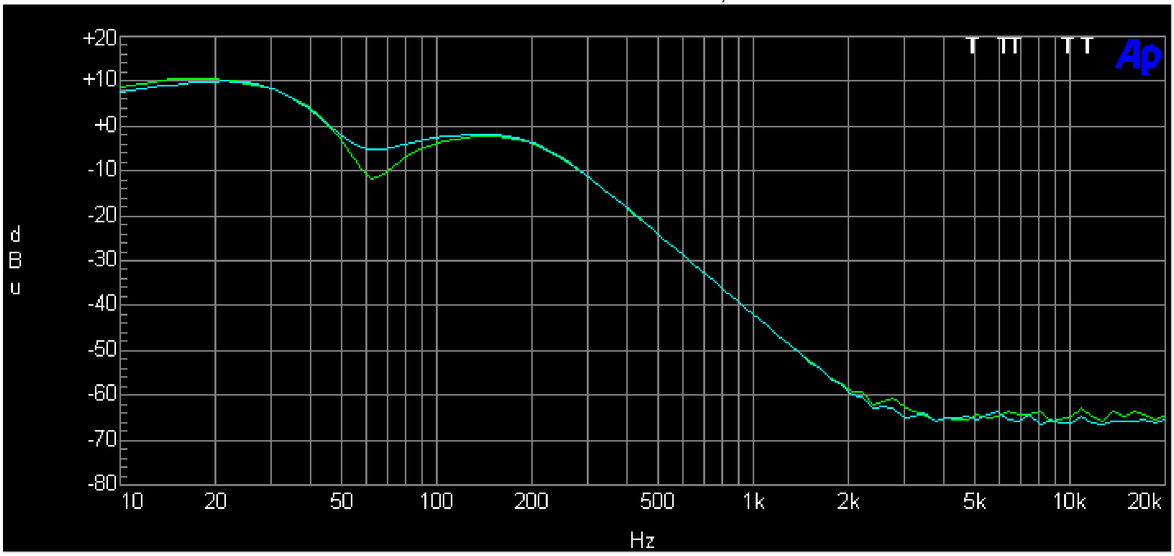
When using the ***Standard*** option, it is possible that the compression function will sound "rough" under some circumstances. This is due to the relatively fast attack & decay times allocated to the peak detector in the compressor causing "gain pumping" and harmonic distortion. Select the ***Long Time-Constant*** option as an alternative. This and other design options are discussed in Section 4.1.

6 Performance

☞ *Stop-band attenuation limits in the following figures do not reflect the performance of the FPAA. These limits arise from impedance mismatch effects with the instrumentation.*

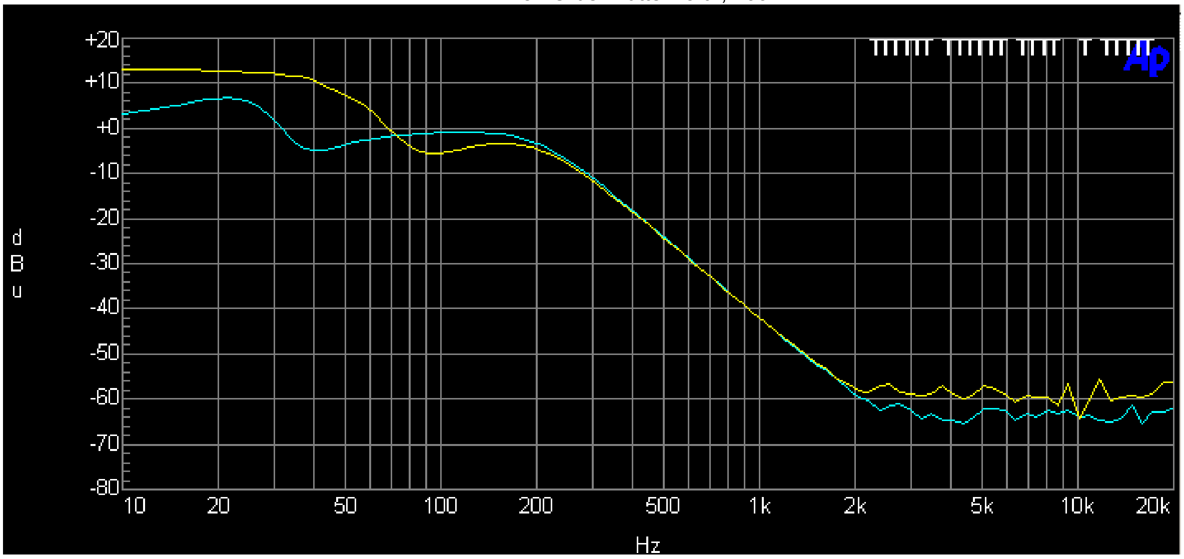
6.1 Varying Q of the Linkwitz Transform

Circuit:	*Subwoofer Conditioner Peak.ad2*
Input:	0dBu
Compression:	None
Sub-Sonic Filter	Gain = 1
	External Cap = 10nF (pole=11.6Hz)
Linkwitz Transform	Pole=31Hz, Q=1.0
	Zero=58Hz, **Q=1.8 and 4.0**
Low-Pass Filter	3rd Order Butterworth, 200Hz



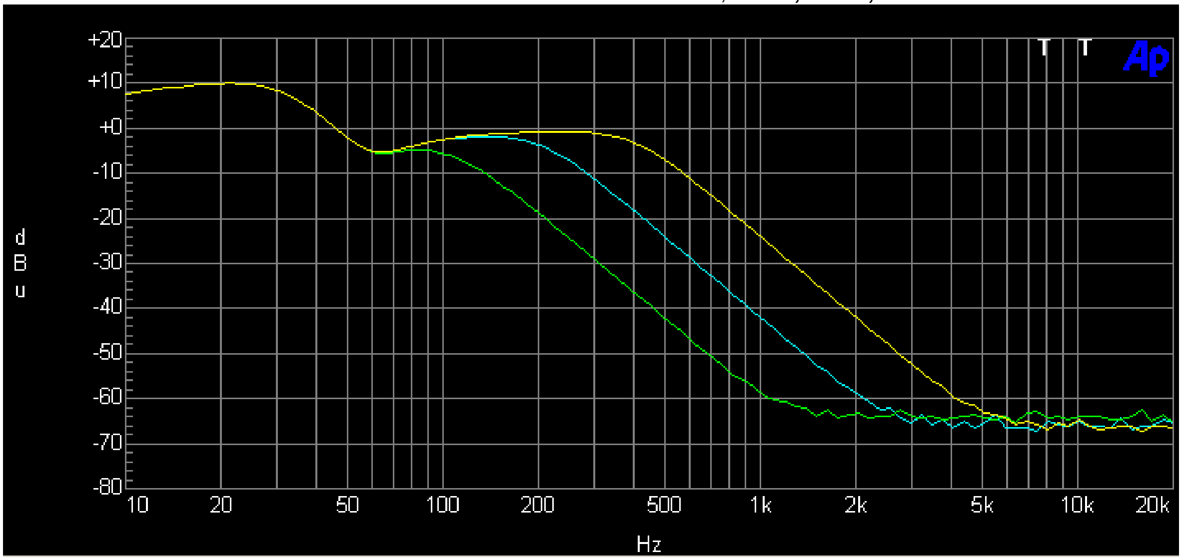
6.2 Varying the Zero Frequency of the Linkwitz Transform

Circuit:	*Subwoofer Conditioner Peak.ad2*
Input:	0dBu
Compression:	None
Sub-Sonic Filter	Gain = 1
	External Cap = 10nF (pole=11.6Hz)
Linkwitz Transform	Pole=26Hz, Q=1.6
	Zero=37Hz, Q=2.4; Zero=87Hz, Q=1.8
Low-Pass Filter	3rd Order Butterworth, 200Hz



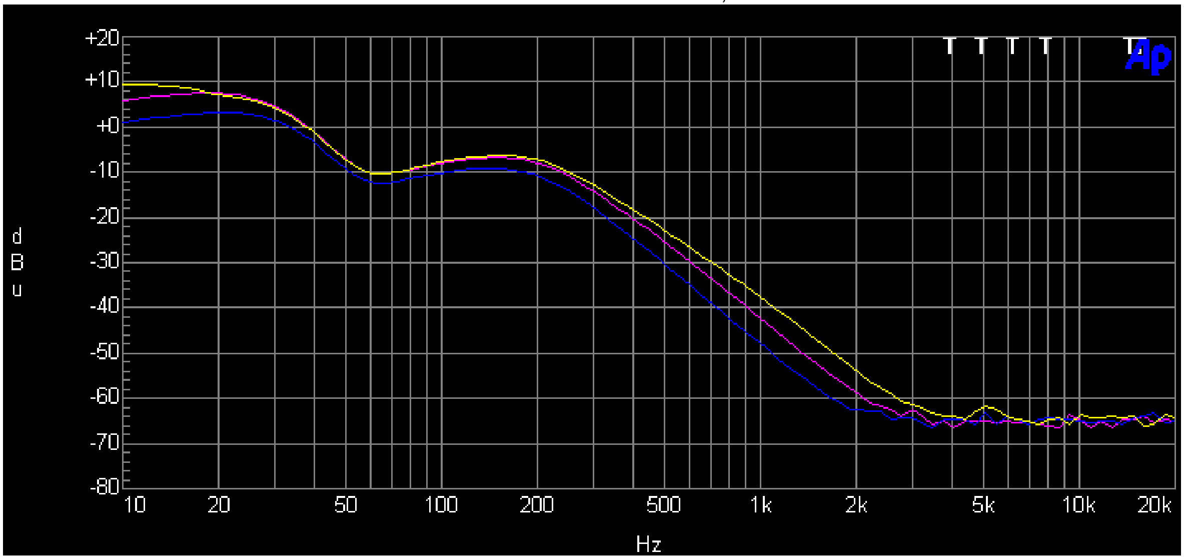
6.3 Varying Pole Frequency of the Subwoofer Low Pass Filter

Circuit:	*Subwoofer Conditioner Peak.ad2*
Input:	0dBu
Compression:	None
Sub-Sonic Filter	Gain = 1
	External Cap = 10nF (pole=11.6Hz)
Linkwitz Transform	Pole=31Hz, Q=1.0
	Zero=58Hz, Q=1.8
Low-Pass Filter	3rd Order Butterworth, **100Hz, 200Hz, 400Hz**



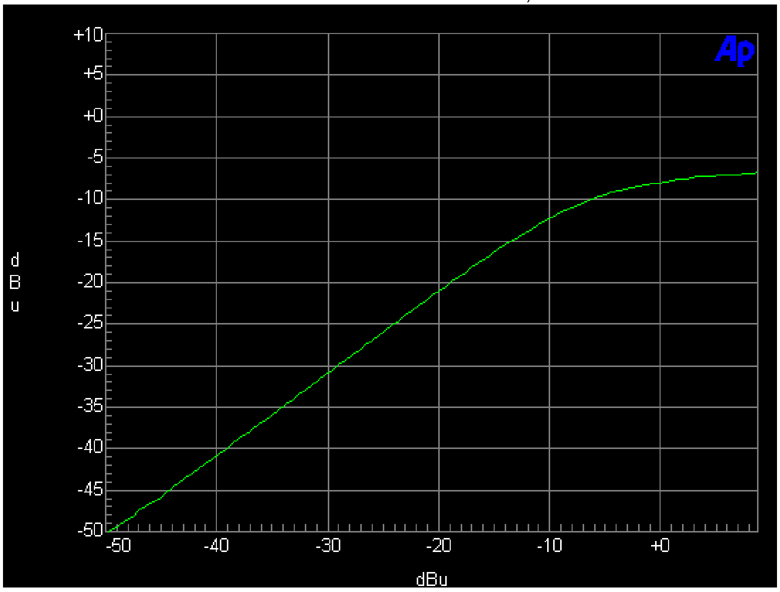
6.4 Effect of the Compressor

Circuit:	*Subwoofer Conditioner Peak.ad2*
Input:	**-6dBu** (blue), **0dBu** (magenta), **6dBu** (yellow)
Compression:	**-6dBu threshold, rounded characteristic** (see Section 3.2)
Sub-Sonic Filter	Gain = 1
	External Cap = 10nF (pole=11.6Hz)
Linkwitz Transform	Pole=31Hz, Q=1.0
	Zero=58Hz, Q=1.8
Low-Pass Filter	3rd Order Butterworth, 200Hz



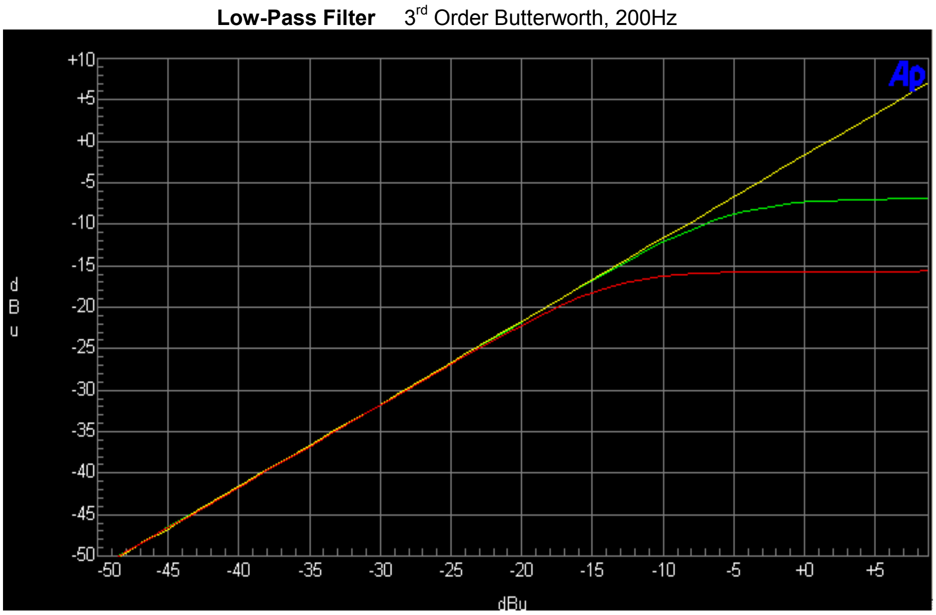
6.5 Compression Characteristic: -6dBu threshold

Circuit:	*Subwoofer Conditioner Peak.ad2*
Input:	**-50dBu to 8.7dBu, 200Hz**
Compression:	**-6dBu threshold, rounded characteristic** (see Section 3.2)
Sub-Sonic Filter	Gain = 1
	External Cap = 10nF (pole=11.6Hz)
Linkwitz Transform	Pole=26Hz, Q=1.6
	Zero=58Hz, Q=1.8
Low-Pass Filter	3rd Order Butterworth, 200Hz



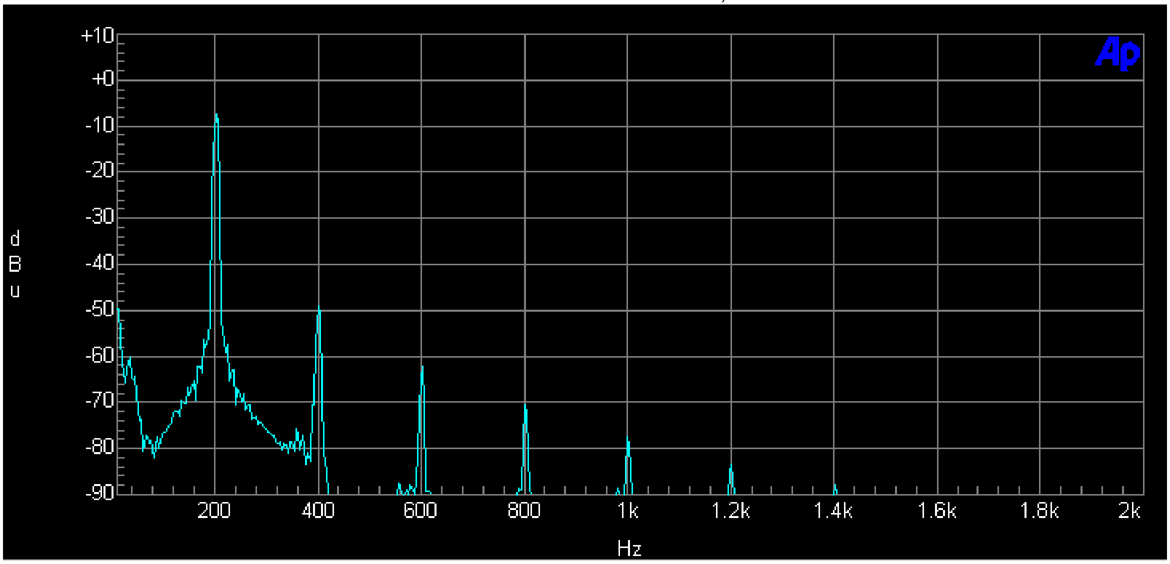
6.6 Compression Characteristic: -15dBu threshold

Circuit:	*Subwoofer Conditioner Peak.ad2*
Input:	**-50dBu to 8.7dBu, 200Hz**
Compression:	**None** (yellow)**, -6dBu threshold** (green)**, -15dBu threshold** (red)
Sub-Sonic Filter	Gain = 1
	External Cap = 10nF (pole=11.6Hz)
Linkwitz Transform	Pole=26Hz, Q=1.6
	Zero=58Hz, Q=1.8
Low-Pass Filter	3rd Order Butterworth, 200Hz



6.7 FFT, Subwoofer Conditioner output (Peak Detector Based)

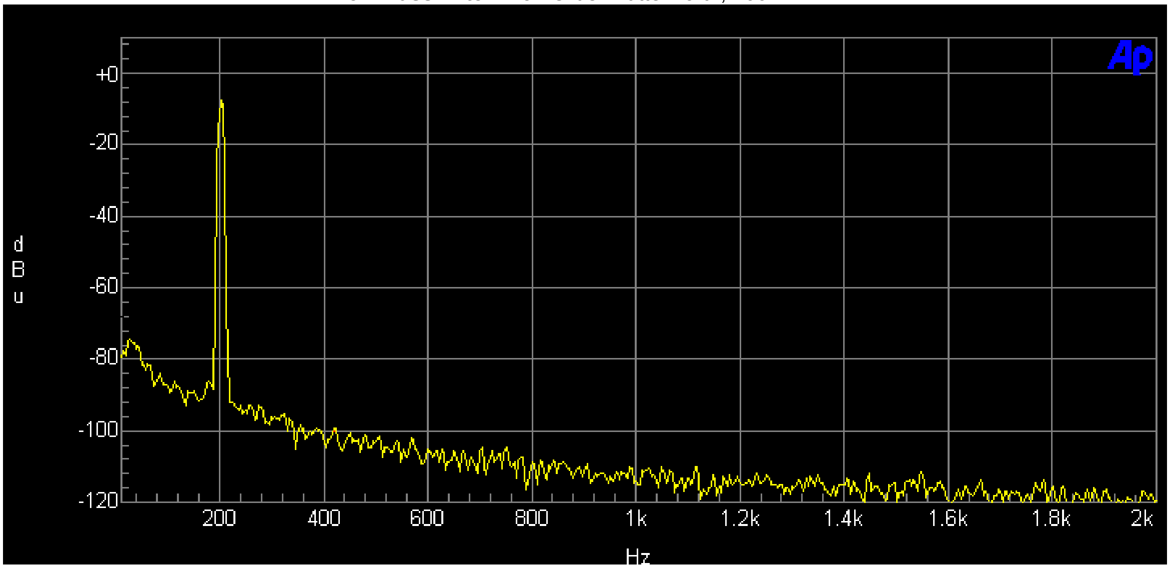
Circuit:	***Subwoofer Conditioner Peak.ad2***
Input:	0dBu, 200Hz
Compression:	**-6dBu threshold, rounded characteristic** (see Section 3.2)
Sub-Sonic Filter	Gain = 1
	External Cap = 10nF (pole=11.6Hz)
Linkwitz Transform	Pole=26Hz, Q=1.6
	Zero=58Hz, Q=1.8
Low-Pass Filter	3rd Order Butterworth, 200Hz



[*See *Note on Harmonic Effects* in Section 3.2]

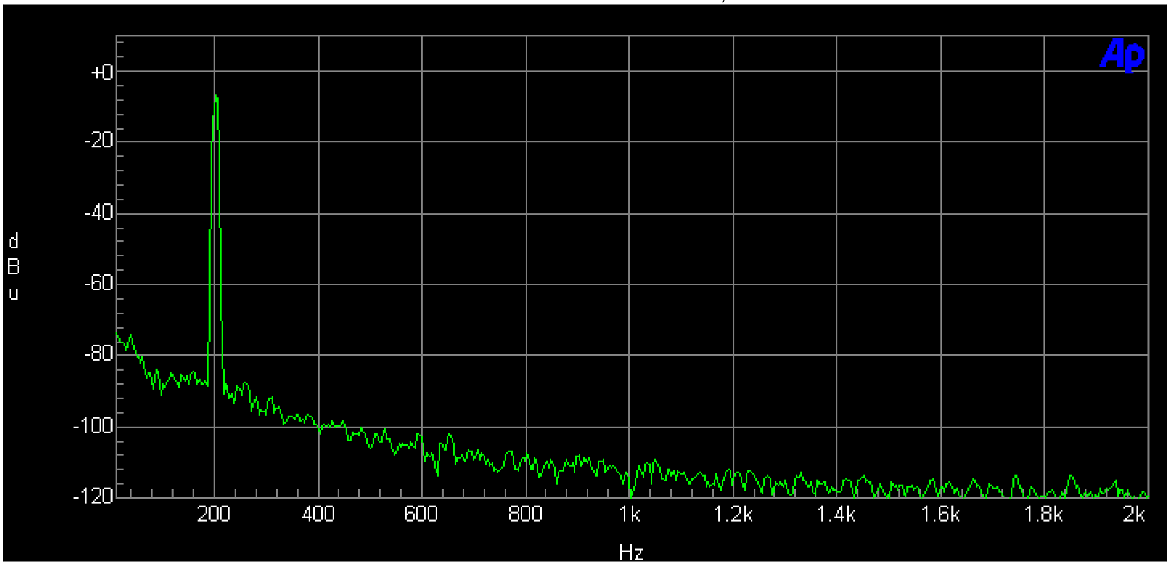
6.8 FFT, Subwoofer Conditioner output (LTC Peak Detector Based)

Circuit:	***Subwoofer Conditioner LTC Peak.ad2***
Input:	0dBu, 200Hz
Compression:	**-6dBu threshold, rounded characteristic** (see Section 3.2)
	Uses **PeakDetectExt** CAM with external **100nF** caps.
Sub-Sonic Filter	Gain = 1
	External Cap = 10nF (pole=11.6Hz)
Linkwitz Transform	Pole=26Hz, Q=1.6
	Zero=58Hz, Q=1.8
Low-Pass Filter	3rd Order Butterworth, 200Hz



6.9 FFT, Subwoofer Conditioner output (Rectifier Based)

Circuit:	***Subwoofer Conditioner Rect.ad2***
Input:	0dBu, 200Hz
Compression:	**-6dBu threshold, rounded characteristic** (see Section 3.2)
Sub-Sonic Filter	Gain = 1
	External Cap = 10nF (pole=11.6Hz)
Linkwitz Transform	Pole=26Hz, Q=1.6
	Zero=58Hz, Q=1.8
Low-Pass Filter	3rd Order Butterworth, 200Hz



APPENDIX A – Alternative DC Input Interfacing Options

Figure 17 gives an example of a single-ended to differential converter for a single-channel audio input.

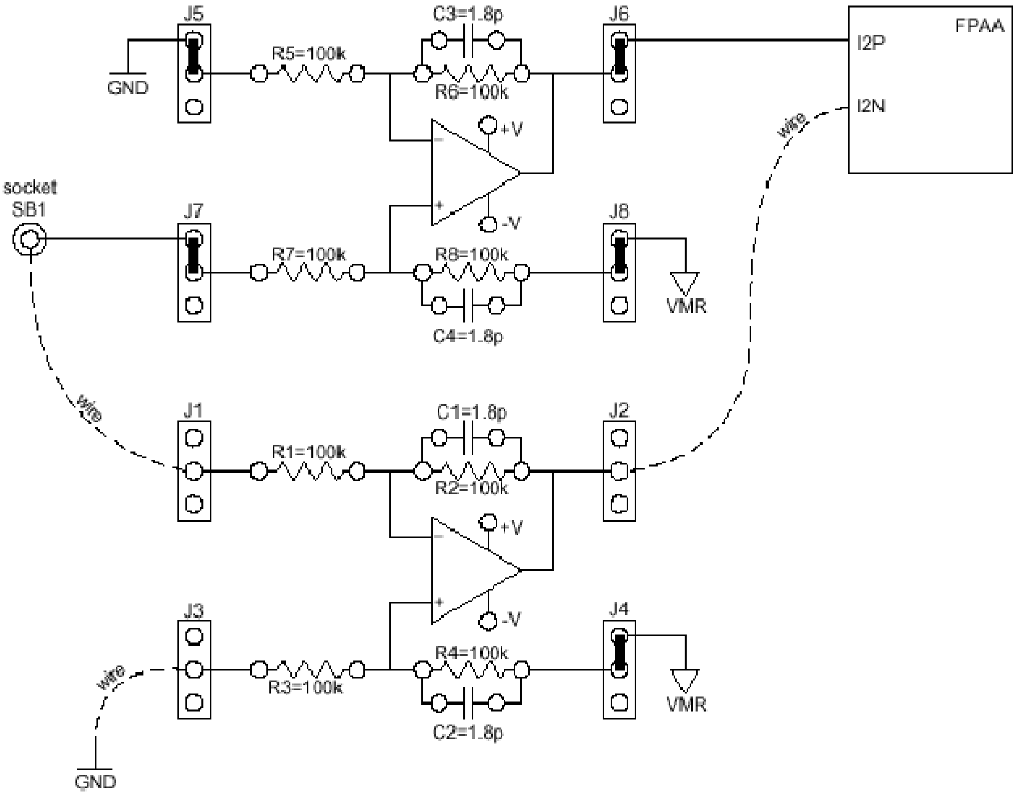


Figure 17

Depending on the nature of the audio signal source output drive (e.g. if it is AC-decoupled), it is possible that the common-mode of 2V will not be correctly established at the input using the circuit of Figure 17. An alternative is given in Figure 18.

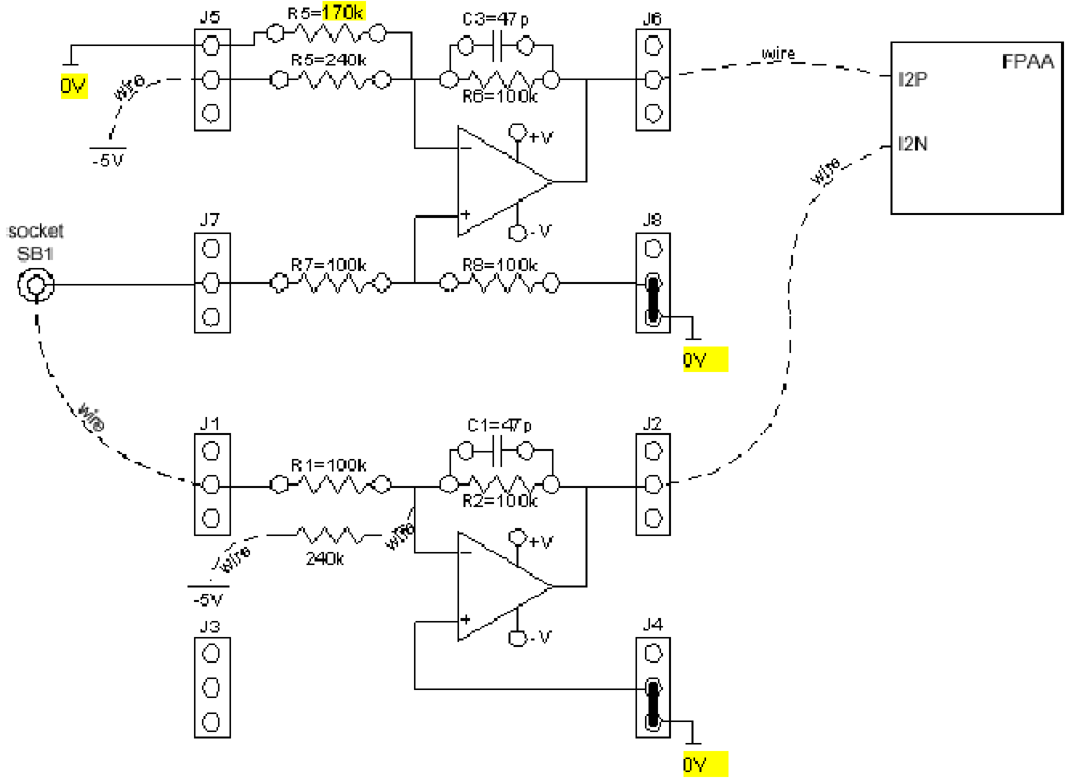

J5
R5=170k
R5=240k
C3=47p
R6=100k
J6
wire
FPAA
I2P
I2N
0V
wire
-5V
+V
-V
socket
SB1
J7
R7=100k
R8=100k
J8
0V
wire
J1
R1=100k
C1=47p
R2=100k
J2
240k
wire
-5V
J3
J4
0V

Figure 18