ACM210X1

www.acmt.co.uk

Manual

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Section 1 - Introduction

1.1 - What is the ACM210X1?

The ACM210X1 plug-in for Windows or Linux PCs and compatible audio workstation applications comprises a 10 band graphical parametric equalizer, with adjustable Bandwidth, Filter-type, Frequency and Gain on each band. It features a graphical interface and custom designed filter algorithms designed to replicate a more natural analogue EQ response, without requiring CPU intensive upsampling.

1.2 - Main Features

- Stereo VST, VST3 and CLAP plug-in for 64Bit Windows or Linux PCs and compatible host applications.
- Accurate zoomable graphical display showing individual filter response and combined EQ curve.
- Comprehensive range of filter types.
- Natural analogue filter response without requiring upsampling. or high sample rates.
- FFT Analyser for graphical display of harmonic content.

1.3 System Requirements

Windows:
A PC running 64Bit Windows 7 or newer and a VST, VST3 or CLAP compatible host application. Requires support for OpenGL 3.x or later.

Linux:
An X11 compatible Linux distribution and a Linux VST, VST3 or CLAP compatible host application. Requires support for OpenGL (GLX) 3.x or later.

1.4 - About the Manual

This manual covers the installation and use of the ACM210X1 equalizer. Features and operation may vary depending upon your operating system configuration and host application. Where appropriate, examples are also illustrated with screenshots of the features being discussed.

1.5 - Conventions Used

Access to menu items are shown as follows:

`Menu -> Item -> Item`

A `Mono-spaced font` is used to illustrate commands as they are typed on the command line.

Section 2 - Installation

2.1 Download Contents

Within the folder that contained this manual you will find Windows and Linux folders containing the plug-in built for **64Bit Windows or Linux systems**. Please refer to section 1.3 for system requirements.

2.2a Installing the Plug-In for Windows

Installing the Plug-In for Windows:

Within the Windows folder you will find installers for the VST, VST3 and CLAP plug-ins. The installers will guide you through the steps required to install the plug-ins.

NOTE: VST3 and CLAP define specific locations for compatible plug-ins. For Windows this is normally:

`Program Files\Common Files\VST3\[CompanyName]`

and

`Program Files\Common Files\CLAP\[CompanyName]`

The installer will permit other locations however you should use only the installer recommended location for the VST3 or CLAP plug-ins. unless you are confident of a specific reason for selecting an alternative.

The installer will only install the files necessary for the plug-in to function. It will not install anything else on your computer.

Uninstalling the plug-in:

To uninstall the plug-in It is recommended to use

`Control Panel -> Add or Remove Programs`

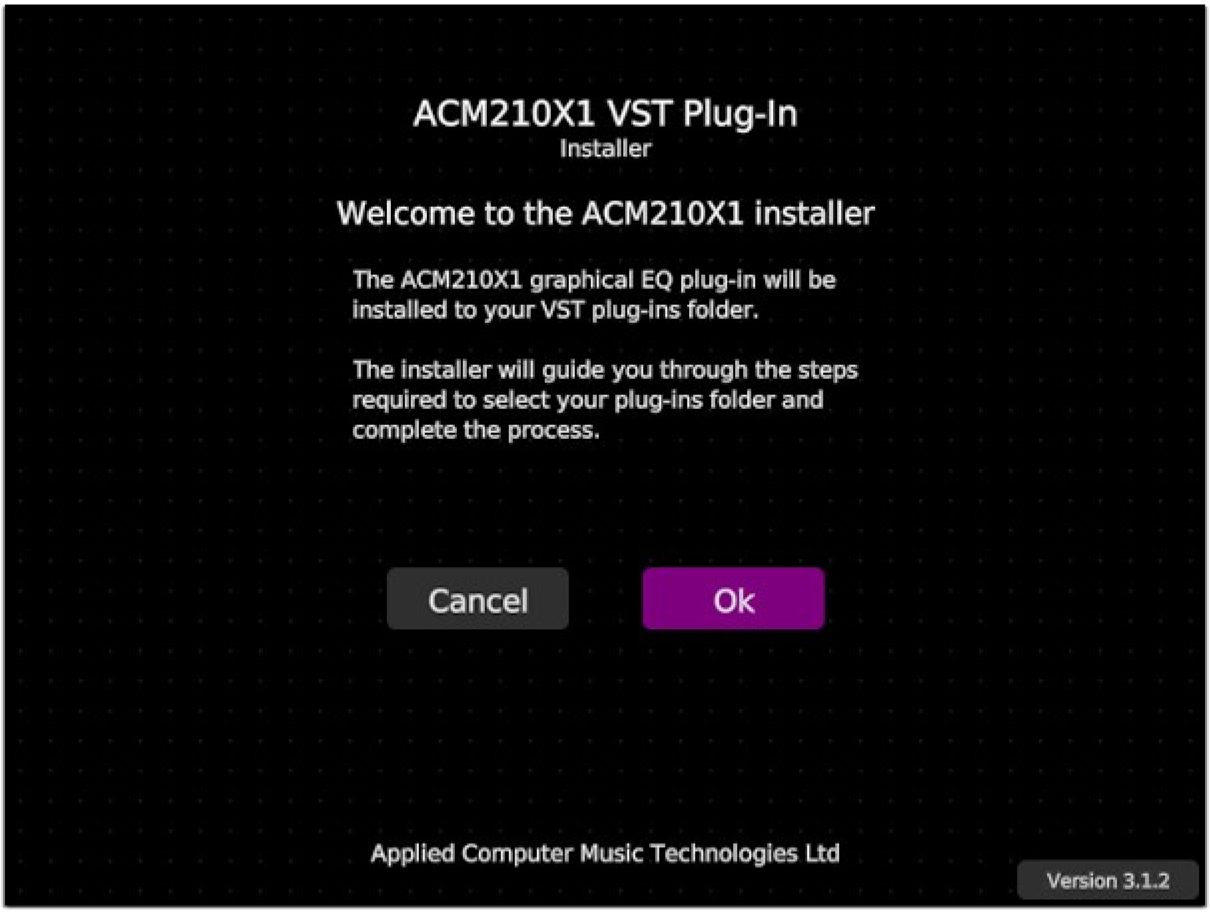
and select **Remove** for the ACM210X1.

2.2 Installing the Plug-In

Installing the Plug-In for Linux:

Within the Linux folder, you will find the x86-64 folder containing the installer executable.

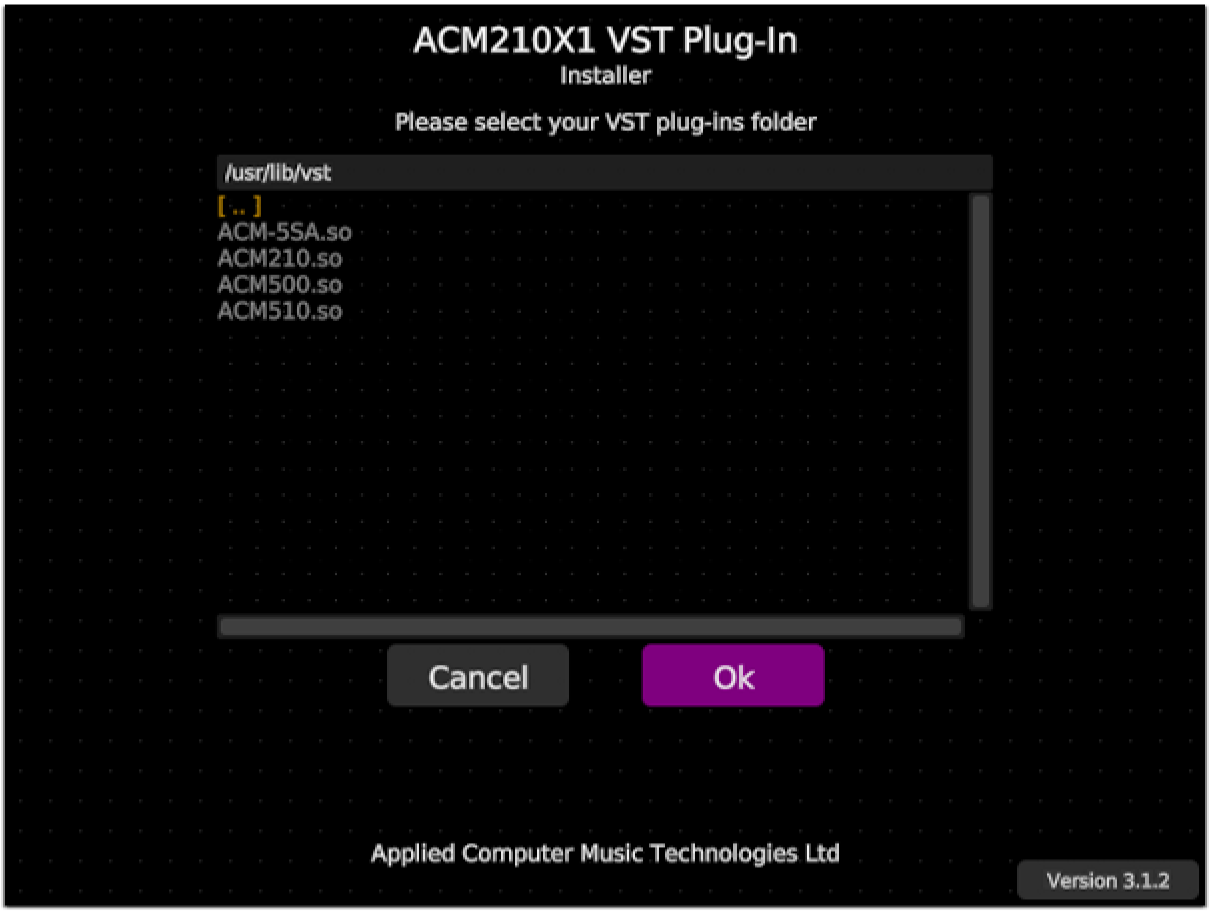
Run the installer executable by (double) clicking it in a file browser, or launching it from the command line. The installer will guide you through the installation process.



Selecting the Install Location

The installer will prompt for your plug-ins folder location. Normally this will be `/home/your-user-name/.vst` or `.vst3` It is recommended to have a single VST or VST3 plug-ins folder, but you can install the plug-ins to as many different locations as you require (just run the installer again and select a new location).

Depending upon system configuration, you may also be prompted for your user or root password if you attempt to install to a system folder, or one to which you do not have write permissions. The installer uses a standard system authentication process `(pkexec)` and does not directly gain elevated permissions.



Troubleshooting

The installer is designed to be self-contained and compatible with most Linux distributions, if you need to backup the installer, the single executable file should be all you need. However, due to the varied and customizable nature of Linux distributions, it is possible that the installer may not be compatible with your system configuration. If this happens, follow these steps to isolate the problem or to install the plug-in manually.

1. Do not try to run the installer as the root / admin user. If you do, there will be a warning message on the console and the installer will exit. The installer is designed to be run as a normal user and will prompt for a password if required.

2. The installer uses the `pkexec` authentication method if attempting to install to a system folder, or one to which the current user does not have write access. (the installer itself never gains root or elevated permissions on your system). If this is not a standard component of your Linux distribution, you will need to correctly install and configure it for your system, or select a different install location with appropriate user access permissions.

3. n some circumstances you may need to mark the installer as 'executable' in order for it to be launched. You can normally do this by right-clicking the installer and selecting:

   `Properties -> Permissions -> Allow executing file as program`

Manually Installing the Plug-In

If your system configuration is not compatible with the installer, you can install the plug-in manually by copying the required files onto your system. You will need to be familiar with command line operations in order to do this.

The plug-in binary files are contained in the `plug-in_binaries.tar.gz` file within the x86 or x86-64 folders. Extract the archive, and you will find it contains VST and VST3 folders.

The VST and VST3 folders contain the plug-in in Linux VST and VST3 format.

There is also a README file which details how to copy the required files onto your system.

2.3 Product Support

If you are unsure how to install the plug-ins, or encounter problems during the installation, please contact:

support@acmt.co.uk

Section 3 - Operation

3.1 - The Graphical User Interface

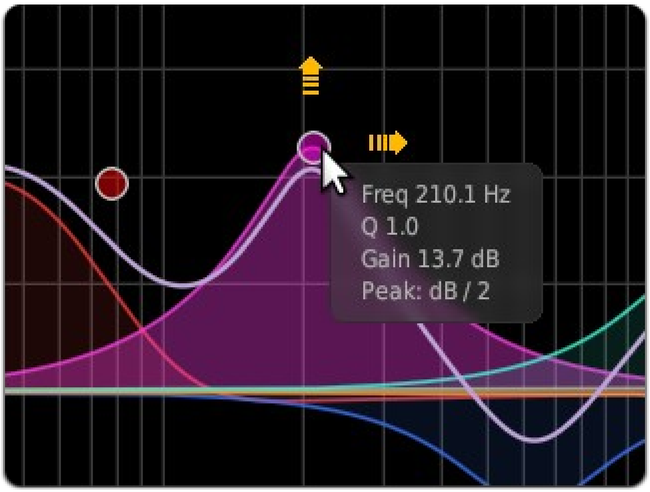
This is the equalizer graphical user interface. Clicking and dragging the marker points for each band will control the filter response for that band. Filter types are selected by right clicking and choosing from the menu. Bands can be individually enabled and disabled by double clicking the marker. Q / Bandwidth can be adjusted using the scroll-wheel with the mouse positioned over the corresponding marker.

The display can be zoomed up to 2X normal size by using the scroll-wheel with the mouse positioned over any part of the graphical display.

3.2 - The Controls

The graphical display adjusts the equalizer response and filter types as follows:

Adjusting Filter Frequency and Gain

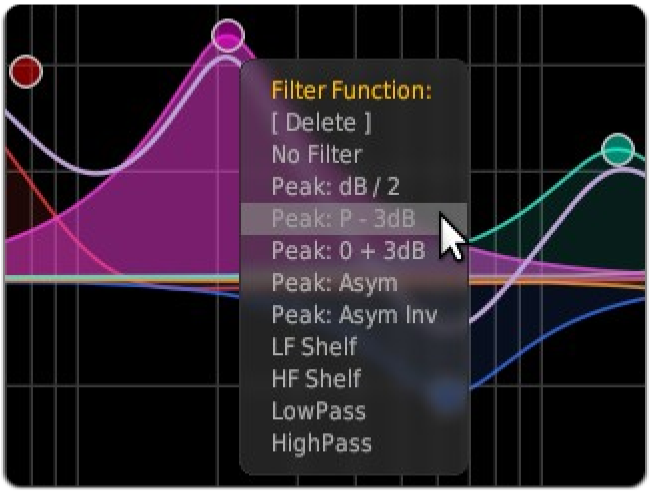


Clicking and dragging the marker associated with a filter band changes the filter response by adjusting the frequency (drag left or right along the frequency axis) and / or gain (drag up and down along the amplitude axis).

As the marker is moved, a context info display will appear showing the frequency, gain, Q and filter type.

These parameters are described in more detail later.

Selecting Filter Types



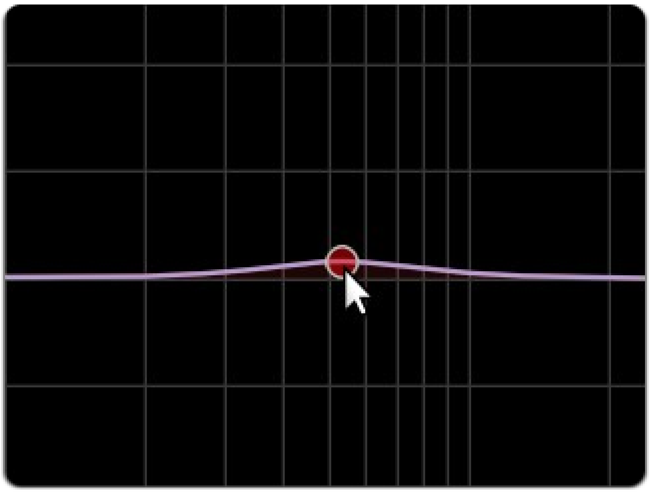
Right clicking on the marker associated with a filter brings up a menu of filter types. The available filter types are discussed in more detail later.

Click on a new type to assign it to the equalizer band.

Clicking [Delete] will remove the band from the graph.

NOTE: When a band is deleted its settings will be lost and it will no longer affect the audio.

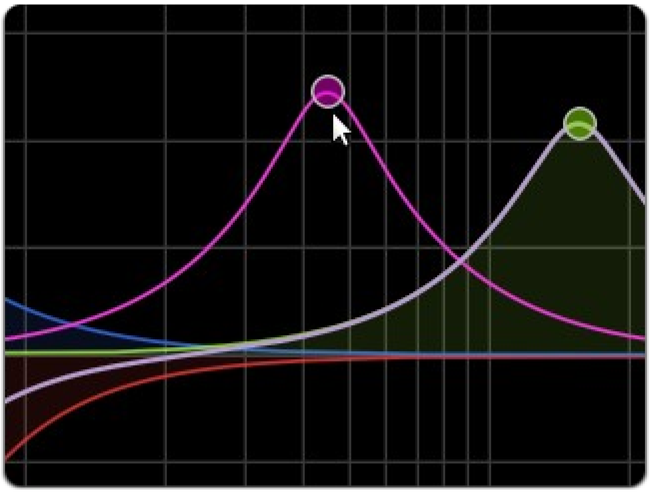
Adding a New EQ Band



Click anywhere along the 0dB line (x-axis) and drag, a new filter band will automatically be added. The filter defaults to a dB / 2 peak, but can be changed if required, as described previously.

NOTE: A maximum of 10 bands can be active on the graph at any time.

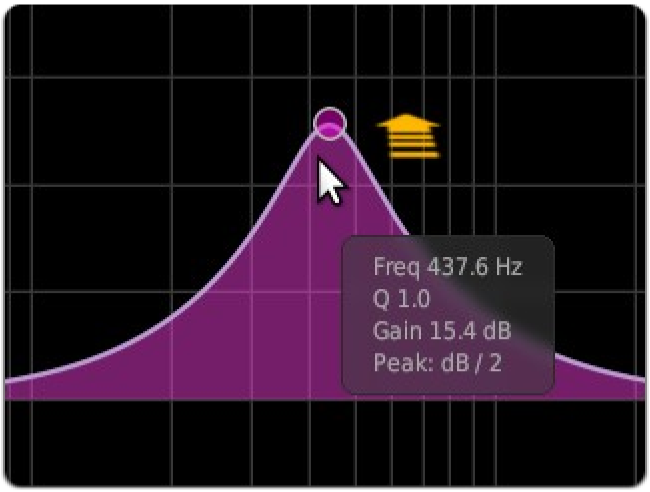
Enabling or Disabling a Band



To switch a band on or off without removing it from the graph, double click the marker. The band will be shown in outline and the audio will pass through it unaffected. The composite graph will also change to show the new EQ response.

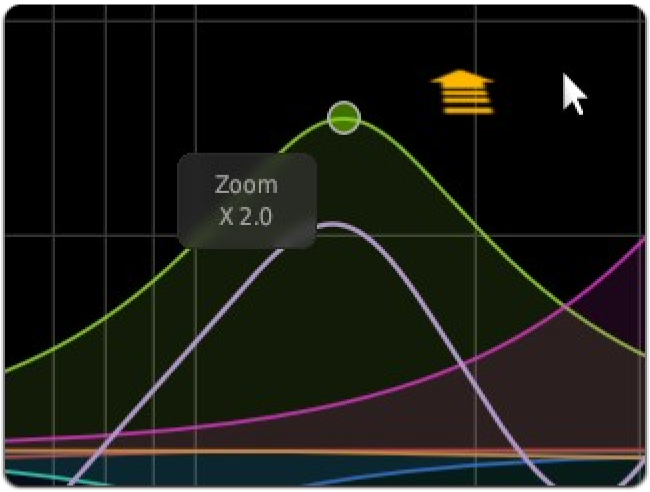
The filter parameters can be altered in the normal way, by adjusting the marker position, even if it is disabled.

Adjusting the Filter Q or Bandwidth



To adjust the filter 'Q' or bandwidth, place the mouse pointer over the marker and use the scroll wheel to adjust for narrow or wider filter peak.

Zooming the Graphical Display

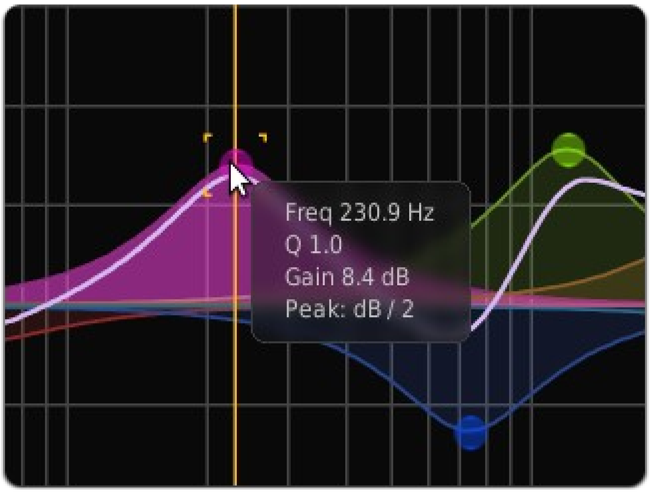


To zoom into the graphical EQ display, place the mouse pointer on an empty area of the graph and use the scroll wheel to increase or decrease the magnification.

The display can be magnified up to X2 permitting finer adjustment of the filter parameters.

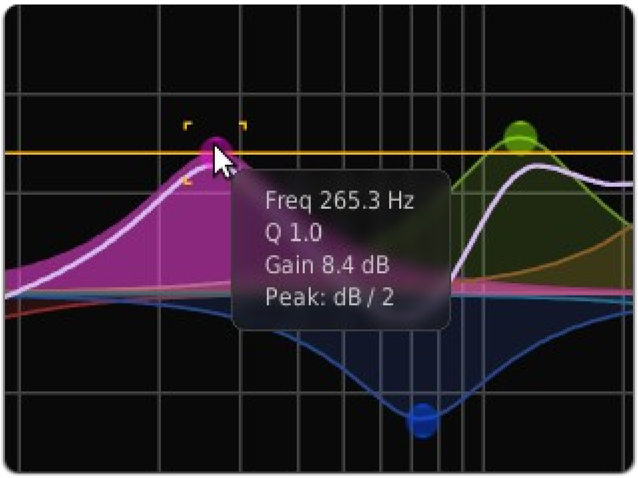
Dragging on the graph while it is zoomed in, will scroll the visible display area.

Freq / Gain Auto-Lock



Pressing [Shift] while dragging on a frequency / gain marker will initiate auto-lock.

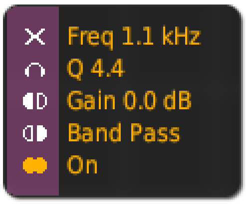
Initiating auto-lock and dragging up or down will lock out movement along the frequency axis, permitting adjustment of filter boost or cut without disturbing critical centre frequency settings.



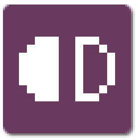
Initiating auto-lock and dragging right or left will lock out movement on the gain axis, permitting adjustment of the filter frequency without disturbing boost or cut.

Auto-lock can be used to good effect to identify a troublesome frequency by boosting a peak filter by a few dBs and sweeping along the frequency axis with gain auto-lock enabled. Once the frequency is identified, switch to frequency auto-lock and drag the peak down to cut at the required frequency.

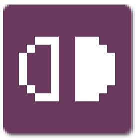
3.2a Stereo Placement



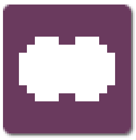
The context info pop-up associated with the active EQ band consists of two parts. The right-hand side shows information about the frequency, gain and bandwidth settings for the selected band, while the left-hand side contains a toolbar for selecting stereo placement and band solo options.



Left
Clicking on the left placement icon assigns the current band to the left side of the stereo image. A tab will be visible on the marker for the selected band indicating its channel assignment, and its effect will be shown in the global **yellow** EQ curve.

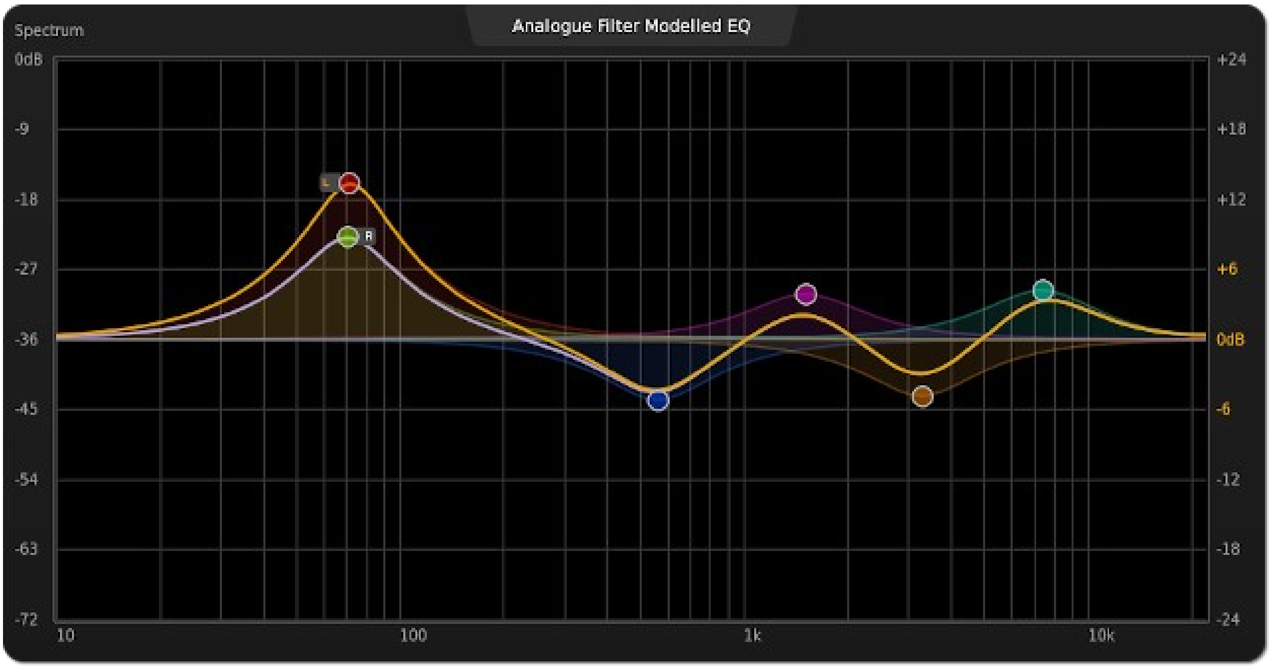


Right
Clicking on the right placement icon assigns the current band to the right side of the stereo image. A tab will be visible on the marker for the selected band indicating its channel assignment, and its effect will be shown in the global **white** EQ curve.

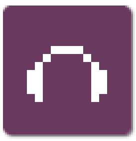


Stereo
Clicking on the stereo placement icon assigns the current band to both the left and right side of the stereo image. The band's effect will be visible in both the global **yellow and white** EQ curves.

When a channel is split to left or right, a separate white global EQ curve will become visible in the graphical display, indicating the cumulative effect of filters assigned to the right-hand channel, while those affecting the left will be indicated by the global yellow EQ curve as normal.



3.2b EQ Band Solo

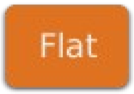


EQ Band Solo
Clicking on the headphone icon places the selected band in solo mode. Only one band can be solo'd at a time.
In solo mode, only the range of frequencies affected by the selected band is audible. For peak filters this equates to a band-pass, for low and high shelf filters this equates to low and high-pass respectively. For low and high-pass (high and low cut) filters, the range of frequencies being cut by the filter is audible.

Bypassing the EQ

Pressing the 'EQ' button switches the processing in or out of the audio path, when the button is on (as shown) the filters are in circuit and will affect the audio. Individual bands can be enabled or disabled by double clicking on their respective markers in the graphical display.

Restoring Flat Settings



The 'FLAT' switch returns the EQ bands to flat settings (0dB boost or cut). The filter types and other parameters are not affected, only the gain.

Clearing the Settings

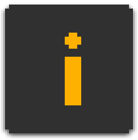
The 'CLEAR switch removes all active EQ bands and clears their settings.
NOTE: All current EQ settings will be lost.

Section 4 – System Toolbars

4.1 - Preset Selectors

In addition to the preset selector options provided by the host application, the plug-in has a pair of preset selector buttons to the right of the status display. Pressing the right or left arrows will step up or down through the factory presets and the four user preset memories.

4.2 - Info Button



Clicking on the Info button will open a pop-up showing the current version, together with a product ID code if the plug-in has been activated with a valid key.

4.3 – Demo Indicator

The red lock icon indicates the plug-in has not been activated with a valid key. To unlock the plug-in and remove the demo limitations, click the button to open the demo / activation key pop-up and enter your key (see section 4.2). Once the key is accepted, the lock will change to an open symbol. **You will need to restart the host application to complete the activation process.**

4.4 – Phase / Polarity

The phase / polarity switch causes the signal at the output to be inverted. When switching between inverted and normal settings, or when bypassing the plug-in with the phase invert enabled, there may be a slight interruption to the audio.

4.5 – Output Trim

The level trim adjusts the output by up to +/- 6dB. Click on the control and drag upwards to increase the level or down to decrease. The mouse scroll-wheel can also be used to adjust the level in +/- 3dB steps. Double clicking on the control will return it to its default 0.0dB setting.

Section 5 - Presets

5.1 – Factory Presets

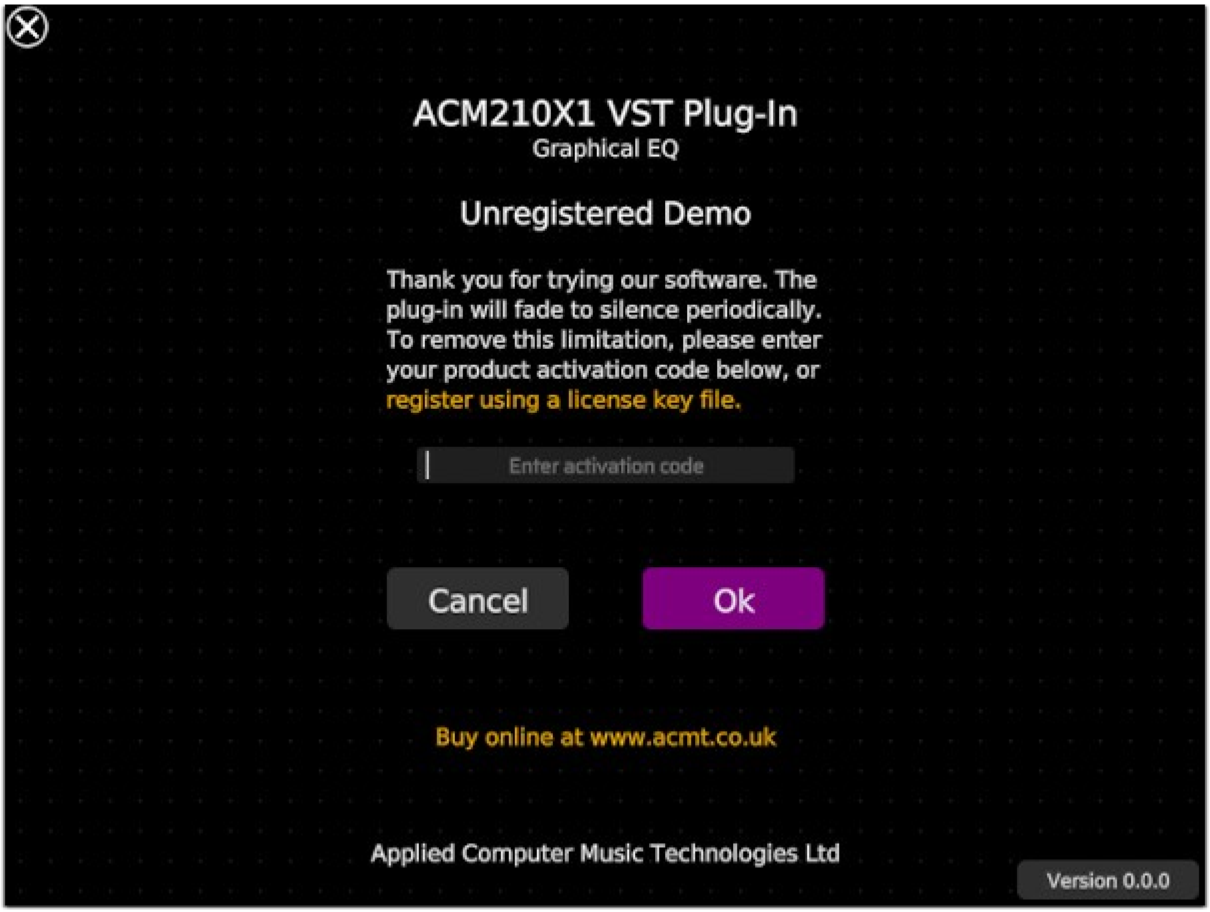
The ACM210X1 has five factory presets - these cannot be overwritten, and are designed to provide a guide to some of the control settings and functionality.

Factory Preset 1 - British EQ NV-1	3-band EQ with mid-parametric boost and low / high frequency shelves with resonance / overshoot similar to some vintage British console EQs.
Factory Preset 2 - British EQ NV-2	Low and High frequency shelving filters with moderate boost and resonance / overshoot similar to some vintage British consoles EQs.
Factory Preset 3 - British EQ NV-3	Low and High frequency shelving filters with selectable LF frequency bands and resonance / overshoot similar to those found in some vintage British console EQs.
Factory Preset 4 - British EQ SL-1	4-band EQ with two mid parametric peak filters. Low and High frequency shelves without overshoot / resonance, giving a broader boost / cut similar to those found in some well known British consoles.
Factory Preset 5 - British EQ SL-2	4-Band EQ with mid parametric peak filters disabled. Low and High frequency shelves with moderate boost and no resonance / overshoot.

Section 6 – Demo Limitations

6.1 - Demo Screen

When the plug-in is first added to a channel / buss, the following screen will appear if it has not been activated by a valid key. This indicates the plug-in is in demo mode and will run with some limitations. To remove these limitations you will need to obtain a valid activation key from the Applied Computer Music Technologies website at: https://www.acmt.co.uk



To activate the plug-in, enter your activation code into the text box (you can also paste it from the clipboard by right-clicking and selecting the 'Paste' context popup). You will need to restart your host application to complete the process. If you do not have a valid key, you can cancel the pop-up and activate it at another time by clicking the lock button in the plug-in's graphical user-interface.

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Appendix

Appendix A – Filter Types

1 - Peak: dB / 2

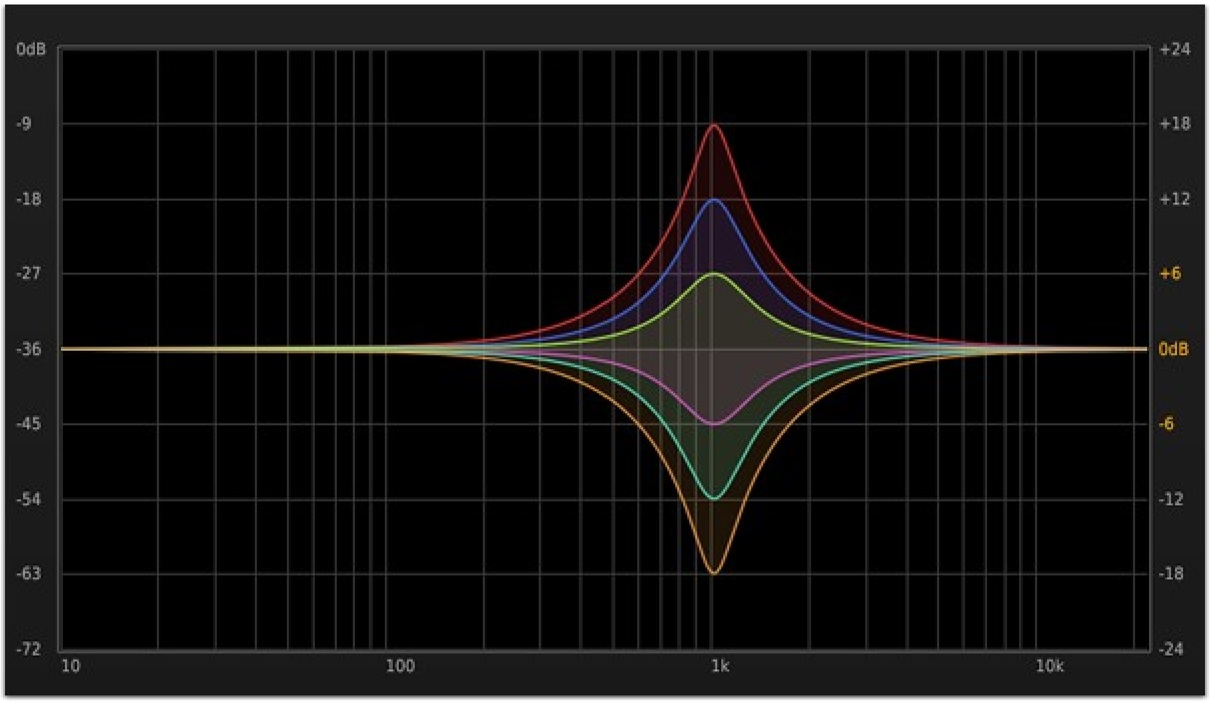


Fig 4.1 Peak dB / 2 filter response with gain settings

This setting provides a standard digital peak filter, where the bandwidth or Q is measured between frequency points at which the output level in dB is half that of the peak boost. The filter provides symmetrical boost and cut responses.

The filter provides an accurate model of an equivalent analogue filter, in particular the gain at Nyquist is defined to match that of the equivalent analogue filter, without requiring high sample rates or internal upsampling.

Filter Type	Peak: dB / 2
Freq Range	20Hz to 20kHz
Boost / Cut	Symmetrical 18dB boost or cut.
Q	0.3 to 5.0

Bandwidth is measured between frequencies at which the filter response in dB is half that of the peak boost (or cut).

2 - Peak: P -3dB

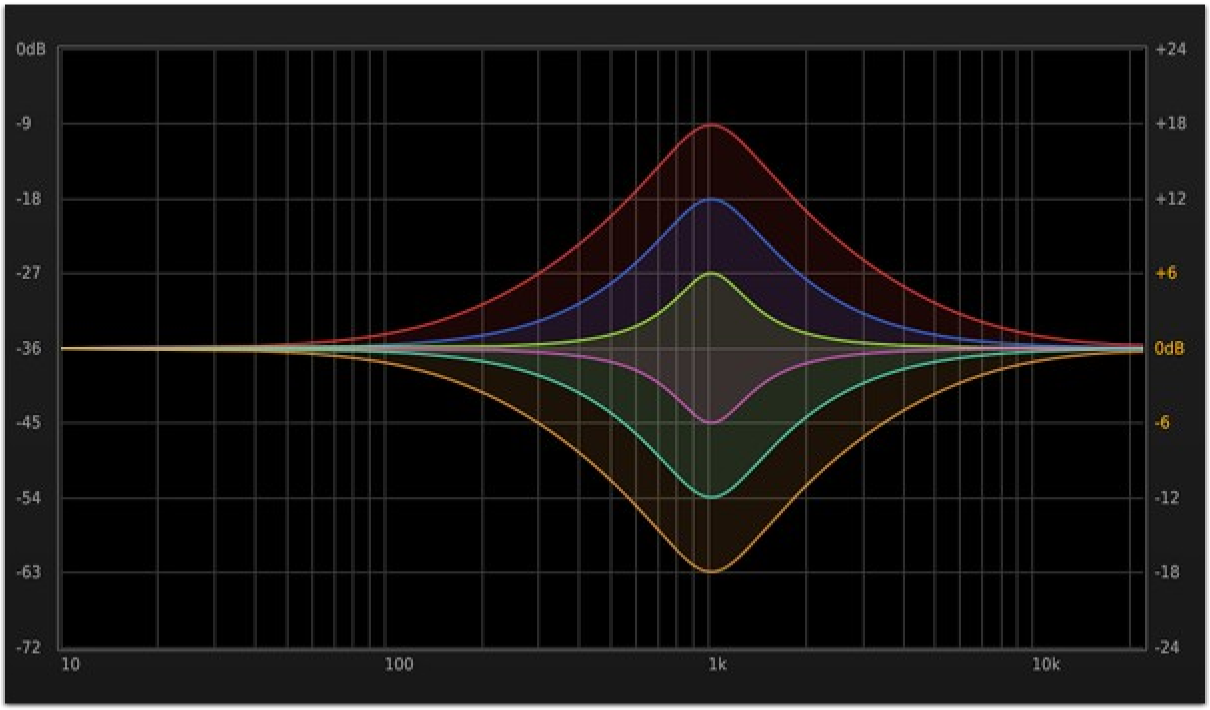


Fig 4.2 Peak P -3dB filter response with gain settings

This setting provides a digital peak filter, where the bandwidth or Q is measured between frequency points at which the output level is 3dB below that of the peak boost (Peak -3 or P -3dB). The filter provides symmetrical boost and cut responses.

The filter provides an accurate model of an equivalent analogue filter, in particular the gain at Nyquist is defined to match that of the equivalent analogue filter, without requiring high sample rates or internal upsampling.

Filter Type	Peak: P -3dB
Freq Range	20Hz to 20kHz
Boost / Cut	Symmetrical 18dB boost or cut.
Q	0.3 to 5.0

Bandwidth is measured between frequencies at which the filter response is 3dB below the peak boost (or 3dB above the peak cut).

3 - Peak: 0 + 3dB

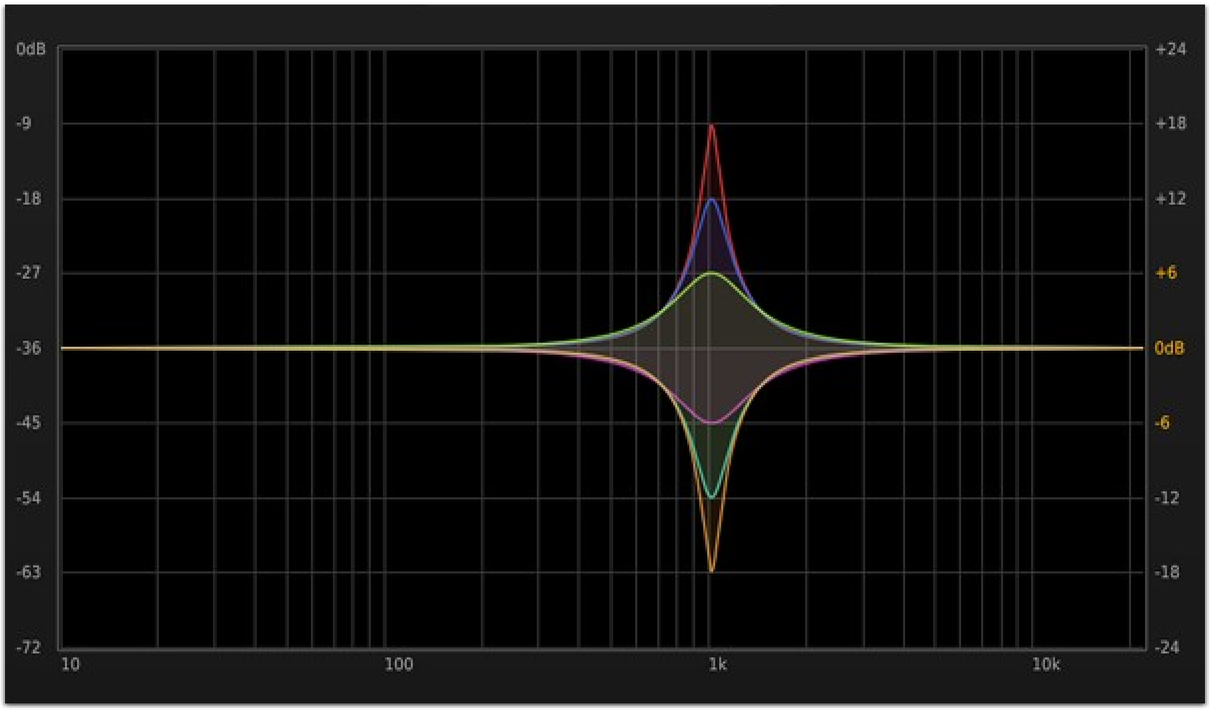


Fig 4.3 Peak 0 + 3dB filter response with gain settings

This setting provides a digital peak filter, where the bandwidth or Q is measured between frequency points at which the output level is 3dB above unity - 0dB - for boost, or 3dB below unity for cut. The filter provides symmetrical boost and cut responses.

The filter provides an accurate model of an equivalent analogue filter, in particular the gain at Nyquist is defined to match that of the equivalent analogue filter, without requiring high sample rates or internal upsampling.

Filter Type	Peak: 0 + 3dB
Freq Range	20Hz to 20kHz
Boost / Cut	Symmetrical 18dB boost or cut.
Q	0.3 to 5.0

Bandwidth is measured between frequencies at which the filter response is 3dB above unity (for boost) or 3dB below unity (for cut).

4 - Peak: Asymmetric

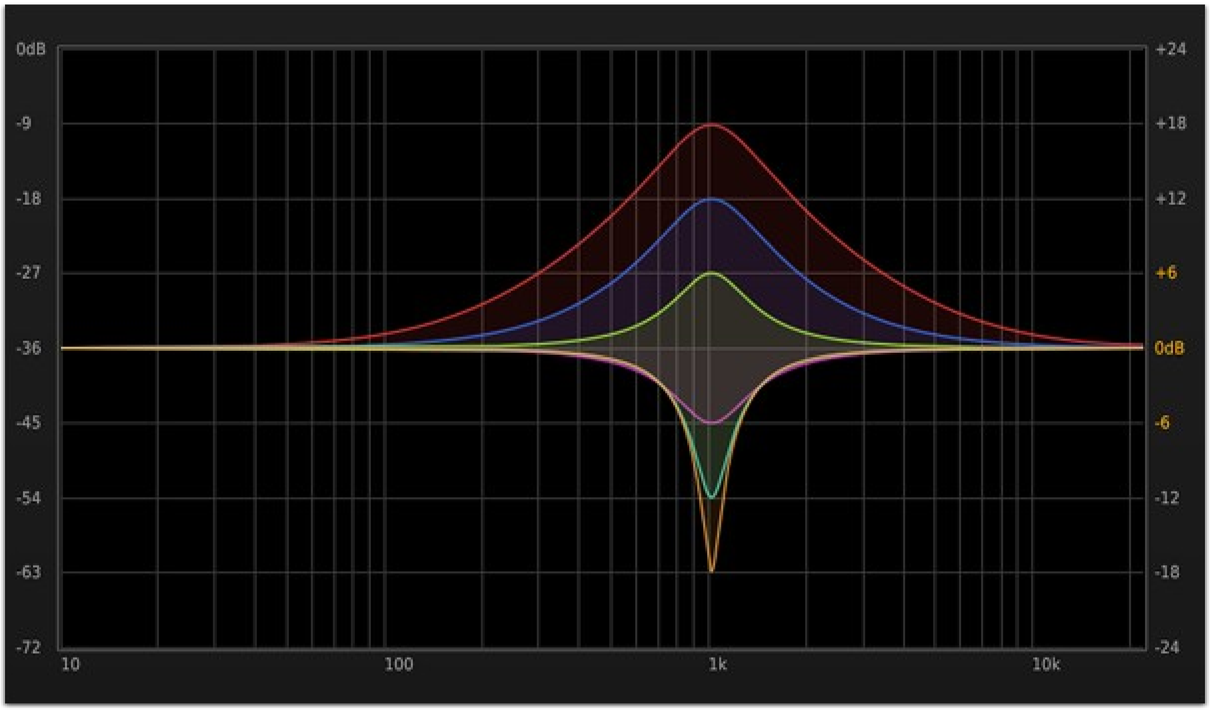


Fig 4.4 Peak Asymmetric filter response with gain settings

This setting provides a digital peak filter, where the bandwidth or Q is measured between frequency points at which the output level is 3dB below the peak (for boost) or 3dB below unity - 0dB - for cut. The filter provides asymmetrical boost / cut responses in which the cut is much narrower than the boost, for the same value of Q.

The filter provides an accurate model of an equivalent analogue filter, in particular the gain at Nyquist is defined to match that of the equivalent analogue filter, without requiring high sample rates or internal upsampling.

Filter Type	Peak: Asymmetric
Freq Range	20Hz to 20kHz
Boost / Cut	Asymmetrical, P -3dB for boost,
	0 - 3dB for cut.
Q	0.3 to 5.0

Bandwidth is measured between frequencies at which the filter response is 3dB below peak (for boost) or 3dB below unity (for cut).

5 - Peak Asymmetric Inverted

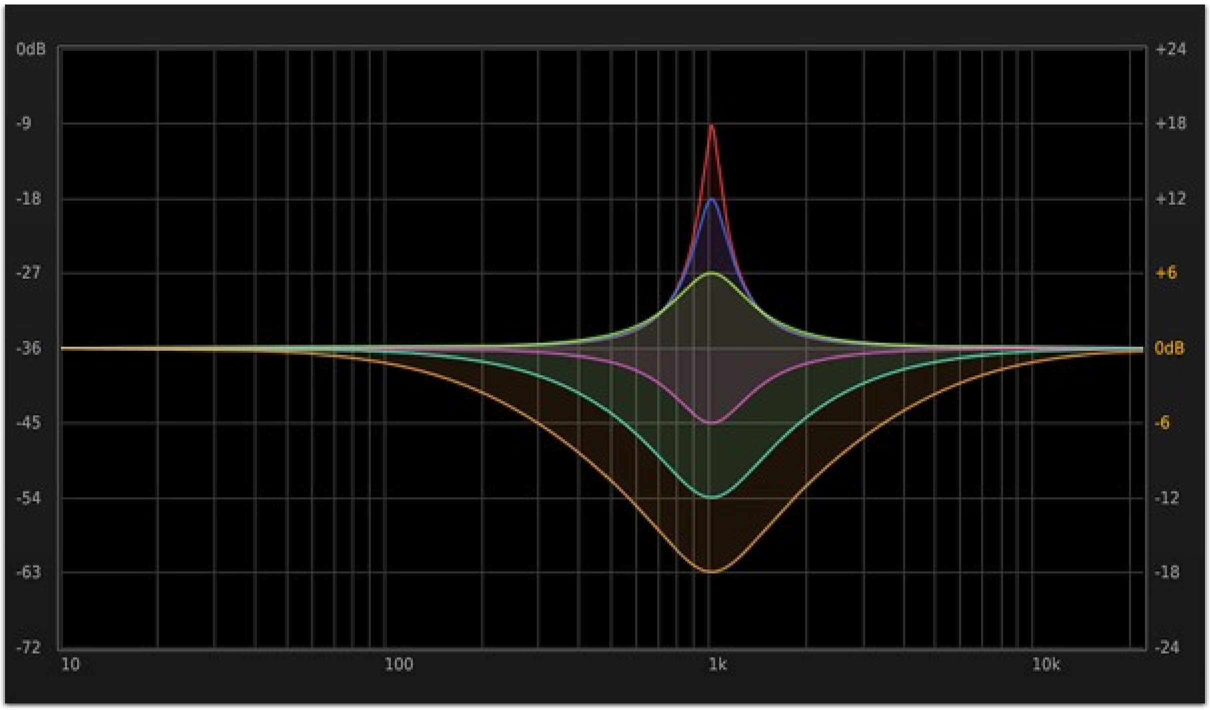


Fig 4.5 Inverse Peak Asymmetric filter response with gain settings

This setting provides a digital peak filter, where the bandwidth or Q is measured between frequency points at which the output level is 3dB above unity (for boost) or 3dB below the peak for cut. The filter provides asymmetrical boost / cut responses in which the boost is much narrower than the cut, for the same value of Q.

Filter Type	Peak: Asymmetric Inverted
Freq Range	20Hz to 20kHz
Boost / Cut	Asymmetrical, 0 +3dB for boost,
	P -3dB for cut.
Q	0.3 to 5.0

Bandwidth is measured between frequencies at which the filter response is 3dB above unity (for boost) or 3dB below peak (for cut).

6 - Low Frequency Shelf

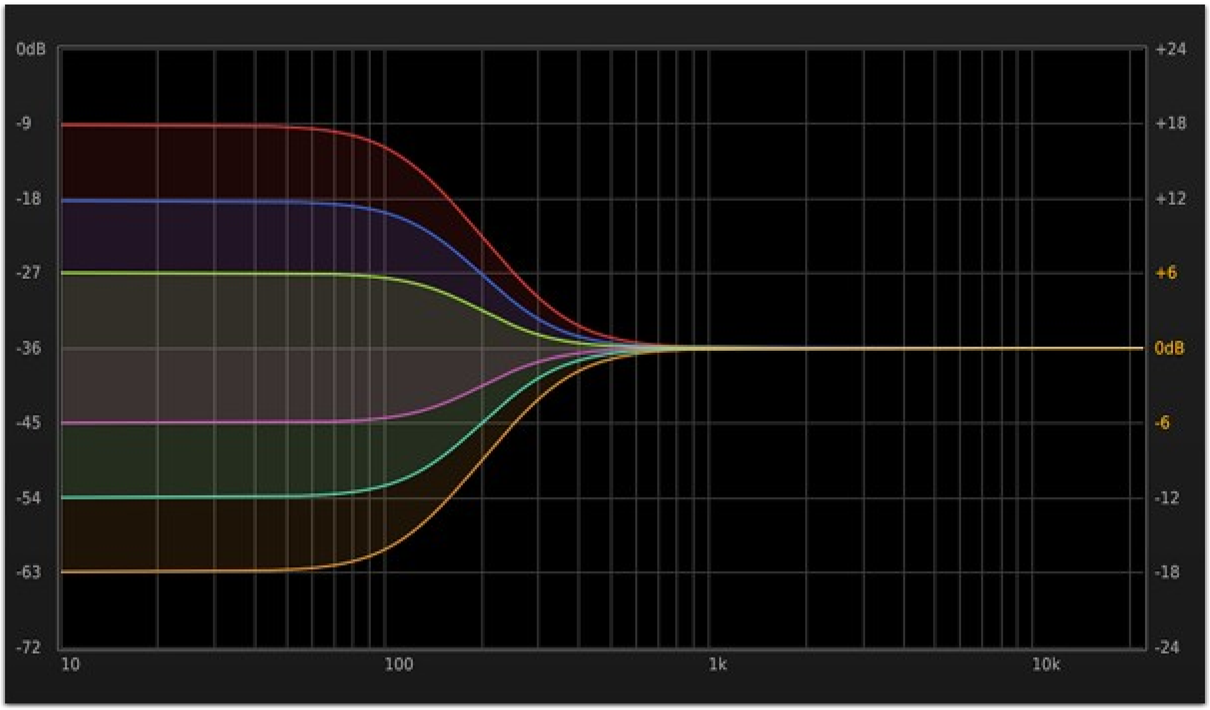


Fig 4.6a Low Frequency Shelf Q = 0.7

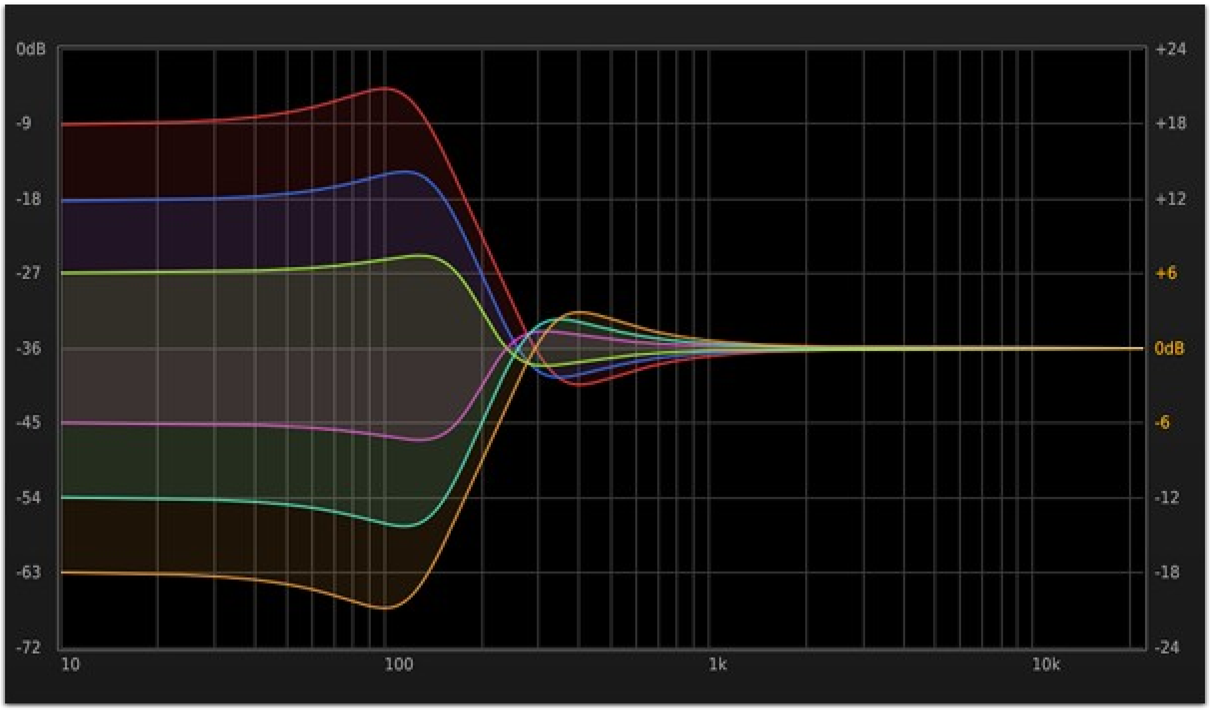


Fig 4.6b Low Frequency Shelf Q = 1.4

This setting provides a low frequency shelving filter, with adjustable 'Q'. In this design, the 'Q' value determines the steepness of the filter slope and the amount of overshoot at the upper and lower corners. The filter frequency is defined as the midpoint between the upper and lower corners. Figure 4.6a and 4.6b show the effect of different Q values for various amounts of boost and cut.

Filter Type	Low Frequency Shelf
Freq Range	20Hz to 20kHz
Boost / Cut	18dB
Slope	Variable

Low frequency shelving function, with variable slope and up to 18dB boost or cut.

7 - High Frequency Shelf

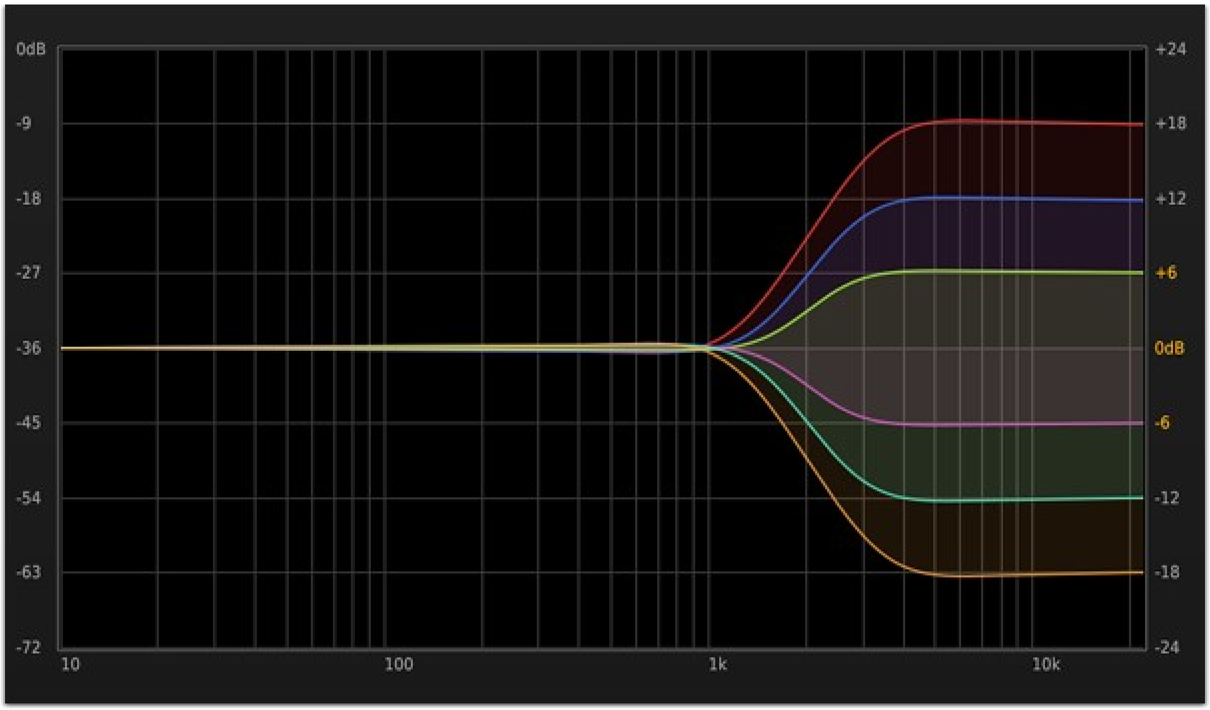


Fig 4.7a High Frequency Shelf Q = 0.7

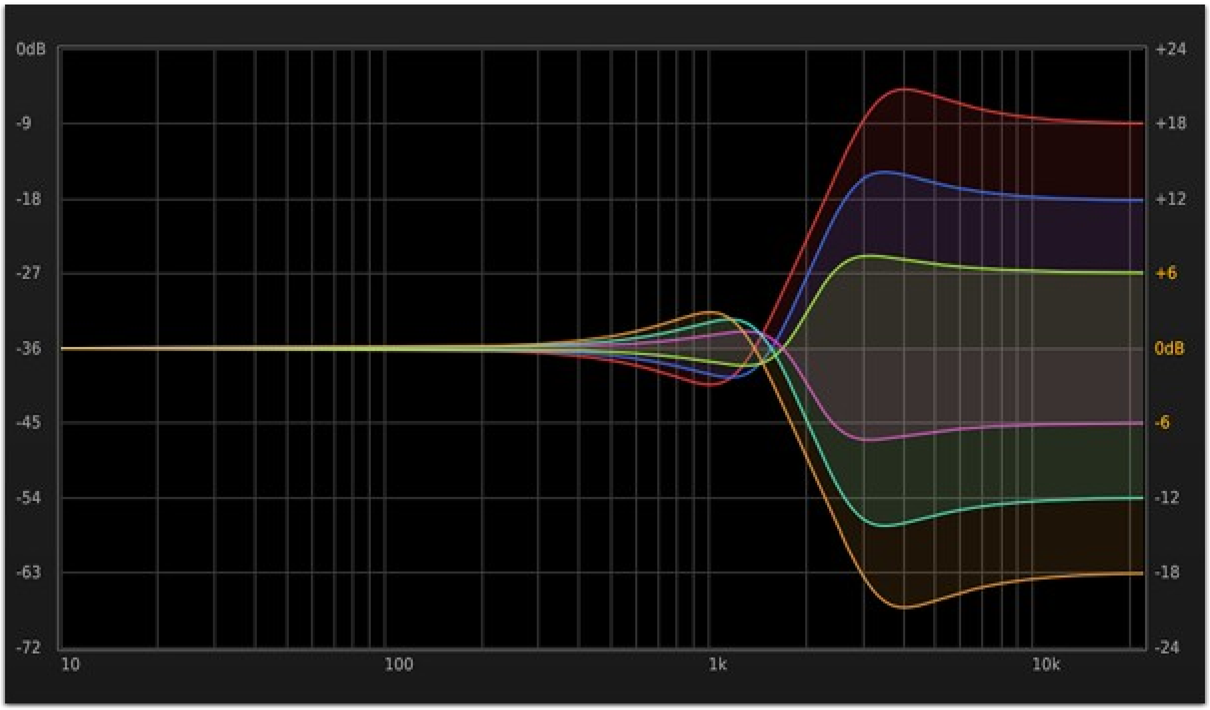


Fig 4.7b High Frequency Shelf Q = 1.4

This setting provides a high frequency shelving filter, with adjustable 'Q'. In this design, the 'Q' value determines the steepness of the filter slope and the amount of overshoot at the upper and lower corners. The filter frequency is defined as the midpoint between the upper and lower corners. Figure 4.7a and 4.7b show the effect of different Q values for various amounts of boost and cut.

Filter Type	High Frequency Shelf
Freq Range	20Hz to 20kHz
Boost / Cut	18dB
Slope	Variable

High frequency shelving function, with variable slope and up to 18dB boost or cut.

8 - Low-Pass

Fig 4.8 Low-pass filter response with frequency and Q settings

This setting provides a 2-pole low-pass filter with variable 'Q' or resonance at the cut-off frequency.

Filter Type	Low-Pass
Freq Range	20Hz to 20kHz
Slope	12dB / octave
Stop-band Attenuation	120dB

Variable 2-pole low-pass filter with variable resonance and 12dB / octave rate of attenuation.

9 - High-Pass

Fig 4.9 High-pass filter response with frequency and Q settings

This setting provides a 2-pole high-pass filter with variable 'Q' or resonance at the cut-off frequency.

Filter Type	High-Pass
Freq Range	20Hz to 20kHz
Slope	12dB / octave
Stop-band Attenuation	120dB

Variable 2-pole high-pass filter with variable resonance and 12dB / octave rate of attenuation.

10 – Band Pass

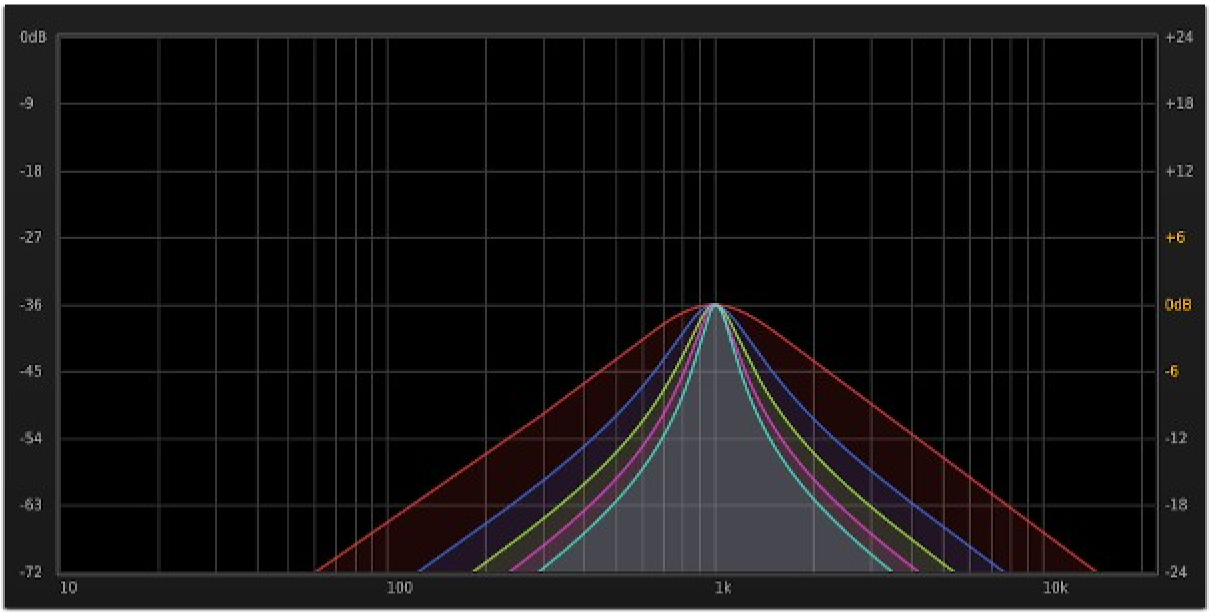


Fig 5.0 Band-pass filter response with Q settings

This setting provides a 2-pole high-pass filter with variable 'Q' or resonance at the cut-off frequency.

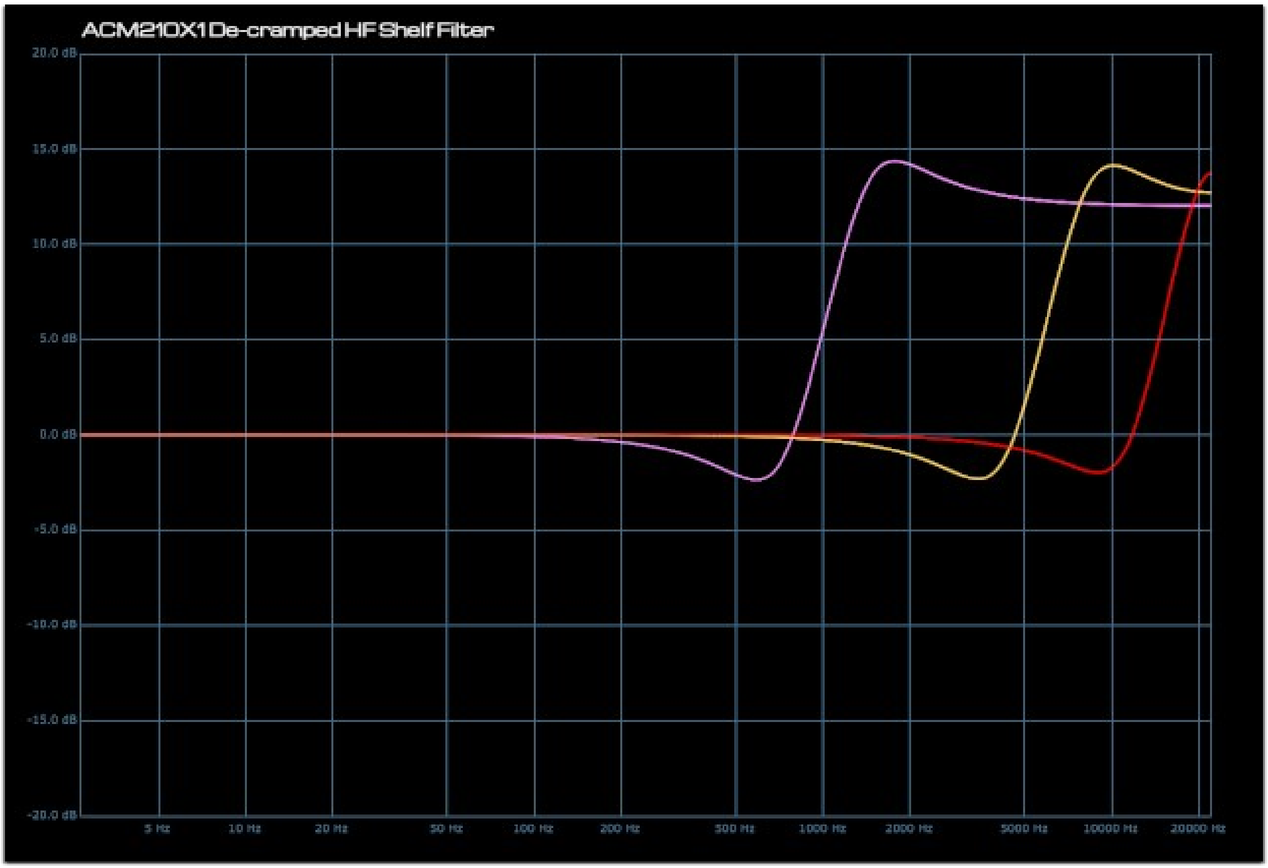
Filter Type	Band-Pass
Freq Range	20Hz to 20kHz
Q	0.3 to 5.0

Bandwidth is measured between frequencies at which the filter response is 3dB below unity.

Appendix B – De-cramped Filters

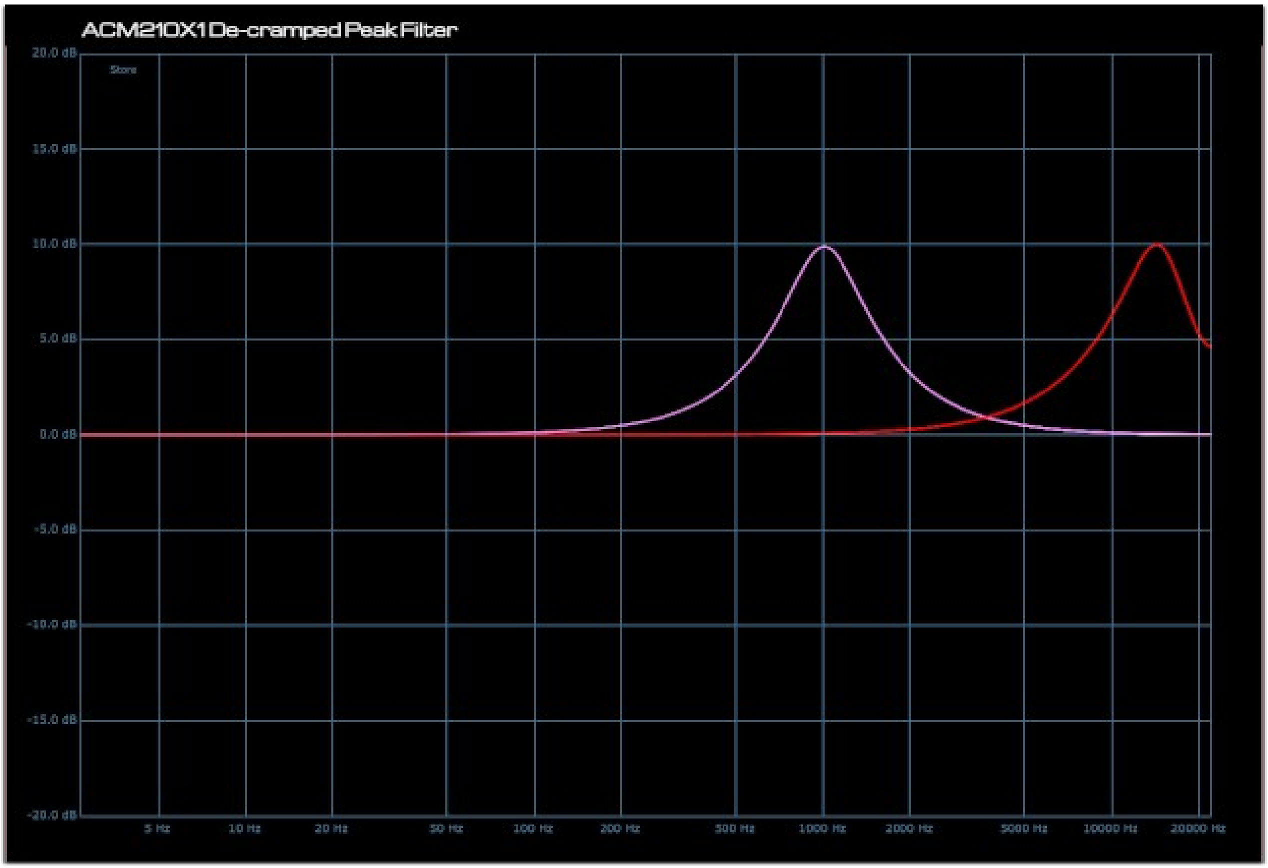
At commonly used sample rates such as 44100Hz and 48000Hz, conventional digital EQs can cause frequency response cramping as the filter frequency approaches its upper limit. Conventional solutions such as oversampling can be CPU intensive, introducing other artefacts and additional latency. The ACM210X1 uses innovative analogue filter modelling technology to provide a zero latency de-cramped response, accurately replicating that of an equivalent analogue filter without requiring oversampling.

De-cramped HF Shelving Filter - Measured Response:



The de-cramped shelving filter response closely matches that of an equivalent analogue filter, even as the filter frequency approaches its upper limit.

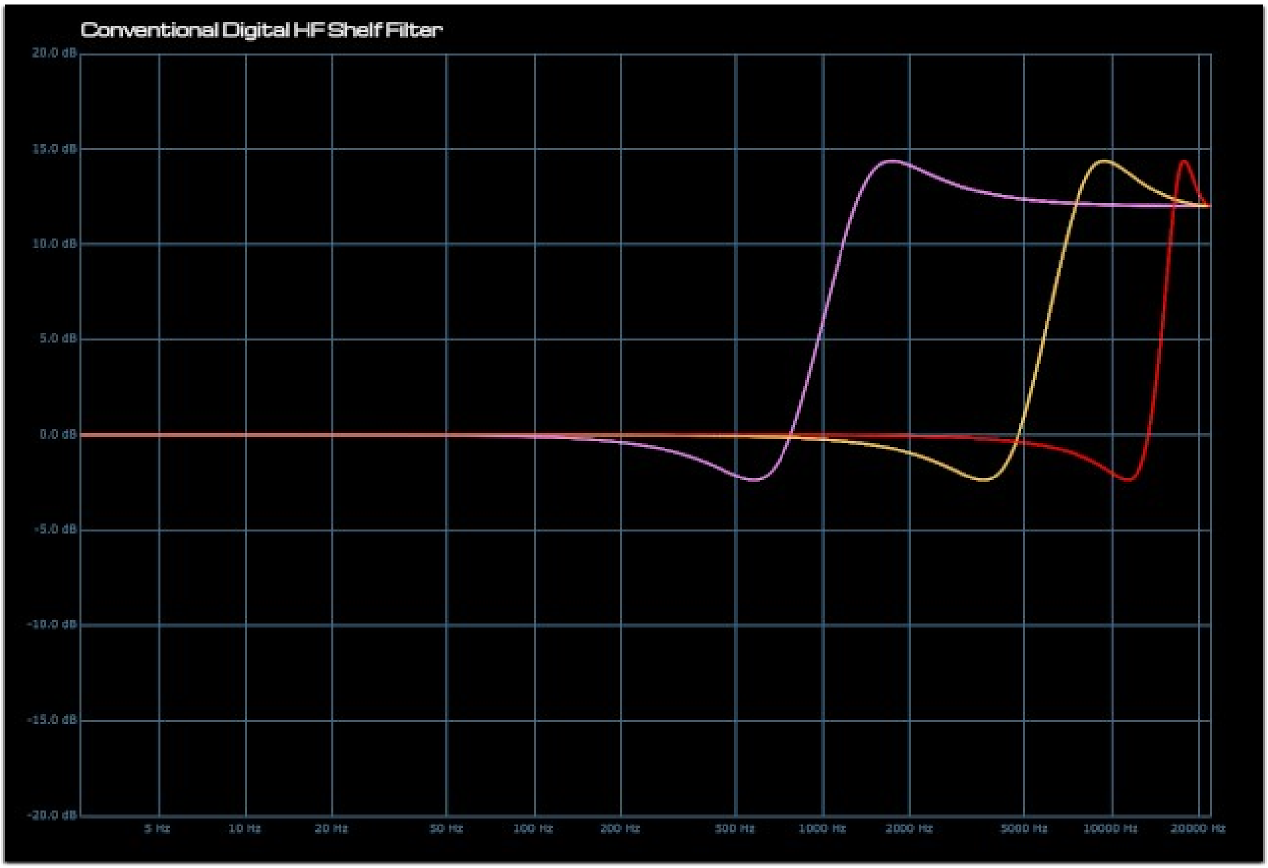
De-cramped Peak Filter – Measured Response:



Similarly the de-cramped peak filter response closely matches that of an equivalent analogue filter as the filter frequency approaches its upper limit.

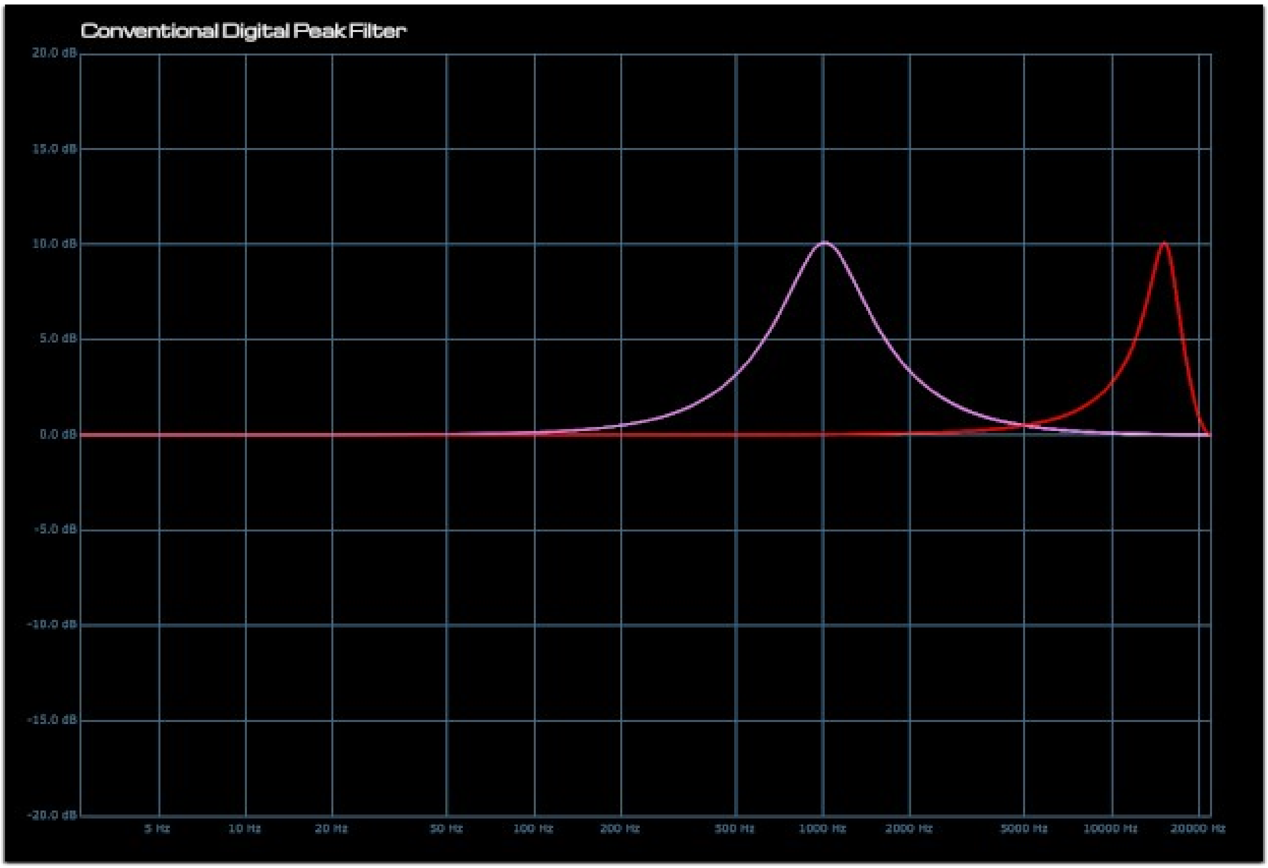
By contrast, the cramping effect of a conventional EQ can clearly be seen in the examples below as the filter frequency is increased (yellow and red traces). In this instance the shelf slope becomes significantly steeper and the resonant peak becomes narrowed.

Conventional Digital Shelving Filter – Measured Response:



In a conventional digital shelving filter, the shelf slope becomes steeper and the resonant peak becomes narrowed as the filter frequency is increased (red trace).

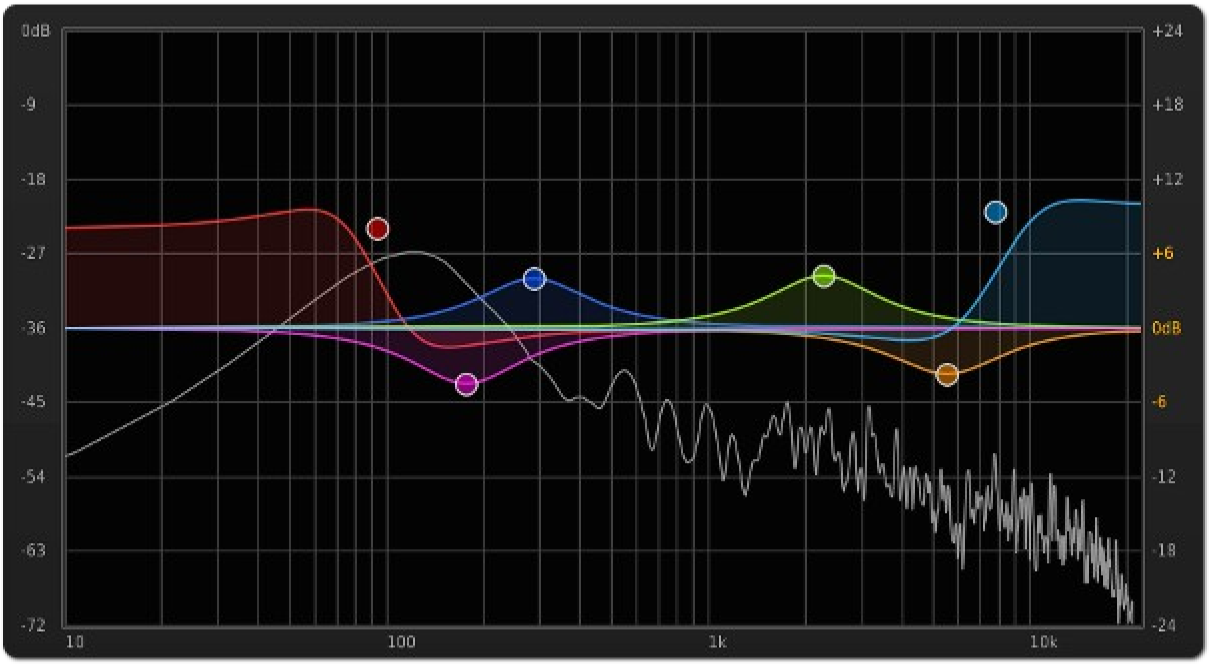
Conventional Digital Peak Filter – Measured Response:



Similarly, in a conventional digital peak filter, the peak becomes narrowed and cramped as the filter frequency is increased.

Appendix C - FFT Analyser

1 - The FFT Display



FFT Display

The ACM210X1 EQ also provides real-time spectrum analysis of the signal via an FFT (Fast Fourier Transform) display. This provides an overview of the signal's harmonic content (shown here behind the EQ graph in grey).

The FFT can be switched on and off - the plug-in will use more CPU when processing and displaying the FFT - and the signal source can be either pre or post the EQ processing. The FFT is described in more technical detail later.

The FFT processing is switched on or off by clicking on the 'FFT' button. With the button set to 'On' (shown) the FFT processing is enabled.

With the FFT source selector switched to 'Pre' (shown) the FFT processor will monitor audio *before* it enters the EQ.

With the FFT source selector switched to 'Post' (shown) the FFT processor will monitor audio *after* it leaves the EQ.

Note: If the EQ is bypassed in this mode, the FFT will display the unprocessed (clean) audio. The FFT shows an average of the Left and Right stereo channels.

2 - The FFT in Detail

The FFT display provides a powerful insight into a signal's harmonic content, however it is a complex process which requires some detailed understanding in order to use it to greatest benefit.

The FFT captures time domain audio data and converts it into a frequency domain representation. This is then shown on the display, as a graph of *level* vs *frequency*, illustrating the relative strengths of the frequency components which make up the audio signal.

The FFT algorithm operates over a pre-determined number of audio samples, the number of audio samples used for each FFT update directly determines the resolution of the FFT (that is, how well it will resolve the relative strengths of different frequencies in the signal). For example:

If the FFT algorithm operates on a buffer of 1024 samples captured at a sample rate of 48kHz, the output will be a series of frequency 'bins' - each 'bin' representing the level at a point in the frequency spectrum, linearly spaced between 0 (sometimes referred to as DC) and the sample rate (in this case 48kHz).

So in this example the frequency spacing between each bin will be 48000 / 1024 = 46.88Hz.

A shorter sample buffer, or a higher sample rate for the same size buffer will result in wider spacing between the bins reducing the effective resolution of the output.

Conversely, a longer sample buffer will increase the resolution by providing a closer spacing between the frequency bins, however the need to acquire more data for each FFT update will result in greater latency between the audio and the visual display.

The FFT analyser in the ACM210X1 uses a sample buffer of 1024 samples, which are interpolated to 4096 samples during the FFT process.

2a - Interpolation vs FFT Buffer Size

Interpolating a shorter length FFT is not the same as performing the FFT calculation over a larger number of acquired samples even though the output data set will be the same size.

It is important to realise that interpolation to 4096 samples provides no new data, that is, the effective bin spacing and therefore the resolution is the same as it would be at 1024 samples, however the interpolated data points permit smoother rendering of the information in the graphical display.
This makes it easier to determine the correct profile of the graph, especially at low frequencies where the resolution is at its most coarse (the FFT frequency bins are spaced linearly which means that while the spacing may be small relative to high frequency signals, at low frequencies it may represent as much as one octave or more).

The use of a shorter buffer size also ensures the latency between the audio and the display is kept to an acceptable limit.

3 - FFT Windowing Functions

The FFT algorithm assumes that the input audio waveform is continuous and has existed forever, therefore, when performing the calculation over a finite interval of the waveform (as is the case when applying it to a buffer or n samples) it is necessary to use a 'window' function to taper the level of the audio at the start and end of the capture interval.

The type and nature of the window function directly affects the nature and the accuracy of the FFT results. In short there will always be some compromises necessary in selecting a suitable window function. The FFT analyser in the ACM210X1 uses a Blackman window, which provides a good compromise between dynamic range and resolution for most audio signals.

4 - Sample Accurate Synchronous FFT Algorithm

The FFT process in the ACM210X1 uses an innovative approach which guarantees the FFT is sample accurate (e.g. there is no lost data between display updates, as every audio sample passes through the algorithm) and which also spreads the processing load as uniformly as possible over the available sample acquisition time.

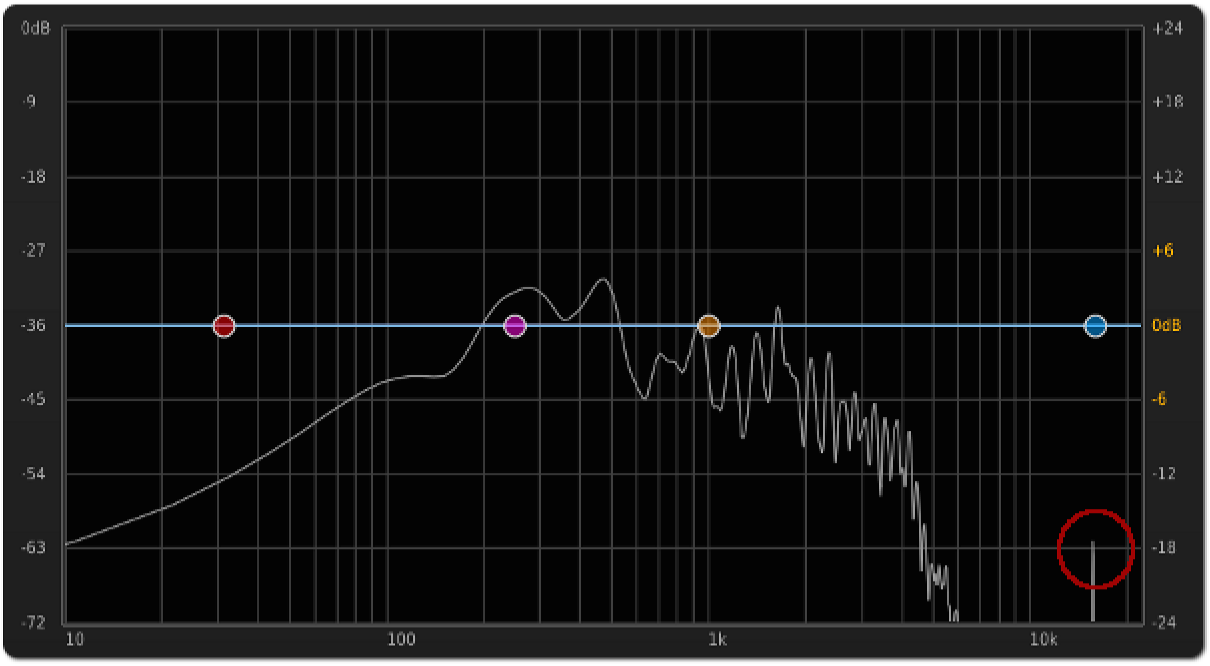
It does this without requiring separate processing threads,which ensures that the audio processing is never deadlocked and eliminates the possibility of priority inversion and / or resulting audio glitches.

This has considerable benefits in ensuring reliable and repeatable performance especially at low latency settings, and also ensures that offline rendering will not suffer buffer over / under-runs.

It should however be noted that though the FFT algorithm is designed to be as efficient as possible, such processing (and the display of its resulting data) will require a powerful CPU. This should not be an issue for modern PC hardware, but it is recommended that the FFT be switched out when not required and / or the longest audio system latency / buffer settings consistent with acceptable performance should be used in order to minimize CPU load.

Empirically it has been shown that adding more instances of the plug-in does not increase CPU load in direct proportion, that is, you will see a smaller increase in CPU load for a second instance of the plug-in than for the first etc.

5 - Using the FFT - An Example



The FFT display can prove useful in detecting unwanted noise or interference in a recording, which may not always be audible but which may cause problems or introduce other artefacts in later processing (audio data compression for example).

In the above screenshot, an FFT of a recording clearly shows a frequency 'spike' above 10kHz (highlighted in red).

In this example, by zooming the display and placing a marker at the peak's location it can be determined that the peak is at 15.625kHz.

The frequency suggests it is likely in this case the interference has come from an external CRT (or television monitor).

Whatever the source, the signal has found its way unnoticed onto the final mix.

Appendix D - Technical Data

1 - Technical Specifications

1a - Equalizer

Frequency Response:	0Hz to Fs / 2 (bypassed or flat) - where Fs is the sample rate.
Number of Filter Bands:	10
Q:	0.3 to 5.0 (peak filters) 0.7 to 1.4 (Shelf / Low and High-Pass)
Maximum Boost:	18dB
Maximum Cut:	-18dB
Filter Types:	High Pass: 2-pole with adjustable resonance. Low Pass: 2-pole with adjustable resonance. Band Pass: 0dB – Q measured at -3dB. HF Shelf: Adjustable resonance. LF Shelf: Adjustable resonance. Peak: (symmetrical - Q measured at 'Peak dB / 2'). Peak: (symmetrical - Q measured at 'Peak - 3dB'). Peak: (symmetrical - Q measured at '0 + 3dB'). Peak: (asymmetrical - Q measured at 'Peak - 3dB' for boost and '0 - 3dB' for cut). Peak: (asymmetrical inverted - Q measured at '0 + 3dB' for boost and 'Peak - 3dB' for cut).
Internal Processing:	64bit floating point.
Algorithm:	Analogue filter modelling coefficient generator provides filter responses which closely match analogue equivalents without requiring high sample rates or internal upsampling.
Reference Level:	0dBu = -18dBFS.
Dynamic Range:	Limited only by internal processing resolution (64bit floating point).
Output Trim	+/- 6dB.
Control Types	GUI operated controls via graphical display of the frequency response with switchable individual band displays and combined transfer function graph.

1b - FFT Analyser

Algorithm:	Windowed Fast Fourier Transform.
FFT Window Function:	Blackman - Harris.
FFT Resolution:	1024 Frequency bins covering 0 to Fs (linear frequency spacing) interpolated to 4096 bins during the FFT process, where Fs is the sample rate.
FFT Latency:	1024 samples.
FFT Slope:	Selectable: +3dB per octave (slope switch enabled) or flat (slope switch disabled).
FFT Source:	Selectable: Pre (input) or post (output). Average of left and right stereo channels.

2 - System Requirements

Memory:	At least 1GB
CPU:	2GHz x86-64 compatible CPU or better (system resource usage dependent on control settings, specifically FFT analyser enabled / disabled).
Operating System:	Windows or Linux. Graphics adapter / drivers with support for OpenGL 3.x or later (GLX / X11).
Host Application:	VST, VST3 or CLAP compatible host application.
Audio Sample Rate:	44.1kHz to 96kHz.

Appendix E - Spare Parts and Service

With regular care and maintenance your new ACM210X1 equalizer plug-in is designed to give long and reliable service. Spare parts and service updates can be downloaded from:

`https://www.acmt.co.uk`

Always ensure it has adequate ventilation and is kept free from dust. **Always use genuine replacement parts.** For service and support information contact:

`support@acmt.co.uk`

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