LIME²

User's manual

LIME²
CRAZY88

User's manual

INTRODUCTION

Thank you for choosing lime2! The newest entry into the Acustica plugin world. Please take the time to read this user manual carefully, to facilitate and ensure a proper understanding of this plugin

LIME2

Lime2 provides our customers with different devices, sampled with Acustica's technology and merged in a single plug-in. It's inspired by a legendary inline mixing desk, and it's aimed for those who are looking for a faithful emulation of that legendary sound.
Without a doubt, Lime is one of the best choice to "mix in a box". This groundbreaking channel strip delivers the quality and versatility to enhance any performance that requires that distinctive British sound.
It features several preamps, two sets of low and hi pass flters, two four-band equalizers, 5 compressors and an external dynamics side chain. You are free to change their routing order in the signal path as appropriate by selecting one of the 14 different configurations.

Lime2 CRAZY88

LIME2 CRAZY88 is a bundle consisting of LIME2 BUS, LIME2 PRE, LIME2 EQ.

To help unleash the incredible potential of LIME2 we have created a BUNDLE including all the single modules, enabling sound engineers to use each component separately, with signifcant savings in terms of CPU. This will help enhance your mixes, ensuring quality, versatility and the distinctive sound peculiar to Acustica's plugins.
We hope LIME2 CRAZY88 will meet our most demanding customers' requirements.

In these plugins you'll find the same features as in LIME2 full channel-strip (please see the OPERATION section below for further details). We guarantee you'll get precisely the same sound performance for each module.
For each plugin included in our Lime2 suite, there is a "Standard" version or an alternative "ZL*" version that operates at *zero latency at the cost of extra processing resources and is thus suitable for use when tracking.

For details refer to page 51.
Also for this release, we included an alternative version of the LIME2 channel strip called LIME2 'AL'.
'Adaptive Latency' technology can reduce plugin latency and resource consumption depending on the number of sections used.
EXAMPLE: if you only use the LIME2 channel-strip equalization module, bypassing the compressor section, you will experience a lower latency. Thanks to this NEW plugin approach, you will be able to optimize the use of your resources without sacrificing the distinctive quality of Acustica products.

About the company

Acustica Audio is a leading company specializing in analog hardware virtualization.
Since the birth of Nebula in the summer of 2005, there has been an active collaboration between forward thinking developers, beta testers, audio engineers and equipment samplers from around the world. The research and development has progressed through many stages and employs many innovative processes and technologies as yet unheard of in other products or devices.
The company's goal is to provide the most authentic reproduction of sampled vintage gear and other high-end hardware devices, using the revolutionary technology Vectorial Volterra Kernels Technology (V.V.K.T.) without the negative artifacts created by the current convolution technology.
After many years of fruitful labor, this creative forward thinking group has evolved into a team of experts in knowing what it takes to serve the "best of both worlds" (digital & analog).
Acustica... Audio Renaissance

Control section

Input knob

It sets the input level from -24dB to +24dB and is used to control the plugin's input signal level.

Output knob

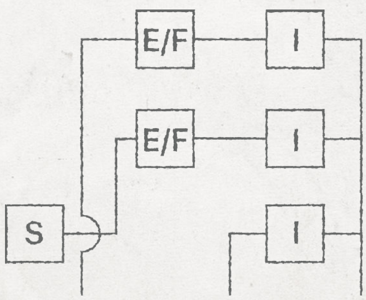
It sets the output level from -24dB to +24dB and is used to control the plugin's output signal level.

Routing

Routing is controlled with 14 buttons.
Each button sets a different block configuration displayed on the "control monitor".
Each block configuration is explained below.
NOTE:
-Lime2 'AL' plugin (unlike the standard version) is based on Dynamic Routing™ technology Dynamic Routing™ technology so we've reduced the number of routing combinations of Lime2 channelstrip but at the same time we improve it in terms of flexibility, responsiveness, and CPU consumption. In other words, this NEW version includes a simplified and optimized dynamic routing with 3 combinations that will make the plug-in react even faster when switching from a routing to the next.

LIME Channel Routing Diagrams

C: Compressor
F: Filter
E: Equalizer
S: Sidechain
I: Selector



Load

It displays the number of NEBULA instances loaded. A higher value indicates higher CPU usage.

Internal / external sidechain

Each compressor may use an external signal, instead of the main input, to control the amount of audio compression.
The different routing configurations allow to choose between the internal (channel n.1 & 2), or the external side chain input (channel n.3 & 4).
We use the term “internal sidechain” to specify that the compressor is fed internally (it means it has an internal routing) when the input signal is at the same time the control signal.
The “external sidechain” derives the control signal from another source (entering the plugin, but not necessarily the compressor, as it can be filtered or EQed before the dynamic section). This control signal then goes through the sidechain channel and sets the amount of the gain reduction.

Explaination of the Sidechain behaviour in different plugin formats:
VST2: Not recommended for Steinberg host (Cubase and Nuendo). Each sequencer has a sidechain connection method that could be different one from another, please check the relative daw’s operating manuals for details.
VST3: Recommended for Steinberg host as (Cubase and Nuendo). Each sequencer has a sidechain connection method that could be different one from another, please check the relative daw’s operating manuals for details.
AAX: the external sidechain routing controls a key input parameter to selectively modify the sidechain in PT. Audio channels (1-2), Sidechain channels (3-4).

NOTE: The sidechain are mono signal in Pro Tools

Some hosts do not support the channels 3 - 4 to avoid any crash: you can set the TAG: LEGACYSTEREOCHANNEL in the plugin's XML Master which allows the old hosts to correctly transmit the requested channels (when the tag is set at 0 = 2 channels, when it's set at 1 = 4 channels). Users may contact our technical support for more details.

AUDIO UNIT (AU):

In Logic Pro X you may not see the Side Chain input at the top of the Plug-in window. To display the Side Chain control you need to enable Show Advanced Tools in the Logic Pro X preference panel.

The external sidechain is not yet supported in AUDIO UNIT (AU). It will be implemented in the future.

INTERNAL/EXTERNAL SIDECHAIN

Basically a sidechain compressor uses another signal other than the main input, the first one going through a sidechain channel, to control the amount of audio compression.

The dynamic compressor has two type of channels, the audio channels (1-2 input) and the sidechain channels(3-4 input). The routing control configurations allow to choose between the internal, or the external side chain input.

We use the term "internal sidechain" to specify that the compressor is fed internally (it means it has an internal routing) and the signal comes from the plugin itself.

It is called "external sidechain" when the input (entering the plugin and not necessarily the compressor, as it can be filtered or EQed before the dynamic section) goes through the sidechain channels (not the plugin's channels), the signal sent to the sidechain is then the "control signal"..


ACUSTICA
LIME²
ON A B
FREQ
BW
GAIN
HMF
HF
LF
LMF
1 B 3
4 5 6
7 8 9
10 11 12
13 14 OFF
PROG: 1
LOAD: 2 INTERNAL
LINE A
MIC A
MIX A ST
LINE B
MIC B
COMP B ST
MIC F
MIX F ST
OFF
INPUT
COMP
OUTPUT
DRY
WET
FILTERS
ATTACK
RELEASE
SHMOD
THRESHOLD
RATIO
KNEE
MAKE UP


ON A B
FREQ
BW
GAIN
HMF
HF
LF
LMF
1 B 3
4 5 6
7 8 9
10 11 12
13 14 OFF
PROG: 1
LOAD: 2 INTERNAL
LINE A
MIC A
MIX A ST
LINE B
MIC B
COMP B ST
MIC F
MIX F ST
OFF
INPUT
COMP
OUTPUT
DRY
WET
FILTERS
ON
COMP
ATTACK
FAST
RELEASE
SHMOD
THRESHOLD
RATIO
KNEE
MAKE UP
A
B
C
D
E
ACUSTICA
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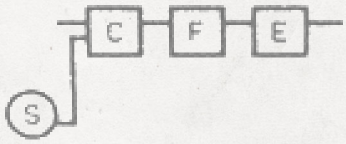
Routing

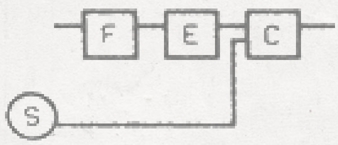
Button 1:
input 1+2 > compressor
This routing is included in LIME2 AL version

Button 2:
input 1+2 > compressor input (audio signal)
input 3+4 > compressor sidechain (control signal)



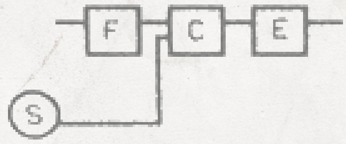
Button 3:
input 1+2 > filter > equalizer > compressor input
This routing is included in LIME2 AL version

Button 4:
input 1+2 > filter > equalizer > compressor input (audio signal)
input 3+4 > compressor sidechain (control signal)

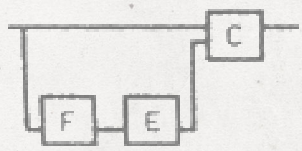


Button 5:
input 1+2 > filter > compressor
This routing is included in LIME2 AL version

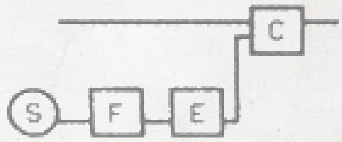
Button 6:
input 1+2 > filter > compressor input (audio signal)
input 3+4 > compressor sidechain (control signal)



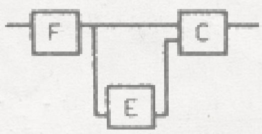
Button 7:
input 1+2 > compressor input (audio signal)
input 1+2 > filter > equalizer > compressor sidechain (control signal)



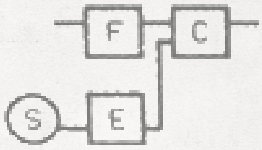
Button 8:
input 1+2 > compressor input (audio signal)
input 3+4 > filter > equalizer > compressor sidechain (control signal)



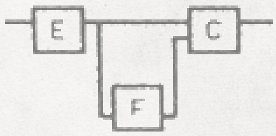
Button 9:
input 1+2 > filter > compressor input (audio signal)
input 1+2 > filter > equalizer > compressor sidechain (control signal)



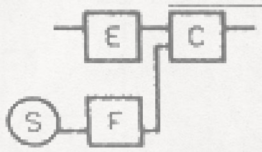
Button 10:
input 1+2 > filter > compressor input
(audio signal)
input 3+4 > equalizer > compressor sidechain
(control signal)



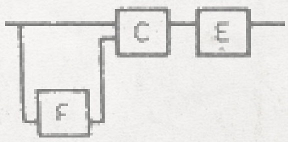
Button 11:
input 1+2 > equalizer > compressor input
(audio signal)
input 1+2 > filter > compressor sidechain
(control signal)



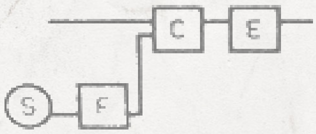
Button 12:
input 1+2 > equalizer > compressor input
(audio signal)
input 3+4 > filter > compressor sidechain
(control signal)



Button 13:
input 1+2 > compressor input (audio signal)
input 1+2 > filter > compressor sidechain
(control signal)



Button 14:
input 1+2 > compressor input (audio signal)
input 3+4 > filter > compressor sidechain (control signal)



Dry / Wet

The DRY/WET control determines the mix proportion between the original (dry) and 'effected' (wet) signals.

This is a very powerful and simple to use feature. It significantly changes the compression character (parallel or sidechain compression), and it is strictly connected to the signal routing control.

The right routing configuration and the correct use of the DRY/WET control are essential in order to achieve radically different sound qualities.

Preamp section

Lime2's preamps are based on a collection of various hardware units; they emulate the phase and the harmonic and frequency response of their corresponding circuits.
This carefully chosen combination of preamps is intended to provide the user with a virtual console emulation with many flexible options. The descriptions provided below should only be taken as a guide, not as rule. The best use of these tone colours can only be evaluated by the user on a case per case basis; the engineer's final judgement should always depend on his/her own ears and imagination only.

NOTE:
-Device A is inspired by a legendary 80s British console.
-Device B is inspired by a modern British flagship console.
-Device C is the new entry of this suite, it is a british console built by the same renowned manufacturer of the previous units and defined as the culmination of 20 years of professional audio equipment design and production by one of the most celebrate pro gear genius and pioneer. This desk was developed in the early 1980s and was the first fully IC-based console by this famous U.K. company;

This module equipped by 4 different mutually exclusive preamps banks allows you to switch between them and an OFF button to disable this stage. We are very proud to provide to our customers a complete virtual console emulation that reproduces a lrge number of channel pre-amps of some iconic desks.

In order to select a specific bank, just press the relevant button. As expected, each button is mutually exclusive; as a consequence, only one preamp emulation at a time can be activated.
You can choose between: LINE, MIC, BUS or MISC.
Each preamp bank contains different emulations, MONO or STEREO depending on the bank chosen, details below.

- You can select the preamp emulation for each bank using the relative stepped knob (labelled SELECTION).

1- LINE bank - MONO PREAMPS

This group of emulations consists of all the new LINE preamps derived from the sampling of a vintage console from which we derived also the C model of equalizer.

1- LINE 01 MONO
2- LINE 02 MONO
3- LINE 03 MONO
4- LINE 04 MONO
5- LINE 05 MONO
6- LINE 06 MONO

2 - MIC bank - MONO PREAMPS

This group of emulations consists of all the new LINE preamps derived from the sampling of a vintage console from which we derived also the C model of equalizer.

1- MIC 01 MONO
2- MIC 02 MONO
3- MIC 03 MONO
4- MIC 04 MONO
5- MIC 05 MONO
6- MIC 06 MONO
7- MIC 05 MONO
8- MIC 06 MONO

3. BUS bank - STEREO PREAMPS

This group of preamplifiers consists of all the STEREO preamps (1-8) included in the previous/first version of lime EQ plus new STEREO preamp emulations (9-12) derived from the sampling of a vintage console from which we derived also the C model of equalizer.

1- MIX A ST
2- COMP C ST
3- MIX F ST
4- COMP B ST
5- BUS B ST
6- MIX B ST
7- LIME ST
8- RETRO ST

9- LINE 01 ST
10- LINE 02 ST
11- MIC 01 ST
12- MIC 02 ST

4. MISC bank - MONO PREAMPS

This group of preamplifiers consists of all the MONO preamps included in the previous/first version of lime Equalizer standalone plugin (LIME EQ).

1- LINE A
2- MIC A
3- COMP A
4- MIX A
5- MIC F
6- LINE B
7- MIC B
8- DI
9- SLATE

NOTE: Lime2 Channelstrip includes only 8 preamps, already mentioned above, in order:
-LINE A
-MIC A
-MIX A ST
-LINE B
-MIC B
-COMP B ST
-MIC F
-MIX F ST

Equalizer section

"ON" button

The ON buttons activate each section of the EQ. When illuminated, these sections are enabled, when the leds are off the sections are bypassed.

A mode

Details
-Device A is inspired by a legendary 80s British console.

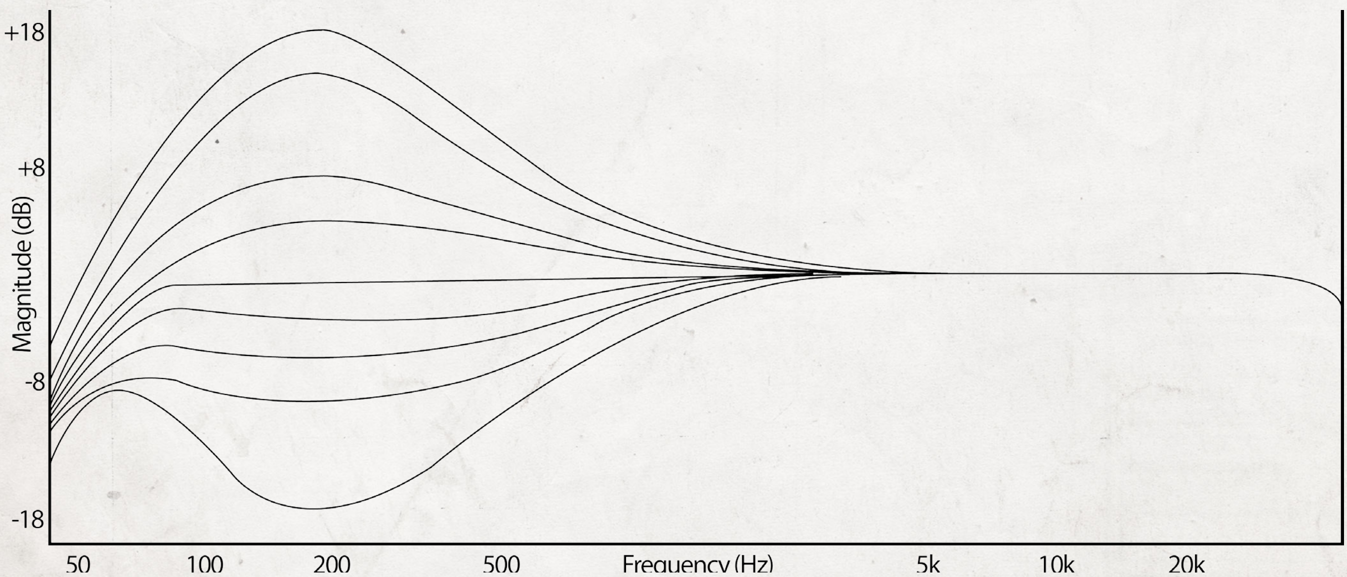
Low frequency band

The LF Hz knob sets the low frequencies between 33Hz to 370Hz in 21 steps.

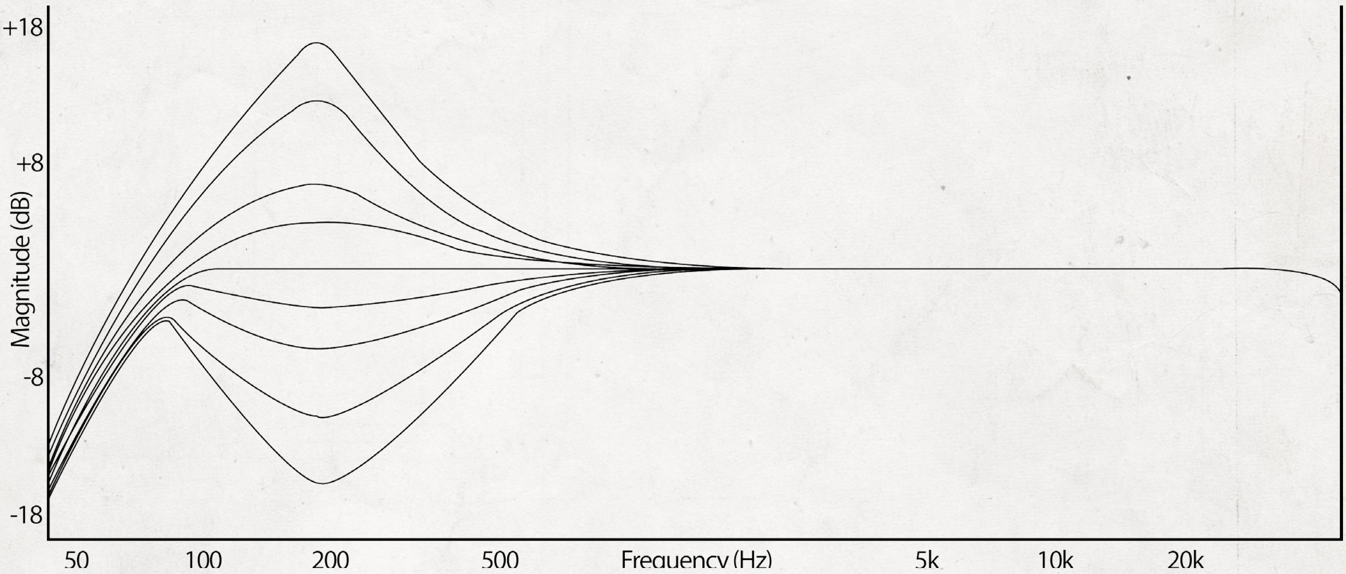
The LF Gain knob sets the boost or cut for this band with a range of +/-18dB.

The LF Q knob switches the Low Frequency band between Peaking and Shelving modes.
This knob allows gradual adjustments from Peak (Low Q) , Peak (High Q) to Shelving mode.
Low Q is 0.7 and High Q is 2.0.

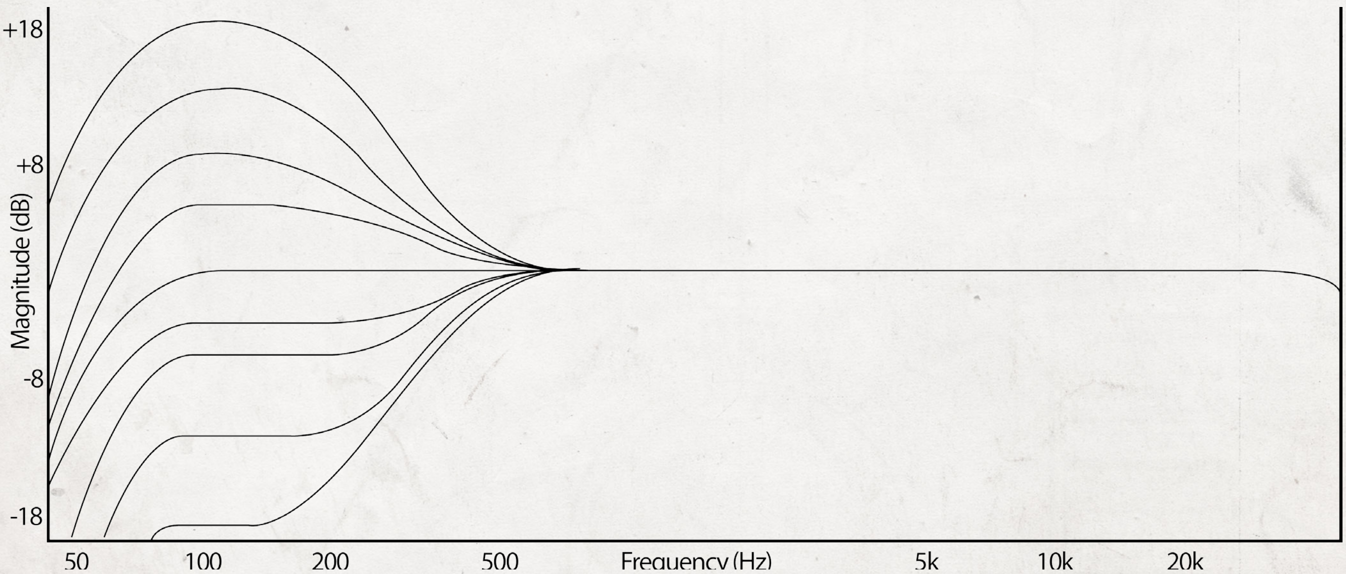
Eq A - LF - 33 Hz - broad peak Q



Eq A - LF - 33 Hz - narrow peak Q



Eq A - LF - 33 Hz - shelf Q



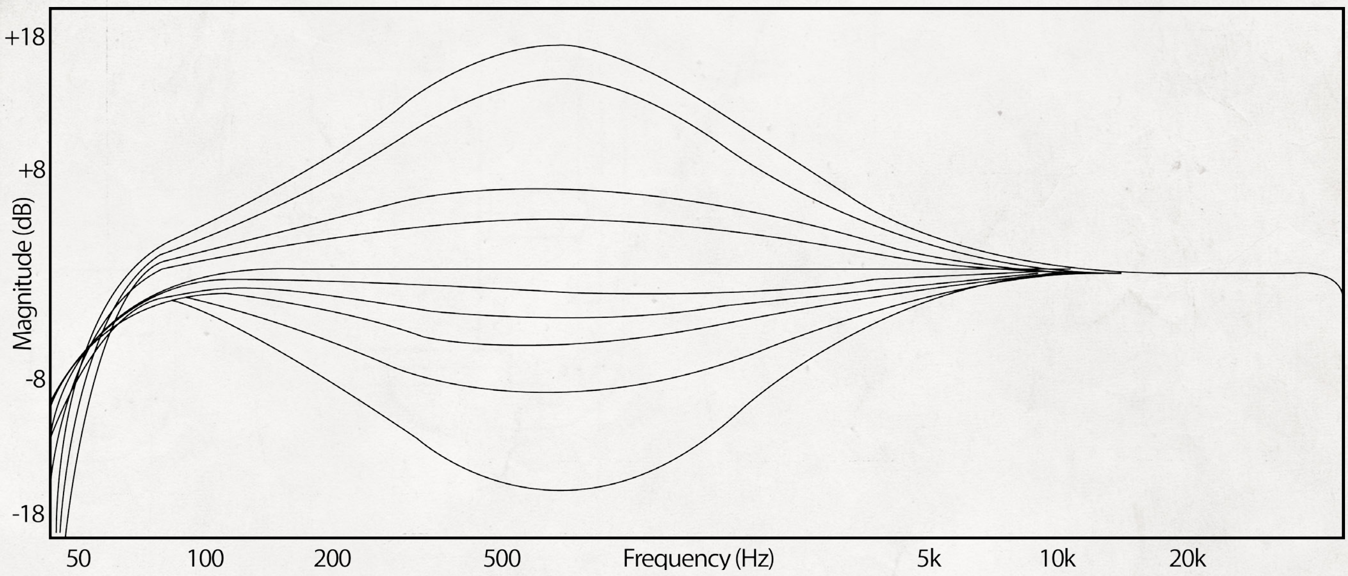
Low Mid Frequency Band

The LMF Hz knob sets the low frequencies between 190Hz to 2kHz in 21 steps.

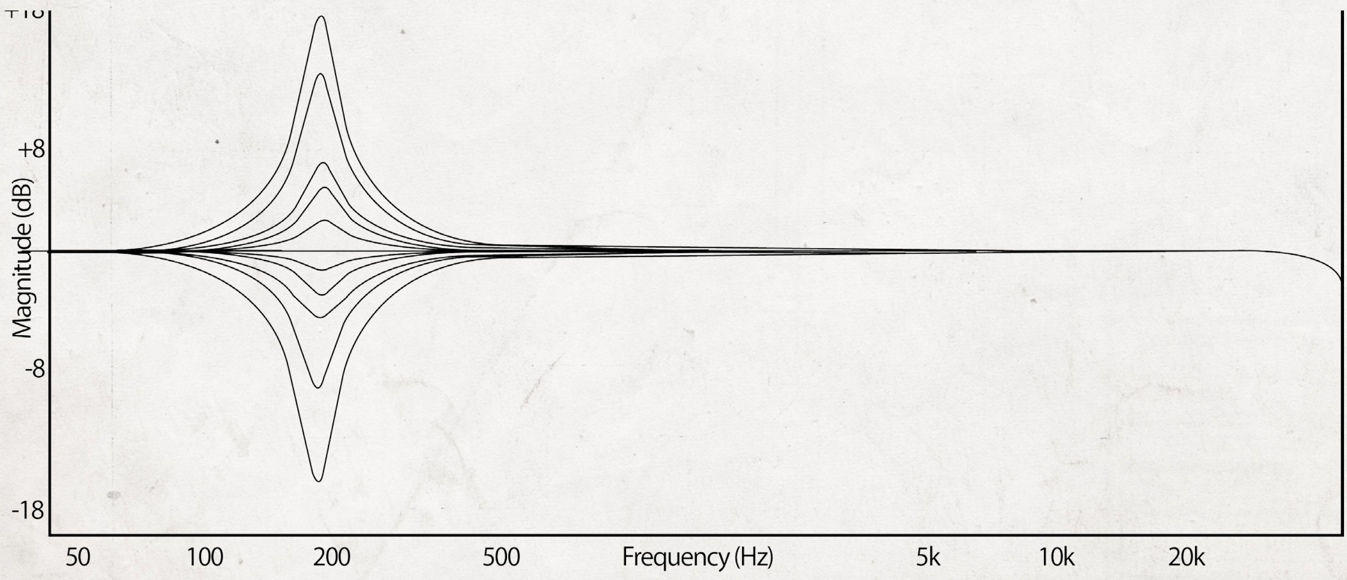
The LMF Gain knob sets the boost or cut for this band with a range of +/-18dB.

The Low Mid Q knob adjusts the continuously Q between 0.4 and 10. Q is variable.

Eq A - LMF - 190 kHz - broad peak Q



Eq A - LMF - 190 kHz - narrow peak Q



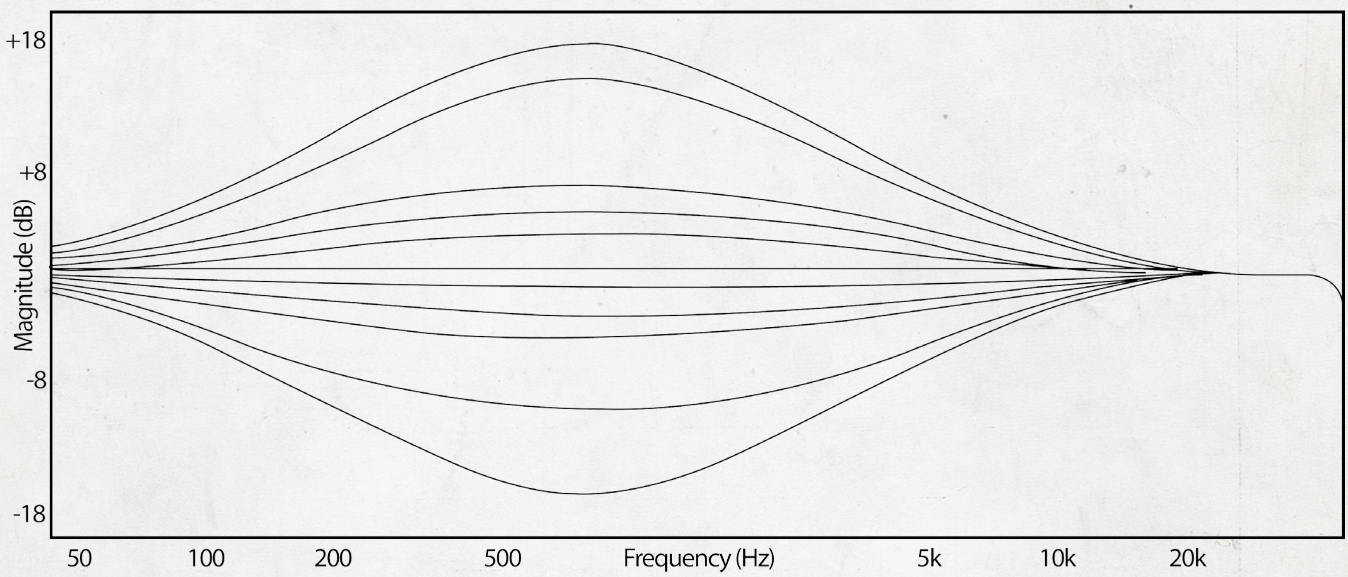
High Mid Frequency Band

The HMF Hz knob sets the low frequencies between 0.8kHz to 9kHz in 21 steps.

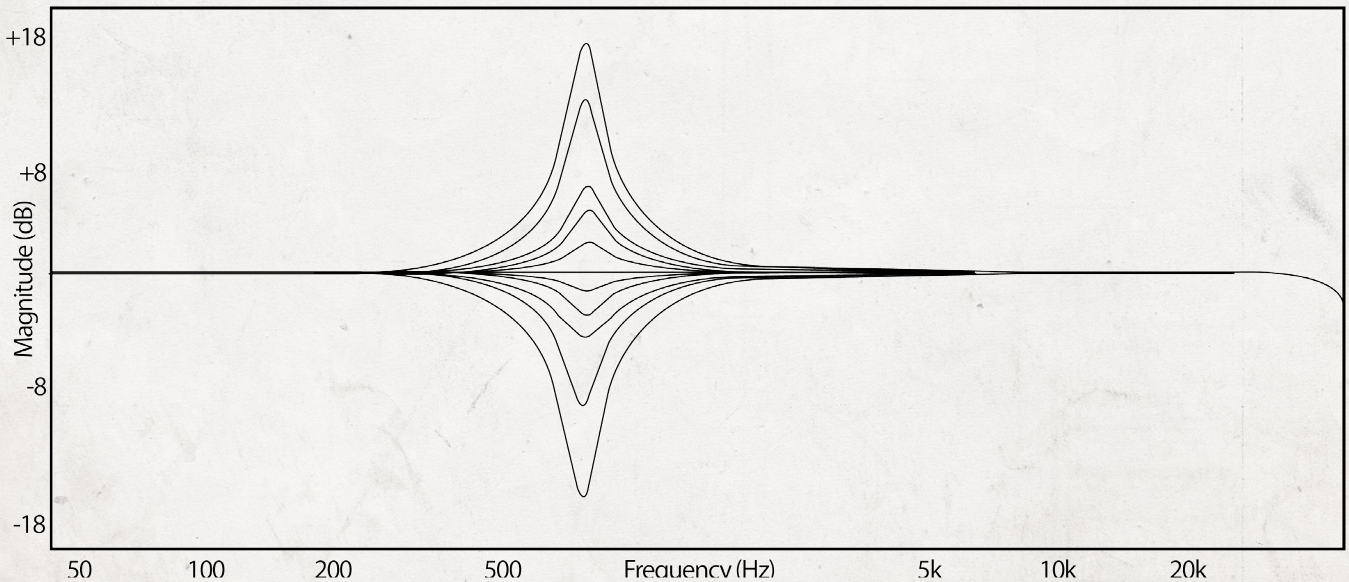
The HMF Gain knob sets the boost or cut for this band with a range of +/-18dB.

The High Mid Q knob adjusts the continuously Q between 0.4 and 10. Q is variable.

Eq A - HMF - 0.8 kHz - broad peak Q



Eq A - HMF - 0.8 kHz - narrow peak Q



High Frequency Band

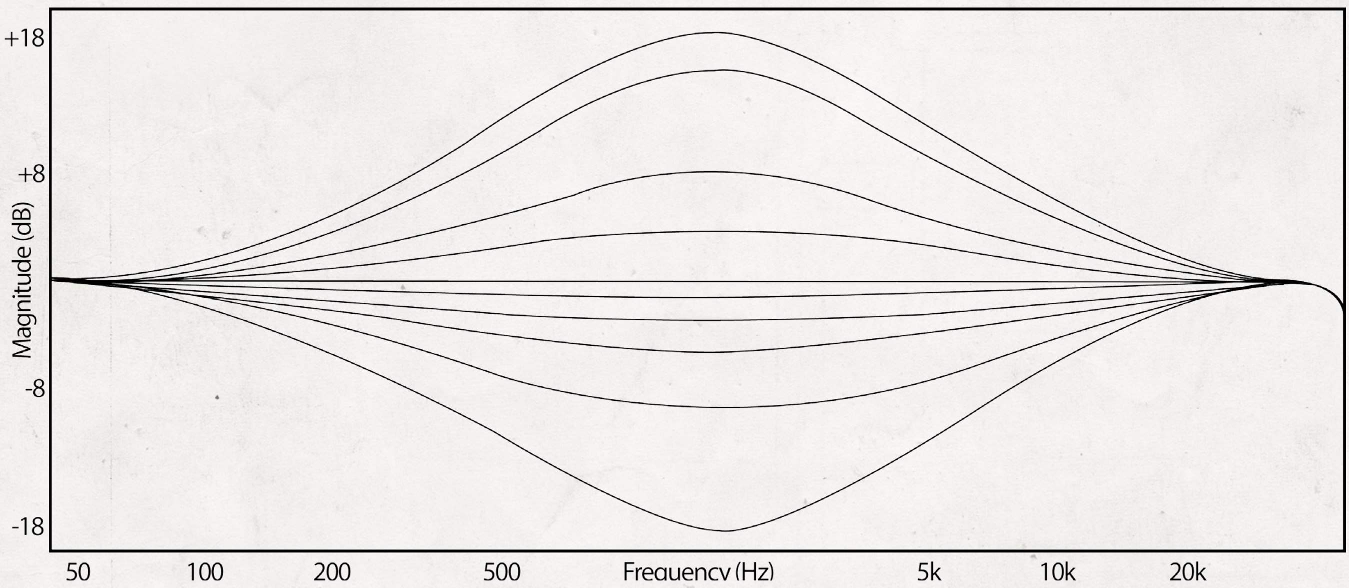
The HF kHz knob sets the low frequencies between 1.5kHz to 18kHz in 21 steps.

The HF Gain knob sets the boost or cut for this band with a range of +/-18dB.

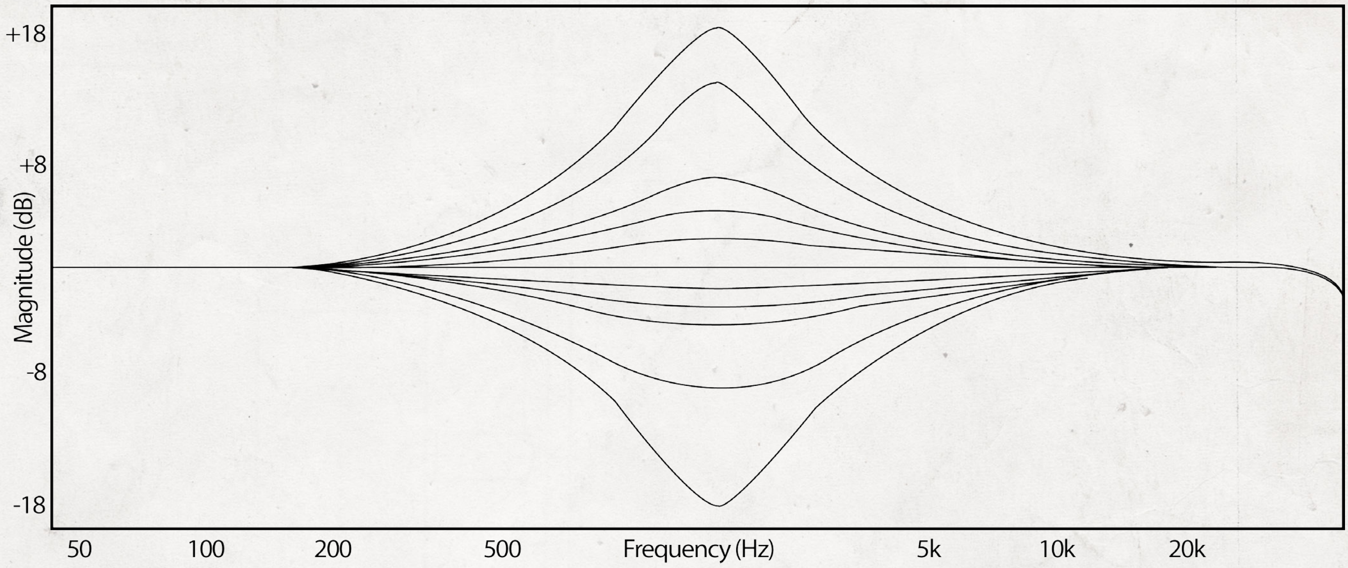
The HF Q knob switches the Low Frequency band between Peaking and Shelving modes.

This knob allows gradual adjustments from Peak (Low Q) , Peak (High Q) to Shelving mode .
Low Q is 0.7 and High Q is 2.0.

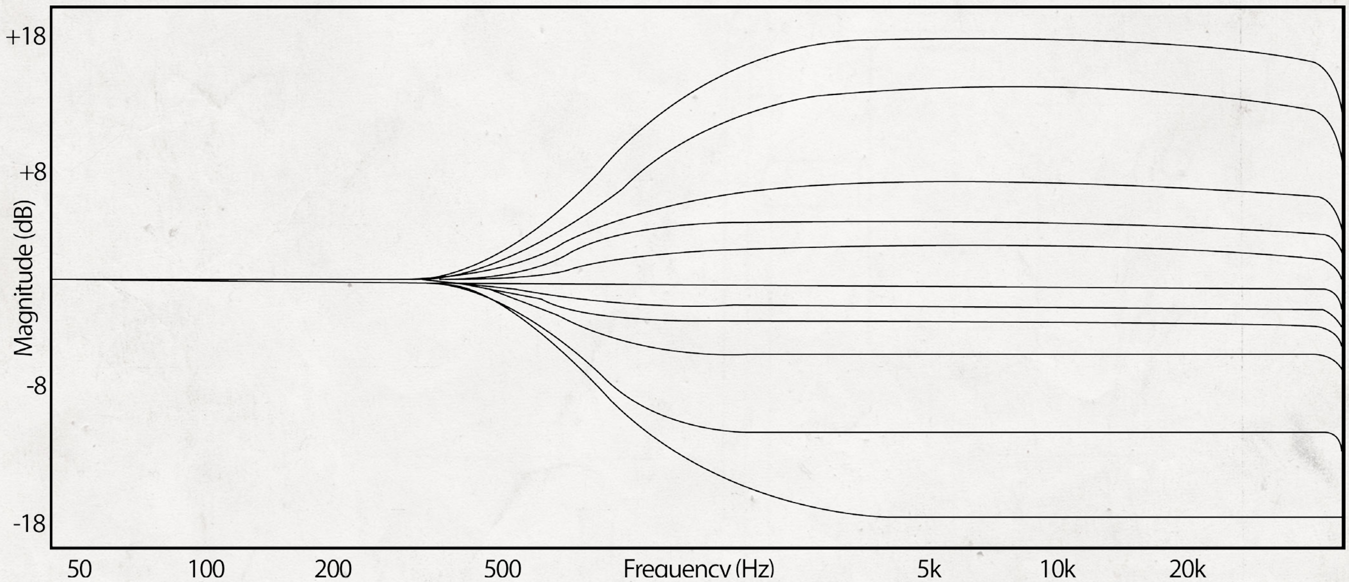
Eq A - HF - 1.5 kHz - broad peak Q



Eq A - HF - 1.5 kHz - narrow peak Q



Eq A - HF - 1.5 kHz - shelf Q



B mode

Details
-Device B is inspired by a modern British flagship console.

Low frequency band

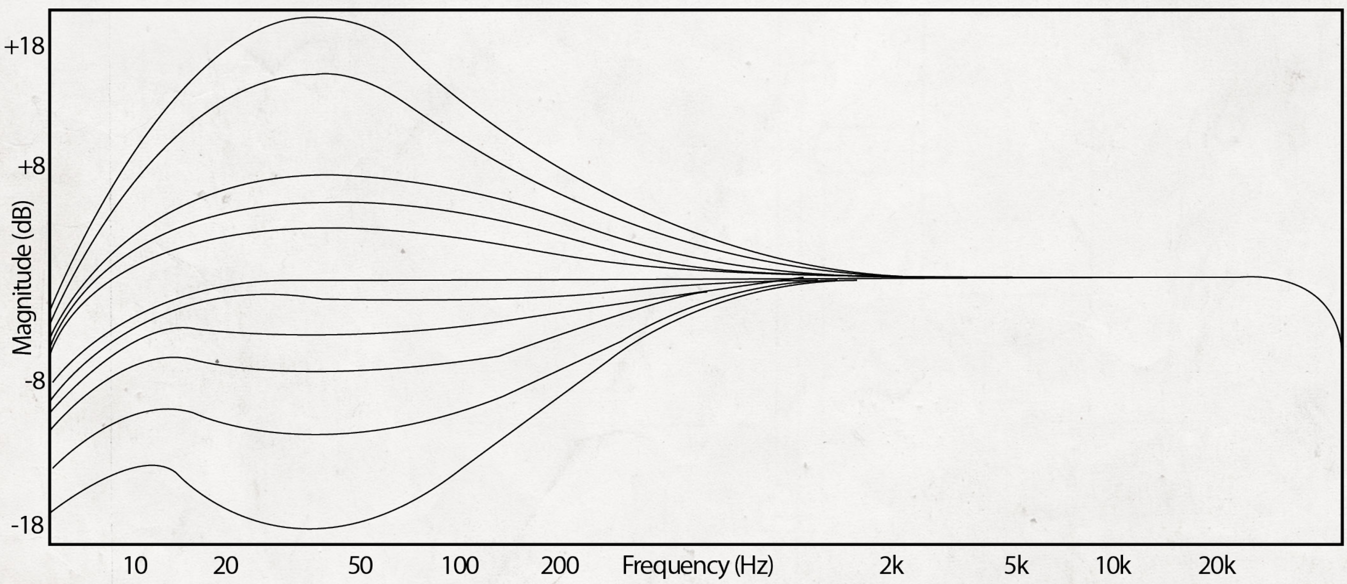
The LF Hz knob sets the low frequencies between 33Hz to 440Hz in 21 steps.

The LF Gain knob sets the boost or cut for this band with a range of +/-18dB

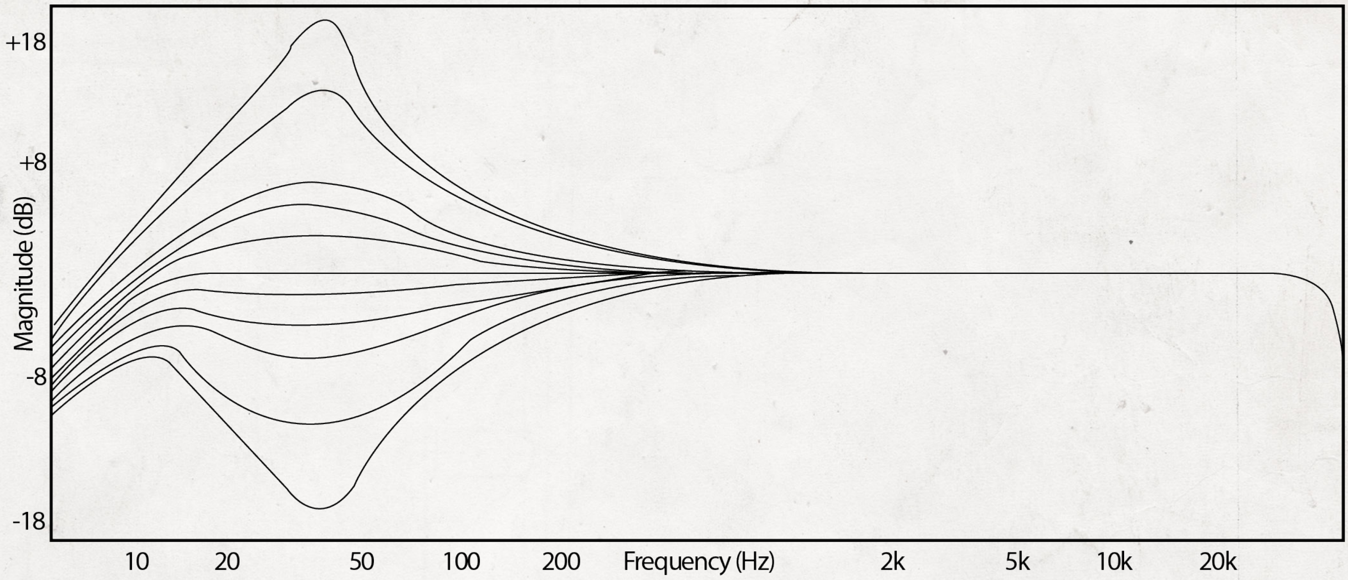
The LF Q knob switches the Low Frequency band between Peaking and Shelving modes.

This knob allows gradual adjustments from Peak (Low Q) , Peak (High Q) to Shelving mode.
Low Q is 0.7 and High Q is 2.0.

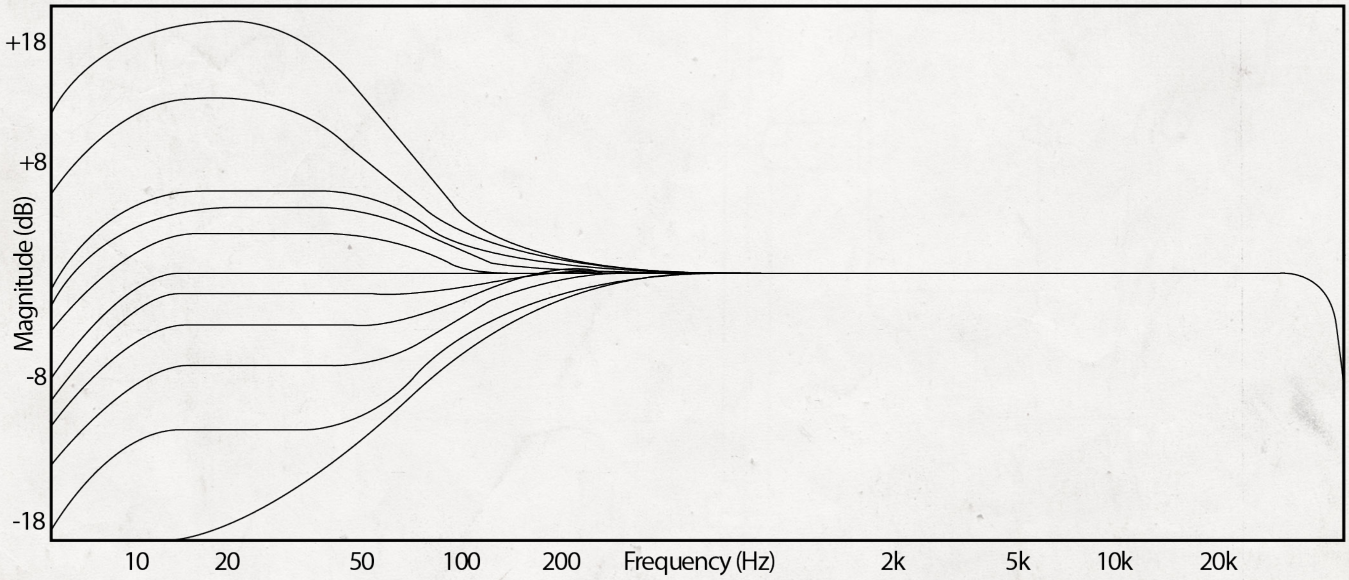
Eq B - LF - 33 Hz - broad peak Q



Eq B - LF - 33 Hz - narrow peak Q



Eq B - LF - 33 Hz - shelf Q



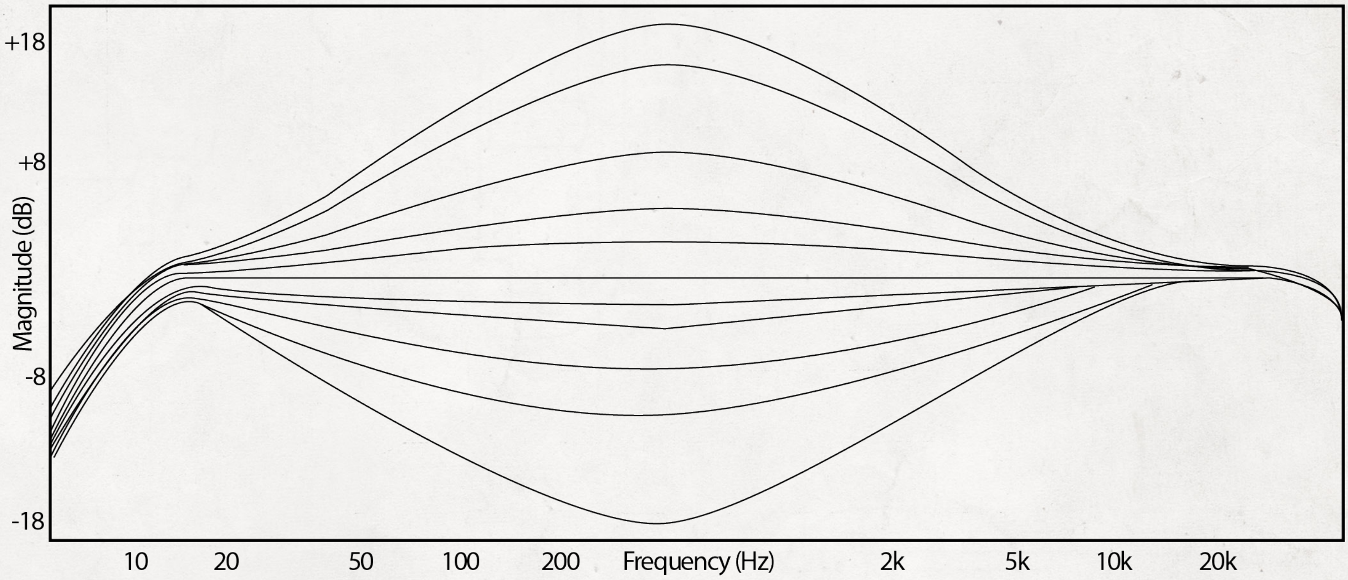
Low Mid Frequency Band

The LMF Hz knob sets the low frequencies between 120Hz to 1.95kHz in 21 steps.

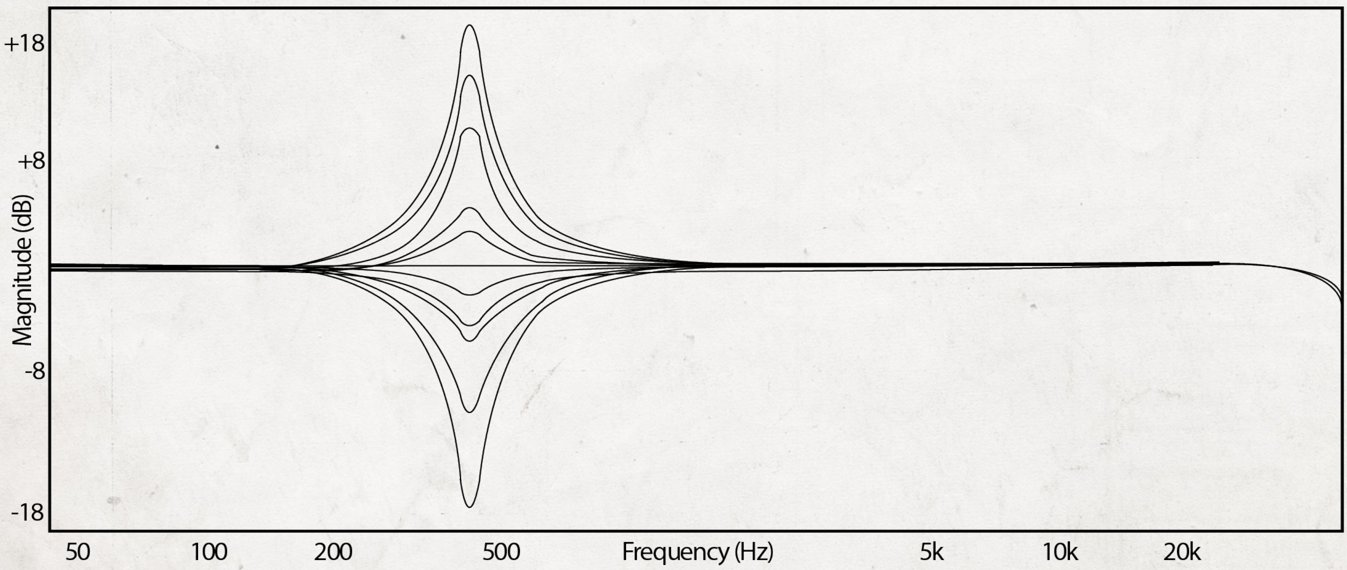
The LMF Gain knob sets the boost or cut for this band with a range of +/-18dB.

The Low Mid Q knob adjusts the continuously Q between 0.4 and 10. Q is variable.

Eq B - LMF - 420 kHz - broad peak Q



Eq B - LMF - 420 kHz - narrow peak Q



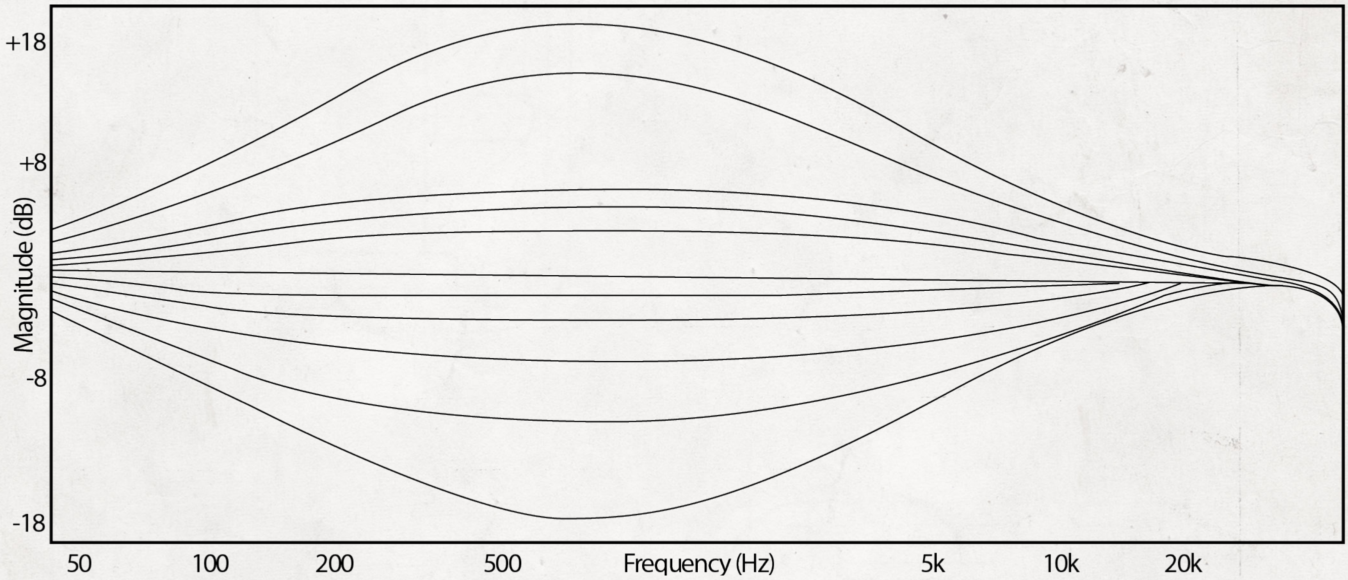
High Mid Frequency Band

The HMF Hz knob sets the low frequencies between 0.8kHz to 9.5kHz in 21 steps.

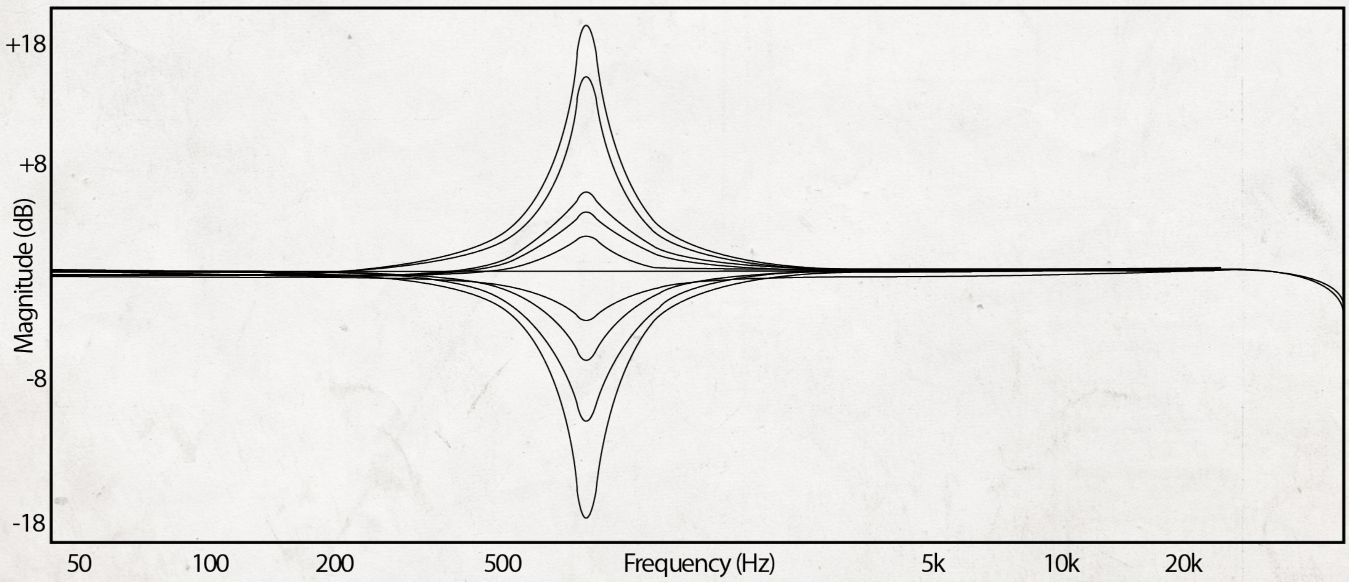
The HMF Gain knob sets the boost or cut for this band with a range of +/-18dB.

The High Mid Q knob adjusts the continuously Q between 0.4 and 10. Q is variable.

Eq B - HMF - 0.6 kHz - broad peak Q



Eq B - HMF - 0.6 kHz - narrow peak Q



High Frequency Band

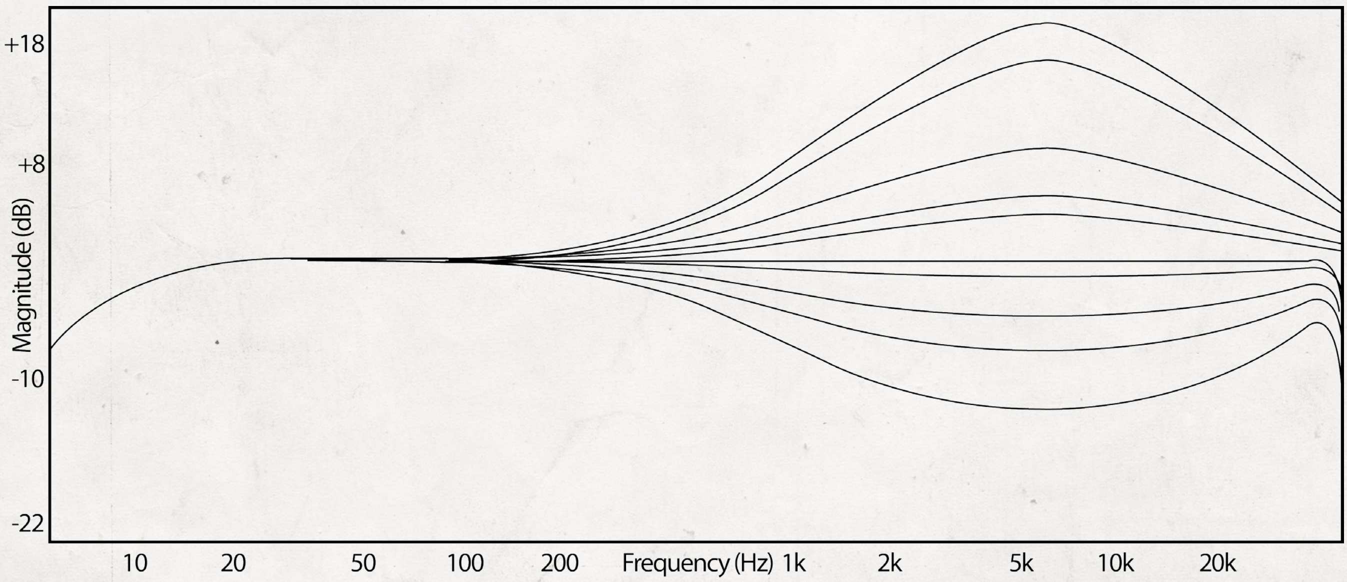
The HF kHz knob sets the low frequencies between 1.6kHz to 19kHz in 21 steps.

The HF Gain knob sets the boost or cut for this band with a range of +/-18dB.

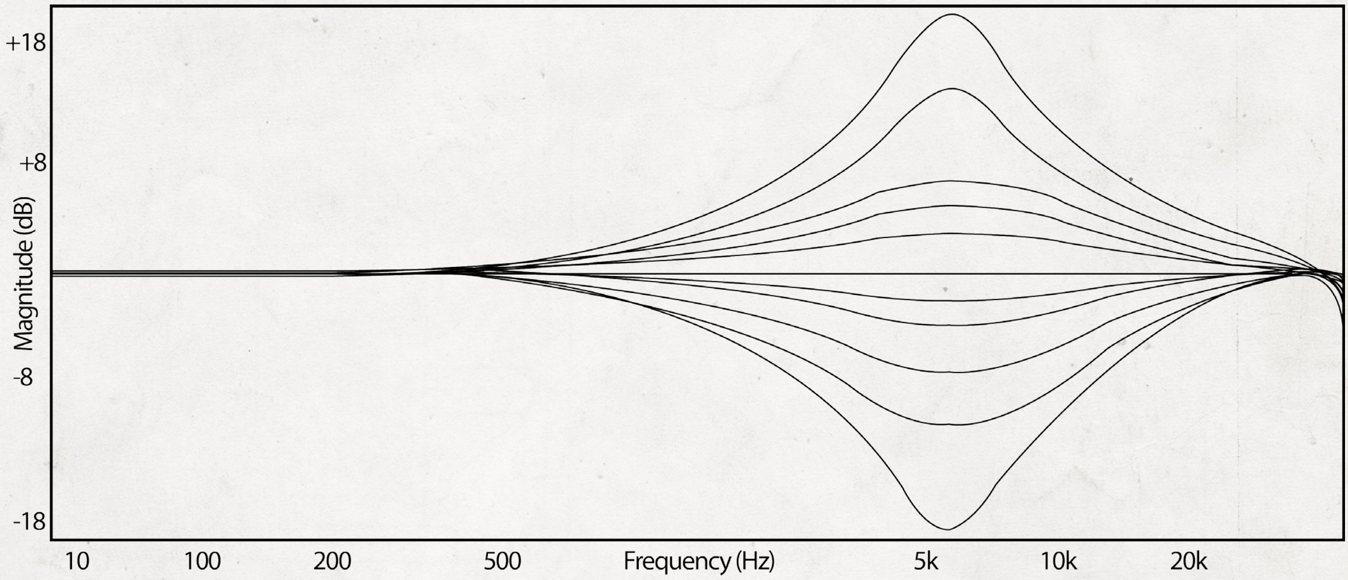
The HF Q knob switches the Low Frequency band between Peaking and Shelving modes.

This knob allows gradual adjustments from Peak (Low Q) , Peak (High Q) to Shelving mode .
Low Q is 0.7 and High Q is 2.0.

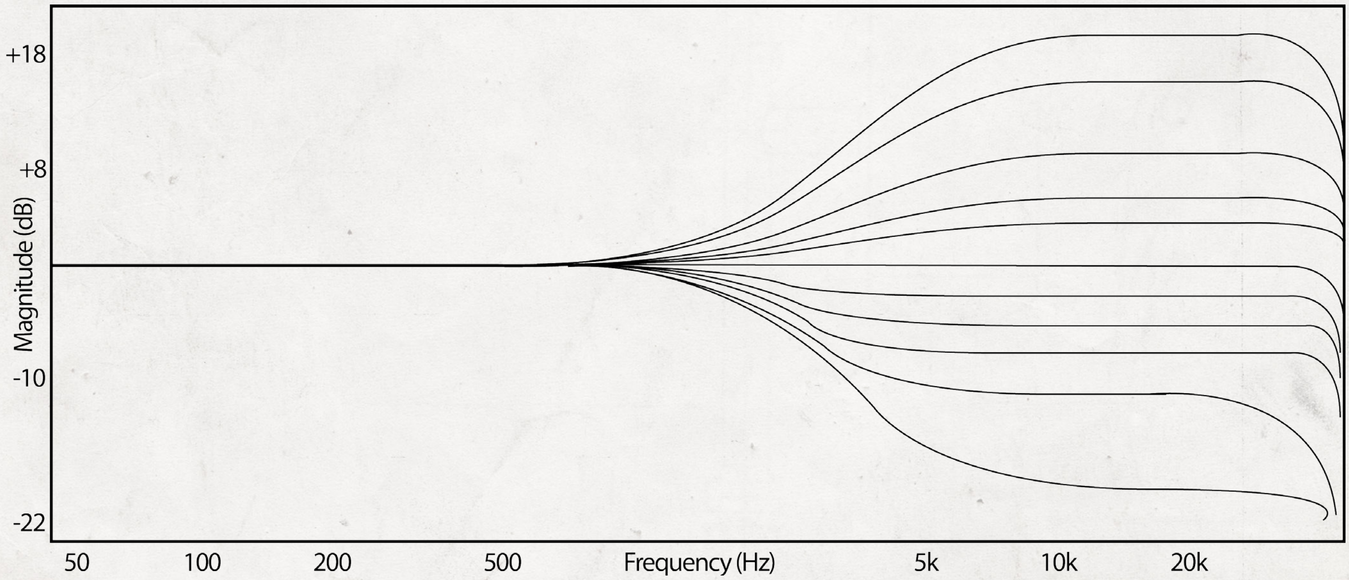
Eq B - HF - 5.0 kHz - broad peak Q



Eq B - HF - 5.0 kHz - narrow peak Q



Eq B - HF - 5.0 kHz - shelf Q



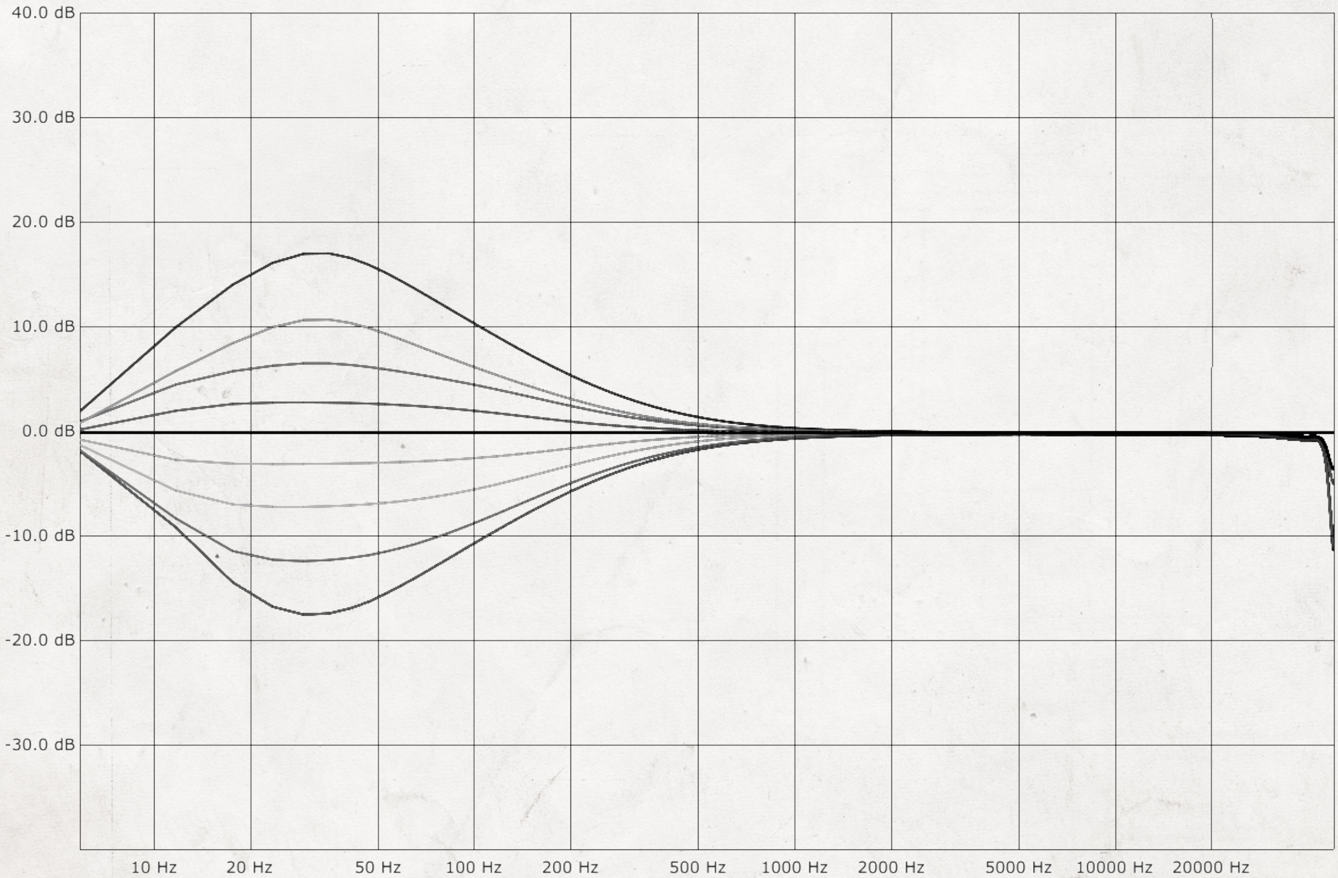
C mode

Details:
Device C inspired by a british console developed in the early 1980s, it was the first fully IC-based console by this famous U.K. company;

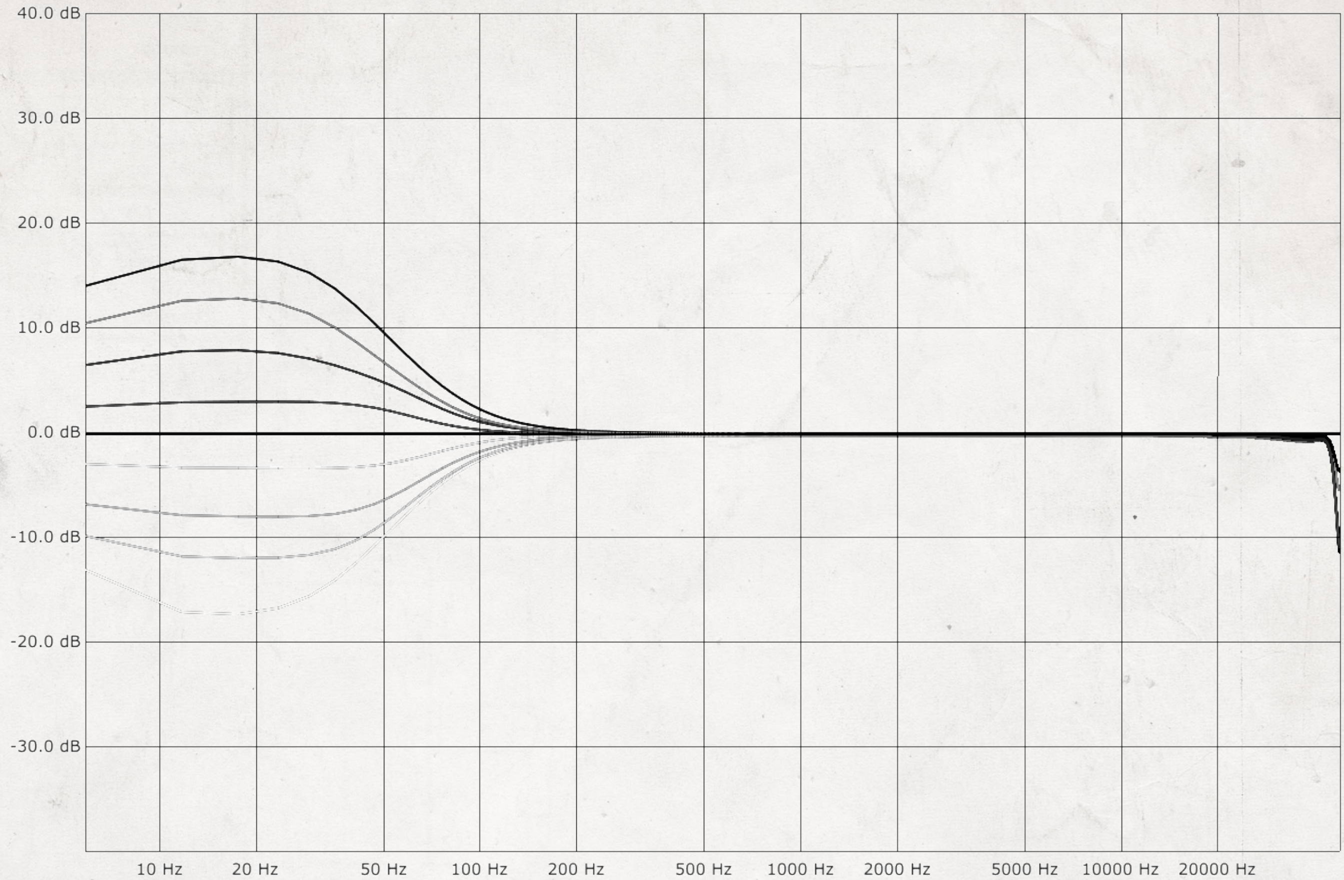
Low frequency band

The LF Hz knob sets the low frequencies between 31.5 Hz to 315 Hz.
The LF Gain knob sets the boost or cut for this band with a range of +/-18dB.

The LF Q knob switches the Low Frequency band between Peaking and Shelving modes.
This knob allows gradual adjustments from Peak (Low Q) , Peak (High Q) to Shelving mode.



Eq C - LF - 31.5 Hz - peak Q



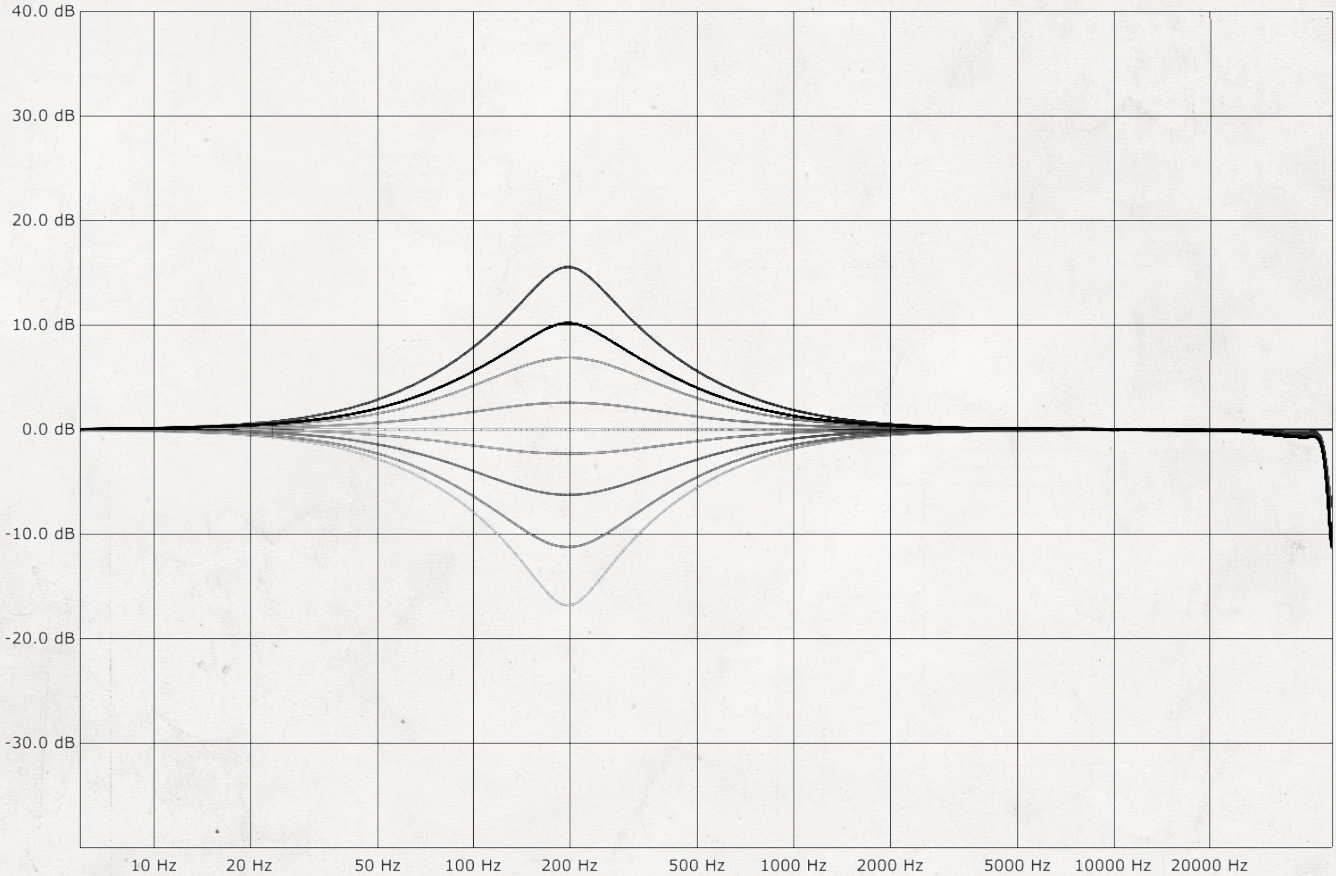
Eq C - LF - 31.5 Hz - shelf Q

Low Mid Frequency Band

The LMF Hz knob sets the low frequencies between

200 Hz to 2kHz.

The LMF Gain knob sets the boost or cut for this band with a range of +/-18dB.
The Q is fixed (Peaking mode)



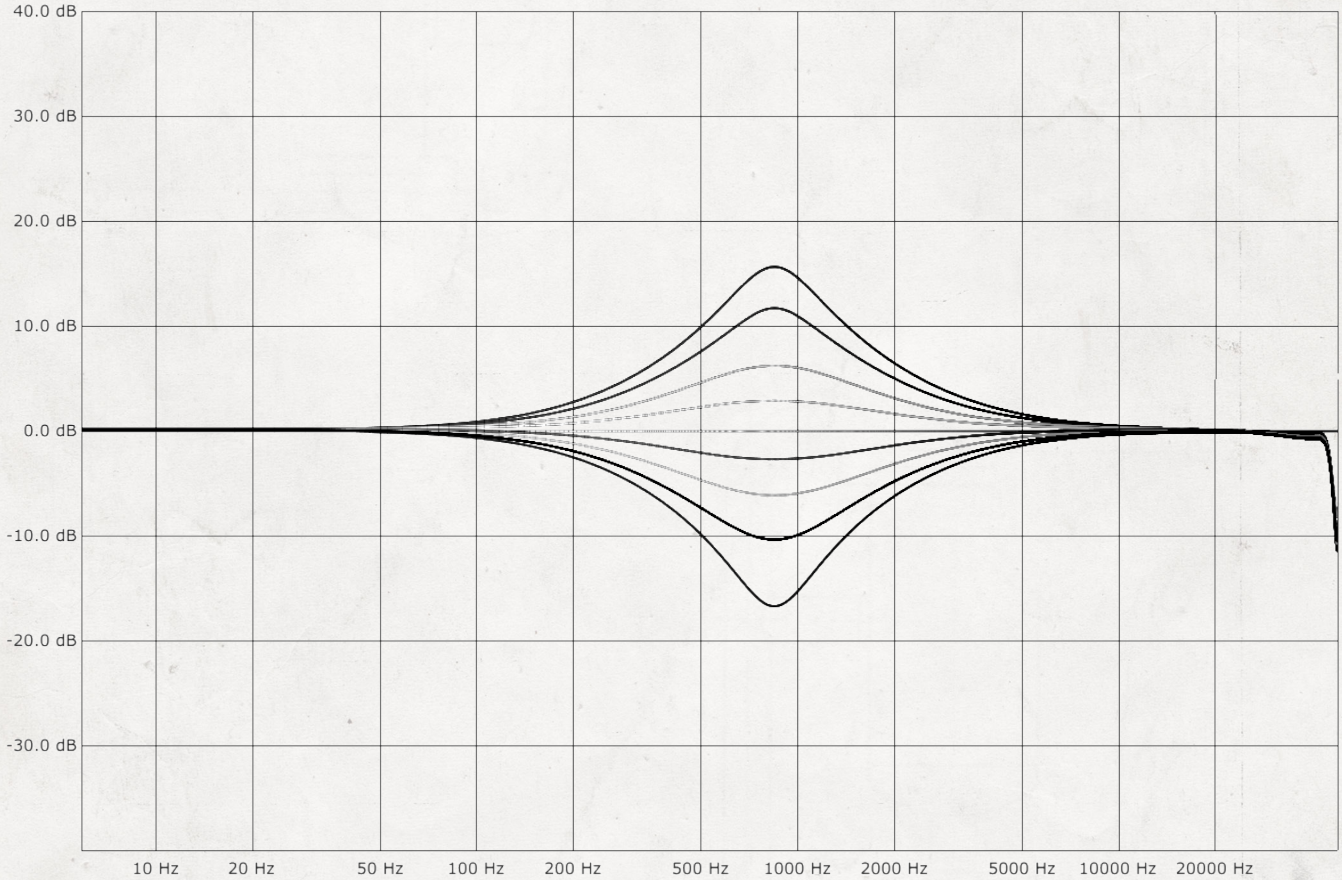
Eq C - LMF - 200 Hz - peak Q

High Mid Frequency Band

The HMF Hz knob sets the low frequencies between

0.8kHz to 8kHz.

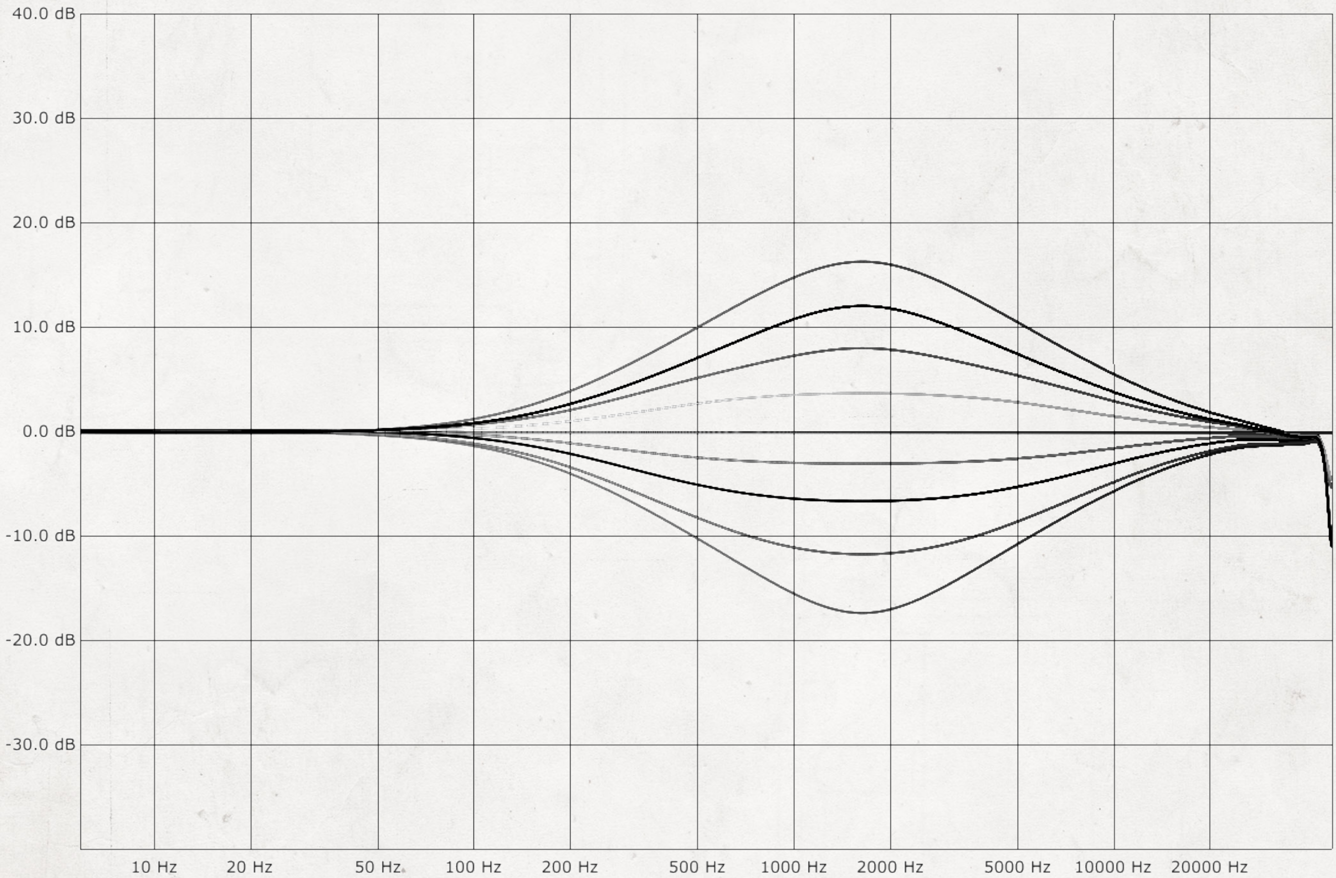
The HMF Gain knob sets the boost or cut for this band with a range of +/-18dB.
The Q is fixed (Peaking mode)



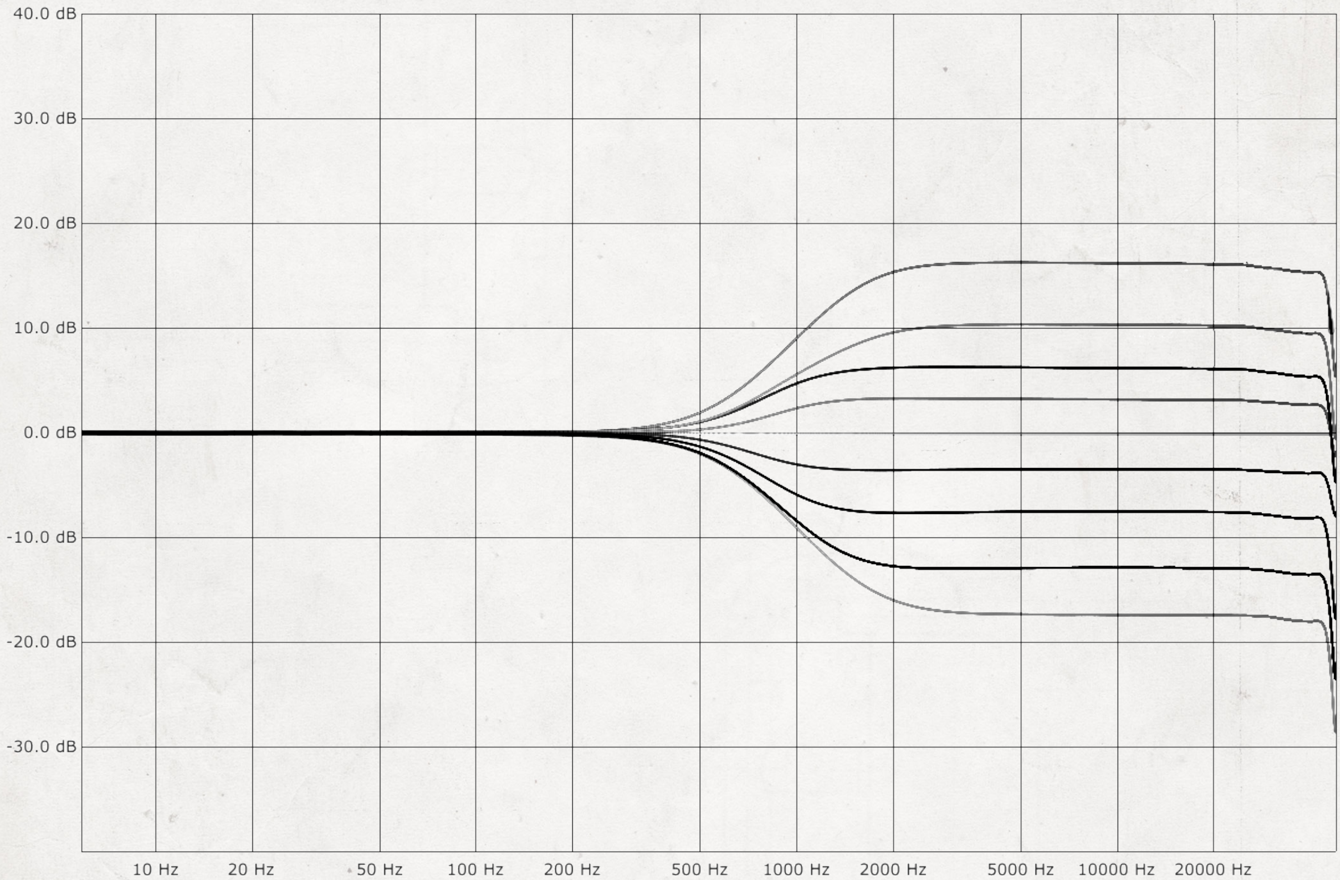
Eq C - HMF - 0.8 kHz - peak Q

High Frequency Band

The HF kHz knob sets the low frequencies between 1.6 kHz to 16 kHz.
The HF Gain knob sets the boost or cut for this band with a range of +/-18dB.
The HF Q knob switches the Low Frequency band between Peaking and Shelving modes.
This knob allows gradual adjustments from Peak (Low Q) , Peak (High Q) to Shelving mode .



Eq C - HF - 1.6 kHz - peak Q



Eq C - HF - 1.6 kHz - shelf Q


GAIN
0
dB
+
FILTERS
80
150
50
240
315
Hz
INPUT
6
0
12
18
ON
A
B
C
GAIN
0
dB
+
10
9
13
8
7.5
OUTPUT
6
0
6
12
18
24

Filters section

"ON" button

The ON buttons activate both Filter sections. When illuminated, these sections are enabled, when the leds are off the sections are bypassed.

HP filter

Hp section includes 2 different highpass filters, A and B mode. They are derived from 2 different devices.

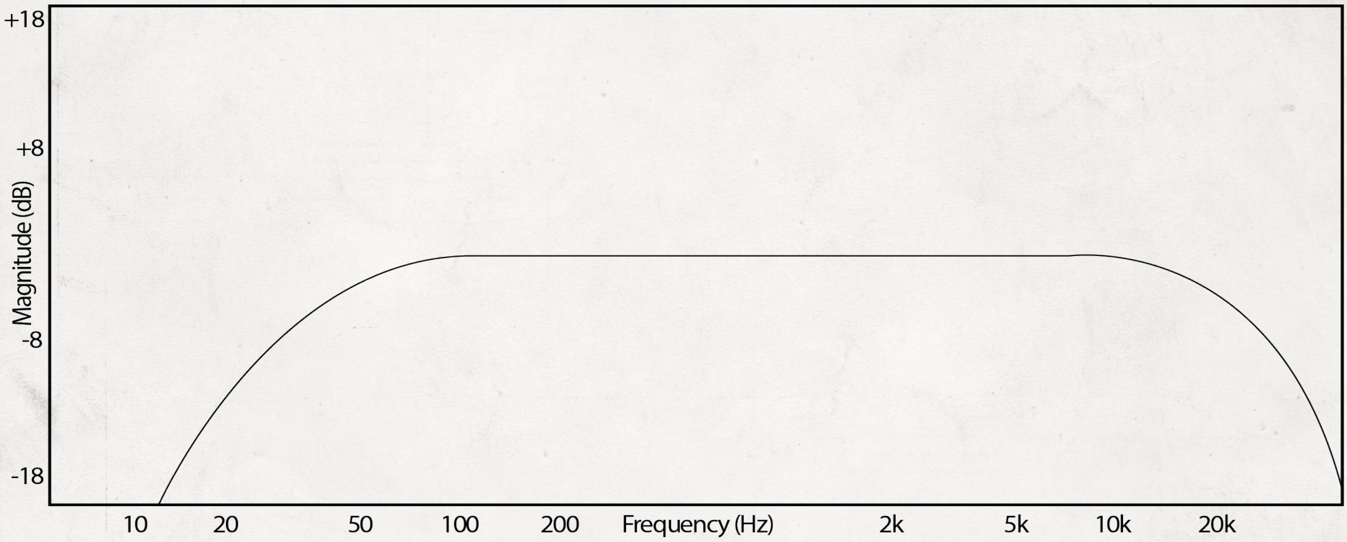
- A mode
The high-pass filter has a 12dB per octave (20dB per decade) slope and a frequency range from 31.5Hz to 315Hz.
- B mode
The high-pass filter has a 12dB per octave (20dB per decade) slope and a frequency range from 30Hz to 300Hz.
- C mode
The high-pass filter has a frequency range from 31.5 Hz to 315Hz.

LP filter

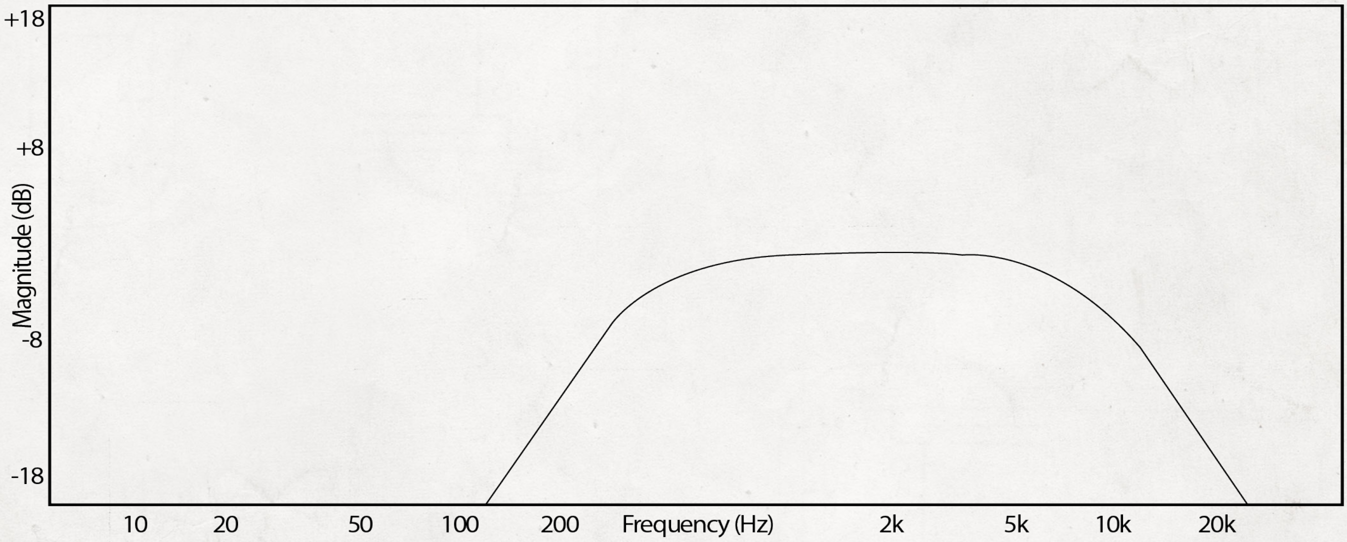
Lp section includes 2 different highpass filters, A and B mode. They are derived from 2 different devices.

- A mode
The low-pass filter has a 12dB per octave (20dB per decade) slope and a frequency range from 7.5kHz to 18hHz.
- B mode
The low-pass filter has a 12dB per octave (20dB per decade) slope and a frequency range from 1.5kHz to 18kHz.
- C mode
The low-pass filter has a frequency range from 4 kHz to 12.5 kHz.

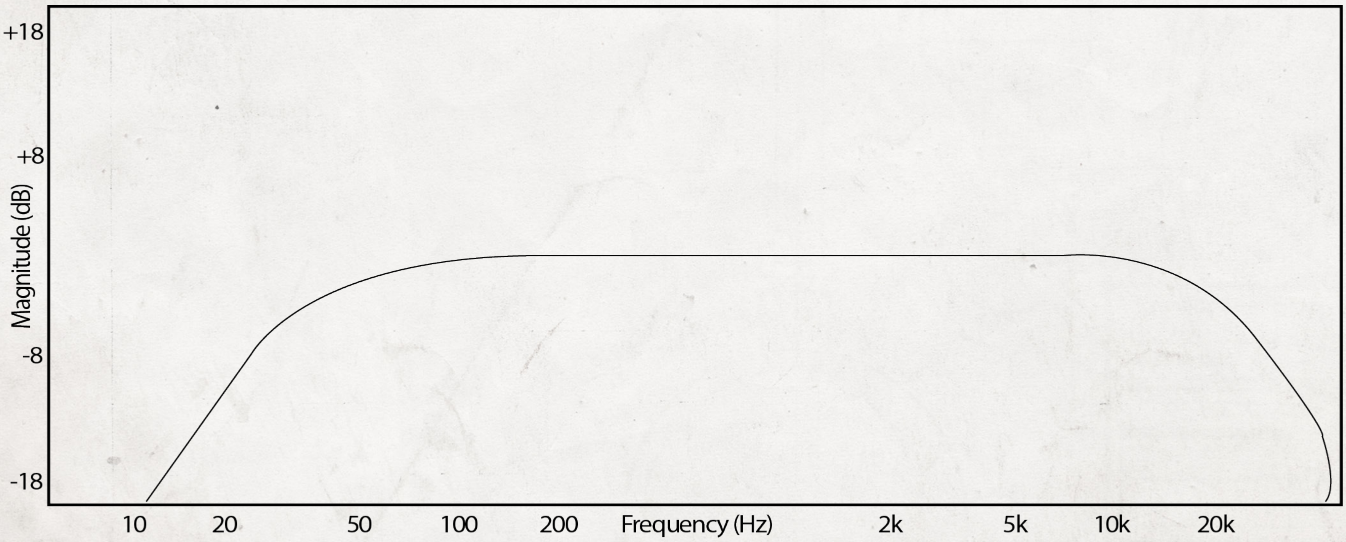
A filters: HP 31.5 Hz / LP 18 kHz



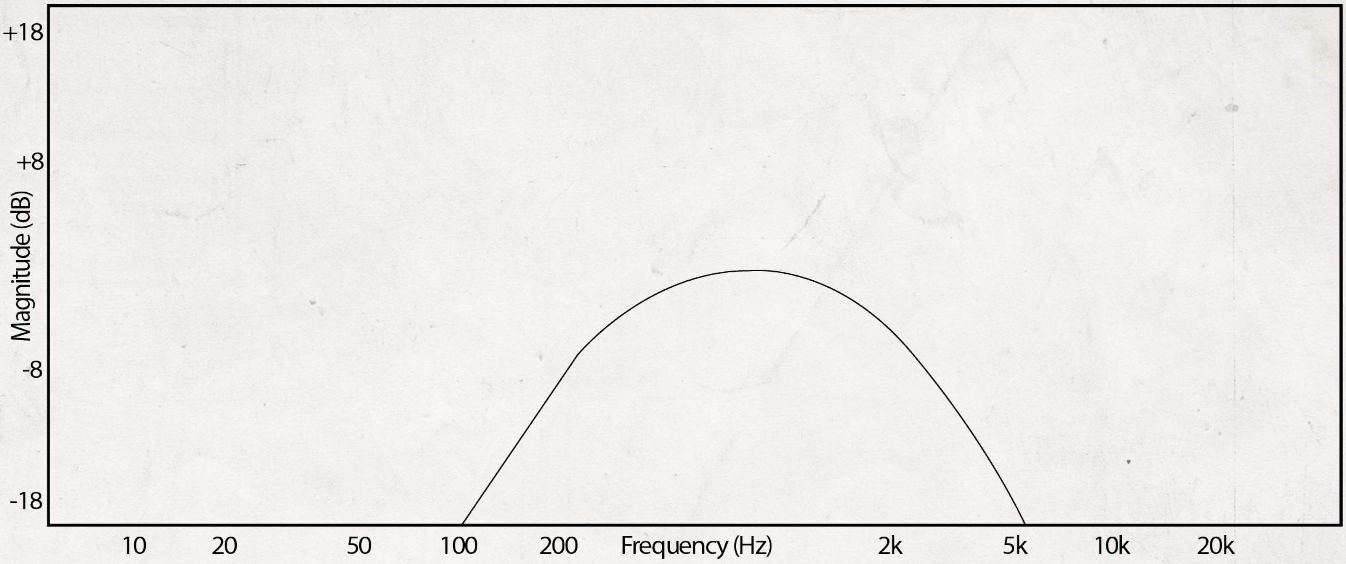
A filters: HP 315 Hz / LP 7.5 kHz



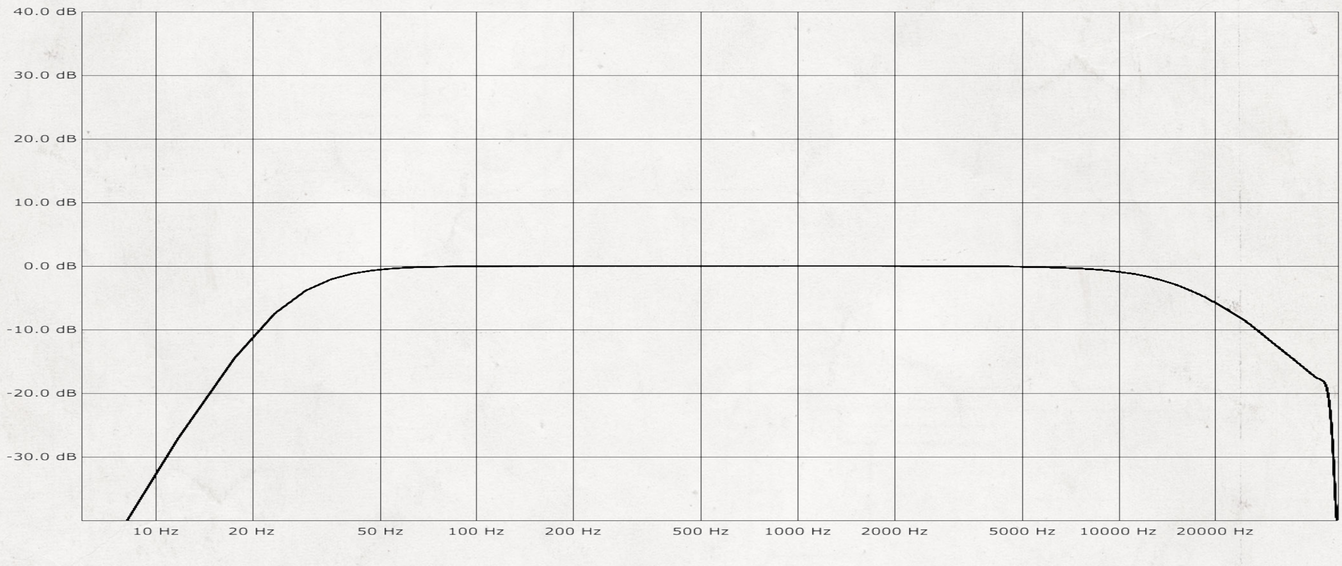
B filters: HP 30 Hz / LP 18 kHz



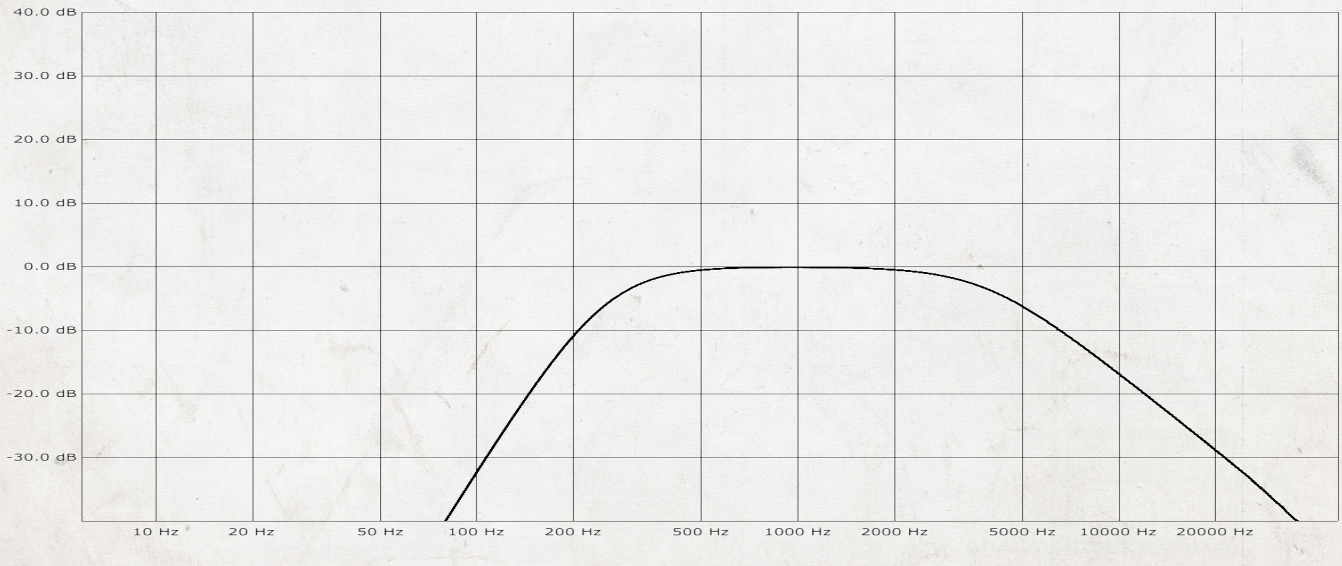
B filters: HP 300 Hz / LP 15 kHz



C filters: HP 31.5 Hz / LP 12.5 kHz



C filters: HP 315 Hz / LP 4 kHz

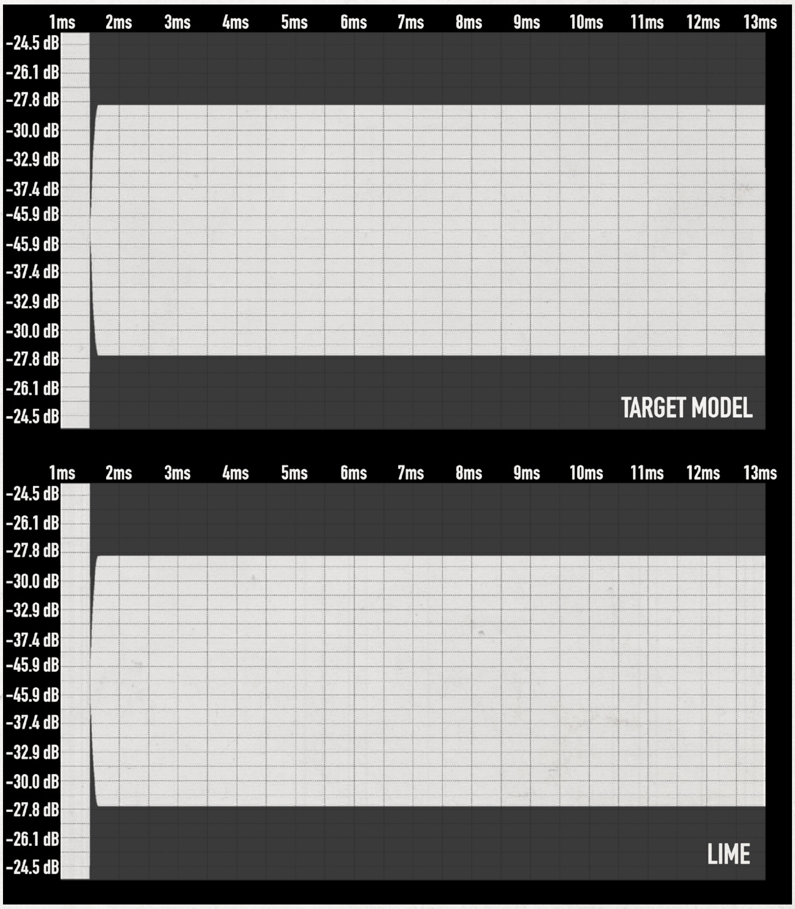


Compressor section

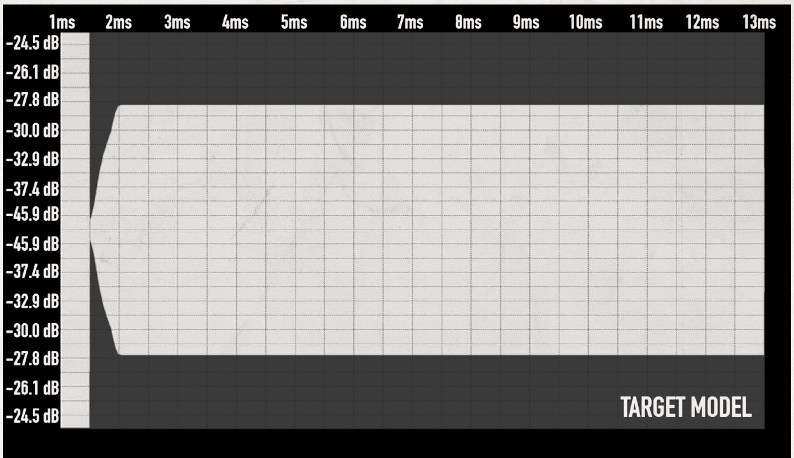
Just a little bit of history: the birth of Ultramatch

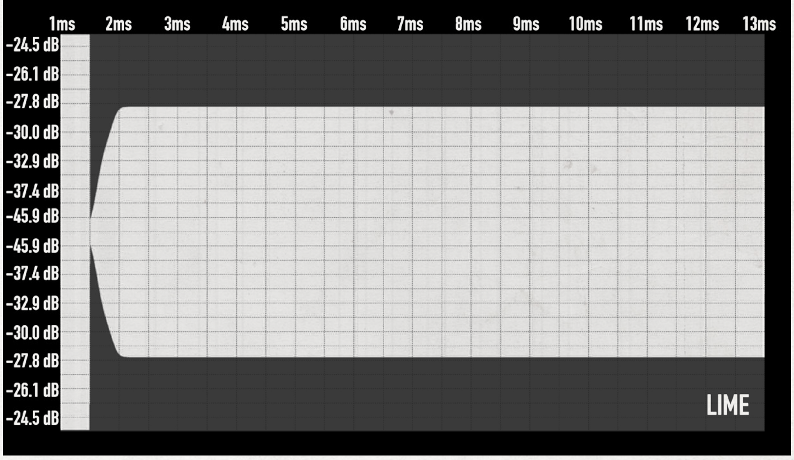
This technique is called “Ultramatch”.
It is a new technique for minimizing the errors: in practice, through the use of a dedicated Nebula instance it’s able to examine and simulate the time variables of the original curve, ensuring a high precision level.
In comparison, the previous technique just used to decompress the signal by calculating the envelope followers of the attack and release shapes.
As an example, below we show a comparison between the target model signal and the processed signal of “C” compressor.

Release 0

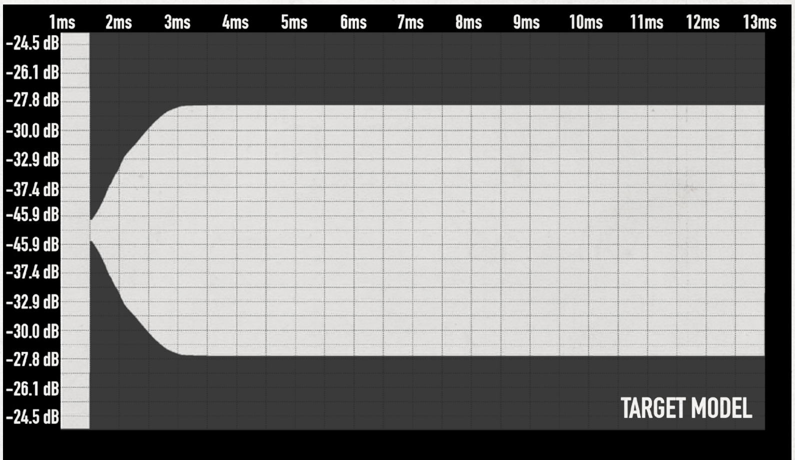


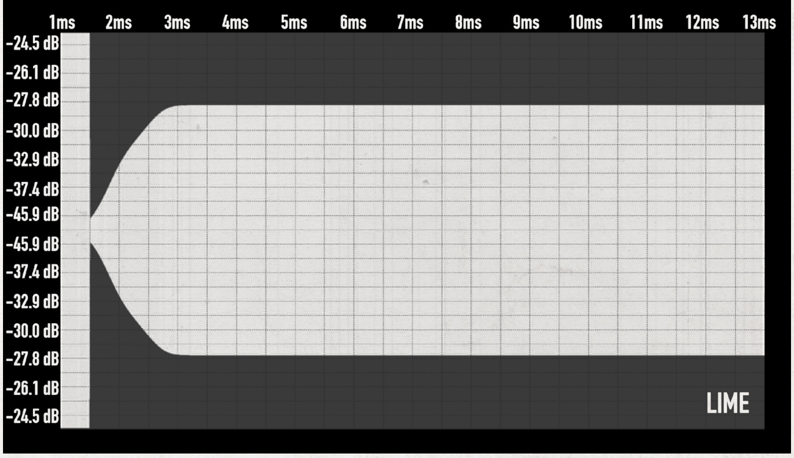
Release 1



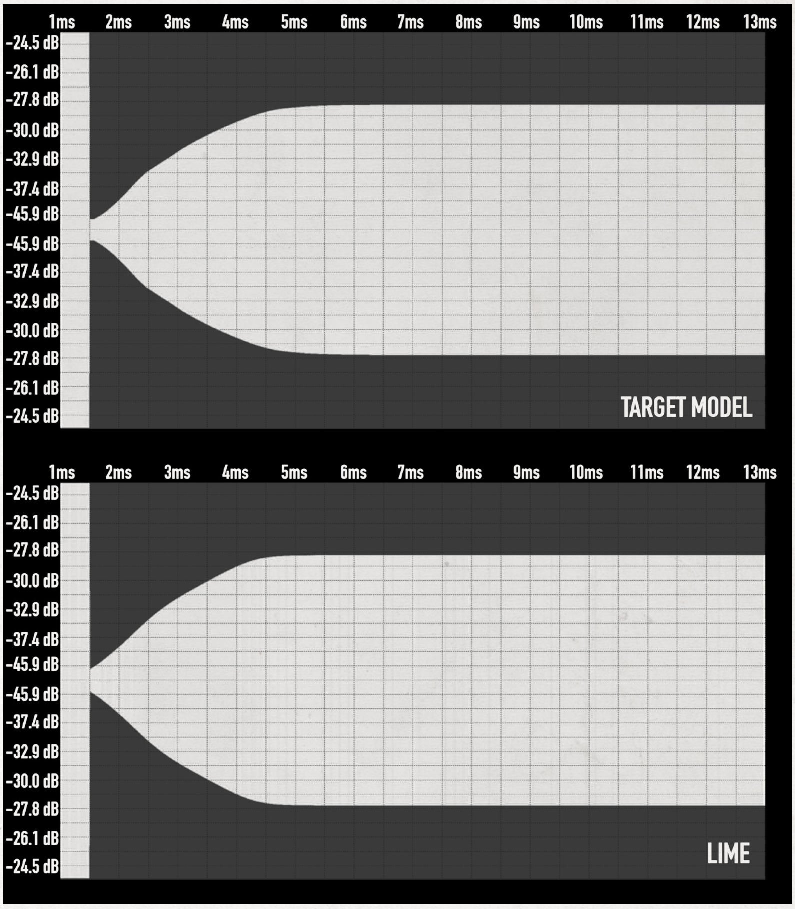


Release 2

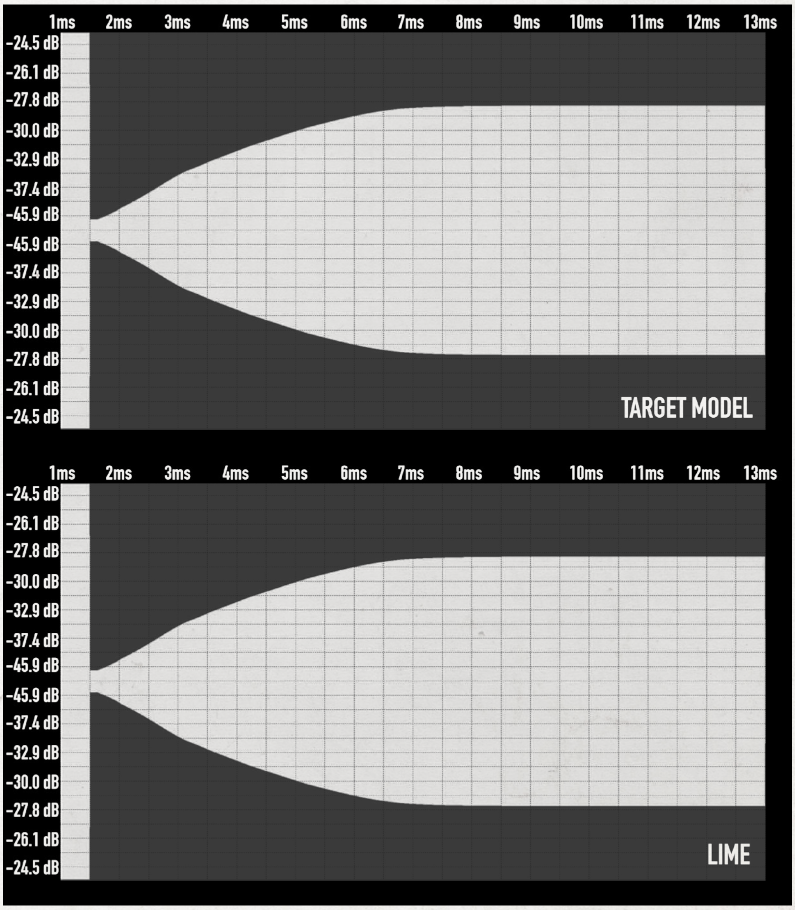




Release 3



Release 4



Compressors

The COMPRESSOR section is characterized by a modern design, great performance and reliability.
The inclusion of a compressor module in this plug-in provides a complete processor capable of delivering the character and sound of the original device, while incorporating some new exciting features.
LIME2's compressor section is a · technoloical masterpiece; it includes 5 different compressors in one plugin! Each one is characterized by a unique sound. We are especially proud of C Compressor, which is an emulation of a rare, powerful and expensive hardware!
In our opinion it's an exceptional compressor with an exceptional sound!

"ON" button

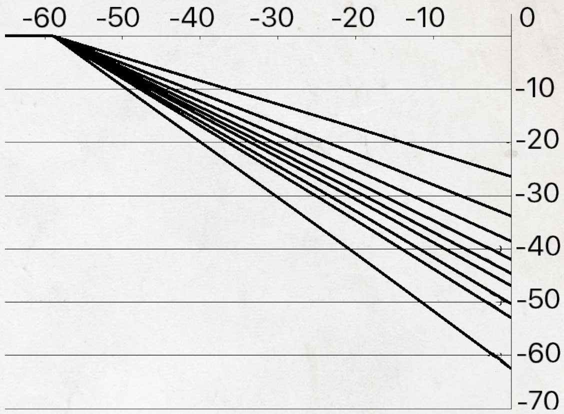
The Compressor is activated by pressing this button. In bypass all controls (green knobs) return to default position.

Gain reduction meter

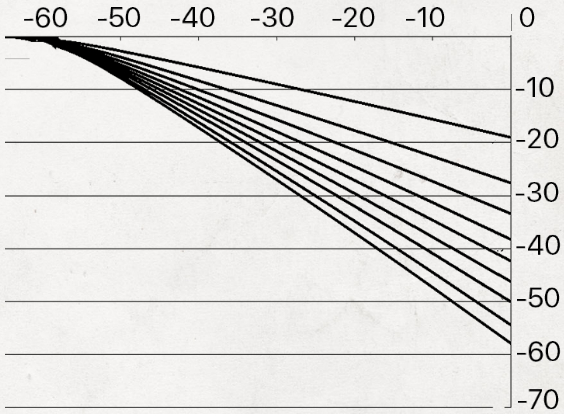
This measures the reduction level applied by the compressor. The meter indicates '0' in the absence of any input signal or gain reduction. If the signal exceeds the compression threshold or limit level, the amount of gain reduction is displayed.

Ratio knob

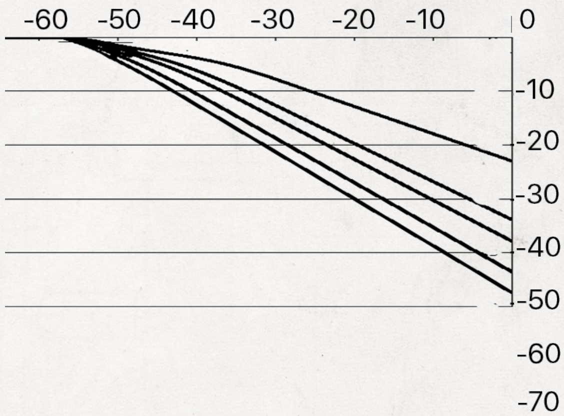
This knob sets the compression ratio.
A, B, E Compressors: available range values from 1:1 to MAX:1
C, D Compressors: available range values from 1:5 to MAX:1



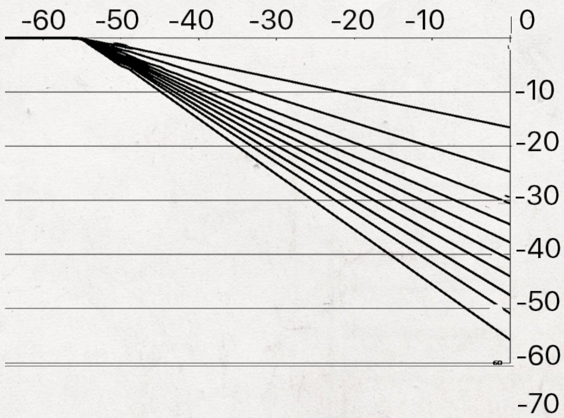
Ratio A



Ratio B



Ratio C-D



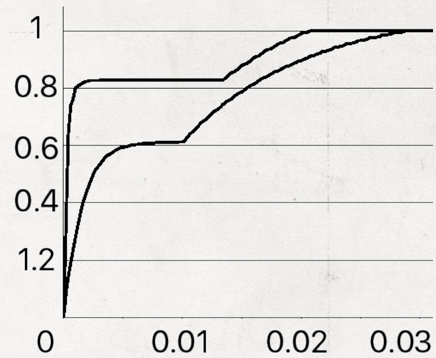
Ratio E

Knee knob (only comp B)

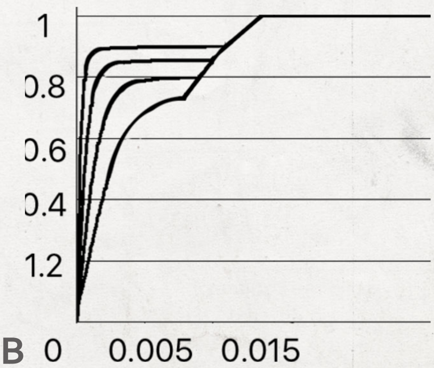
This knob sets the shape of the compression curve.
It controls the KNEE of this module, changing the way the compressor begins to reduce the signal level.
SOFT : compression begins gradually as the signal exceeds the threshold.
HARD: compression begins immediately at the chosen ratio.

Attack knob

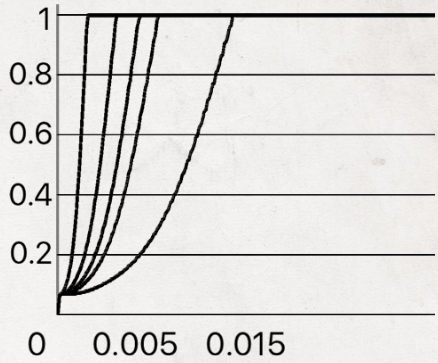
This knob sets the compressor's attack time, ranging from X* ms (fast) to X* ms (slow) – see table below.



Attack A



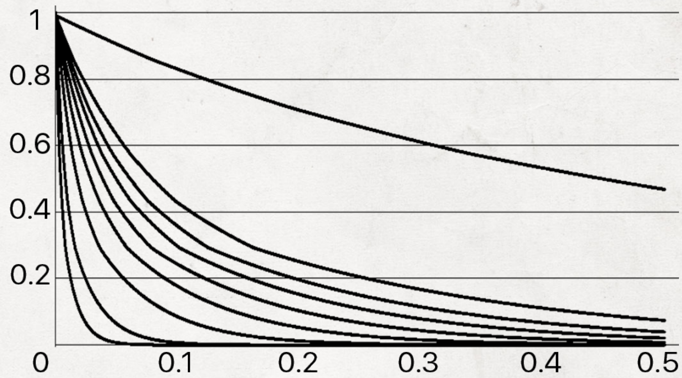
Attack B



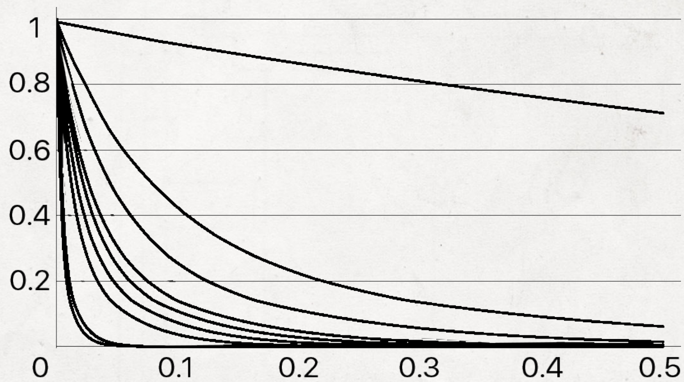
Attack C-D

Release knob

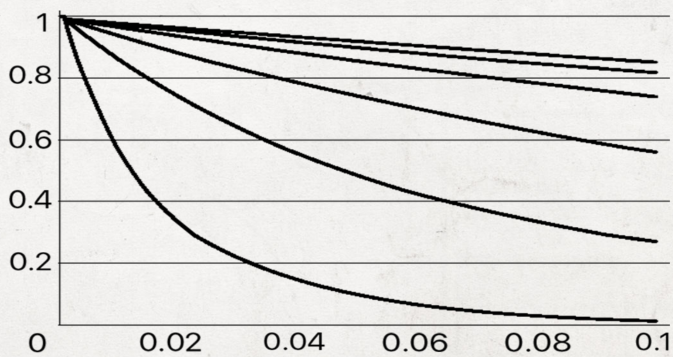
This knob sets the compressor's release time, ranging from X* ms (fast) to X* ms (slow) – see table below.



Release A



Release B



Release C-D

N°	Compressor	Attack (ms)*					Release (ms)*								
		1 step	2 step	3 step	4 step	5 step	1 step	2 step	3 step	4 step	5 step	6 step	7 step	8 step	9 step
1)	A	3	10				0.046646	0.092375	0.211781	0.337760	0.471063	0.611177	0.754792	0.989729	5.816219
2)	B	3	4,5	6	8		0.037094	0.047302	0.155927	0.217344	0.279510	0.345729	0.561281	0.972292	7.700167
3)	C	2.6	5	7	9	16	0.108281	0.452594	1.211635	1.211635	4.722844	6.955125			
4)	D	2.6	5	7	9	16	0.108281	0.452594	1.211635	1.211635	4.722844	6.955125			
5)	E	3	10				0.046646	0.092375	0.211781	0.337760	0.471063	0.611177	0.754792	0.989729	5.816219

SHmod knob

This knob controls the shape of the compressor's attack curve. It allows you to fine-tune the attack shape, allowing you to really optimize the compressor's behaviour for any audio source.

Threshold knob

This knob sets the threshold of the compressor. Range: from -64dBu to 0dBu.

Make up knob

This knob sets the gain compensation and is designed to boost the compressed signal in order to match the level of the uncompressed signal.

Mix knob

This control determines the mix proportion between the original (dry) and 'effected' (wet) signals. It's a very powerful and simple to-use feature.

Contents

System Requirements

Acustica Audio has been working in high-quality analog hardware device software modeling for over fourteen years now. The audio rendering engine, Acqua, embodies state of the art, sample-based technology, and has set a new quality standard in the professional audio plug-in market. Acustica Audio, in a bold move, even for a cutting-edge company like us, have cre¬ated something great and we are now bringing it to you in the form of this ground¬breaking and incredible sounding Acqua plug-in. Of the current software plug-ins available on the market, none come close to the sound of the Lime2 suite. This plug-ins bundle is based on our new CORE 15 technology.

	PC Windows		Apple macOS	
	MINIMUM	RECOMMENDED	MINIMUM	RECOMMENDED
OPERATING SYSTEM	Windows 10 64 bits	Windows 10 64 bits	macOS 10.9	macOS 10.14
CPU	Intel i5 Broadwell 3.1 GHz*	Intel i9 Coffee Lake 3.5 GHz*	Intel i5 Broadwell 3.1 GHz*	Intel i9 Coffee Lake 3.5 GHz*
RAM	4 GB of RAM (1)	64 GB of RAM(1)	4 GB of RAM (1)	64 GB of RAM(1)
SSD	4300 MB	4300 MB	4300 MB	4300 MB
SCREEN RESOLUTION	FHD (1920x1080)	UHD (3840x2160)	FHD (1920x1080)	UHD (3840x2160)
PLUG-IN FORMAT	VST & AAX (2)	VST & AAX (2)	VST, AAX & AU (2)	VST, AAX & AU (2)
AQUARIUS	Mandatory	Mandatory	Mandatory	Mandatory
INTERNET CONNECTION	Mandatory	Mandatory	Mandatory	Mandatory

All technical specifications of Acustica Audio products provided are intended to be estimates or approximations.
Due to numerous variables, no guarantees of compatibility or performance can be made.
The end-user is solely responsible for, prior to purchase, ensuring that the end-user's devices are compatible and meet the system requirements for Acustica Audio products.

* AMD or Intel Xeon CPUs are not recommendable and the CPU speed is more important than the number of CPU cores.
(1) In order to run more plug-ins instances it is always necessary to increase the amount of RAM.
(2) 64-bits supported only.

During the modeling process we used the best converters and cables in existence, we measured the units in excellent conditions, and employed skilled experts in the sampling process using our self-developed sampling application. Now you have one of the best, high-quality professional audio plugins in your audio workstation.
We spend countless hours developing these no-compromise plug-ins to give you only the best sound and feel that is as close to real hardware as can be imagined. We are confident that this plug-in will help you make your production sound more professional.

IMPORTANT:
- It is the user's responsibility to configure correctly the operating system, drivers and the DAW application
- The computer system should be optimized to work at high CPU load and low audio latency.

Product Registration

After you have purchased a product from our webshop, product registration is automatic. Your newly purchased product will be available for downloading using our installation assistant application Aquarius. For more details about product registration, please refer to the Aquarius user manual on our website.

Product Authorization

Product authorization and de-authorization is an on-line automatic process that creates a product license based on your computer's identification code.
This procedure is automatically performed by our installation application, Aquarius.

Its purpose is to simplify and automate the authorization, installation and uninstallation process of your purchased Acustica products.
For more details about installation/authorization, please refer to the Aquarius user manual on our website.
RECOMMENDATION:
Please always update Aquarius to the latest version available. In case of authorization problems with an Acqua plug-in, we recommend you to proceed with a product uninstall and then re-install through the latest version of the Aquarius app.

Performance Caution

In order to maximize the performance and usability of the Lime2 suite on your computer, we suggest you follow some precautionary rules that will help you save precious CPU cycles.
-First of all, set your buffer size setting as large as possible.
There is generally no specific reason for using a low buffer size setting during mixing or mastering sessions. Increasing buffer sizes (consequently latency) highly decreases required CPU power.
-You should also consider only using the necessary features. We do not ensure the complete absence of bugs or the perfect operation of the product. Before purchasing, we suggest you download the Trial version to verify the behavior of the plug-in with your system.
Trial products are fully-functional versions of the relative commercial plug-in. The trial period expires 30 days after activation.
We do not take any responsibility for the misuse of the product, or collateral problems derived from it.
Normally the Introductory price period ends within 30 days from the publication on the product page, but this period may vary at our discretion.
This manual includes a description of the product but gives no guarantee for specific characteristics or successful results.

The design of our products is under continuous development and improvement. Technical specifications are subject to change.

NOTE: Please keep in mind that for each plug-in in the Lime2 suite we recommended that you calibrate your input levels to the usual Acqua/Nebula convention: -18dBFS = 0VU; this way you will avoid any unwanted distortion or unpredictable behavior due to an excessively high input level.

What is a ZL instance for?

Acustica Audio's plug-ins come in two versions: ZL (zero latency) and normal (non ZL). While the ZL version does not introduce any latency to your system, the standard version does.
This buffer varies in size for each plug-in and helps to significantly reduce the CPU and system load of your computer. For this reason we recommend that you use a ZL instance whilst tracking.
Keep in mind that anything that can reduce the CPU load on your system should be considered. For example the track count of your session, the number of plug-in instances used, sample rate, etc. You could also consider direct monitoring or doubling the buffer/hybrid audio engine in your host if available. Basically both plug-in instances are identical but the current Acqua engine can work with a long audio buffer or without any audio buffer. The instance without audio buffers, "ZL", or zero latency, does not have any audio buffer pre-loaded, and will process the audio without any delay, but at the same time the CPU load will be higher compared to the standard non ZL instance. The idea behind a ZL instance is to give users the option to run Acqua Effect products with minimal latency, which is useful for tracking or direct monitoring.

AI presets

Presets Management

PRESET MANAGEMENT
The LIME2 EQ (standalone version) includes AI (Artificial Intelligence) Presets.
By clicking the "PRESET" drop down menu on the left hand side of the LIME2 EQ you can select a preset from the displayed list. You can choose between several presets. You may find a detailed list of presets in the following Chapter "AI PRESET LIST & CREDITS".
A normal preset would simply load the same settings each time you use it. Our AI Presets are based on a huge amount of data sampled from real-life mixing sessions by renowned engineers. Any AI Preset will assess the audio being fed into the plug-in and then, based on the data stored in its memory, it will automatically modify the EQ settings, emulating what the referenced engineer would have done in the same situation.
Here's the procedure to obtain the best results:
- loop a short section of audio that you deem is most significant for the AI evaluation. The analyzed time frame is quite short (only a couple of seconds) so different points in the audio will obviously produce different results;
- click the preset you would like to use;
- sit back and watch as the eq settings change.
This brand new technology works very well on individual tracks and groups, whereas results on the master bus may vary.

01 - AMIEL REUVEN

General Purpose EQ suitable for individual audio sources, group busses and mixbus.

Grammy Winning Mixer Reuven Amiel is an eclectic and versatile Mixing Engineer/Producer/Sound Designer.
He applies his modern, edgy and vibey sound to Indie Music, Modern Rock, Electro-Pop as well as Latin Pop moving thru World Music and everything in between.
Reuven is also a sound designer and programmer for many prestigious audio software and electronic music software/loops companies.
He has worked with a kaleidoscope of Artists and genres as his life is eclectic, having lived in many hem-ispheres of the world. From his beginnings, studying under the wings of Yoav Gera (Ofra Haza, Yehudit Ravitz) and moving all the way to Canada to receive knowledge from Top Producer Bob Ezrin (Pink Floyd, Peter Gabriel, Kiss etc.) to working with Israeli/Scottish underground Rock Band Mushroom Symphony, Cult Indie artists as Rouckfour to his upcoming project with European Rock Band, Pony Asteroid. He has also worked with acts like, PVRIS, Cadaver Exquisito, Canadian Electronic Band NOIA, Prime Ministers among many others.
He also has worked with Top Latin Grammy Winning/Nominees such as Ricardo Arjona, Shaila Durcal, Gian Marco, Susana Baca, Cristian Castro among others. He received a Latin Grammy for his mixing of Artist Felipe Pelaez and several other awards in different territories.

AmielMix
www.amielmix.com

02 - OLEG YORSHOFF

Preset Sampling Technique (used in the AI capture phase)
- General Purpose EQ suitable for individual audio sources, group busses and mixbus.

Oleg "Yorshoff" Yershov - mixing and mastering engineer, pro audio journalist and respected audio mentor and educator from Ukraine. Former classical piano player, then heavy metal touring vocalist Oleg now focuses on studio work for different artists producing different genres and styles of music - from synth-pop and Indie to EDM, atmospheric black metal and countless Hip-Hop artists all over East Europe. In 2013 Oleg launched YorshoffMix, a Youtube-channel with the aim of helping Russian speaking audio engineers to grow and become better educated in music production, mixing and mastering. In addition, Oleg writes for Future Music Russia magazine.

Yorshoff Mix | Mixing & Mastering Services
www.yorshoffmix.com

03 - JAMIE DOCKERTY

Preset Technique (used in the AI capture phase)
-EQ for single instruments, Drums, Strings, keys, Synth, Guitars, Male, Female Vocals.

Born under the oceanic sun of England in 1984, Jamie went on to become an Audio Visual/Crestron Technician in the Royal College of Surgeons in Ireland, He is a Studio Mix Engineer in his own Studio as well as being a Musician, Songwriter, Audio plugin/Hardware consultant, and most importantly a Word spreader of vital information. Jamie is the Creator of Acustica Audiophiles on Facebook as well as the Facebook Page Plugin Review.
Jamie likes long walks on the beach, purple M&M's and Vanilla Lattes.

04 - MATTHIAS FLEISCHMANN

General Purpose EQ suitable for individual audio sources, group busses and mixbus.

Matt Fleischmann, born in 1968 in Germany, started playing playing piano and guitar at the age of 6. A relative introduced him to sound technology at the early age of 14 by taking him to his studio on a regular basis. At the age of 19 Matt left for Ireland and the UK where he worked as a musician and live-sound technician for more than 10 years while studying studio sound engineering in the UK. This gave him the chance to work with some notable folk and rock artists from Ireland and the UK, both live and in his first own studio.
After moving back to Germany in the late 90s he worked as a musician and freelance sound engineer. At the time he was primarily involved in live recordings. He reopened his own recording studio which is now located near Stuttgart and Ulm, Germany. Matt´s widespread musical interests include the recording, mixing and mastering of hand made folk, blues, jazz, rock and also classical music in the same way as world music and experimental elec¬tronic music. His studio services also include audio restoration.
Today Matt is mainly running his own studio while still maintaining the live side of things on the side. He never lost his passion for live mixing and recording, he plays in a couple of bands himself and enjoys supporting new talent. He´s also distributor and product specialist for Fuchs Audio Technology guitar amplifiers as well as some high-end recording microphones and outboard gear. On top of that he´s beta-tester for some DAW and plugin makers, and gives classes in audio engineering and workshops in guitar technology and guitar recording.

www.pro-suite-audio.de

05 - EMRE RAMAZANOGLU

Preset Sampling Technique (used in the AI capture phase)

- General Purpose EQ suitable for individual audio sources, group busses and Master bus.

Emre is an incredibly versatile songwriter, producer, mixer, and programmer who loves everythingfrom filmic landscapes, 80s pop, beats and jazz. In two words he'd describe his sound as "Dysfunctional Pop".
On one end he's a founder member of Jazz band 'Ill Considered' and on the other he co-wrote and co-produced "Making the Most of the Night" by Carly Rae Jepsen. (with Sam Dixon and Sia).
Somewhere in the middle would be his work on the latest #1 Noel Gallagher album "Who Built the Moon" He was brought in by David Holmes to engineer but ended up drumming, programming and mixing the whole lot. He was an integral part of the sonic.
He's mixed RITUAL feat Tove Styrke, John Newman, Aqualung, Rita Ora, Jack Savoretti, Kylie (Music's Too Sad Without You, from her #1 album) and has just produced/ mixed four Lily Allen tracks from her latest album including the single "Lost My Mind". He's co-produced and mixed Richard Ashcroft's new album 'Natural Rebel'. He co-produced "Warrior' from Paloma Faith's number 1 album and mixed Jimothy and Tom Grennan recently. He's engineered for U2 and played drums for Michael Jackson. He's been writing with Sampha, Foxes, Theon Cross, Actress, Emily Burns, Pasteur, Morgxn (including single "Holy Water' which he also produced and mixed), LUME, JONES, Sophia Alexa, Chelcee Grimes, Nilufer, Grace Barker, Lauren Aquilina, Barny Lister amongst others.
Emre worked with David Holmes on the movie Logan Lucky. He's currently working on the new Killing Eve sound track (having worked on the first and second series) and the new Steven Soderbergh movie "The Laundromat".

www.emremusic.com

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