Engineer

User's Guide

MARTIN AUDIO
LONDON

The Martin Experience

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1. Important Safety Warnings

It is extremely important to read ALL safety information and instructions provided in this manual and any accompanying documentation before installing and operating the products described herein.

Heed all cautions and warnings during installation and use of this product.

Keep this instruction manual for future reference.

This unit does not contain any user serviceable parts.

Do not open this unit. Doing so will void warranty and might present a risk. Servicing must be performed by qualified personnel only. Servicing is required when the apparatus has been damaged in any way, such as power-supply cord or plug being damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.

This unit is designed for indoor use. Do not use this unit in a wet or damp environment or near water.

Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.

This unit is not designed for residential use.

Do not block any ventilation openings. Install in accordance with the manufacturers instructions.

Do not defeat the safety purpose of the polarized or grounding-type plug. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.

Unplug this apparatus during lightning storms.

CAUTION
RISK OF ELECTRIC SHOCK
DO NOT OPEN
DO NOT EXPOSE TO RAIN, MOISTURE,
DRIPPING OR SPLASHING

ATTENTION
RISQUE DE CHOC ELECTRIQUE
NE PAS ENLEVER
NE PAS EXPOSER A LA PLUIE NI A L'HUMITE

2. Unpacking the unit

After unpacking the unit, please check it carefully for any damage. If any is found, immediately notify the carrier concerned - you, the consignee, must instigate any claim. Please retain all packaging in case of future re-shipment.

Please think of our environment and don't bin any materials, including this manual. When the product has reached the end of its useful life, please dispose of it responsibly through a recycling centre.

3. Introduction

Thank you for choosing the Martin Audio Engineer for your application. The Martin Audio Engineer is a powerful, advanced DSP platform designed for audio installations. It features 4 inputs and 8 outputs with extensive, flexible routing and x-over functionality.

On top of the 'normal' x-over functionality it features two unique highly advanced specialist audio algorithms. The first one being the dream of every installer: an automated sound engineer-in-a-box called 'The Engineer', working 24-7 to keep a pleasant, consistent sound in the venue. The second one being the Basscreator algorithm, a psycho-acoustical effect to make small speakers sound like they are a lot bigger, with unexpected amounts of perceived low-frequency output from a small speaker.

The unit also features an advanced scheduler, to automate preset recall and make sure the correct preset is triggered at any time.

These combined features make the Martin Audio Engineer an outstanding one-box problem solver for any installation where high quality and "no-fuzz" operation are important factors.

Please take the time to read this manual carefully, as it will enable you to get the best out of the product.

3.1 Features

- 4 inputs and 8 outputs with complete & freely routable signal path.
- 8-band fully parametric EQ on every input.
- 8-band fully parametric EQ on every output.
- Highpass and Lowpass filters on every output with slopes up to 24 dB/oct. featuring L-R, Butterworth and Bessel filters
- Speaker alignment delays of up to 10ms on every output.
- The most musical-sounding speaker protection limiters in the business on every output.
- Very high, musical sound quality throughout.
- Unique Engineer DSP algorithm, working 24-7 to keep a consistent sound in your venue.
- Unique Basscreator algorithm, gives small speakers the perceived low frequency response of a large speaker.
- Flexible built-in scheduler for automated preset recalling.
- RS-232/485 interface for extensive computer-based control.
- Programmable remote control with RS-485 interface on industry-standard RJ-45 connector, for long cable runs.
- Up to 4 remote controls can co-operate on one Engineer through the use of a HUB.

4. Connections

4.1 Power

The power connection is located on the back of the unit in the right corner. This device should always be used with an earthed power connection. The blue LED on the front of the device indicates that the device is powered. The power switch is located on the front of the device behind the small hole on the left side. Operate the power switch carefully by the use of a small screwdriver.

4.2 Audio

The audio connections are located on the backside of the device. The connections are balanced "Phoenix" connectors. From left-to-right are OUTPUT 1-8 and INPUT 1-4. The connections are labelled:

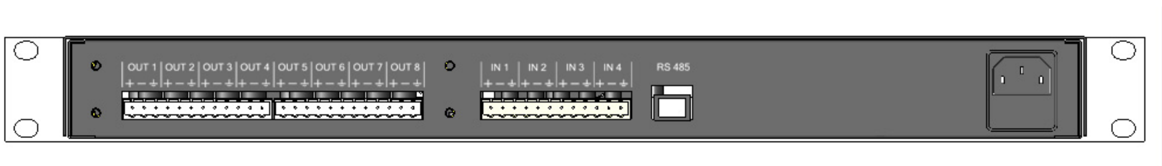
+ for audio hot
- for audio cold
⏚ for audio ground

4.3 RS-485

On the back of the device, next to the audio inputs, is the powered RS-485 connection for connecting the Engineer Remote on a RJ-45 connector. For wiring scheme, please see the chapter 'Remote Control'.

4.4 RS-232

On the front of the device, behind the cover plate in the middle, is the RS-232 connection for remote control of the device with a PC. For connection with a PC use a fully wired 1:1 DB09-DB09 M/F cable only. The cable may have a maximum length of 45 feet.



5. Using the software

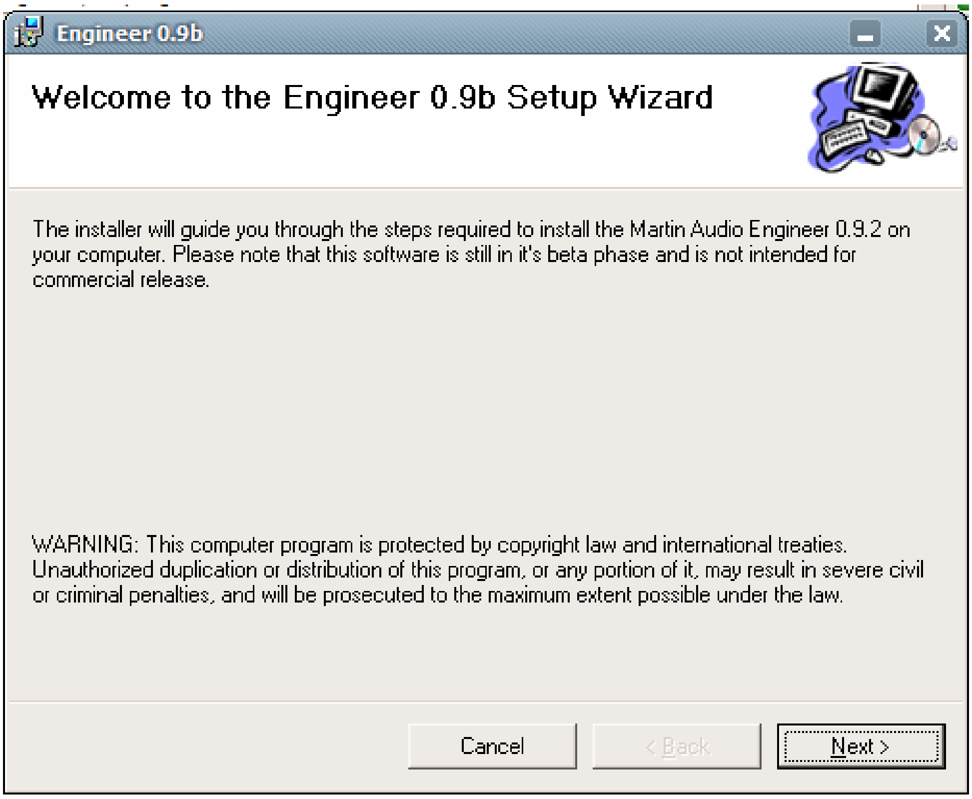
To use the software, your computer has to meet the following requirements:

- Intel x86 Pentium 3 compatible computer with RS232 port or RS232 USB dongle
- Windows XP with service pack 2
- Microsoft .Net framework version 2.0 or higher

5.1 Installation

To install the Martin Audio Engineer control software, insert the CD-ROM and follow the Autostart EngOpen.htm instructions.

If the disk doesn't autostart, open the "my computer" dialog, located on your desktop or start menu. Select the CD or DVD drive containing the Martin Audio Engineer software and double-click on the "Engineer.msi" or "Setup.exe" file. You should now see the installation screen. Please follow the instructions on your screen to complete installation.

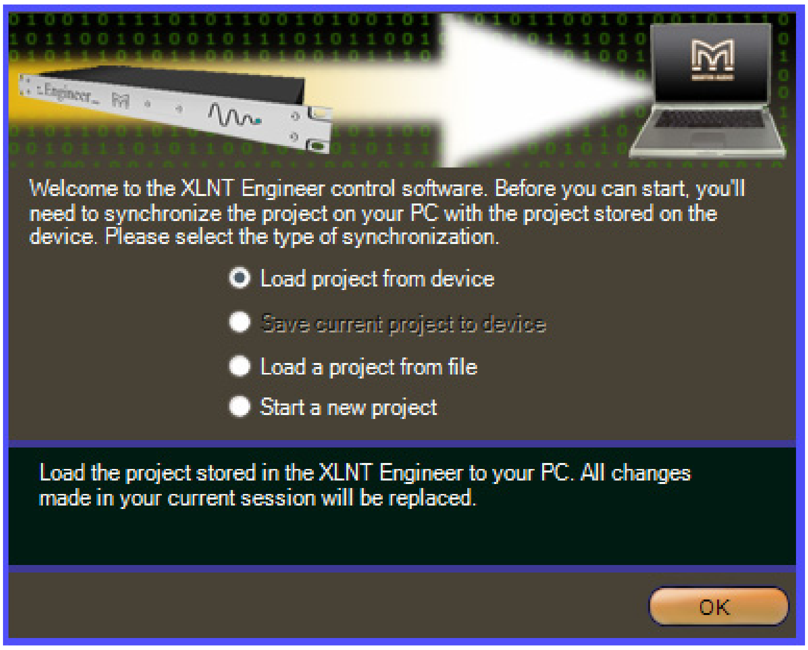


After installation, a new program group called "Martin Audio Engineer" will be available from the programs folder in your start menu. To start using the Engineer software for the first time, simply click the Engineer icon.

5.2 Synchronization

As soon as the Engineer software starts up, it will ask in which way you want to synchronize the software. Synchronization is an important concept within the software: Always make sure your software is synchronized with the device when working with the system. This ensures correct functioning of the device (more importantly, it ensures that what you hear is actually a representation of what you see on your screen). Also, when used correctly, this will prevent you from losing that precious device setting that you worked on for hours.

On start up, you're presented with the following screen:

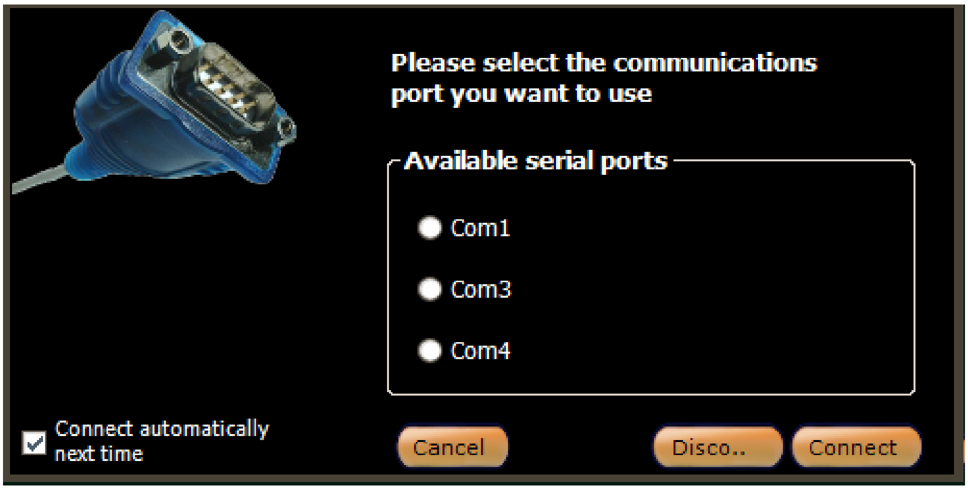


Loading the current project from the device is the safest way when it comes to making sure the current project stored in the device isn't lost: The Engineer will send the currently loaded settings to the software, and you can take it from there. In case you have an existing project in an Engineer file, select the "load project from file" option. This will load the stored project in the software and send it straight to the device, discarding all data previously stored in the unit.

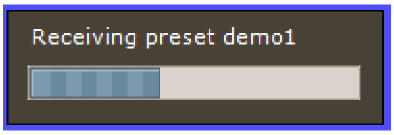
As you can see in the image above, saving the current project to the device is disabled on start-up, for the obvious reason that there's no project yet. The last option, starting a new project, is mainly available for convenience when working offline. The device will not be synchronized with the project in your software, so this isn't recommended. Synchronization will have to be handled manually, which can be done from the file menu. For now, please select the first option.

5.3 Initializing communication

Once you've chosen in which way you want to synchronize, you'll be presented with the communications dialog:

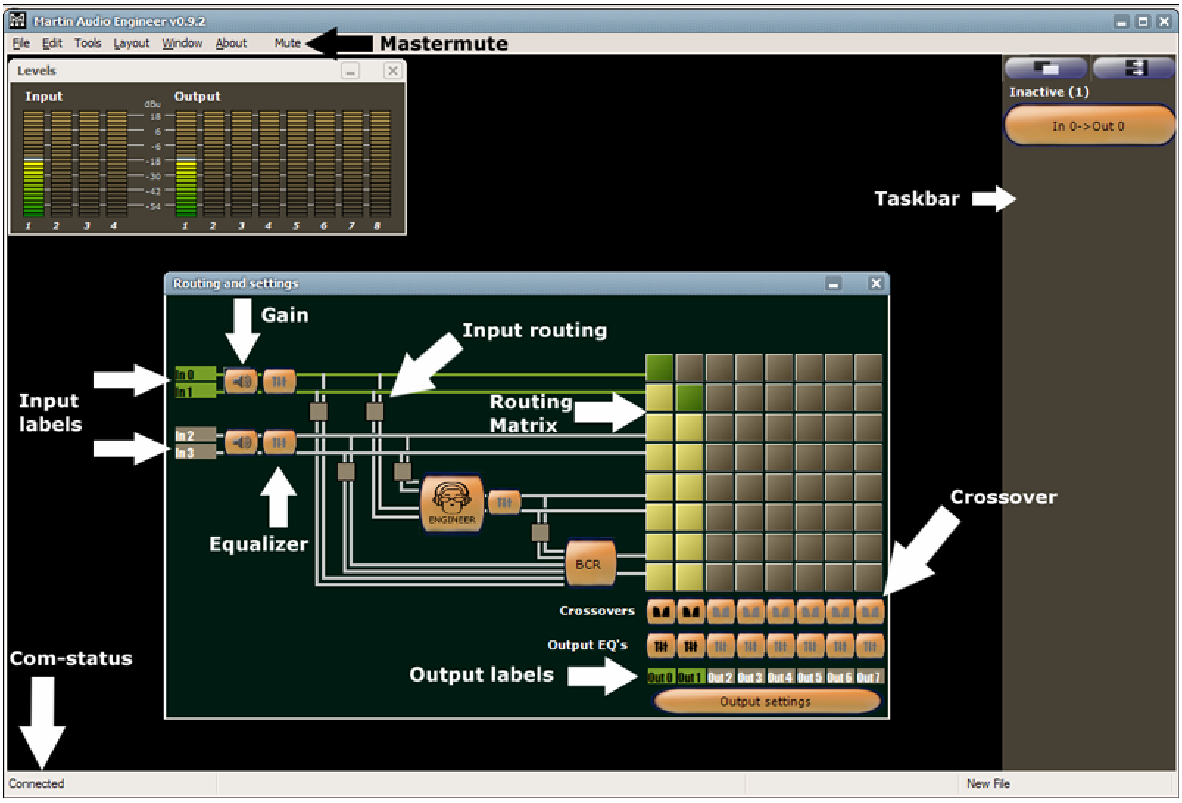


All available RS232 ports will be shown here, in this examples port 1, 2 and 4. Note the check box in the bottom-left corner. Selecting this will prevent this dialog box from showing up in the future, and set the communications port you select as the default. Make sure your Engineer is connected to the port you select and hit "Connect" to start the synchronization. During the synchronization process a progress bar will be shown. Please be patient while this operation completes; this can take a few minutes in the case of many stored presets. Since this is the first time though, only the factory defaults will be downloaded, which will be relatively quick.



6. Software features

Now that we're done synchronizing we can really start using the software. You should be seeing the following screen (without the arrows of course).



Beside the "Levels" window which -obviously- represents the current signal on the Engineer in- and outputs, there's a window called "Routing and settings". Take some time getting to know this window: Most of your work within the software will start from here. After synchronization it will always show the currently active settings in the engineer.

6.1 Master Mute

Before you start experimenting with the software, you'll want to be sure to know where the mute button is located. After all, everyone makes mistakes and you wouldn't want to have to unplug the whole device. Hit the mute button in the top menu-bar to mute all outputs on the system. This button will always be above any other window, so you won't have to look far when you need it. The background of your screen will turn red if the master mute is engaged.

6.2 The communications status indicator

If you're not sure whether you're connected to the Engineer, you can check the status of your connection at any time in the bottom-left corner of your screen. Don't worry about it too much though: Should the connection drop for any reason, a notification window will appear informing you of the problem.

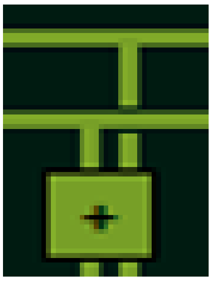
6.3 The taskbar

See the grey bar to the right side of your application? This is where minimized windows will appear. You'll want to make use of this when configuring complex projects where dozens of windows might clog your workspace with excess information, preventing you to focus on the job at hand. When you minimize a window it'll appear in the taskbar as an orange button, which you can click on to restore its window. Use the small blue buttons on the top to rid your workspace of all windows at once, or to restore them all.

6.4 Input routing

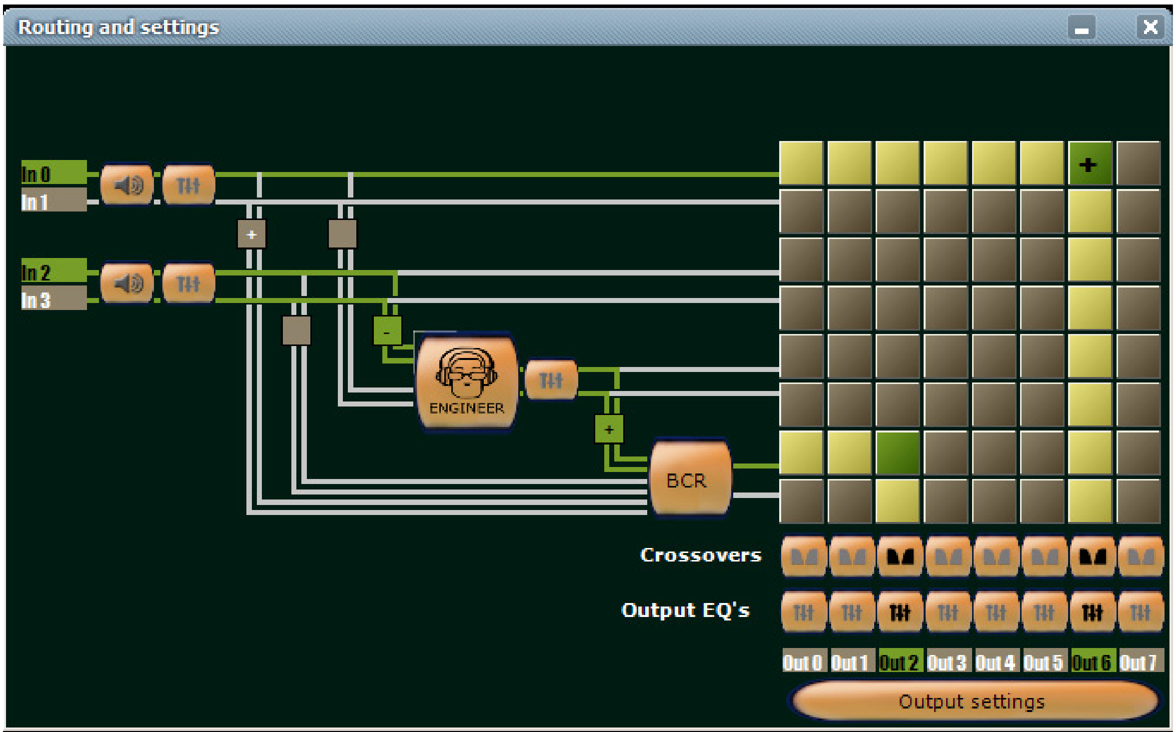
Each of the orange buttons in the routing and settings window opens up a separate window, allowing you to control the specified feature. These buttons are placed within a routing grid (marked input routing in the image above), and they allow you to route the input signal through a number of junctions towards the main routing matrix. Simply press a routing junction to enable the source audio signal to flow through. The route travelled by your audio signal will light up in green.

Note that you can also adjust the gain of your signal on each routing junction: Right click the junction to open up a small gain adjustment dialog. Use the keyboard or the scroll-wheel on your mouse to make an adjustment. Boosting the signal will be represented by a small plus within the junction marker, while cutting it will produce a minus.



6.5 The routing matrix

When your input routing has been connected to the routing matrix, you can select to which output(s) you route using this nifty control. Just like with the input routing, simply click a cell in this matrix to route the audio signal from the inputs on the left toward the outputs on the bottom right of the screen. Note that the path of your audio signal lights up in the matrix. If configured correctly the whole audio path through the system will now be marked in green, including the input and output labels. An example of 2 complete audio paths can be seen below.



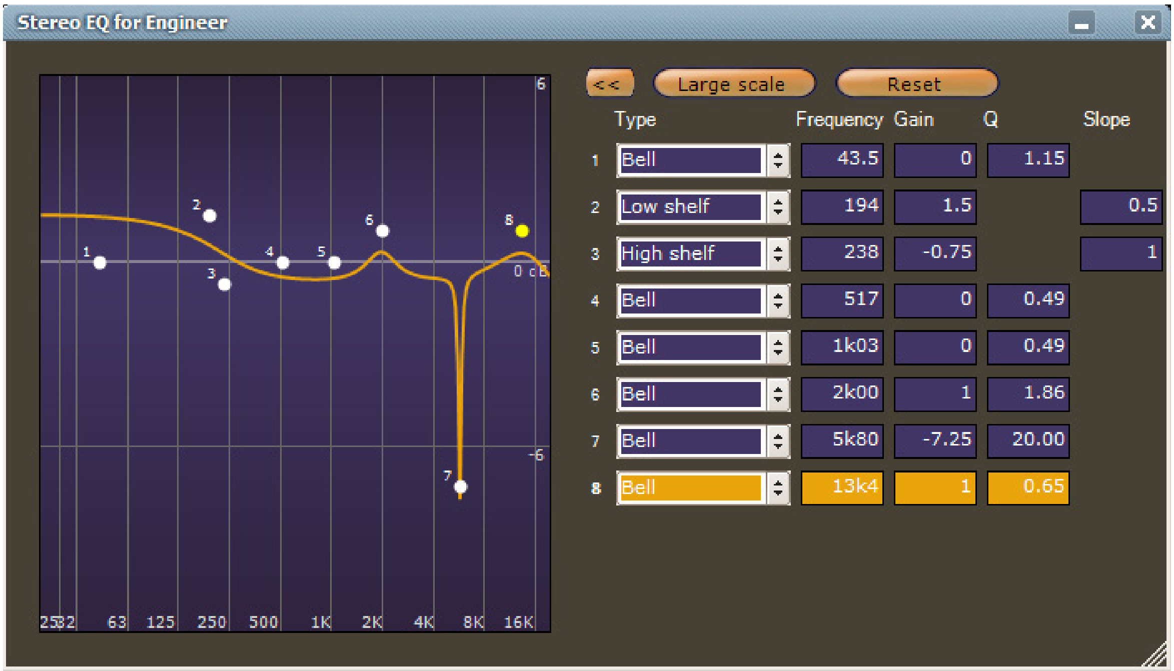
Each separate matrix cell can be right clicked to produce dialog that controls the gain of your signal. As with the input routing, a plus represents an audio boost while a minus represents a cut.

6.6 Input and output labels

For your convenience, the input and output labelling can be changed to whatever name you want: Simply click on the label and type any name you like, with a maximum of 12 characters. The labels will be saved when saving your project to file or synchronizing it to the Engineer.

6.7 Equalizers

The Engineer features stereo equalizers on each input and the Engineer algorithm, as well as mono equalizers on each output. Open one by simply clicking on the button that represents it.



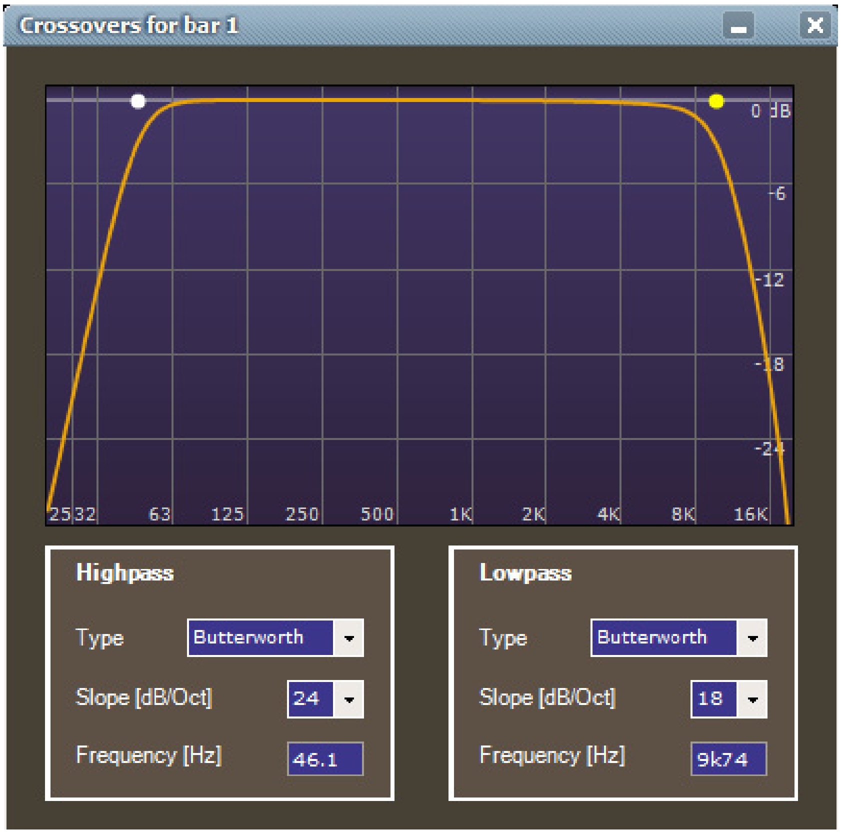
As you can see, the 8-band parametric equalizers can be configured by either text input or by using your mouse. The type of equalizer can be selected by clicking on the arrows beside the type name. You can select Bell-filters, shelving filters and band-pass/reject filters. Select "none" to bypass the selected filter. Frequency, gain and Q/bandwidth values can be selected by typing them in the boxes. Note that the band you're working on lights up in yellow.

When using the mouse to configure a filter, simply drag the graphical markers around while holding the left mouse-button to select the gain and frequency. While dragging, **you can use the scroll-wheel of your mouse to change the Q/bandwidth/slope factor of the selected filter**. While dragging, **holding down the CONTROL key locks the gain value, and holding down the SHIFT key locks the frequency value.**

You can use the reset button to bypass each filter's type. All other parameters will remain the same, meaning you only have to change each filters type to get your original settings back. By default, the graphical view is set to a maximum of -12/+6 dB. Click the "Large scale" button to change this to -45/+12 dB. Clicking the button a second time will revert back to the original view. Note that you can resize the entire window to get a better view of your settings by dragging the arrow mark in the bottom right corner. You can even hide the text input on the right of the screen by pressing the "<<" button. This way you can easily leave it in a corner of your workspace, as a reminder of your settings. You might want to copy the settings for a specific EQ to another channel. To do this, simply right click anywhere in the window, or even the button that opens the window, and select paste in the target window.

6.8 Crossovers

For every output channel a crossover filter can be set. Click on one of the crossover buttons from the routing and settings window to open the crossover window.



This window is pretty straightforward: Select a high and/or low-pass filter from a variety of filter-types and set its frequency by using mouse or keyboard. The steepness of the filter can be selected in the "Slope" selection box. Like equalizers, crossovers can be copied and pasted by right clicking on their respective windows or buttons.

6.9 Output settings

You may have already taken a look at the output parameters screen accessible from routing and settings window, if you haven't please do so now.

As you can see above, all important settings to make sure your speakers stay intact and sound their best are here: Gains, delays, phase inverts and limiters.

The limiters come with reduction indicators and the current level of each channel is also visible. Each channel also features an output mute, click the speaker-icon beside the channels label to (de)activate them. Note that the attack time for the limiter is set as a function of its release time.

All output parameters can be copied and pasted to any of the other outputs in the same way as the Crossover and EQ parameters. Place the cursor over the label at the top of the channel you wish to copy, right click and select Copy. Move to the output channel to which you would like to paste the settings and again right click on the label and select Paste.

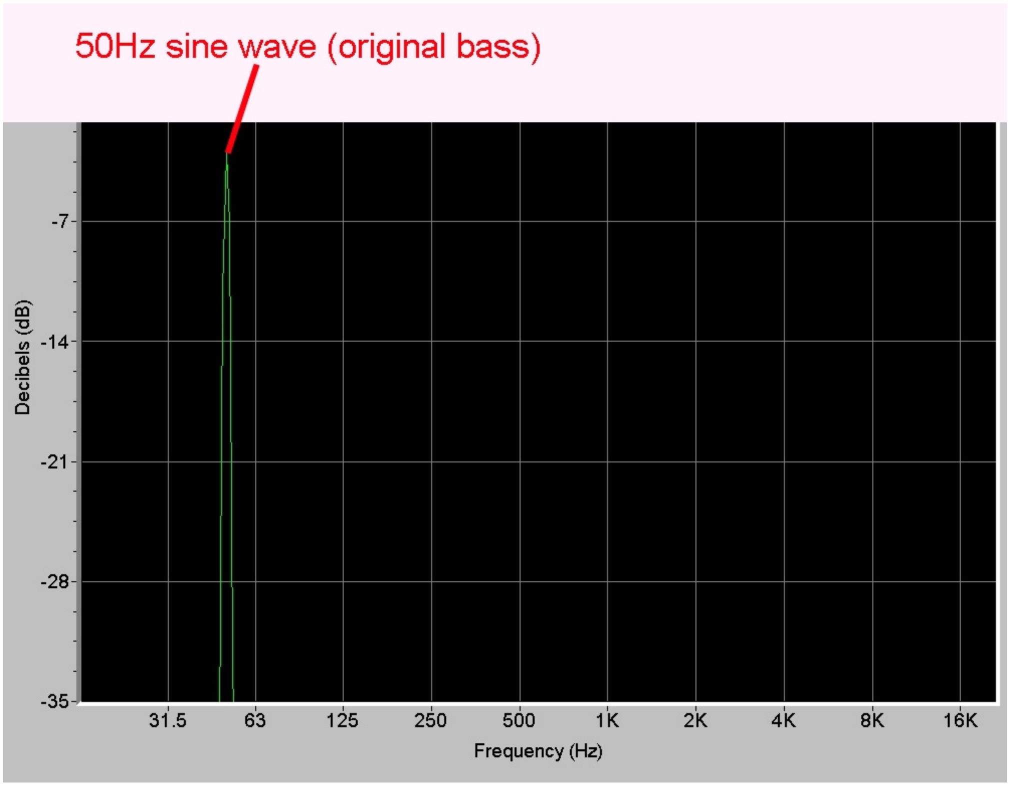
In addition it is possible to copy and paste all output parameters; crossover, EQ, gain, delay and limiter simultaneously. Hold the Control button down before right clicking over the label of the output channel to be copied and select Copy Channel. Keep the Control button down and right click over the label of the channel to which you would like the parameters pasted and select Paste Channel.

7. Basscreator

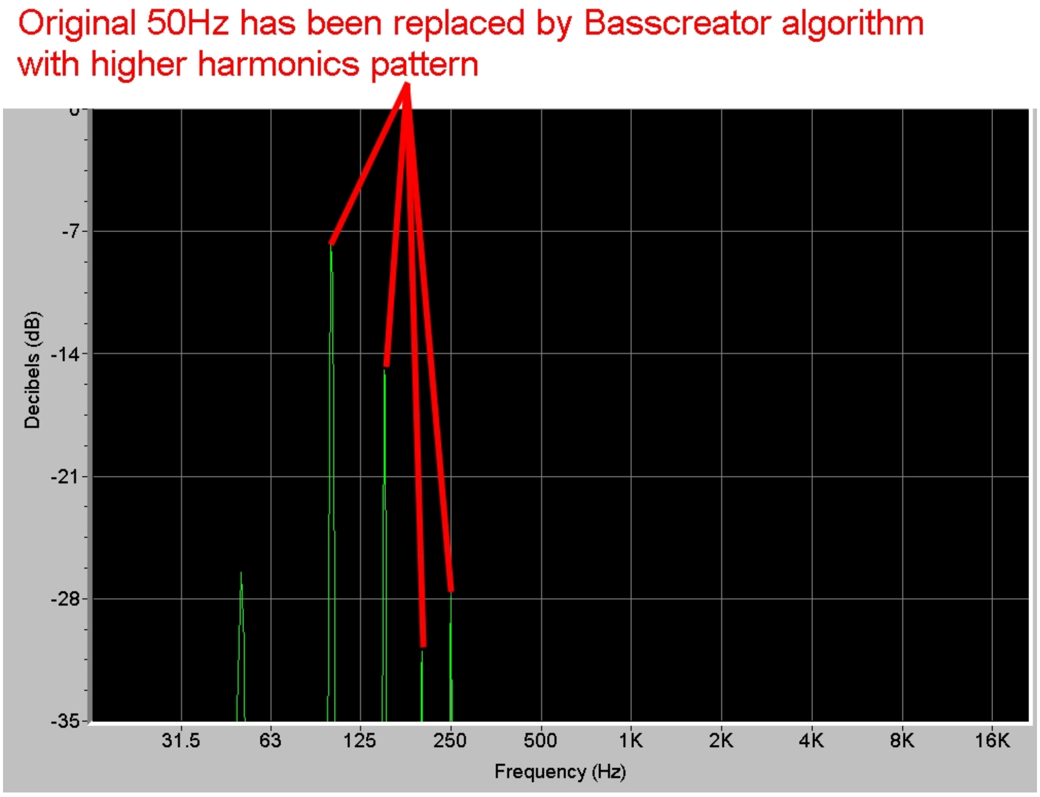
The Basscreator algorithm creates a virtual bass based on the "lost fundamental" principle: It creates a range of higher harmonics of a narrow frequency band around a used specified centre frequency. This is a psycho-acoustical effect that fools your brains to think something is going on in the bass frequency range. There are numerous advantages to the algorithm, you can:

- Replace some of the original bass in the signal with virtual bass to avoid noise emission problems
- Beef up small speakers to give a sense of low end that couldn't be achieved normally on the same speaker
- Reduce the amount of sub-woofers needed to save space and money
- Use the Basscreator for infill systems to maintain a consistent sound field with the added advantage of avoiding interference problems with the main systems sub-woofers (the perceived bass is in a different frequency range)
- Use the Basscreator on small line-array systems to bring the bass in the frequency range where the line-array is actual capable of behaving as a line-array.

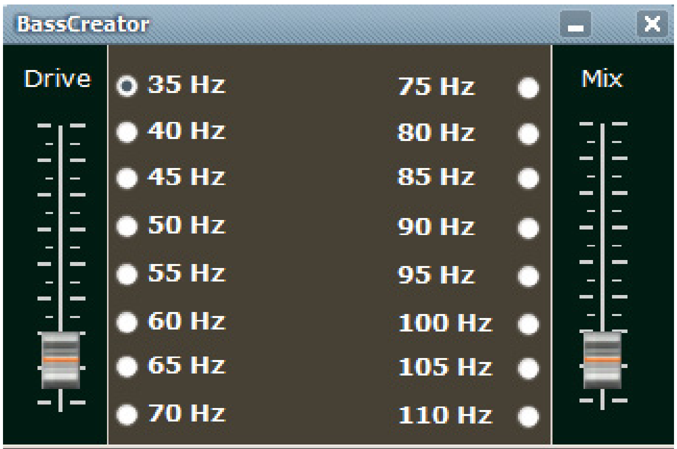
The pictures below give an impression of what happens within the Basscreator algorithm. If we put in a sine wave with a frequency of 50 Hz and we put the output of the device in an analyzer, we see the sine wave as a single spike.



Now if we put that same 50 Hz sine wave through the Basscreator algorithm and analyse its output we then see the picture below. We see that the original 50 Hz sine wave has been replaced with higher harmonics on 100,150,200 and 250Hz in a certain ratio. To our brains, this harmonic pattern will "sound" as if there was a 50 Hz tone.



To start using the basscreator you should first route the inputs you want to use it on toward the Basscreator module in the Routing and settings window. Then click the "BCR" button in the same window to open up the Basscreator control window.



First, select the frequency you want to set as the **perceived** bass. The best value for this setting depends on the genre of music you'll be playing most regularly and the size of your speaker system.

A rule of thumb for this is to choose the frequency about one octave below the low frequency cut-off of your speaker. This means that if you have a speaker which is capable of producing 100 Hz, you should use the 50HZ setting on the Basscreator as a starting point.

If you have set the frequency, set the drive and mix levels. The drive level determines the amount of harmonics generated by the algorithm and determines the sound it will produce for a large part. The mix level sets the amount of added effect (the amount of virtual bass added to the original signal).

8. Engineer

One of the main features of the Martin Audio Engineer is the Engineer algorithm itself. Consider the Engineer algorithm as a real-life sound engineer who has been told what kind of sound you'd like from your sound system and will then make sure that this will be the case. The underlying technology is quite complex and for this reason the system has been split up into a simple and an advanced mode.

8.1 Concept

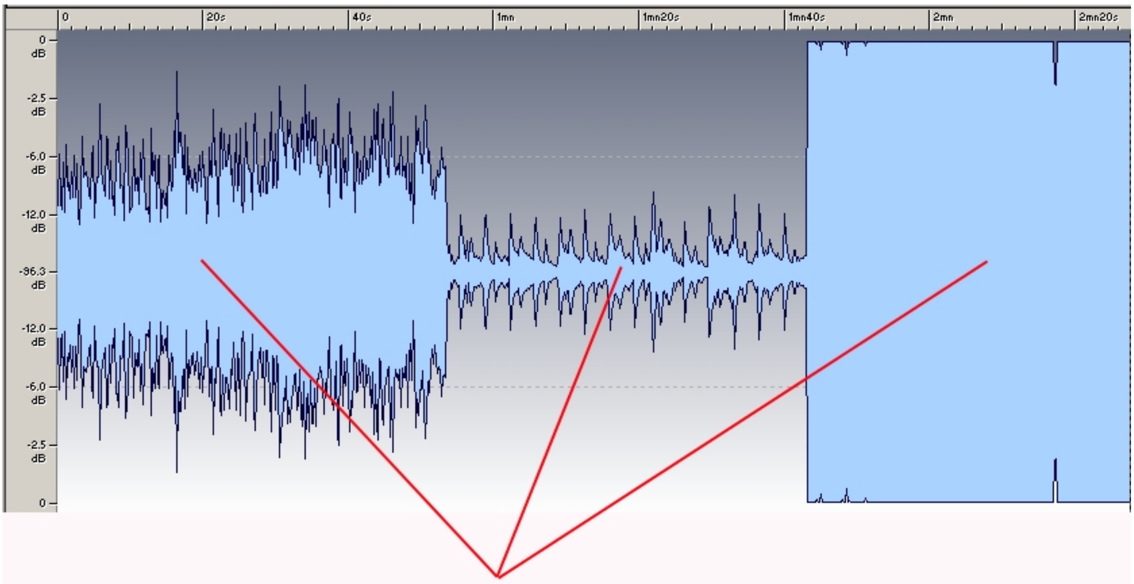
First, let's look at the reason why one would want to use the Engineer algorithm.

Say we have a situation where we have a venue (maybe a bar or a restaurant) with a MP3 player taking care of the music. We tell the machine to play us a nice lounge background program and the following happens:

-First we get a nicely produced music track which has nice dynamics and a proper mastering volume, everything is fine and we set the total volume to the level we like in the venue.

-Then the next track is very quiet with very little bass and treble so suddenly the music in the venue is 'gone', totally drowned in the noise of talking people. A little annoyed we turn the volume up to the desired level and add some bass and treble until we reach the desired sound again.

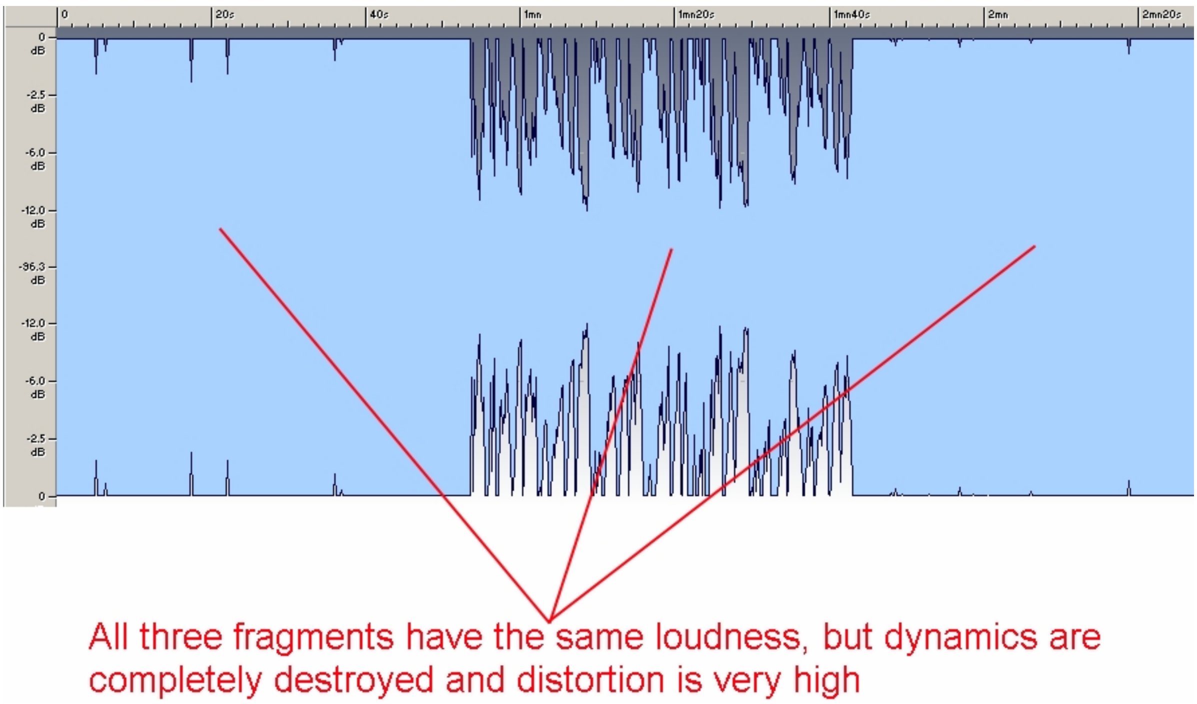
- Then suddenly a new song starts and this song is a very modern track which is mastered in such a way that no dynamics are left at all. So suddenly the music feels 8 times as loud as you intended it and everybody in the venue is putting their fingers in their ears and looking disturbed and due to the massive amount of bass in this track their drinks are falling off the table…



Three fragment have large differences in loudness and tone

This example describes a very common situation which happens every day in venues all over the world. This problem is actually not anybody's fault; it's just the way things evolve through the years and the result of different opinions of how things should sound. But it is a problem that we can control.

The way to overcome this problem is what we call the 'radio station solution'. What happens at the radio stations is that very strong multiband-limiting is applied to all the program material, so that every song will sound the same and is as loud as the rest, no matter if it is a middle of the road ballad or a techno track. This technique works quite well, but the drawback is that the music loses all it's dynamics, large amounts of harmonic distortion are added and the carefully mixed material is losing it's balance completely (hey, I didn't know that the tambourine in this song was louder than the lead vocal....). See the resulting waveform from our example treated with the "radio solution" in the picture below.



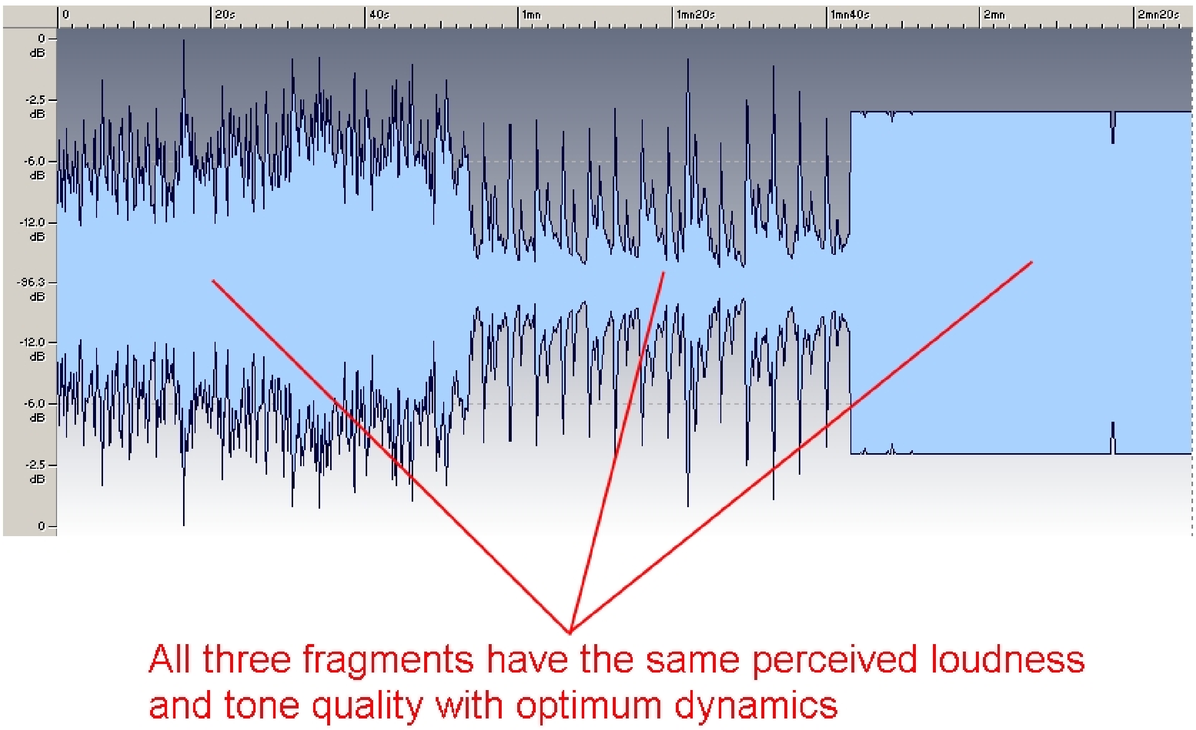
The best way to overcome these problems is to have a real-life flesh and blood sound engineer standby 24-7. This way we would always have someone monitoring the program material who would fix every change in loudness or tone immediately. Too bad that this is way too expensive for 99% of the venues.

What we have done to solve this issue for you is create an algorithm that reacts in the same way as a normal sound engineer would do, hence the name ‘Engineer’. The Engineer works with human-based perceptive algorithms which react the same way as a normal human being would do. This means that the algorithm reacts to perceived loudness instead of electrical loudness, so that songs with a lot of compression which sound loud will be turned down more than more dynamic songs, so that the overall perceived level stays the same.

The Engineer algorithm features an automated volume control, an automated low EQ, an automated high EQ and a multiband end-stop limiter to catch extreme peaks in the program material.

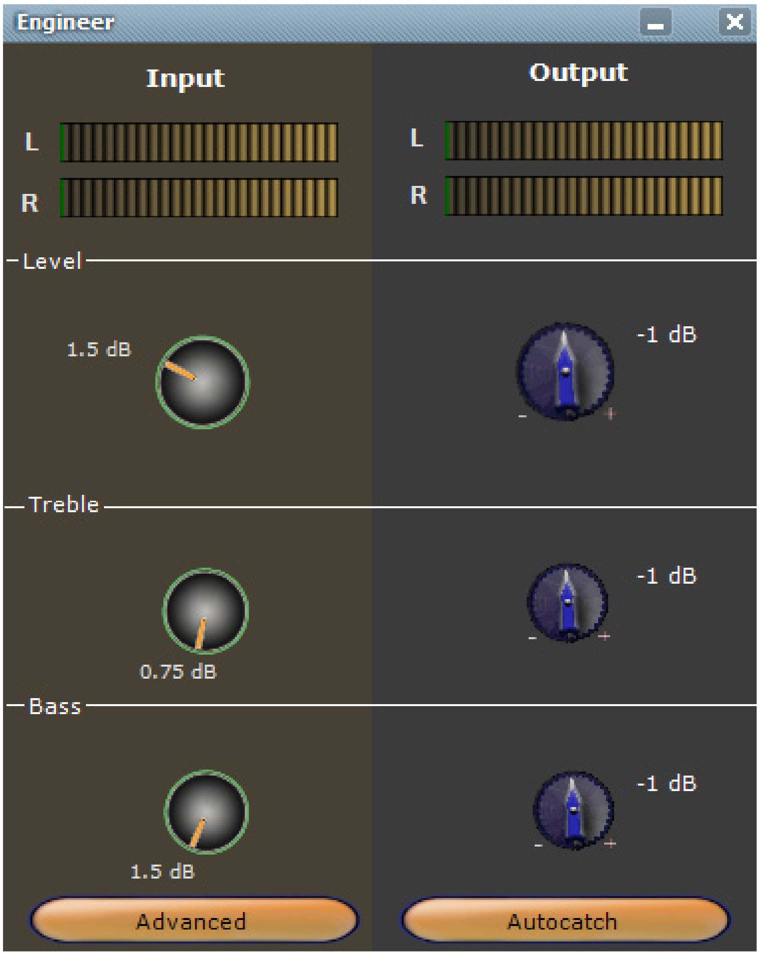
Just like a normal human being would do, the algorithm listens to the incoming signal for a couple of seconds, then decides if it sounds too loud or too weak, adjusts the volume, listens to the amount of bass and treble in the material and, if necessary, changes it. If the volume of the material has a sudden large boost in volume (for instance if a DJ is fooling around), these peaks will be handled by the end-stop limiter so that the algorithm has the time to slowly lower the volume.

The picture below is the output of the engineer algorithm with our example input. In this picture we see that the level and tone differences have been solved in an elegant way without destroying the dynamics or adding distortion, and that the last piece of heavily compressed music has been lowered in volume to give it the same perceived volume.



8.2 Engineer simple

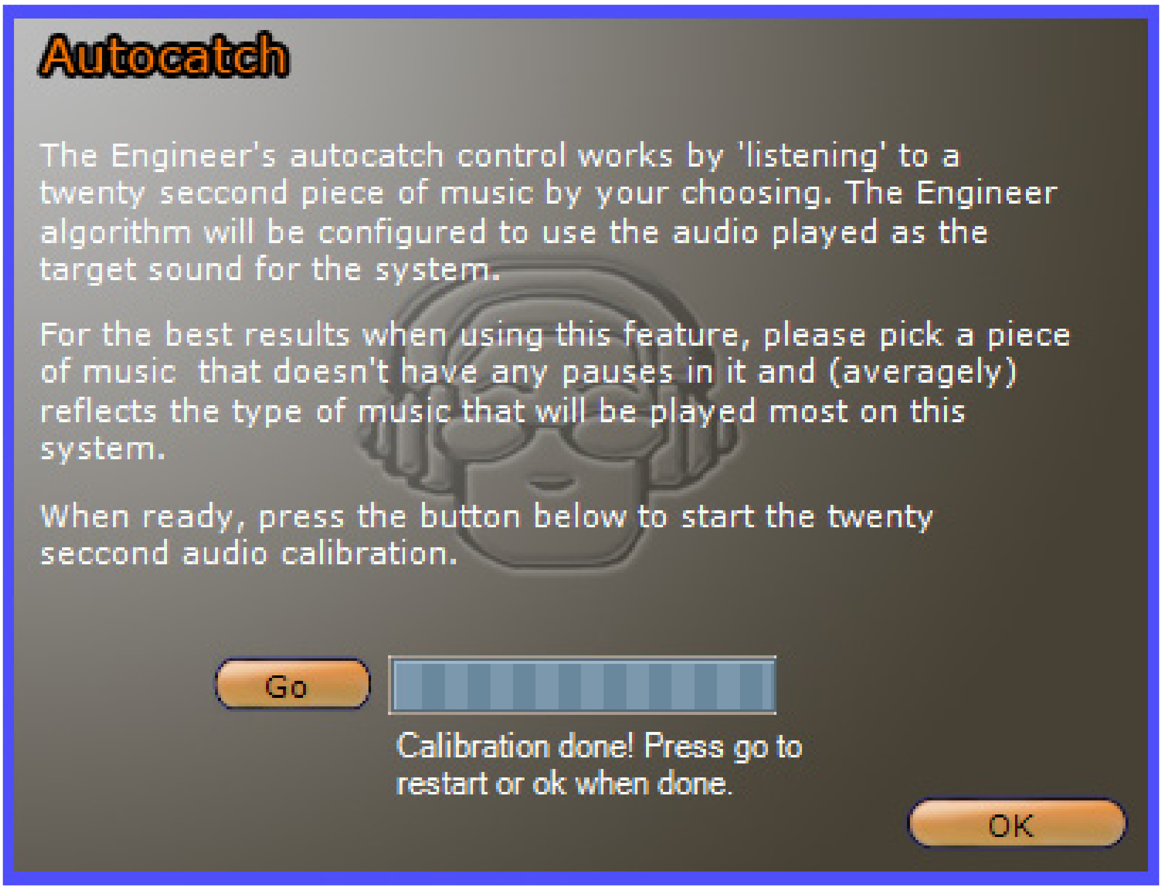
To try out the algorithm, route a sound signal to the Engineer in the routing screen. Make sure that the routed signal enters the algorithm at a minimum level of at least -30dBU. Then click the picture of the sound engineer to open the engineer simple control window.



This windows shows you the input and output levels and gives you a basic feel of what's going on in the algorithm: The three compasses on the right side of the window show the amount of gain boosted or cut from the input signal, for the overall level as well as specifically for the high and low tones.

The three knobs on the left side of this screen give you the opportunity to give a user offset to the sound made by the engineer, so you can decide that you like the setting but you would like to have a bit more bass or treble. You can consider these knobs a the 'User Control' knobs, because the algorithm keeps the sound consistent, here the user can determine that overall the sound needs some more bass or treble.

The large "Autocatch" button sets the absolute offset based on the current signal. Press it to open up the Autocatch window:

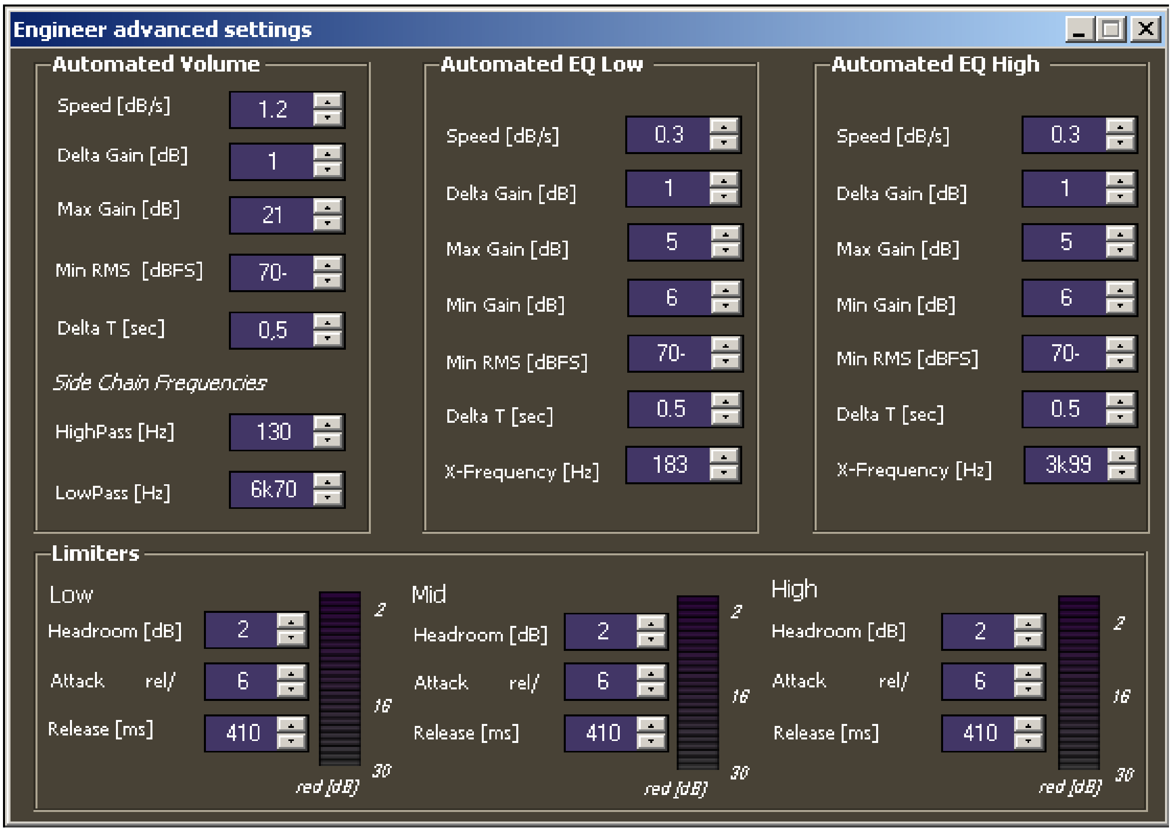


As the window above explains, you can use this feature to set the target sound for the algorithm, it's best to use the type of music that will be played most on your system. If the advanced settings for the algorithm are set to their defaults this will usually be correct for the Engineer to function as it should. If not, we can tweak the advanced settings for the algorithm via the "Advanced" button in the Engineer simple window. **Be sure to hit the Autocatch again after any changes made in the advanced screen.**

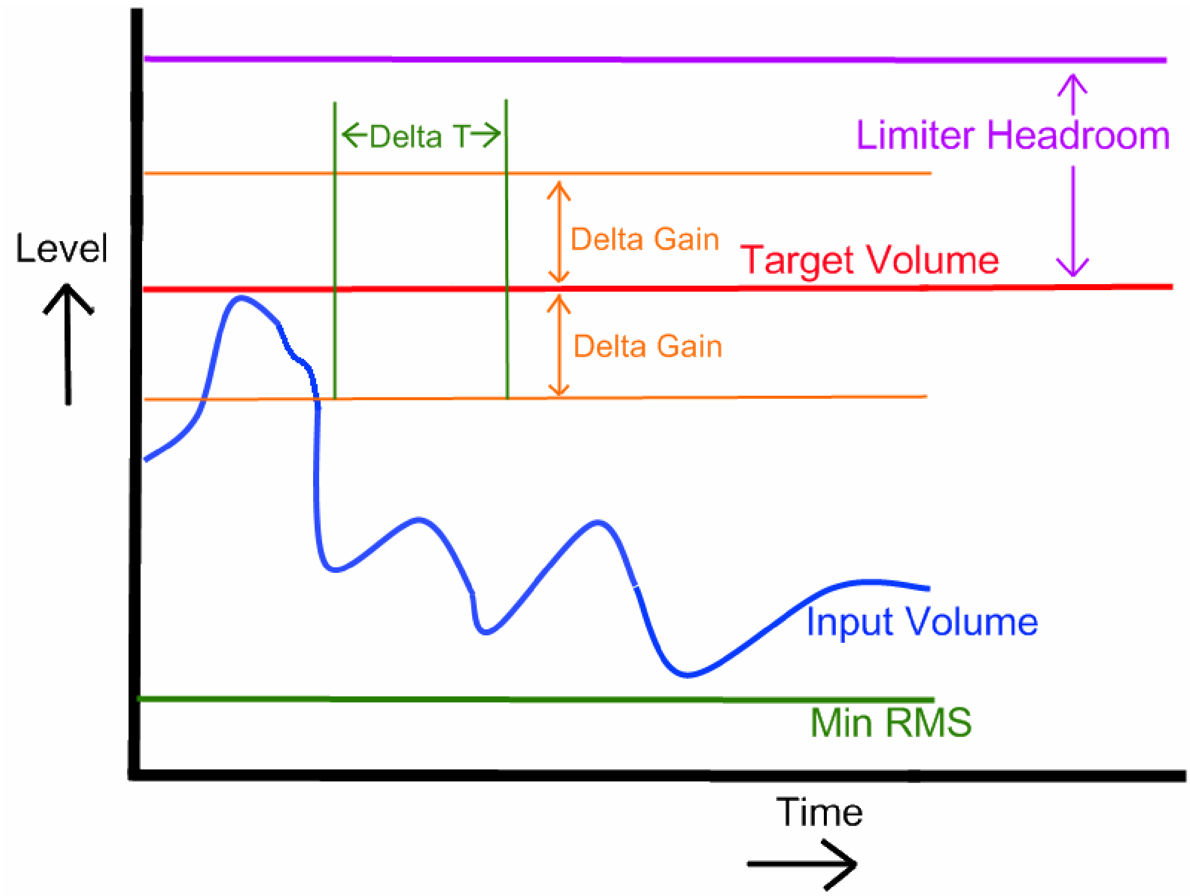
8.3 Engineer advanced

The Engineer advanced window consists of four separate components:

-Automated volume control
-Automated low EQ control
-Automated high EQ control
-Endstop limiters



Let's take a moment to go through all the parameters to see what they actually stand for.

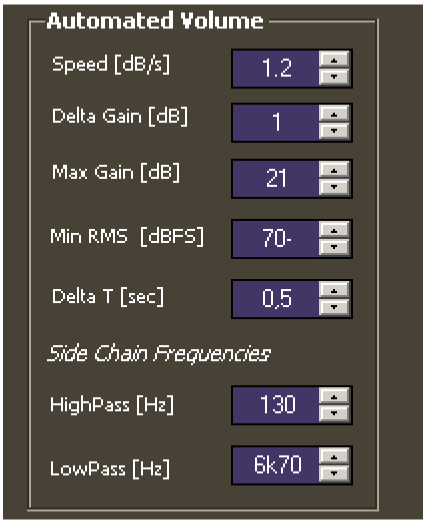


In this simplified diagram we can see what the functions of the Delta Gain, Delta Time, Min RMS are.

-The Blue line represents the RMS level of the incoming program material.

-The Red line marked 'Target Volume' is set by pressing the Autocatch button in the Engineer simple screen, this is the level the algorithm uses as a reference.

8.4 Automated Volume



8.4.1 Speed

The Speed parameter determines the speed with which the algorithm changes it's volumes

8.4.2 Delta Gain

The Delta Gain parameter determines the offset to the Target Volume needed for the algorithm to start altering the volume. E.g. when the Delta Gain is set to 3, the input RMS volume has be more than 3dB over or under the Target Volume for the algorithm to change the volume.

8.4.3 Delta T

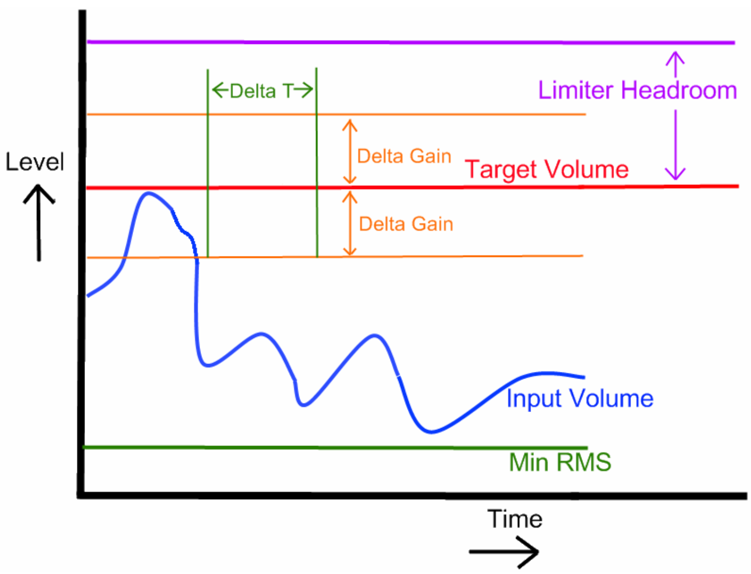
The Delta T co-operates with the Delta Gain parameter. This means that the difference between the Input volume and the Target Volume has to be larger than Delta Gain for a Delta T amount of time before the algorithm starts altering the volume. E.g. When Delta Gain is set to 3dB and Delta T is set to 2 seconds, than the difference between the Input Volume and the Target Volume has to be more than 3dB for at least 2 seconds before the algorithm starts altering the volume.

8.4.4 Min RMS

The Min RMS parameter determines the threshold in volume before the algorithm starts working. When the input volume is below this value the volume will be set to 0dB. This is necessary in case there is no input signal present and you don't want the algorithm to boost the volume all the way up because it thinks that it's just a very weak signal.

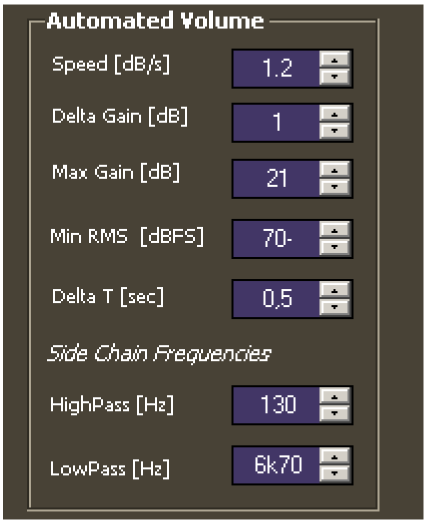
8.4.5 Max Gain

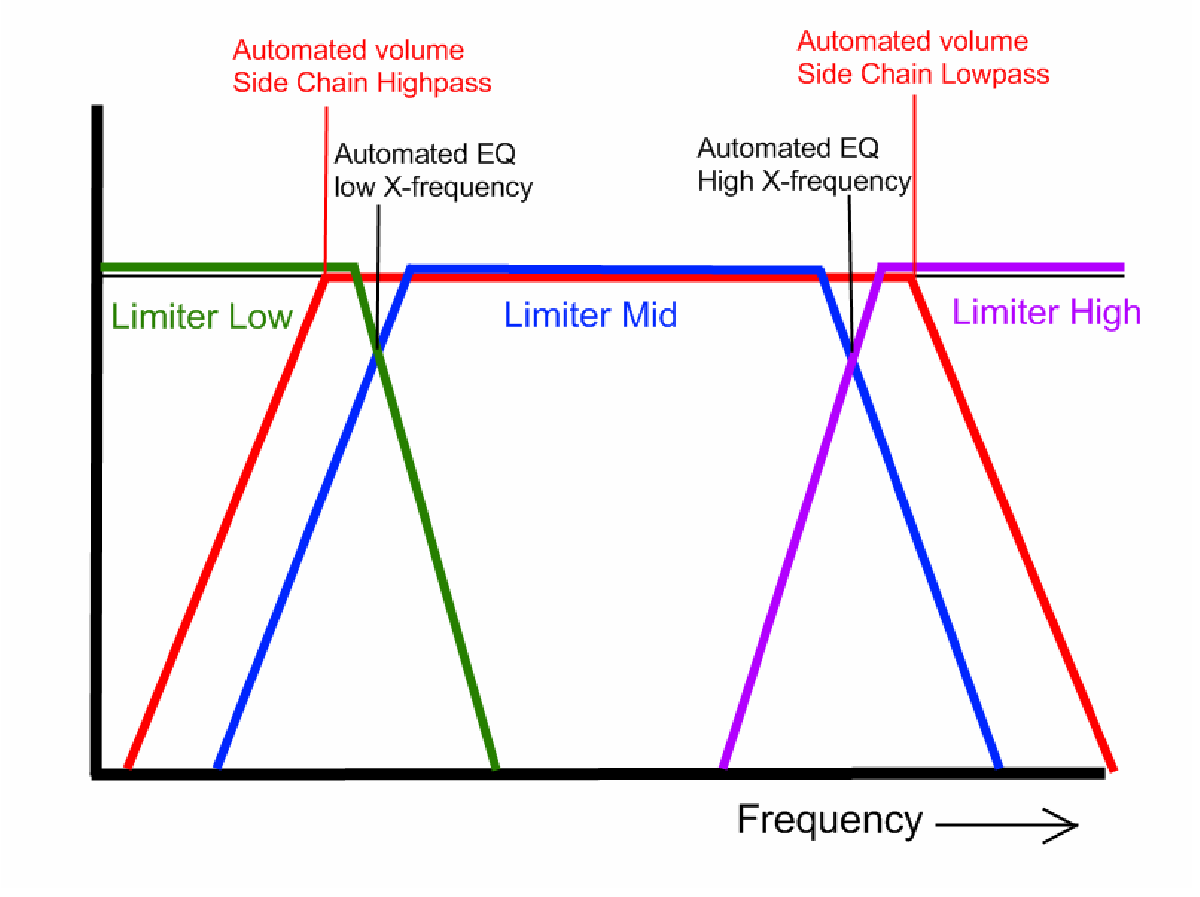
The Maximum amount of Gain the algorithm is allowed to boost the input signal. E.g. if you set this parameter to 0dB, then the algorithm will only take down the volume and not boost it.



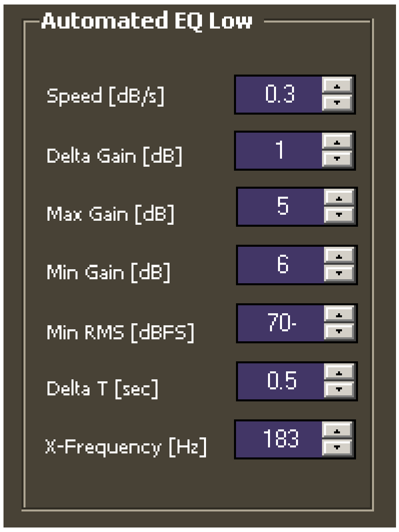
8.4.6 Side Chain Frequencies

The Side Chain Frequencies determine the frequency range to which the Automated Volume control 'listens'. E.g. when you set the HighPass parameter to 130Hz and the LowPass parameter to 6k7Hz, then the algorithm will listen only to frequencies between 130Hz and 6k7Hz.





8.5 Automated EQ Low



8.5.1 Speed

The Speed parameter determines the speed with which the algorithm changes its gain.

8.5.2 Delta Gain

The Delta Gain parameter determines the offset to the Target Volume needed for the algorithm to start altering the gain. E.g. when the Delta Gain is set to 3, the input RMS volume has be more than 3dB over or under the Target Volume for the algorithm to change the gain.

8.5.3 Delta T

The Delta T co-operates with the Delta Gain parameter. This means that the difference between the Input volume and the Target Volume has to be larger than Delta Gain for a Delta T amount of time before the algorithm starts altering the gain. E.g. When Delta Gain is set to 3dB and Delta T is set to 2 seconds, then the difference between the Input Volume and the Target Volume has to be more than 3dB for at least 2 seconds before the algorithm starts altering the gain.

8.5.4 Min RMS

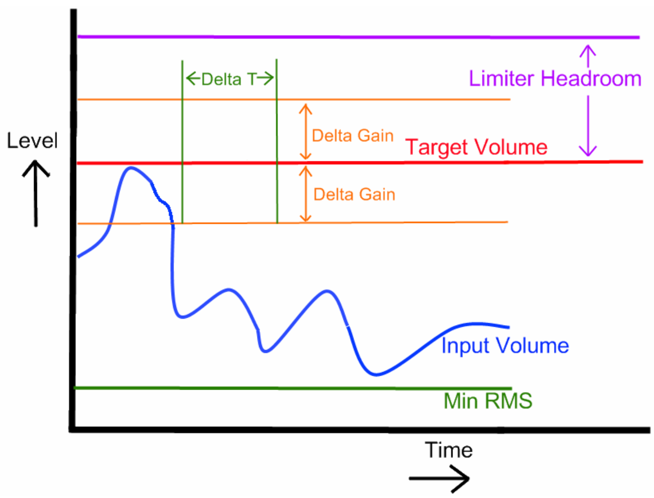
The Min RMS parameter determines the threshold in volume before the algorithm starts working. When the input volume is below this value the volume will be set to 0dB. This is necessary in case there are breaks in the music or when the DJ makes an extreme cut on his EQ, we don't want the algorithm to compensate for this but instead just go back to 0 dB.

8.5.5 Max Gain

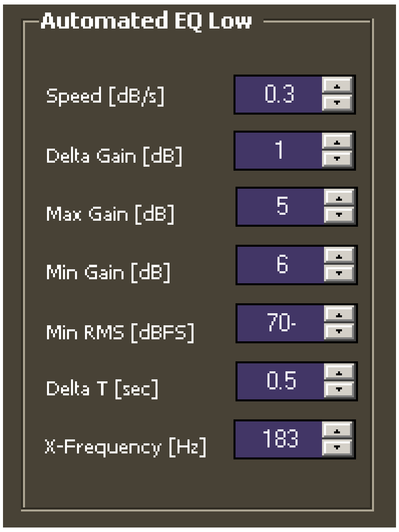
The Maximum amount of Gain the algorithm is allowed to boost the EQ gain.

8.5.6 Min Gain

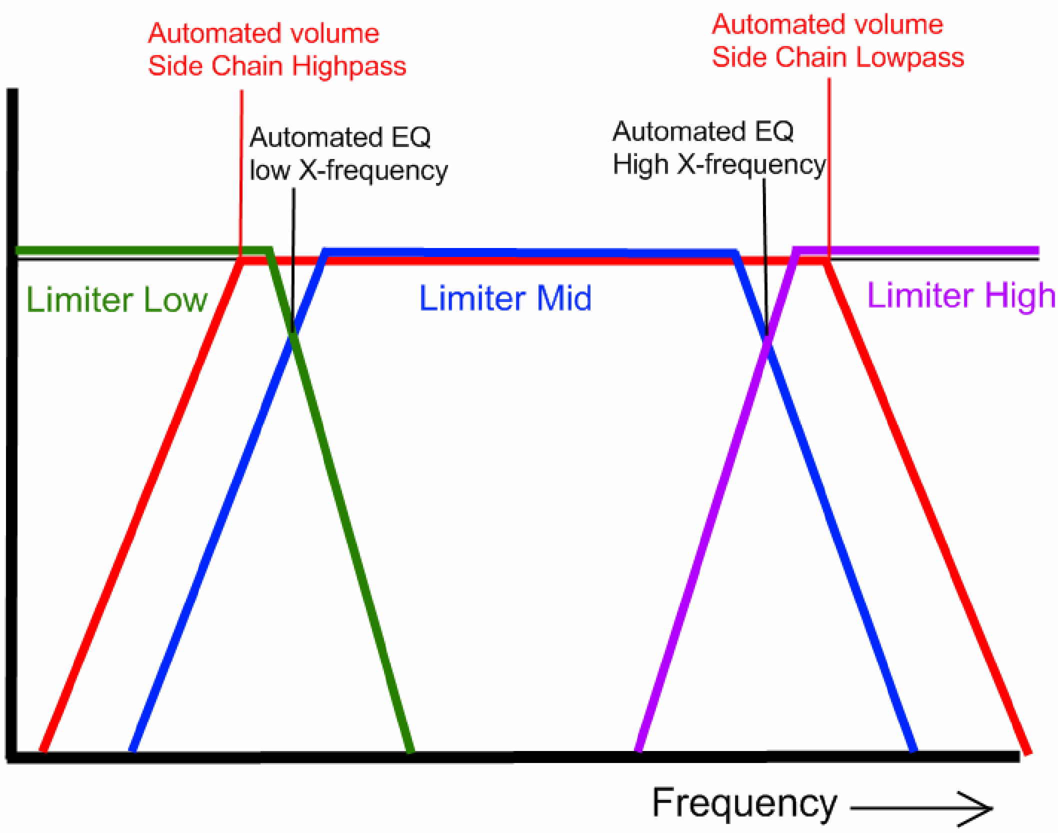
The Maximum amount of Gain the algorithm is allowed to cut the EQ gain.



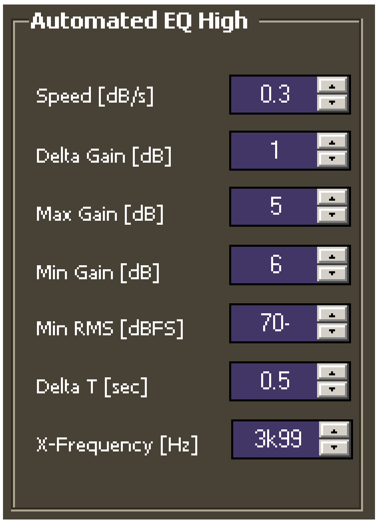
8.5.7 X-frequency



The X-Frequency determines the frequency range the Low EQ will listen to and alter. E.g. When this parameter is set to 183Hz, the Low EQ will work on frequencies below 183Hz. This frequency also represents the X-over frequency for the limiters Low-to-Mid band.



8.6 Automated EQ High



8.6.1 Speed
The Speed parameter determines the speed with which the algorithm changes its gain.

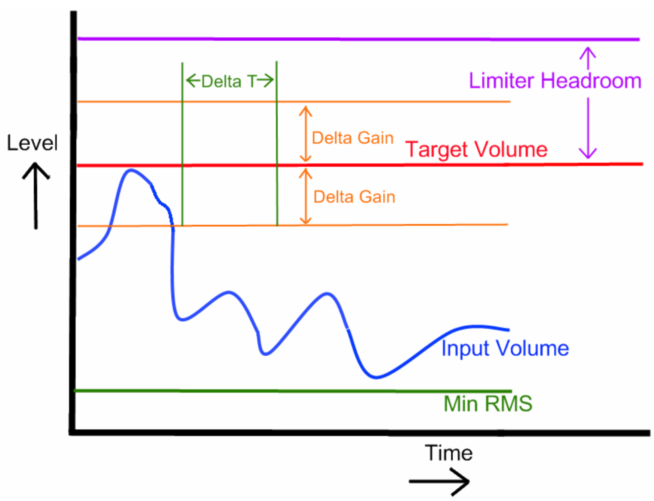
8.6.2 Delta Gain
The Delta Gain parameter determines the offset to the Target Volume needed for the algorithm to start altering the gain. E.g. when the Delta Gain is set to 3, the input RMS volume has be more than 3dB over or under the Target Volume for the algorithm to change the gain.

8.6.3 Delta T
The Delta T co-operates with the Delta Gain parameter.
This means that the difference between the Input volume and the Target Volume has to be larger than Delta Gain for a Delta T amount of time before the algorithm starts altering the gain. E.g. When Delta Gain is set to 3dB and Delta T is set to 2 seconds, than the difference between the Input Volume and the Target Volume has to be more than 3dB for at least 2 seconds before the algorithm starts altering the gain.

8.6.4 Min RMS
The Min RMS parameter determines the threshold in volume before the algorithm starts working. When the input volume is below this value the volume will be set to 0dB. This is necessary in case there are breaks in the music or when de DJ makes an extreme cut on his EQ, we don't want the algorithm to compensate for this but instead just go back to 0 dB.

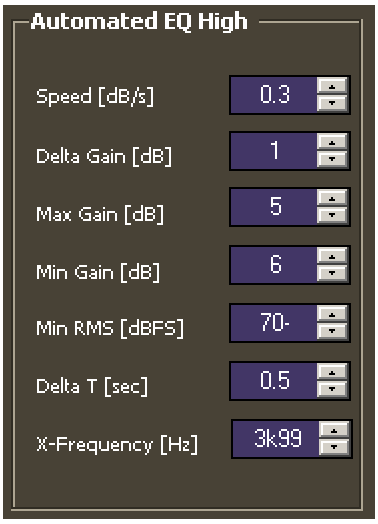
8.6.5 Max Gain
The Maximum amount of Gain the algorithm is allowed to boost the EQ gain.

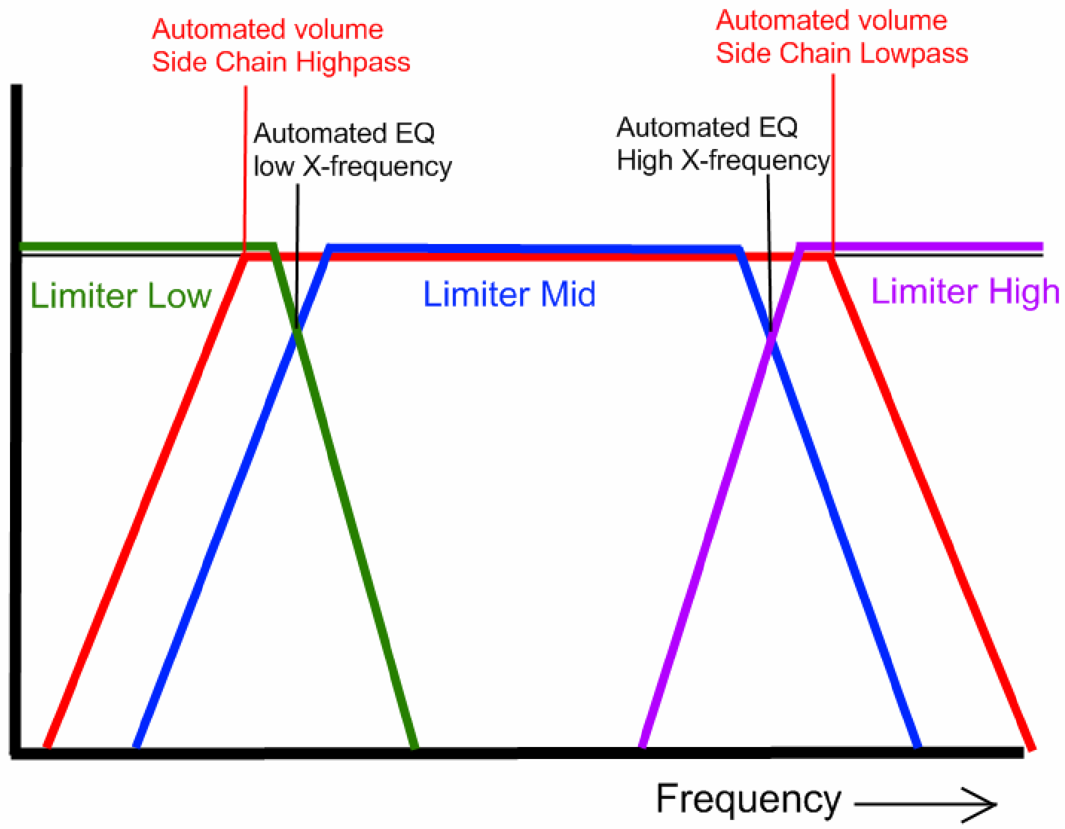
8.6.6 Min Gain
The Maximum amount of Gain
the algorithm is allowed to cut
the EQ gain.



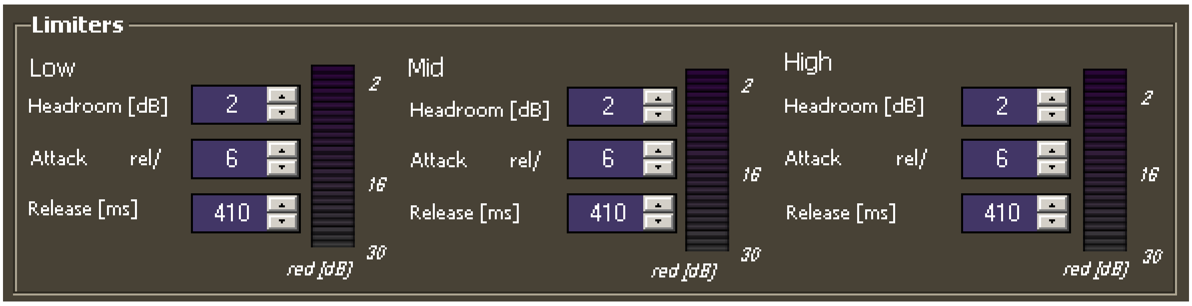
8.6.7 X-Frequency

The X-Frequency determines the frequency range the High EQ will listen to and alter. E.g. When this parameter is set to 3k99Hz, the Lo EQ will work on frequencies above 3k99Hz. This frequency also represents the X-over frequency for the limiters Mid-to-High band.

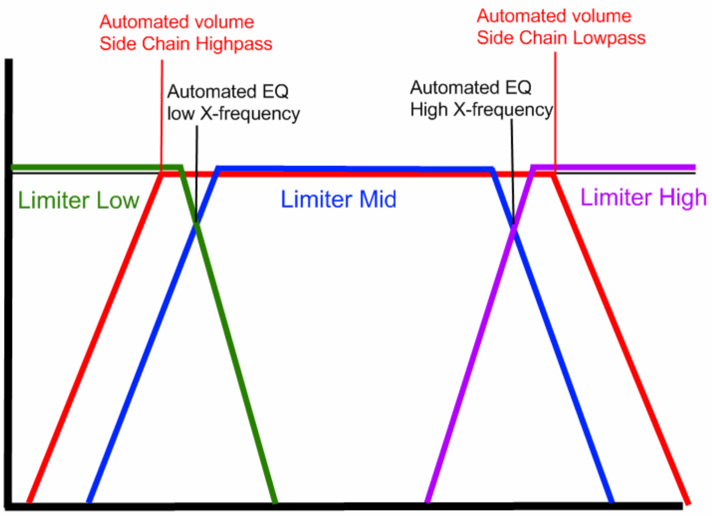




8.7 Limiter

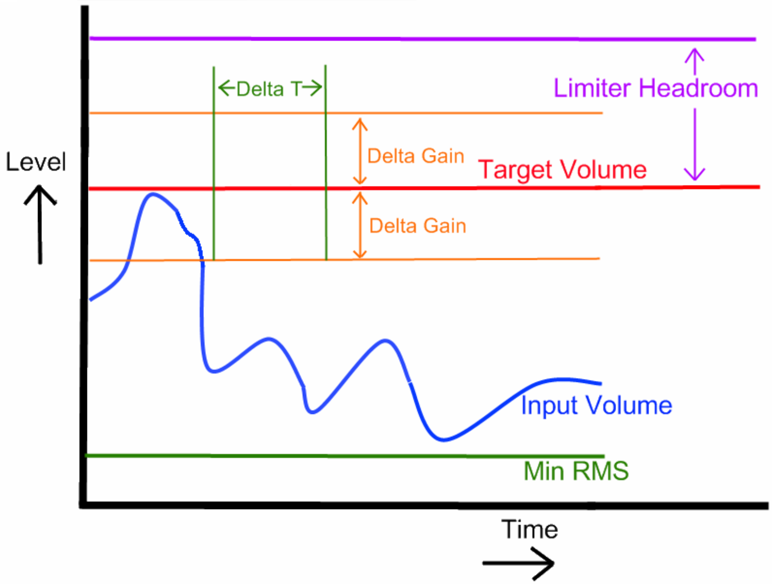


At the end of the Engineer Algorithm there is a three-band limiter for the Low-Mid-High bands.
The X-over frequencies between the bands are determined by the X-frequencies from the Automated EQ section.



The Headroom parameter sets the limiter threshold as a function of the Target Volume. E.g. when the Headroom parameter is set to 3dB, then the level is allowed to be 3dB higher than the target volume before the limiter starts working.

The limiters release time is set in milliseconds, the limiters attack time is set as a function of the release.

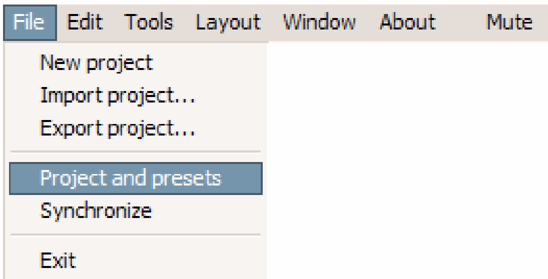


9. Project-and preset management

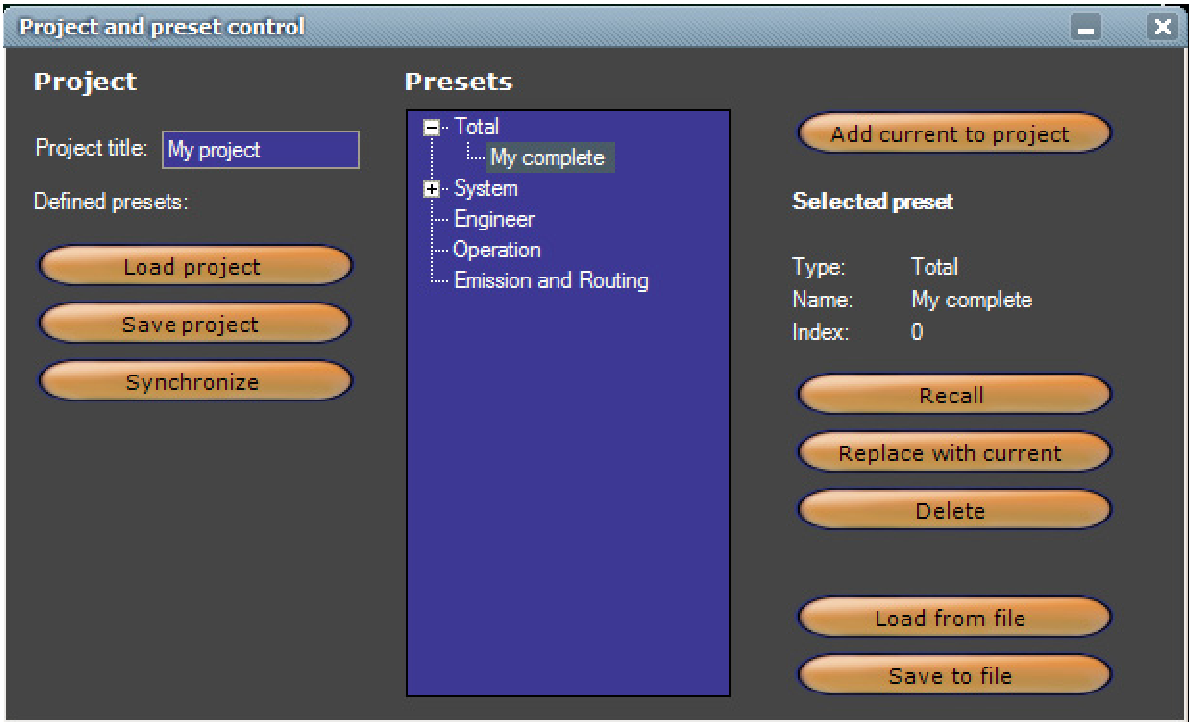
An Engineer project can be defined as a set of multiple presets that can be recalled at will. Each preset contains a set of configuration parameters, as described above. Presets can be made up out all of these parameters, or a limited set of parameters for a specific part of the device:

Preset type	Definition
System settings	All output parameters: Crossovers, output EQ's, output delays, gains and limiters
Engineer settings	All parameters described in the Engineer advanced control window
Operation settings	All input parameters: Input gains, EQ's, simple settings for the engineer (bass, treble and gain)
Noise emission settings	All routing settings and basscreator settings

Of course there's always the possibility of storing presets as a whole by using "total" presets. Presets can be recalled using either the software or the Engineer remote control. In the software, you can access the project and preset window from the file menu:

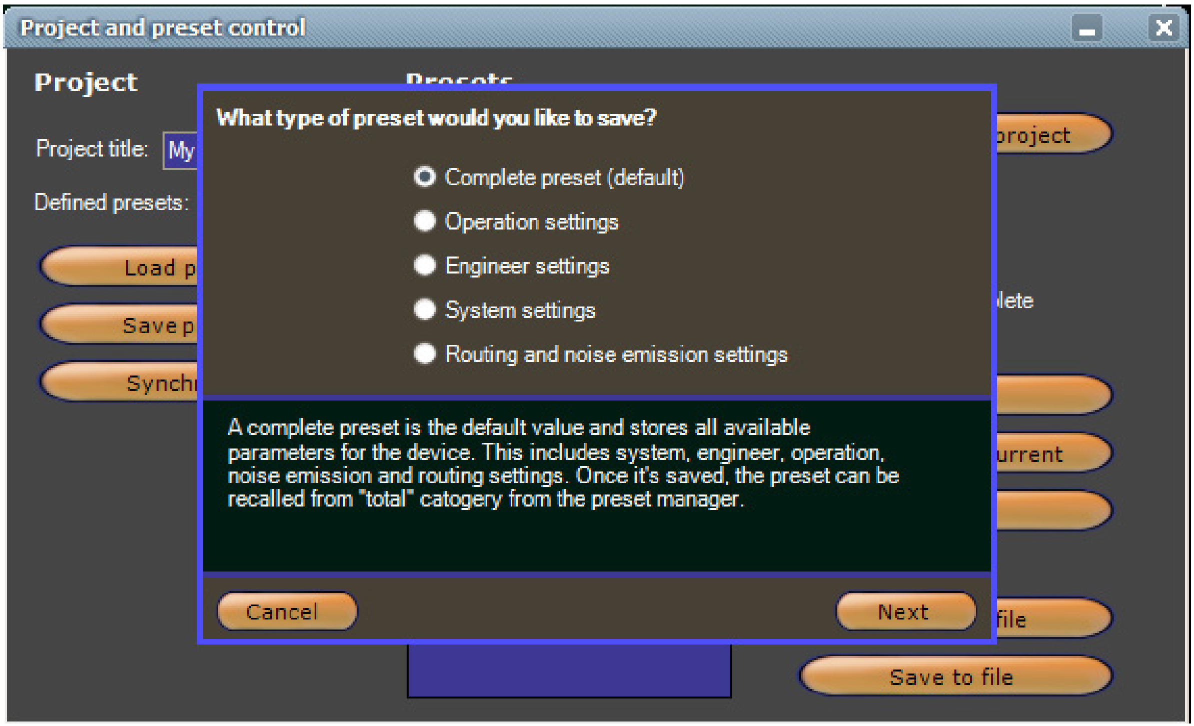


From this menu, shortcuts to save (export) or load (import) an existing project from file are also available, as well as an option to synchronize your device to the software. Selecting "project and presets" will open up the following window:



This window allows you to manage up to 28 presets. The presets are presented in the centre of the window and are displayed in a tree-view. Each of the nodes in this view represents one of the preset types as described above. A plus sign before the type name indicates one or more presets of this type are present in your project. Click on it to display each preset of this type, and click on the preset to view the details on the right of the window. The buttons on the right can be used to recall, delete or replace the selected preset with you current workspace. The project will be immediately updated locally and on the device after clicking them.

The button on the top-right of the window adds your current workspace to the project as a new preset. After clicking it, you will need to select the type for the new preset.



The short description in the bottom half of this window describes exactly which parameters will be stored in the new preset. After selecting a name for the new preset, it will be stored to the Engineer and you can use it straight away.

Besides storing your preset to the Engineer, you can also export or import it from a file. Please note that preset files cannot be loaded as a project: They can only be imported to an existing project.

The project settings to the left of the window are quite straightforward: You can save or load a collection of presets from/to a file or synchronize the project with the device. The project title is there for convenience, it will be stored to the device so you can recognize the project stored on it easily.

Martin Audio Presets

The Engineer Software CD ROM also contains a folder containing a library of System Presets with parameters for all Martin Audio cabinets. These are also available on the Martin Audio User Guides and from the Martin Audio website; www.martin-audio.com.

All Presets contain Crossover, EQ and Limiter settings where applicable and all have the parameters for one cabinet only. Using the copy and paste function it is very easy to duplicate settings over to additional channels.

It is also possible to combine 2 or more different presets to build a project using a number of different Martin Audio speakers. To do this you use a combination of the Channel Copy and Paste function and the Replace Current button in the Project and Preset Control window.

As and example we will create a project using the parameters for an AQ210 sub in channel 1, AQ6's in channels 2 and 3, AQ12's in channels 4 & 5.

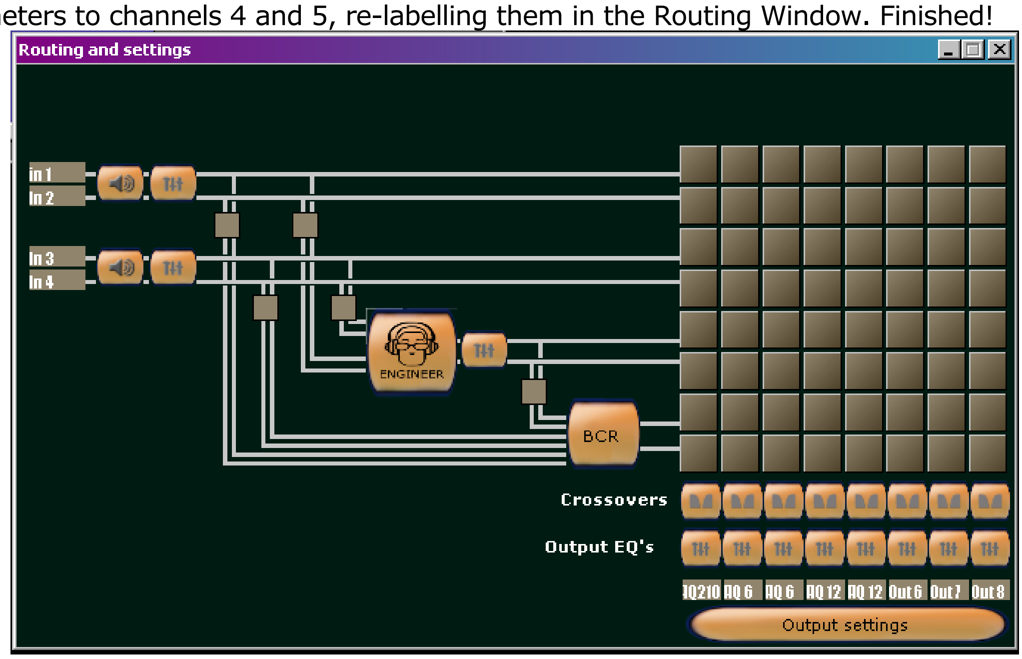
First, open the Project & Preset Control window and click on Load from File. Navigate to the CD or file location containing the Martin Audio Presets and select on AQ210 + AQ6,8,10,12 and click on Open. Engineer will ask if you wish to use the preset now, select 'No'. You will notice that the System Presets will now have a ⊞ by it to indicate that System Presets are present.

Repeat the operation for AQ6 and AQ12 Presets. Expand the System Presets by clicking on the ⊞ and select AQ6. Click Recall. Now open the Output Settings Window and holding Control down, right click over the Channel 1 AQ6 label and select Copy Channel.

Now Recall the AQ210 Preset, go back to the Output Settings Window and again holding down Control, right click over first channel 2 and then channel 3 labels and select Paste Channel. Note that pasting output channels does not copy the channel label so you will need to manually change this in the routing window.

Next go back to the Project and Preset Control Window and make sure that the AQ210 Preset is still highlighted, click on Replace with current. The AQ210 Preset now contains AQ6 parameters on channels 2 & 3. Now recall AQ12 and again copy the channel settings from channel 1.

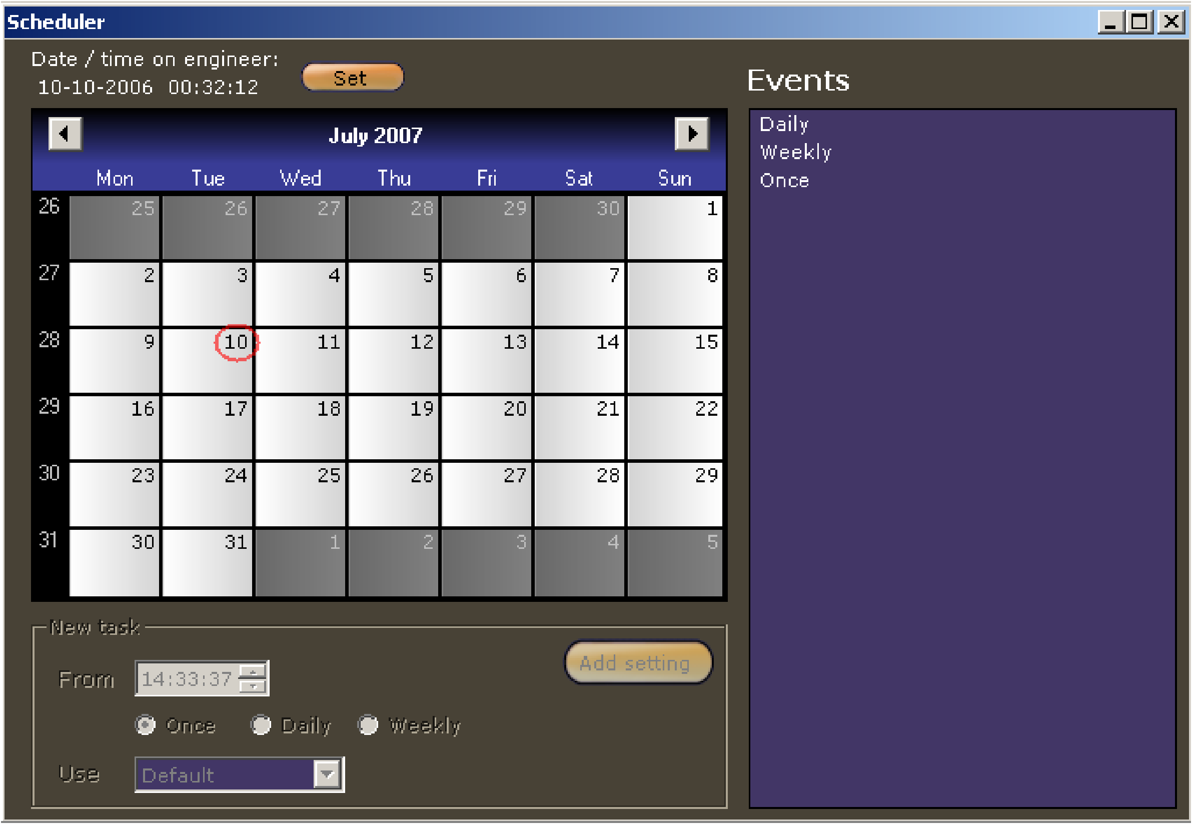
Again recall the AQ210 Preset and finally with the Control button down paste the AQ12 parameters to channels 4 and 5, re-labelling them in the Routing Window. Finished!



Whilst it is a little laborious to build a project this way, it is certainly quicker and easier than manually loading all parameters and potentially it is possible to design a system with parameters for different Martin Audio speakers on every output.

10. The Scheduler

The Engineer's scheduler module allows you to recall a specified preset to be activated at a fixed day and time. You can access it from the **tools** menu in the menu bar.



On the top of the window you can see the currently set date and time on the Engineer. Use the "set" button to set this to the time and date corresponding to the PC you're working from.

There are three different scheduler events you can create:

- ONCE, these are events which will be triggered ones per year on the corresponding date and time
- DAILY, these are events which will be triggered every day at the same time
- WEEKLY, these are events which will be triggered on a specified day in the week at the same time

By clicking on a date in the calendar, you can see what events are scheduled for that day.

10.1 One-time events

One-time events, called ONCE in the scheduler are meant for events you want to be triggered one time per year at a given date. This can be useful for e.g. a special party on New Years Eve.

To add a one time event to the scheduler:

-Click on the calendar to select the date you want the event to be triggered .

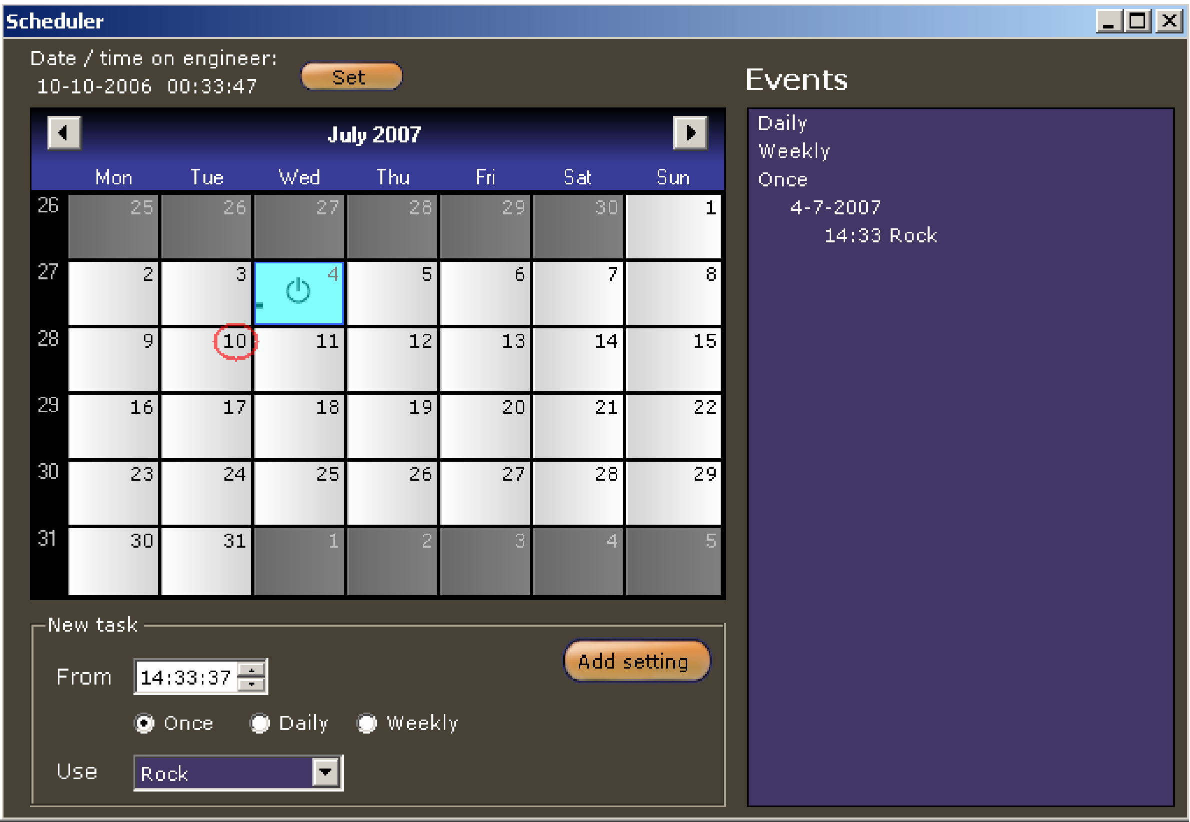
-Select the 'Once' option in the left bottom corner of the screen.

-Select the preset you want to be triggered.

-Click on the 'Add setting' button.

You now see the preset appear in the events list on the right side of the screen under 'Once'. You will also see the clock logo and a little black dot appearing in the calendar, indicating that a one time event is stored on that date.

To remove the event from the device, just click on the event in the events list and click 'delete'.



10.2 Weekly events

Weekly events are meant for events you want to be triggered weekly at a given day. This can be useful for e.g. a Drum & Bass evening the venue has every Thursday evening.

To add a weekly event to the scheduler:

-Click on the calendar to select the date you want the event to be triggered.

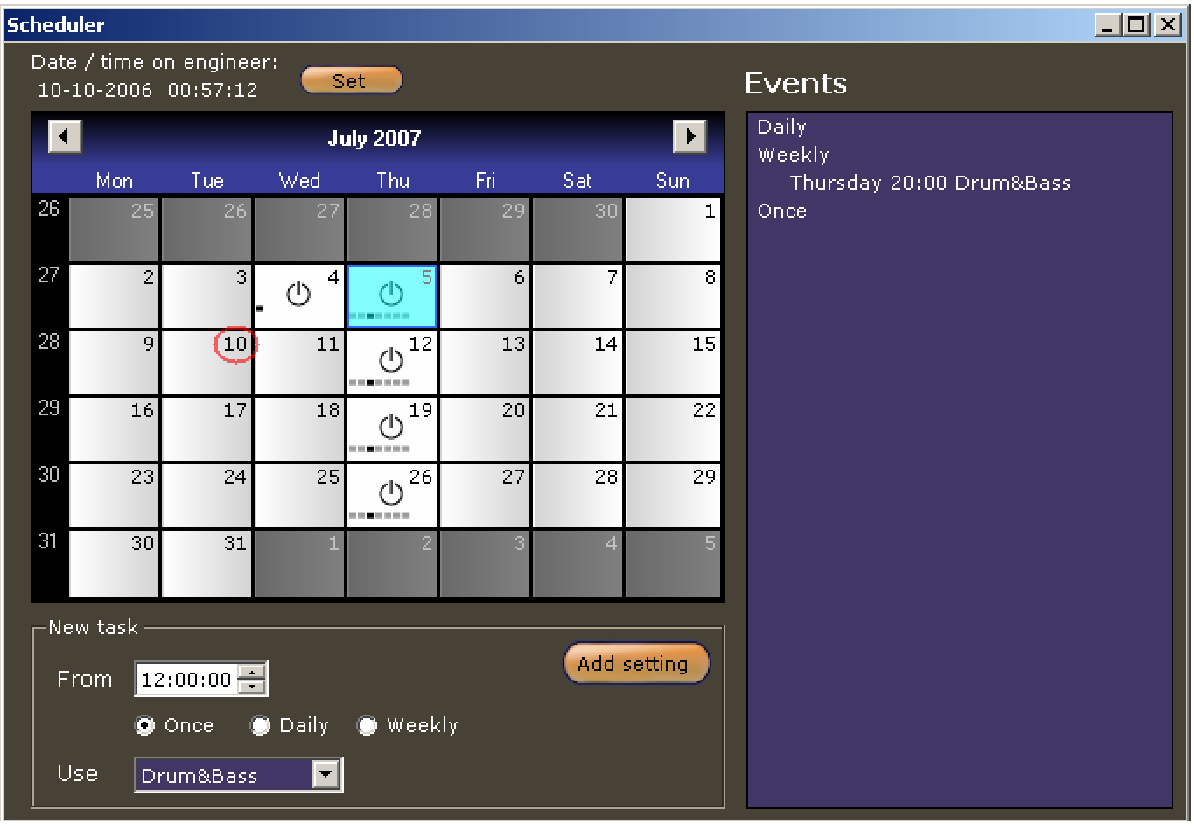
-Select the 'Weekly' option in the left bottom corner of the screen.

-Select the preset you want to be triggered.

-Click on the 'Add setting' button.

You now see the preset appear in the events list on the right side of the screen under 'Weekly'. You will also see the clock logo and a row of little grey dots with a little black dot appearing in the calendar, indicating that a weekly event is stored on that date.

To remove the event from the device, just select the date in the calendar, click on the event in the events list and click 'delete'.



10.3 Daily events

Daily events are meant for events you want to be triggered Daily at a given time. This can be useful for e.g. a restaurant where every day the background music for the lunchtime has a certain preset.

To add a daily event to the scheduler:

-Click on the calendar to select the date you want the event to be triggered.

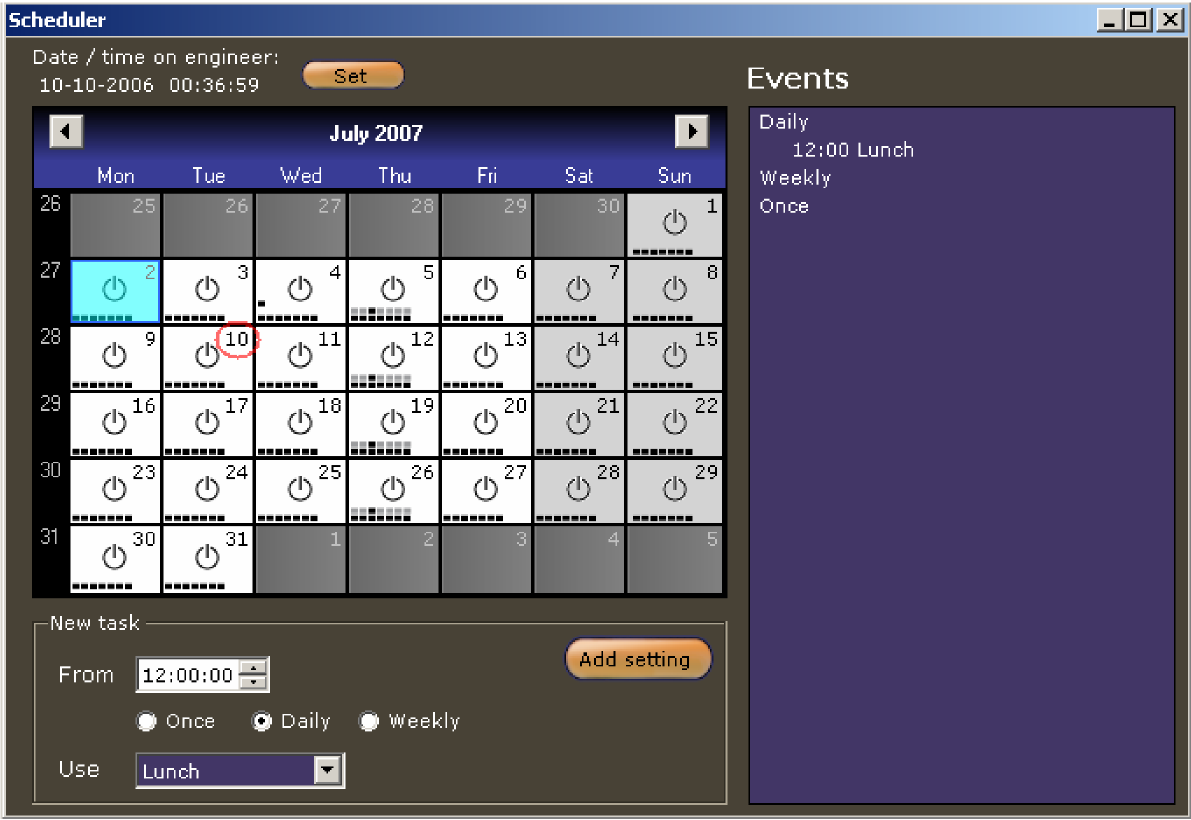
-Select the 'daily' option in the left bottom corner of the screen.

-Select the preset you want to be triggered.

-Click on the 'Add setting' button.

You now see the preset appear in the events list on the right side of the screen under 'daily'. You will also see the clock logo and a row of little black dots appearing in the calendar, indicating that a weekly event is stored on that date.

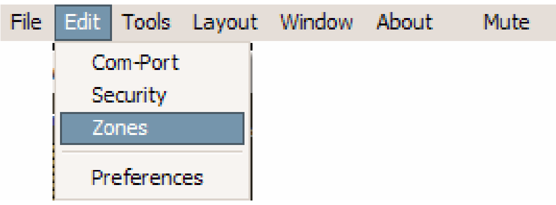
To remove the event from the device, just select the date in the calendar, click on the event in the events list and click 'delete'.

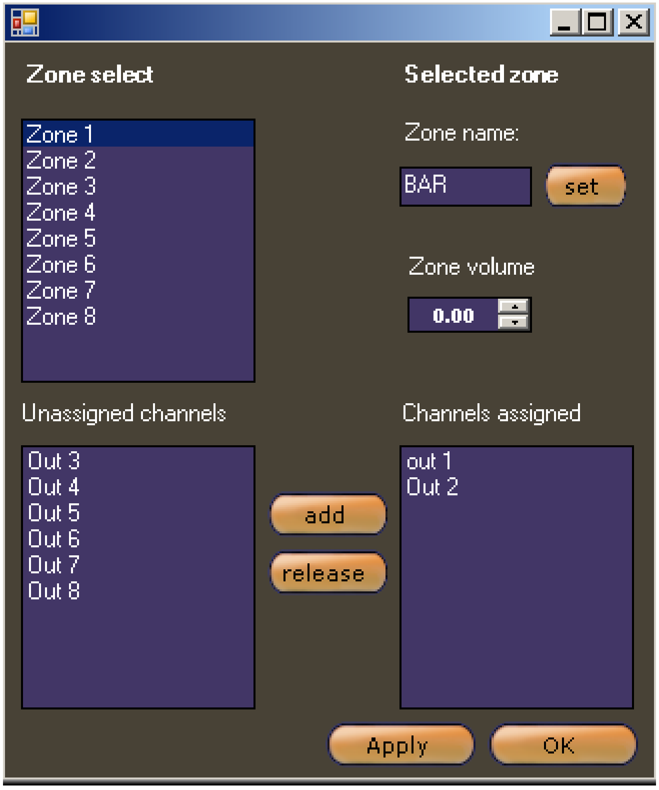


11. Zoning

From the Engineer's remote you can control the gains for several "zones". A zone is defined as a collection of up to eight outputs. When controlling the gain for a zone from the remote, the gain-settings will be applied to each output the zone contains.

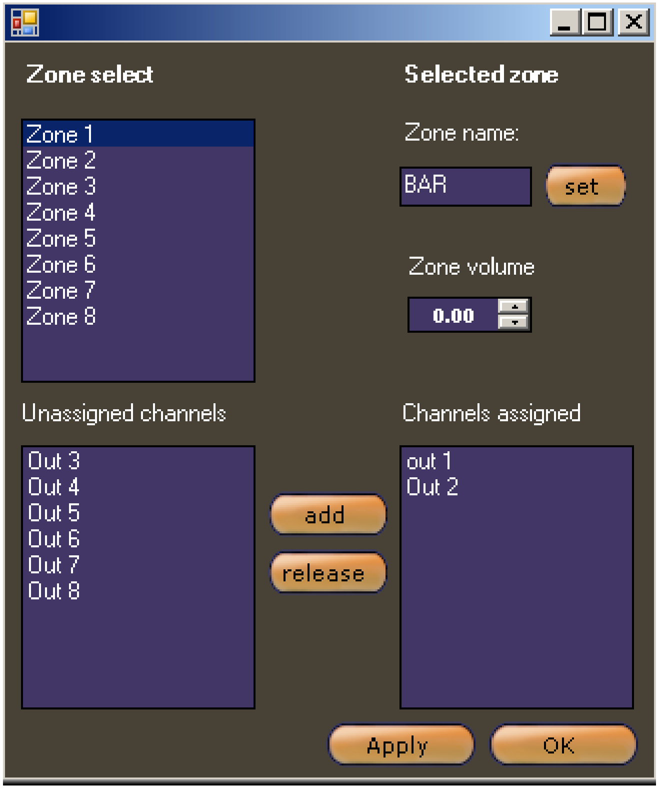
You can open the zone-settings window from the "edit" menu located in the menu bar.





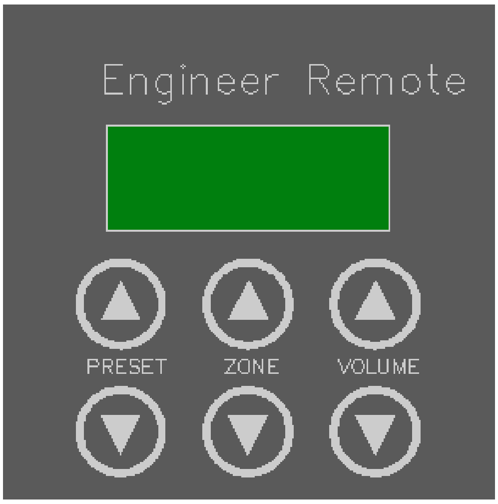
You can select any of the eight zones from the top-selection list. The lists on the bottom of the window show the outputs in the currently selected zone and the outputs that don't have a zone assigned to them. Select any of these outputs by clicking on them, and use the "Add" button to add them to the selected zone. You can use the "Release" button to release the specified output from the zone. Type in a Zone's name in the Zone Name field and click 'SET'. After you have finished, click on apply in the bottom of the screen.

Outputs can only be assigned to one zone.



12. Remote control

When you have created zones on the Engineer, you can use the Engineer Remote to control the volume of the zones you have created in the software and the master volume of the device. Beside that you can recall presets on the Remote Control.



12.1 Recalling presets

To recall a preset from the Engineer Remote, just press the 'Preset' up or down arrow, the display will show 'selecting' and the name of the preset. After 3 seconds the currently selected preset is recalled.

12.2 Changing volumes

To change a volume of a zone, just press 'Zone' up or down arrow. The display will show the zones you have created and the MASTER volume. Once you have selected the correct zone you alter the volume through the 'Volume' up down arrows.

12.3 Connection

The Engineer Remote is connected through the RS-485 connector using an RJ-45 plug on the rear panel of the device. This allows you to make cable runs of up to a 1000 ft on standard CAT-5 cable. Up to four remote panels can be connected to the Engineer by using the Engineer Remote Hub.

ATTENTION:To use the Engineer Remotes, DISCONNECT THE PC FROM THE ENGINEER

12.4 Wiring

The Engineer Remote is wired through standard CAT-5 cabling, with a RS-485 bus-powered protocol. Please make sure that the wires are properly connected according to the wiring scheme below:

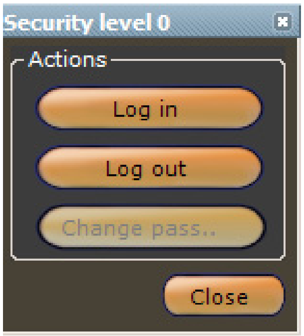
PIN	FUNCTION	RJ-45 CONNECTOR	REMOTE HEADER
1	VCC	ORANGE/WHITE	ORANGE/WHITE
2	RO	ORANGE	ORANGE
3	DI	GREEN/WHITE	GREEN/WHITE
4	GRD	BLUE	GREEN
5	Y	BLUE/WHITE	BLUE & BLUE/WHITE
6	Z	GREEN	BROWN & BROWN/WHITE
7	B	BROWN/WHITE	
8	A	BROWN	

13. Security

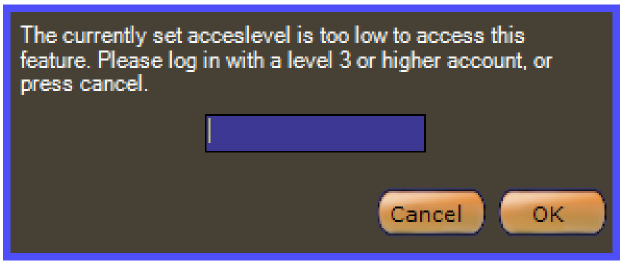
The engineer's security settings can be accessed via the edit menu in your toolbar. The security system has been divided into four access levels, each allowing you to control a fixed amount of Engineer parameters:

Level	Permissions
0	Zone volumes, bass, treble, preset recall
1	Zone volumes, bass, treble, preset recall, preset store, project save and load
2	Zone volumes, bass, treble, preset recall, preset store, project save and load, engineer settings
3	Zone volumes, bass, treble, preset recall, preset store, project save and load, engineer settings, system settings, noise emission settings, full preset and project access, password modification

The security dialog allows you to modify the current access level, by logging in or out. It also allows you to change the password for each level (level 3 only).



When logged in with a level lower than level 3, certain features will be locked. For example, when logged in with a level 0 account, trying to open the output settings will be disallowed:



At this point, a user can still enter a level 3 password to access the selected feature, or press cancel to give up.

14. Technical Specifications

INPUTS	-	4 electronically balanced
IMPEDANCE	-	> 25k ohms
OUTPUTS	-	8 electronically balanced
SOURCE IMP	-	< 60 ohms
MIN. LOAD	-	600 ohm
MAX. LEVEL	-	+20dBU
FREQUENCY RESP.	-	+- 0.1dB 8Hz – 20 kHz
DYNAMIC RANGE	-	> 110dB Unweighted
DISTORTION	-	0.0035% @1kHz 0dBU
MAXIMUM DELAY	-	10ms. (increment 2 microseconds.)
OUTPUT GAIN	-	adjustable +12dB to –inf.dB in 0.25dB steps
INPUT GAIN	-	adjustable +12dB to –inf.dB in 0.25dB steps

PARAMETRIC EQUALISATION

FILTERS	-	8 sections per input, 8 sections per output
FILTER TYPES	-	Bell, Lowshelf, Highshelf, Notch, Bandpass
FILTERGAIN	-	+12dB to –45dB in 0.25dB steps
FILTER Q/BW	-	User selectable 0.05 to 20 / 20 to 0.05
FREQUENCY	-	20Hz – 20kHz

HIGH-AND LOWPASS FILTERS

FILTERS	-	One of each per output
FREQUENCY	-	20Hz – 20 kHz
RESPONSE	-	Bessel, Butterworth, Linkwitz-Riley 12/18/24 dB/oct

LIMITERS

LIMITERS THRES.	-	+20dBU to –43 dBU
ATTACK TIME	-	½, ¼, 1/6, 1/8 times the release time
RELEASE TIME	-	49ms to 2000ms

BASSCREATOR ALGORITHM

PLACEMENT	-	freely insertable on any output
FREQUENCY RANGE	-	virtual 30Hz to 120Hz
PARAMETERS	-	drive, mix level

ENGINEER ALGORITHM

PLACEMENT	-	freely insert able on any input
OPERATING RANGE	-	-45dB to +22 Db

PRESETS

PRESET TYPES	-	system, routing, noise emission, engineer
PRESET NUMBER	-	32 user-accessible presets

SECURITY

LEVELS	-	3 security levels with user assignable passwords
STORAGE	-	security settings are stored within the device

CONNECTORS

INPUTS	-	3-pin Phoenix plug-in terminal block
OUTPUTS	-	3-pin Phoenix plug-in terminal block
RS-232	-	9-pin DB9 connector
RS-485	-	RJ-45 connector
POWER	-	3-pin IEC
POWER	-	60 to 250V @ 50/60Hz
CONSUMPTION	-	30 Watts
WEIGHT	-	3.35 kg. (4.35 kg. Shipping)

SIZE	-	(W) 483mm x (H) 44mm x (D) 251mm
		(W) 19inch (H) 1.75 inch (D) 9.89inch

15. Known issues in software release 0.9.5

- If a value is manually entered in ANY number box, it has to be confirmed by pressing the enter key for the software to store the value, even though the device will respond to it.