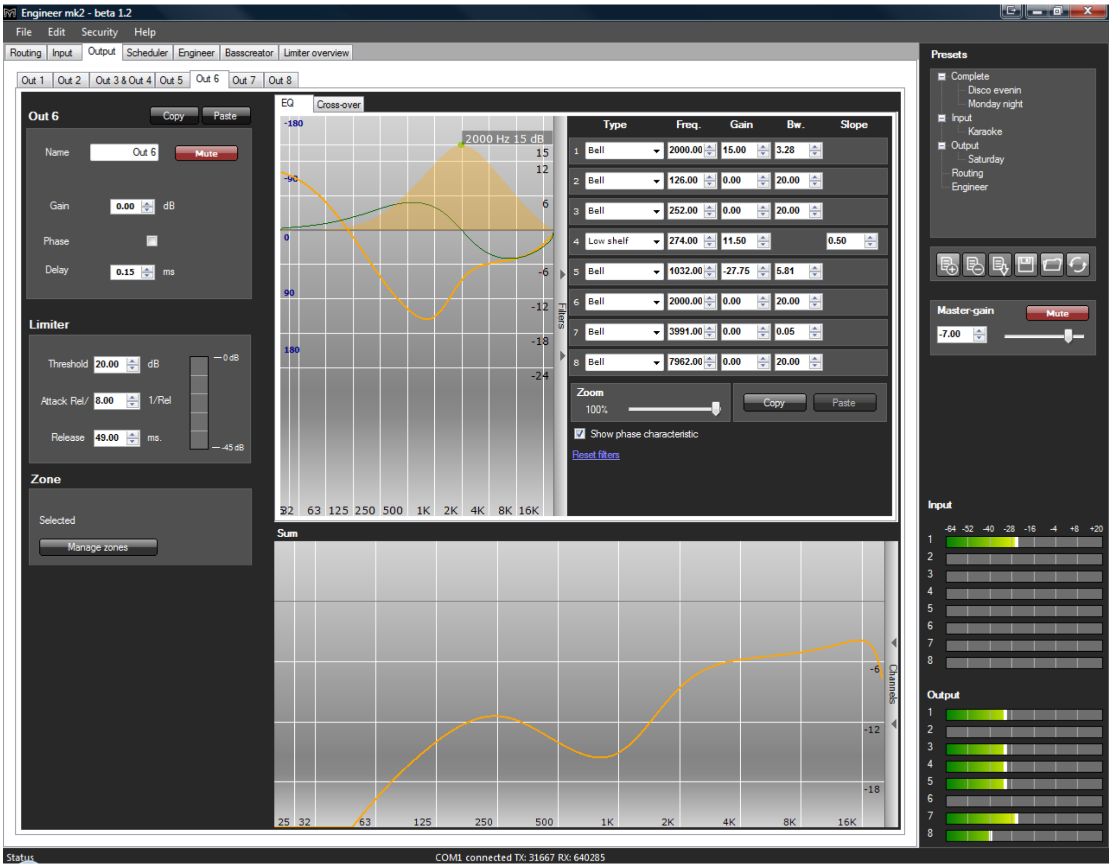
Engineer Control Software Manual



Engineer 418/818 Control Software Manual

By TeamProjects BV for Martin Audio

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BD	2008-06-11	Various changes suggested by KN
BD	2008-06-16	Updated output section for popup feature
BD	2008-06-20	Updated to reflect software v1.00, final draft
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Introduction

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

On top of the ‘normal’ cross-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 nice, 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 compromise operation are important factors.

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

Before you start

The Engineer control software was designed to work with either Windows Vista or Windows XP. Although it might work with older version of Windows this isn’t officially supported.

You’ll need a computer that meets the following specifications to run the software:

- Intel compatible Pentium IV class computer or better
- At least 512mb of system RAM
- RS232 serial port or USB RS232 adapter with drivers installed
- A mouse (scroll wheel recommended)

Before you attempt to install the software please make sure no previous version is installed. You can check this from the “Programs and features” function in the Windows Control Panel. If a previous version is installed, please uninstall it by pressing the “uninstall” button here.

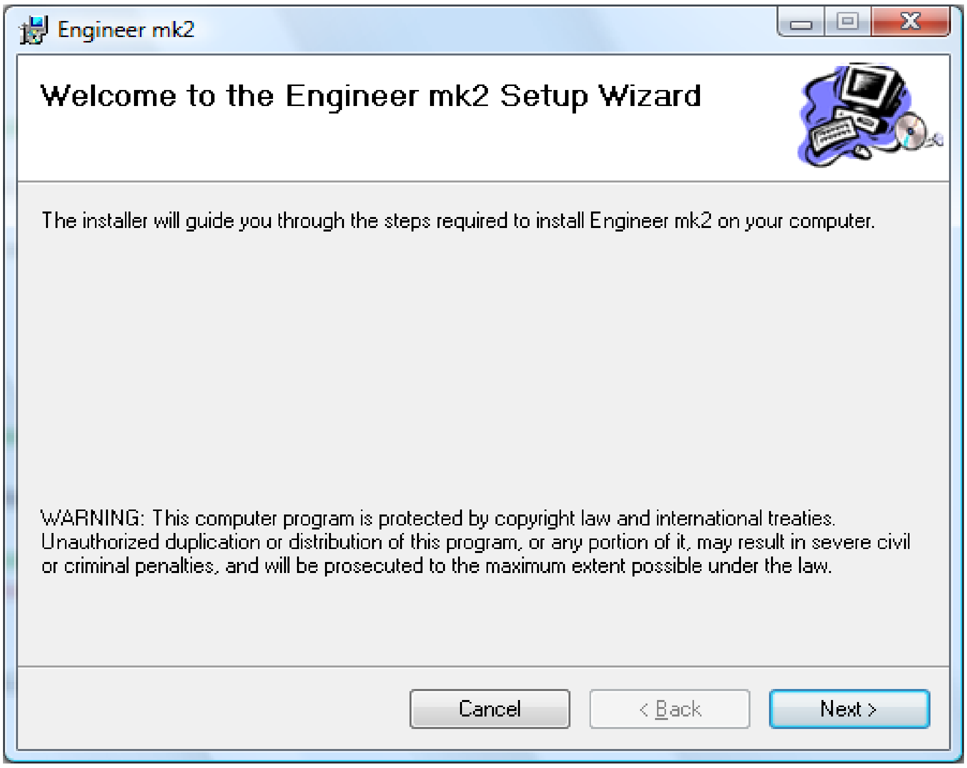
Also, be sure the Microsoft .Net framework version 2.0 or higher is installed. This is a freely available framework which you can obtain from the microsoft.com website. If you don’t have an internet connection, you can use the version included on the Engineer installation cd.

Installation

To install the Martin Audio Engineer control software, simply insert the CD-ROM provided with the device. The installation software menu will automatically start.

If this is not the case, please open the "*My compute*r" 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*" file.

You should now see the installation wizard. Please follow the instructions on your screen to complete the installation.



After installation, a new program group called "Martin Audio" 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 located in this group.

Uninstalling the software

In case you want to uninstall the software, access the "*Martin Audio*" program group in you Windows start menu and select "*Uninstall*". Alternatively, access the Windows Control Panel, select "*Add/Remove programs*". From here, select the entry for the Engineer mk2 control software, and press the remove button.

Concepts and terminology

The Engineer was designed with a few concepts in mind, which we'll try and explain in this chapter.

Presets, projects and the workspace

The Engineer can store up to 28 *presets* at one time. A preset is a predefined combination of settings you've made, which can all be recalled simultaneously with a click of your mouse, using the remote control or the built-in scheduler.

There are five types of presets:

- Complete presets (default): This type of preset stores or recalls all available parameters.
- Routing presets: This type preset only stores or retrieves the routing settings and BassCreator settings.
- Input presets: Stores/retrieves all input settings (input eq's and gains, simple engineer settings)
- Output presets: Stores/retrieves all output settings (output gains, limiters, crossovers, output eq's, delay, phase)
- Engineer presets: Stores/retrieves all advanced Engineer settings

To create a preset, you first have to make the settings you want from the currently active settings. This is called the "workspace", which always represents the settings currently on your screen and –when synchronized properly- what you're hearing.

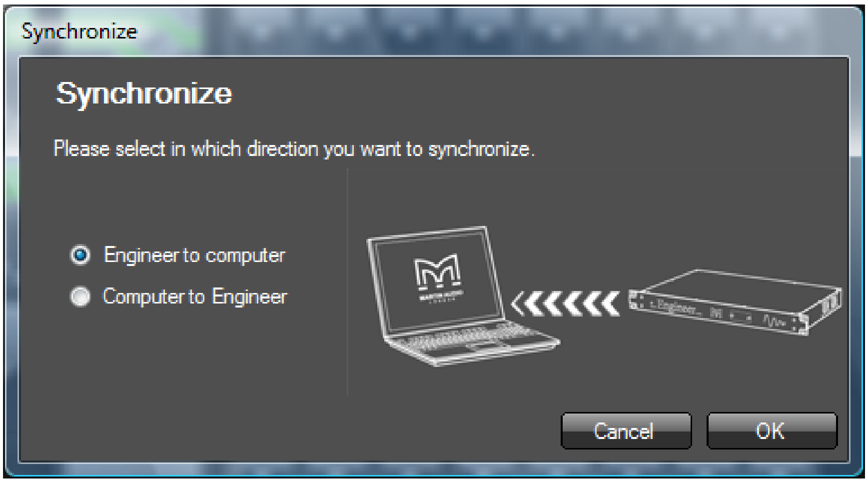
When recalling a preset, its settings will **copied** to the workspace. If a partial preset is recalled, only the settings defined in that preset are overwritten in the workspace; the rest of the active settings will not be changed.

Any number of presets together with the current workspace form a project. A project is defined by the complete content of an Engineer device stored at a given time. You can store a complete project to file (files with the "epf2" extension) for later usage or for back-up purposes, which is something we recommend.

Synchronization

Each time you start the Engineer control software you will be asked to synchronize the software with the project stored on the Engineer itself. Synchronization is an important concept that ensures that whatever it is you're controlling from the software actually represents the settings stored on the device.

You should **always** synchronize before working on a project with an Engineer connected. The choice to skip synchronization is available solely as a means to work on a project when the Engineer isn't connected to your computer.



Zones

One or multiple outputs on the Engineer can be grouped into zones. Zones allow you to define up to eight regions in your venue for which you want to separately control the volume from either the remote control or the software.

This way you save trouble of adjusting each output volume independently. Zones can be named, so they will show up on the remote control and in the software in a convenient way.

Security

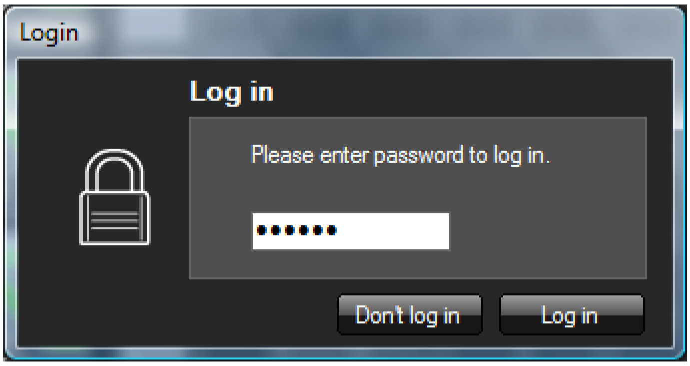
Logging in

If you plan on using the Engineer mk2 Control Software for everyday use, you'll want some way of preventing incapable hands from adjusting crucial system parameters that can potentially destroy an entire PA system.

The Engineer's security features provide just this and offer a limited set of controllable settings for four separate security roles, ascending from say bar personnel to full blown audio experts:

Level	Role	Allowed controls
0	Basic personnel	Master gain (cut only), Engineer bass treble and gain, zone gains
1	DJ	Input gains, input EQ's, preset recall
2	Local sound technician	Routing, zone setup, BassCreator settings
3	Sound guru's	All output settings, full preset control, preset scheduler, Password modification

Each level, except for the most basic one, is password protected and needs to be entered every time you connect the Engineer to the computer. By default, only one password is defined, which gives you access to all functionality. This is set to "**MA**" (case insensitive).

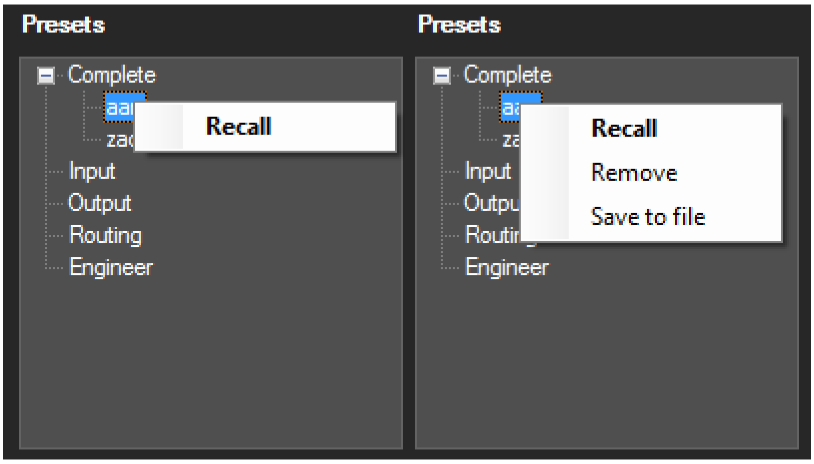


When connected, you'll be presented with this dialog box, prompting you to enter the password that goes with the level at which you wish to access the device. Selecting "Don't log in" results in level 0 access.

You can also log in or out of the Engineer at a later time from the "Security" menu. Logging out sets the access level to 0. When logged in at level 3, you can also change the passwords for each level from this menu.

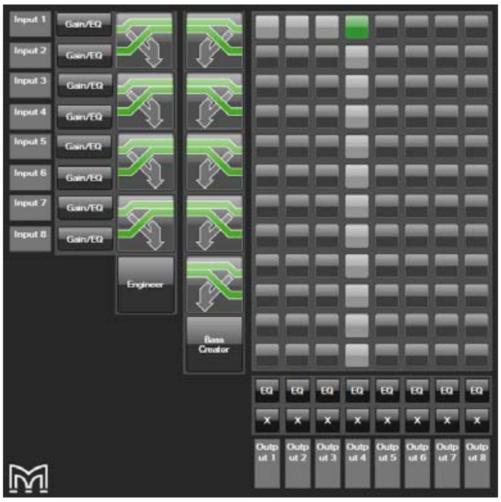
How it works

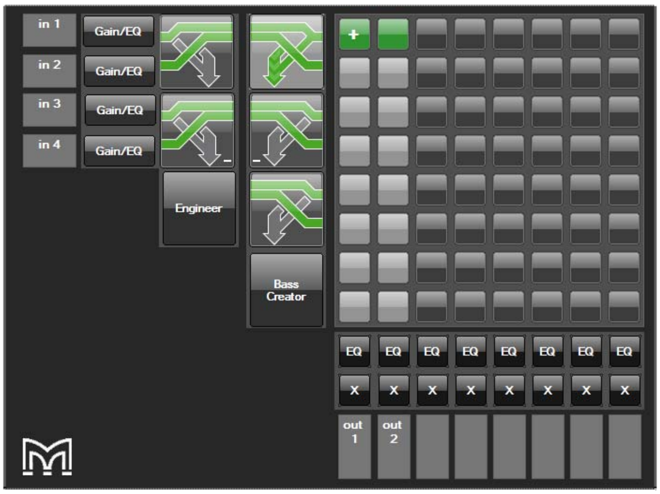
Features unavailable at the current access level are either disabled, or hidden from the user. A good example of this can be found in the preset menu: When right-clicking a preset with a level 1 account, a user will only be able to recall the selected preset, while a level 3 user has more options at his disposal:



Main control

On synchronization completion, you'll be presented with the routing screen. This offers you a nice overview of the features available in the device. Let's take a look.



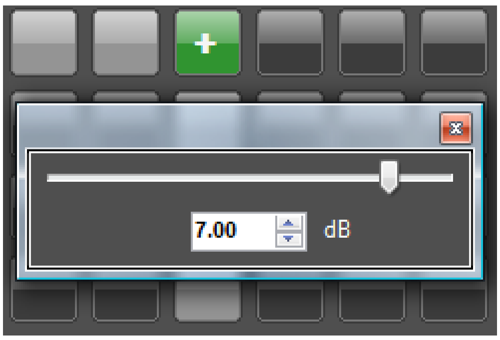


The routing matrix

The largest part of this screen is taken up by the routing matrix, which allows you to control the path you audio signal will travel through the device. All available input channels are positioned to the left of the matrix, while the outputs are shown at the bottom.

When connected to an Engineer, the device type (418 or 818) will be auto-detected, and the matrix will be adjusted to the number of inputs.

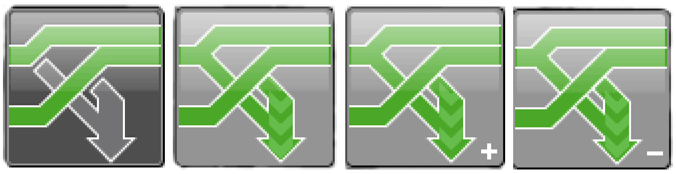
When you press one of the nodes in the matrix you connect one of the inputs to the outputs on the bottom. The route travelled by the signal is highlighted, with the enabled node coloured in green. At this point you can adjust the volume of the signal by right clicking the node and adjusting the gain value.



Effects channels

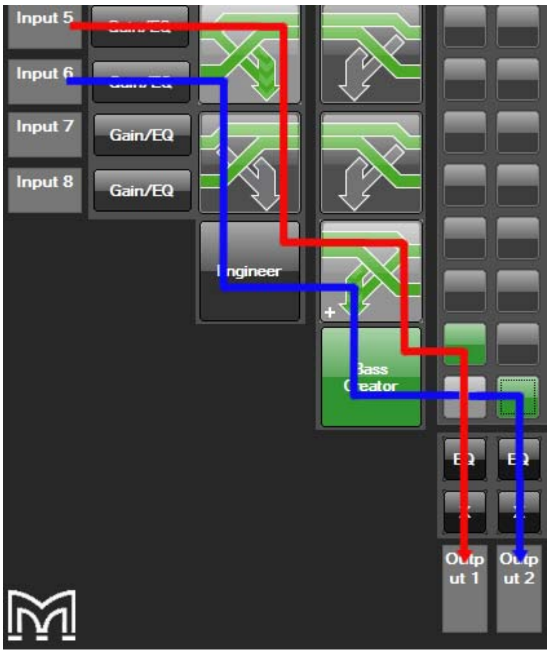
As you might have noticed, there are four extra channels available beneath the input channels. These are return channels for the Engineer and BassCreator effects.

Since the BassCreator and Engineer are stereo effects, all routing towards the effect channels must be done in stereo as well. You can use the large nodes between the routing matrix and input effects for this purpose. Just like the mono-routed nodes, use the right mouse button to adjust the gain of a specific node.



You can route multiple inputs to the Engineer and BassCreator; the inputs will summed before they're processed by the algorithms and will come out merged. Use the stereo-node to the right of the Engineer effect button to route its signal to the BassCreator effect.

Routing your inputs towards the effects channel doesn't prevent you from using the unmodified signal elsewhere: You can still route the unmodified version of the signal to a different target within the matrix.



Labeling your channels

For comfortable usage of your program without the hassle of having to remember which numbered output represents the audio channel you want to control, we recommend labelling your channels. This can be done easily from the routing screen by simply updating the text within the gray boxes beside each in and output channel. The corresponding tabs in the program will automatically be updated with this name and all labels are stored on the device for future usage.

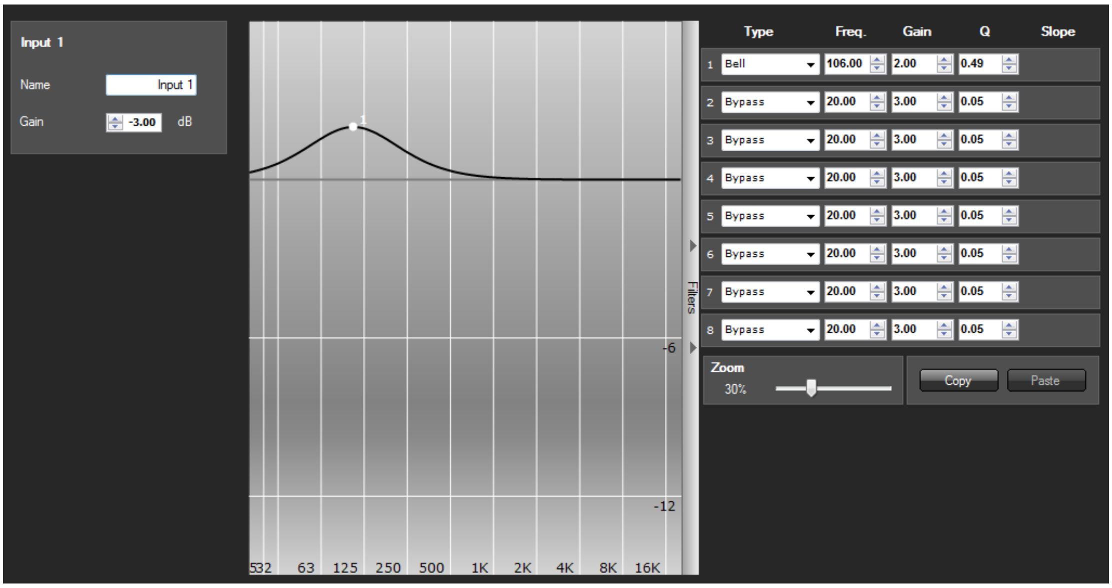
Accessing functionality

Although the Engineer mk2 software is built as a tab-based interface, but you can access most of the effects within the program from the main routing screen. Simply click any of the buttons within the routing overview to jump to the tab it represents.

Input controls

Each input on the Engineer features an eight-band parametric equalizer plus gain adjustment possibilities. These operate in a straightforward manner.

To access one of the input channels select the corresponding tab and channel sub tab or click its button within the routing screen.



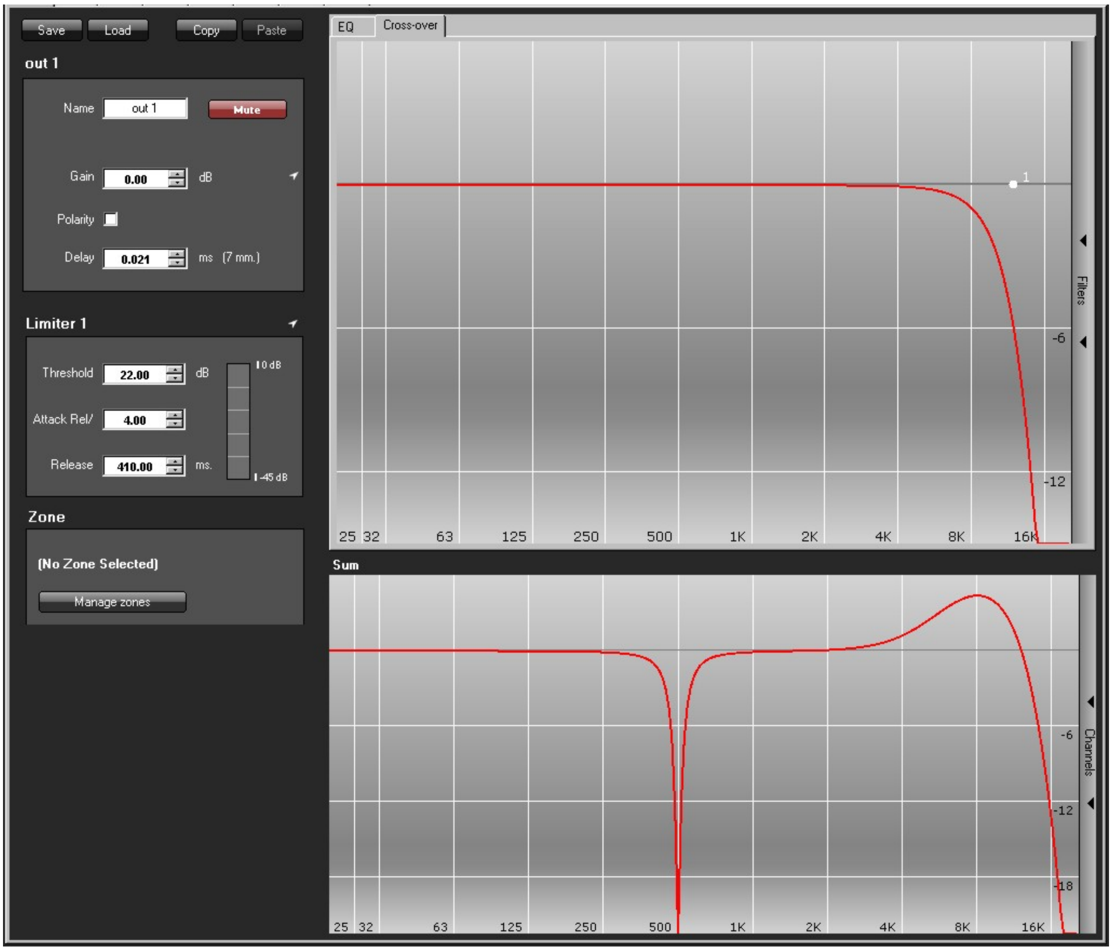
The following types of eq filters are available:

- Bell filters
- Low-shelving filters
- High-shelving filters
- Band-pass filters
- Band-reject filters (notch filters)
- 2nd order all-pass filters (effects phase only)
- Filter bypass (no filter)

The filter graph can be expanded by clicking on the “Filters” bar on the right of the graph. The vertical scale can also be trimmed to get the best view of your EQ using the ‘Zoom’ control.

EQ settings can easily be copied and pasted to other input or output channels with the Copy and Paste buttons.

Output controls



The output channel tab

All output settings are grouped on a per-channel base within the output tab. You can access this through the output settings tab or from the main routing screen by pressing the corresponding button in the routing overview.

Each output channel features an eight-band parametric eq, crossover filers, polarity, delay and gain settings and an output-limiter. The eq and crossover filter settings can be switched from the top-right part of this tab.

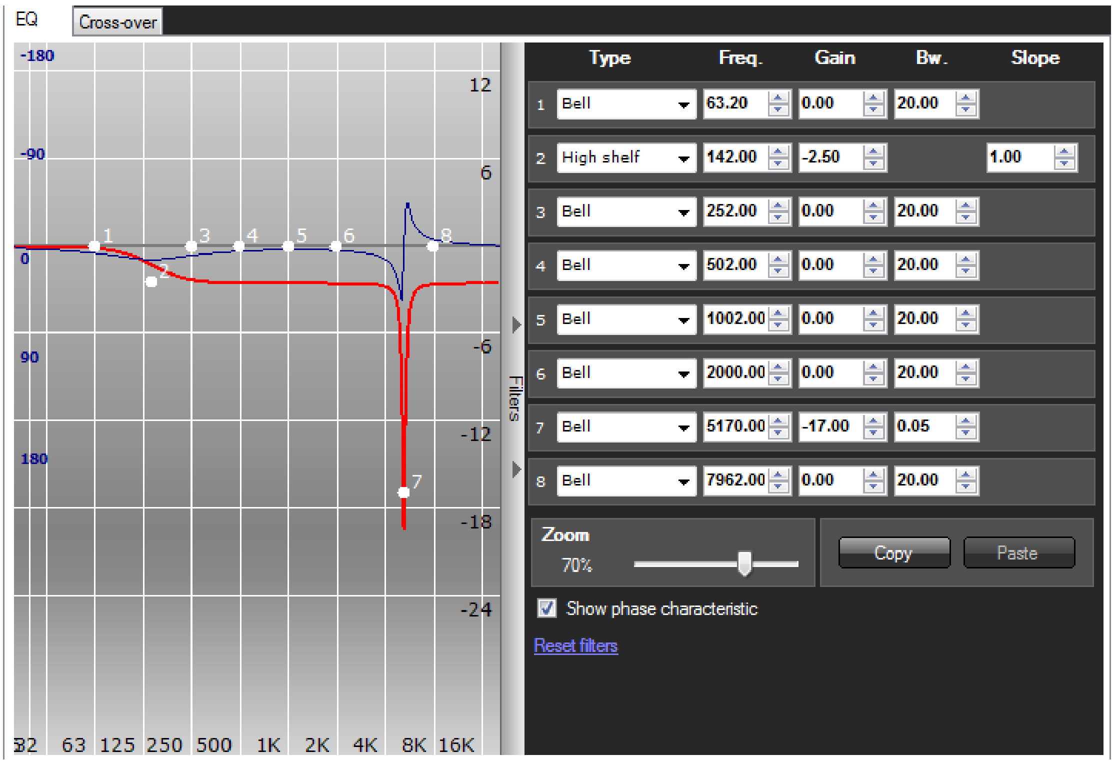
A specific type of filter and the settings that go with it can be changed from the “filters” overview that can be expanded and closed using the vertical bar to the right of this view.

The same types of eq filters are available as the inputs:

- Bell filters
- Low-shelving filters
- High-shelving filters
- Band-pass filters
- Band-reject filters (notch filters)
- 2nd order all-pass filters (effects phase only)
- Filter bypass (no filter)

Crossovers can also be selected from a range of flavours:

- 2nd or 4th order Linkwitz-Riley (12dB/Oct or 24dB/Oct)
- 1st, 2nd, 3rd or 4th order Bessel (6dB/Octave to 24dB/Oct)
- 2nd, 3rd or 4th order Butterworth 12dB/Oct, 18dB/Oct or 24dB/Oct)



To change the gain, q/bandwidth/slope for each filter, use the numerical values next to the filter type. Double clicking each value will pop-up a window in which you can use a slide bar to adjust the selected value.

Another option to change each filter's parameters is by dragging each filter in the visual overview to their position. Use the scroll-wheel of the mouse to adjust the selected filters q/bandwidth/slope. While dragging a filter to its place its specific effect on the eq as a whole is displayed in opaque.

To reset all filters to their default values, you can use the “Reset filters” option located on the right side of the screen.

The filter overview also contains a “zoom” option you can use to zoom in or out from the graphical overview. Beside this button, two buttons allow for copying and pasting from and to other output EQ’s as well as input EQ’s.

The sum (final result) of the effect the crossovers and equalizer have together is displayed directly below the tabs controlling them. You can choose to display additional channels and the colour they’re displayed in from the “Channels” panel located on the right of this control (click to expand or collapse this part).

You may have noticed the checkbox allowing you to show the phase characteristics of all filters. Disabled by default, this feature is meant for advanced users. Setting it to enabled allows you to view the phase characteristic of all filters or only the selected one (click and hold the left mouse button over the filter you want to see).

You can double click each equalizer or crossover’s graphical representation to have a separate window pop up that controls the same parameters. This can be left on top of all windows, so that you can control other parameters while keeping this window handy on the same screen.

There are copy and paste buttons available for the crossover functions as well as the EQ, and the ones located at the top left of the output channel tab which can be used to copy and paste all output parameters simultaneously ;- gain, limiter, crossover and EQ.

Output tab Save, Load, Copy & Paste Buttons

There is also a Save and Load button. These buttons are used for saving or loading all output parameters to a file. This makes designing complex projects with different speakers on each output very simple indeed. The Engineer Mk2 software CD includes output files for all Martin Audio speakers which can easily be loaded into any of the outputs.

If you need to create a custom output this can be saved to file and loaded into other outputs or in future projects. Unlike saving output presets (see the chapter on Projects and Presets on page 34), the output Save function does not save the channel location, just all the parameters so even if a channel setting were created and saved in say output 1, the file can be loaded into any of the 8 outputs.

Engineer algorithm concept

One of the main features of the Martin Audio Engineer is the Engineer algorithm itself. Think of the Engineer algorithm as a real-life sound engineer whom you can tell what kind of sound you'd like from your sound system.

The underlying technology is quite complex and for this reason the system has been split up into a simple and an advanced mode.

The problem

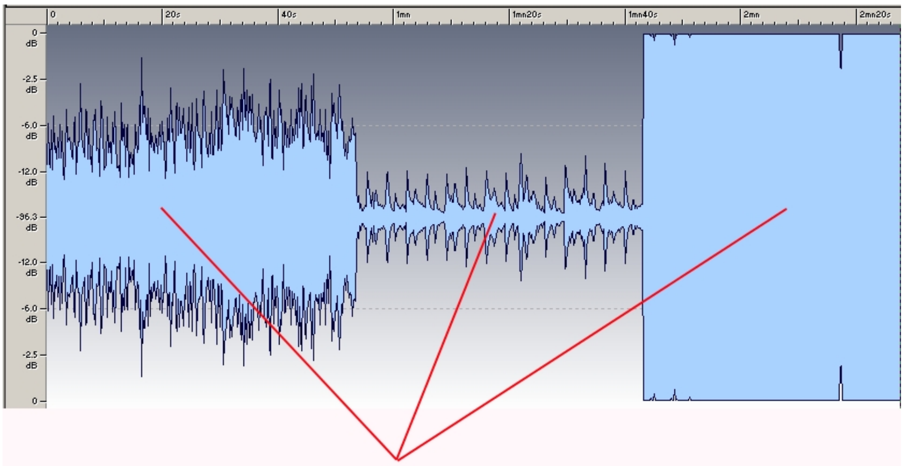
First, let’s look at the reason why one would want to use the Engineer algorithm.

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

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

- Then the next track is very quiet with very little bass and treble so suddenly the music in the venue has all but disappeared, 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, because the massive amount of bass in the venue shakes their drinks off the tables...

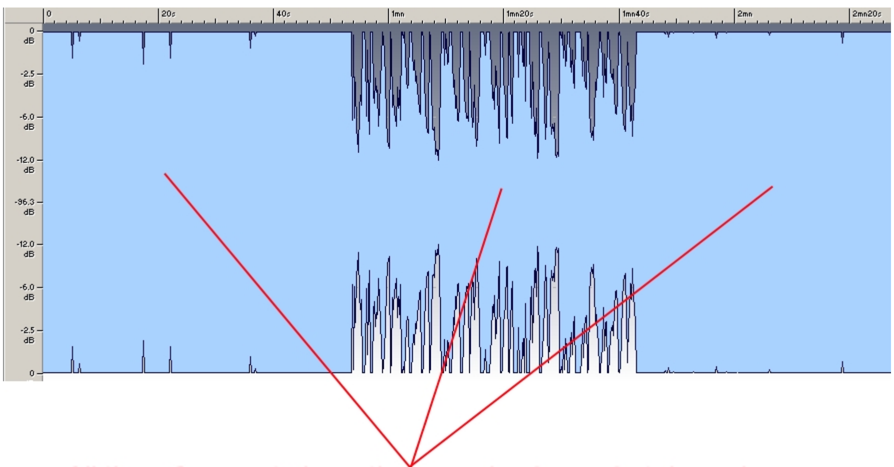


Three fragment have large differences in loudness and tone

This example describes a very common situation which we see happening 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 have to deal with.

On way to overcome this problem is what we call the 'radio station solution'. What happens at a radio stations is that very strong multiband-limiting is applied to all 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, 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 loosing 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.



All three fragments have the same loudness, but dynamics are completely destroyed and distortion is very high

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

The solution

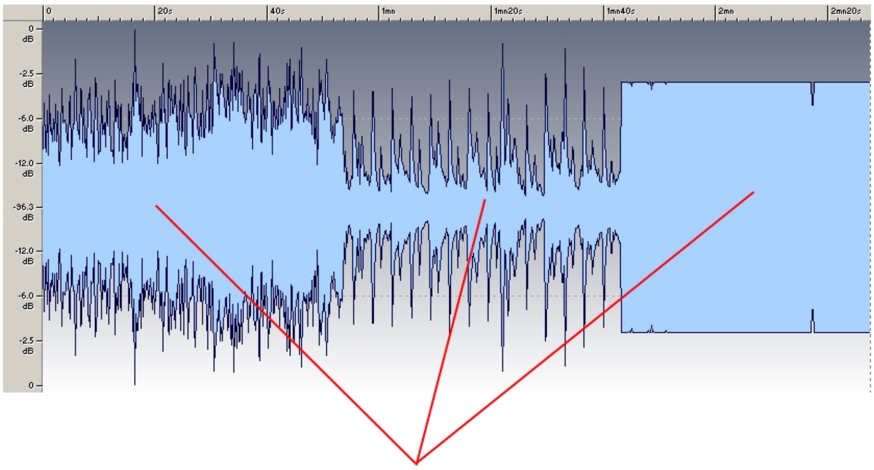
We've solved the issue described above for you by creating an algorithm that reacts in the same way as a normal sound engineer would do, hence the name 'Engineer'. The Engineer works with human-based perceptive algorithms which react the same way as a normal human being would. This means that the algorithm 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 intact.

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, 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 when 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.

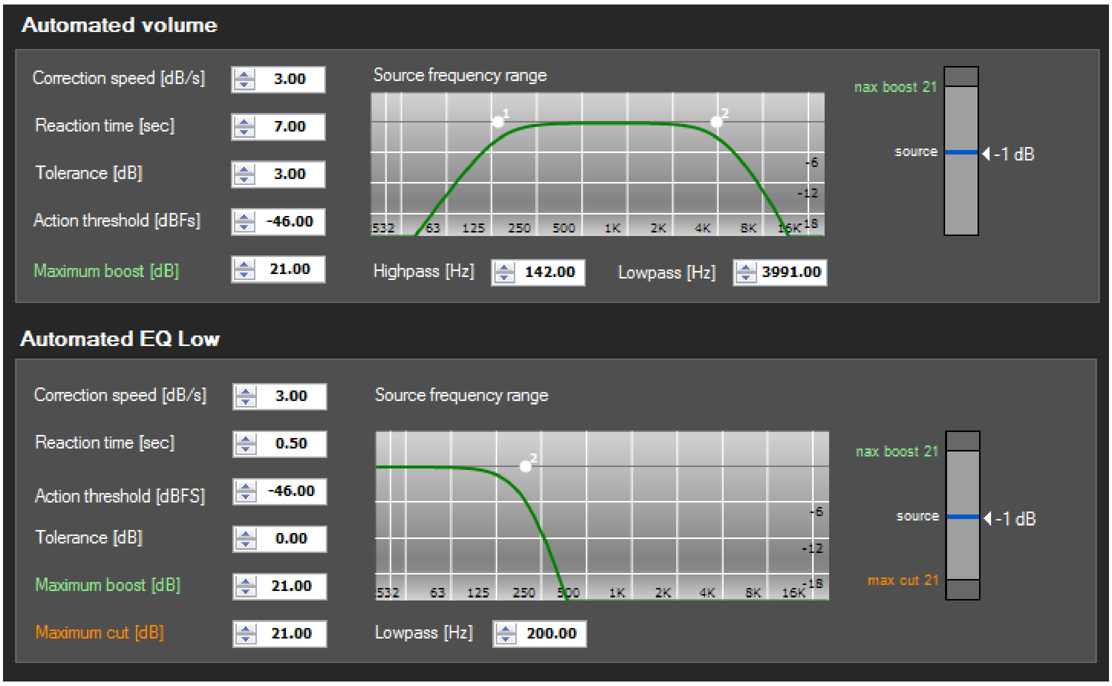
The picture below is the output of the engineer algorithm with our example input. In this picture we see that the level and tone differences have been solved in an elegant way without destroying the dynamics or adding distortion, and that the last piece of heavily compressed music has been lowered in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



All three fragments have the same perceived loudness
and tone quality with optimum dynamics

Engineer advanced configuration

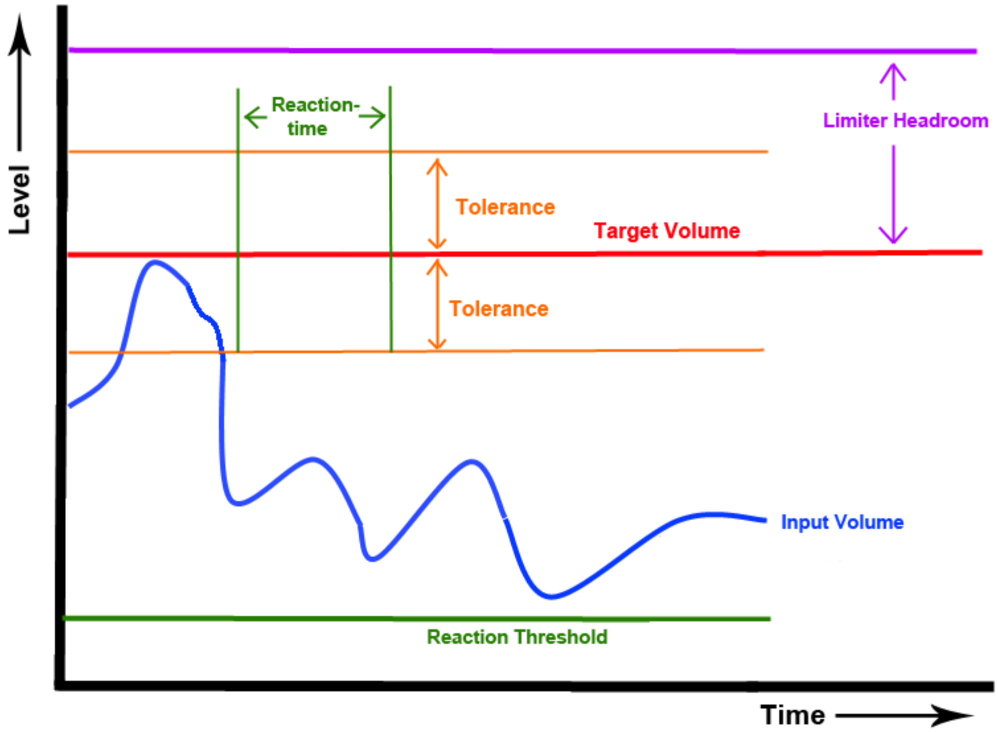
The Engineer can be configured in an advanced mode. Please make sure to read this chapter of the manual if you plan to do so, as it will help you get the most from this feature.



The Engineer features three automated controls; an automated volume control and two automatic EQ’s. It also features three end-stop peak limiters.

The big picture

Before attempting to configure the algorithm, let's take a look at the main parameters that will come into play.



The figure above gives us an idea of the problem to solve and the tools at our disposal: The blue line represents the input volume offered to the algorithm; the red line depicts the overall volume we would like as the output volume.

The goal is to modify our input volume in a way it will be in between the orange lines marked "Tolerance" by the time we're done. The tolerance can be set as an offset to the desired target level. If our virtual sound engineer has low tolerance he'll try to match the target level as accurately as he can: The higher the tolerance, the more slack will be allowed in the source level.

But when to intervene? Sometimes the source level will only be outside of the target area for a short time. The maximum amount of time this is allowed to happen is set by the reaction time. Setting this to a short period of time will result in a nervous engineer, turning the knobs as soon as the volume deviates from the target we've set: The longer the reaction time, the more "lazy" our engineer will become.

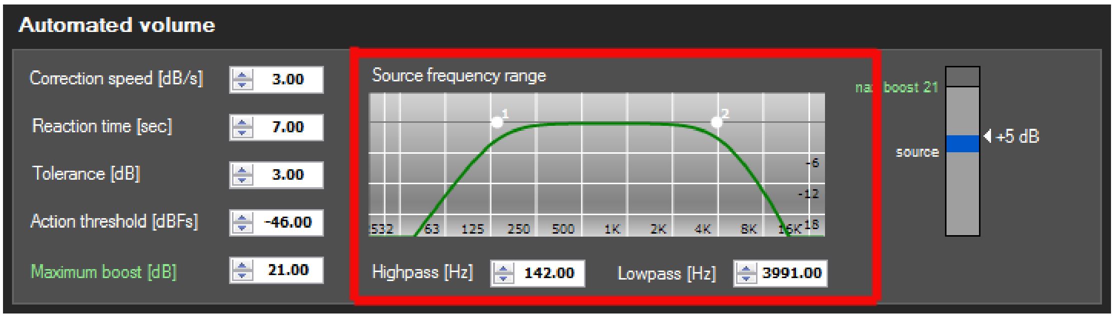
You might be thinking you'll want to set the reaction time as short as possible to counter sudden peaks in the volume, but there's no need: Sudden peaks in the source volume are countered by three limiters (low-, mid- and high tones). The purple line marked "headroom" shows us the maximum level at which these peaks are allowed before these limiters kick in.

Finally, we need to account for situations in which no music is being played. A human engineer wouldn't turn up the level to 10 when there's (almost) no input, so neither will ours. You can set the minimum level he'll work with; it's called the threshold and marked in green in the figure above. If the input level's below the threshold, the engineer will sit back and relax: All knobs will be set to 0, leaving the source signal unchanged.

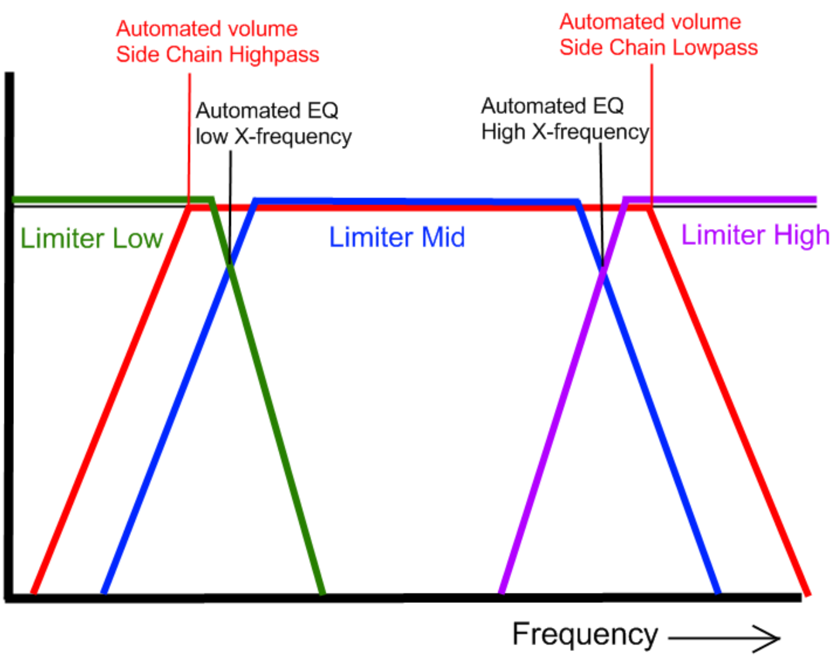
Now that we have an idea of what we'll be configuring, let's get started!

Determining the source frequency range

Each of the automated controls has its own domain of operation: The leveller takes care of the overall volume; the automated EQ's take care of the treble and bass.



The frequency domain you want each of these controls to work their magic on can be determined by setting their respective frequency range. Typically, you'll want to adapt this setting to the type of music you'll be playing most.



We recommend setting the automated volume source frequency to the same values that are used in the automated EQ's. Common settings for different types of music:

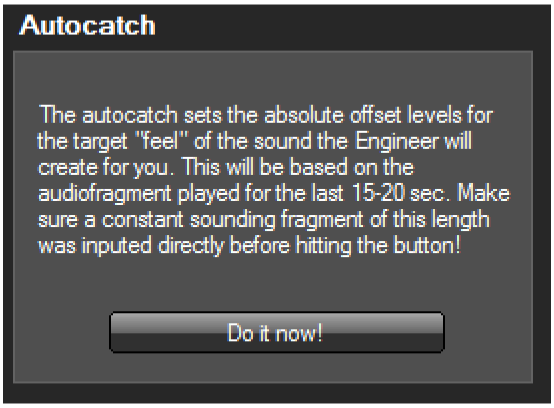
- Dance: 100 Hz - 4 KHz
- Rock: 150 Hz - 4 KHz
- Classical: 100 Hz - 4 KHz
- Blues, Jazz: 120 Hz - 4 KHz

The three end stop limiters have a frequency range that's determined from the frequency range you entered for the automatic EQ's. In the figure above you can see how they correspond to these frequency ranges.

Setting the Autocatch

Remember the target level we discussed on the last page? We haven't talked about the way this is determined yet. To set this to level, we'll need to use the "Autocatch" feature.

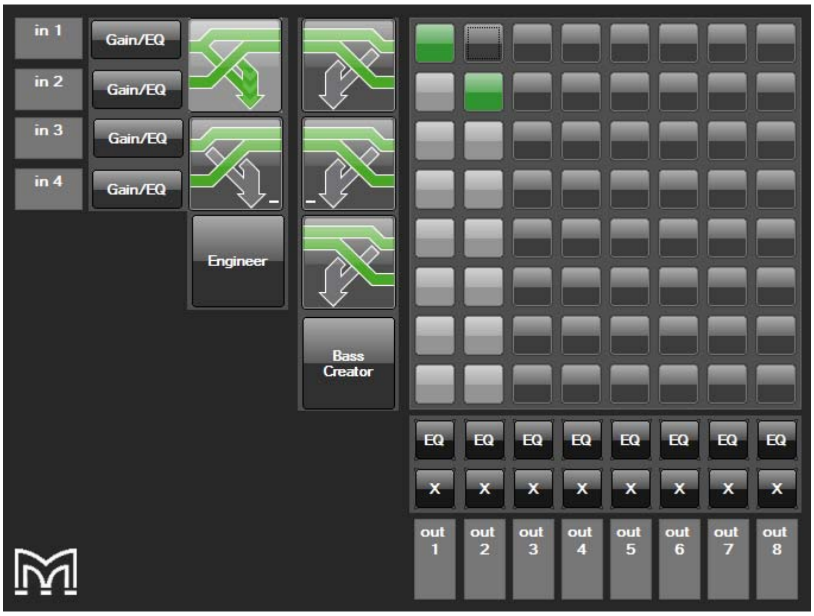
The Autocatch determines the treble and bass sound we want our engineer to look for. In other words: His personal preference when it comes to a specific "feel" to the sound he's creating.



As we would with a human sound engineer, we'll need to let him hear what we're looking for. So, put on your favorite song and make sure it's a good example of the music you'll generally play. Ensure the music you play continuous and doesn't contain any breaks.

Now, before pressing the Autocatch button, make sure our engineer hears the source playing your track, by routing it towards the engineer. Also, make sure your input settings aren't interfering with the signal (no extreme eq or input gain settings).

To ensure the engineer is hearing the same as you are we recommend routing it as pictured below, assuming you've connected your source to inputs 1 and 2.

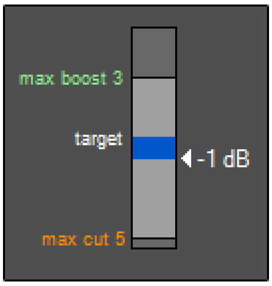


When you're satisfied with the sound you're hearing, press the Autocatch button in the top-right corner of the Engineer advanced screen. It'll remember the sound you want for this preset. You can set different values for the target bass and treble in each preset, so you might want to create separate settings for different types of music.

Please note, that every time you change any of the source frequency parameters (discussed on page 23) the Autocatch needs to repeated!

Max cut, max boost and the correction indicators

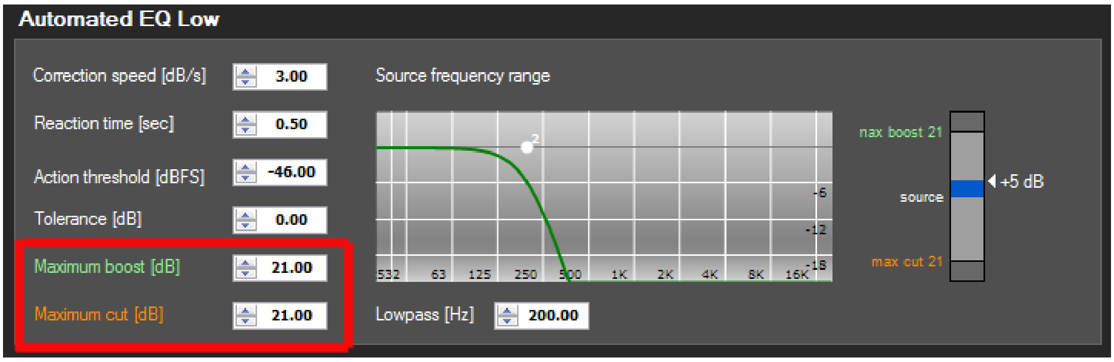
To help you find the perfect settings for your situation, we've added a handy indicator that visualizes what's happening within the device.



This indicator is displayed for the overall level correction (Automated volume) and low and high tone corrections (Automated EQ's Low/High). It shows some of the most important parameters:

- The max-boost is the amount of gain you allow your Engineer to add to the input level to reach the target level you set using the "Autocatch" feature.

- The max-cut is the amount of you allow your Engineer to deduct from the input level to reach the target level you set using the "Autocatch" feature.



Together these two form the boundaries in which the Engineer functions: If you want to give it maximum control over the source material, you should set them as high as possible: Setting them to lower levels restrains the engineer from tampering with the knobs too much.

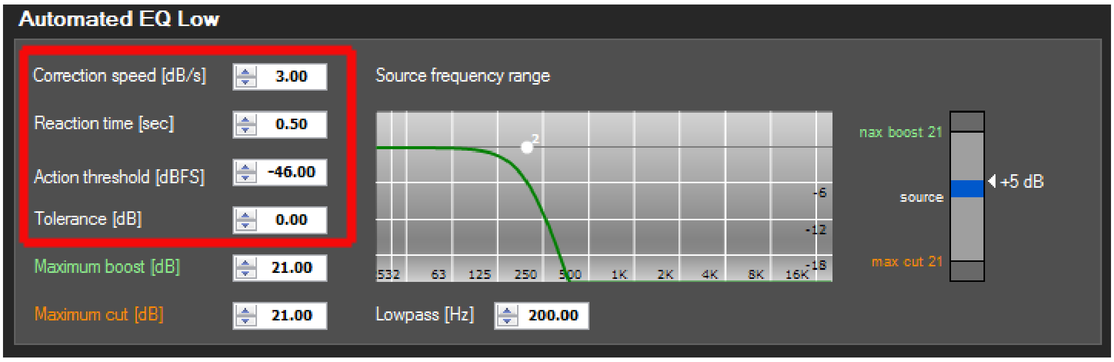
In the case of the leveller there's no maximum to the amount the Engineer can cut. This way, no matter how loud a crazy DJ's input signal is offered, it will always be cut to the maximum level you allow it to be.

The space you allow the Engineer to do its work in is marked in light gray on the indicators. The blue bar indicates how far the engineer is currently turning the knob and this value in dB's is shown beside the arrow on the right. In the picture above he's adjusted the source level by –1 dB in order to match the target sound we defined.

In case the engineer determines an amount of gain should be applied outside of the max-boost/cut boundaries, it will be clipped at their maximum values.

Correction speed, reaction time and threshold

Another important factor in the operation of the Engineer is the speed in which the input level is corrected. This is determined by the correction speed: the maximum amount of dB's per second you allow your sound engineer to correct the source signal towards the target level. In other words: The speed with which the knobs are turned up or down.



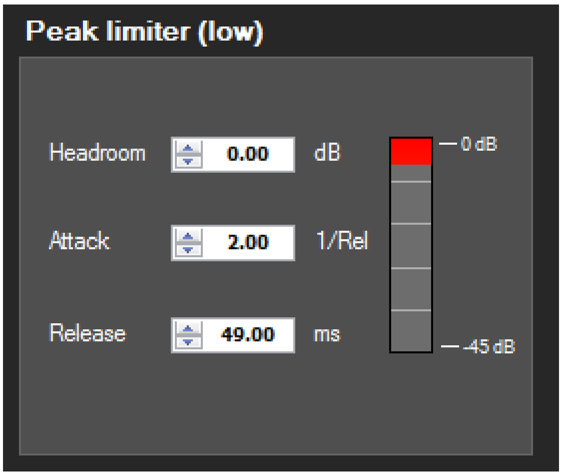
But when should the algorithm start to correct the input level? In some cases you might not mind small variations in the level, or will want to allow for short periods of loudness. This can be set by the reaction time and tolerance.

These parameters do just what their name suggests: The reaction time is the time in seconds a louder or softer level will be allowed, after this your engineer will kick in and correct the signal. The tolerance is the maximum amount of variation (either boost or cut) in dB's you want to allow for.

A problem that could arise while the engineer is at work, is a sudden pause or unusually soft passage in the music. We wouldn't expect a human engineer to turn up the level in this case either, so we need a certain threshold for the engineer to start working. This is set in dBFS.

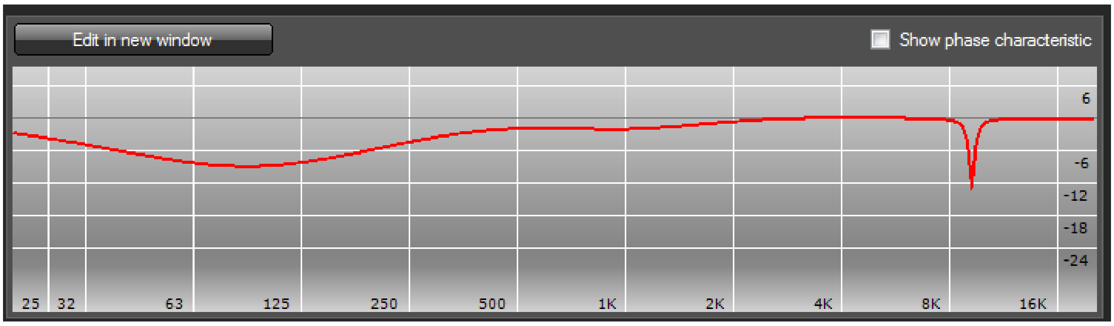
Peak limiters

Sudden level changes are countered by the three peak limiters, so you won't have to account for this when setting the reaction time and tolerance. The peak limiters can be separately set for the low, mid and high tones. They work like any normal limiter you're used to.



Each limiter allows for an amount of "headroom", which – again – allows you tolerate peaks up to a predefined level. The attack is set as a factor of the release time.

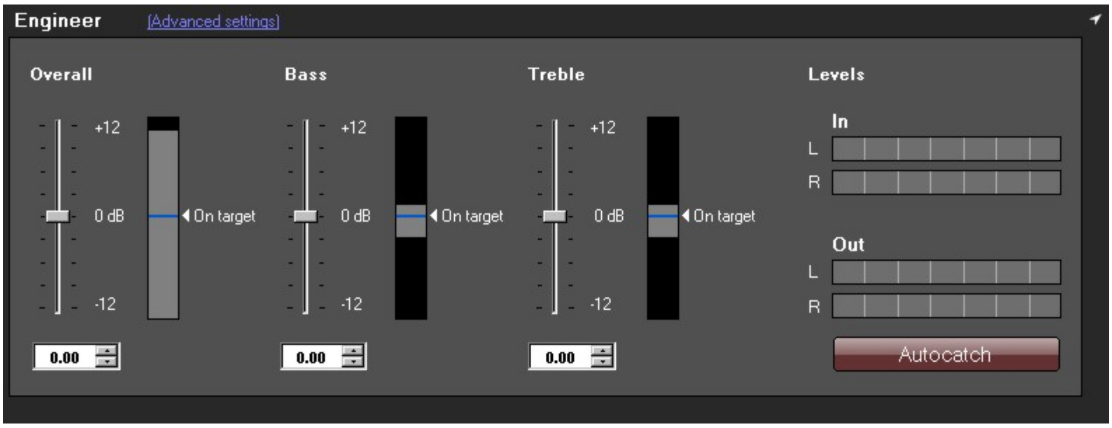
Exit equalizer



Still not satisfied with the results you're getting? The Engineer allows you to make some final adjustments before the routed signal moves on. You can use the exit equalizer for this purpose. Press the "Edit in new window" button to open it up.

Engineer simple control

Of course, the settings made in the Engineer Advanced overview aren't suited for day to day usage. For this purpose, a simple control centre can be found from the "Engineer" tab. It contains only three settings that should look familiar to anyone: A slide for bass, one for treble and an overall volume slide.



The level indicators to the right depict the stereo in- and output levels for the Engineer. The Engineer Simple control is available from access level 0 (meaning to anyone).

Basscreator algorithm

Concept

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 emissions problems.

- Beef up small speakers to give a sense of low end that couldn't be achieved normally on the same speaker

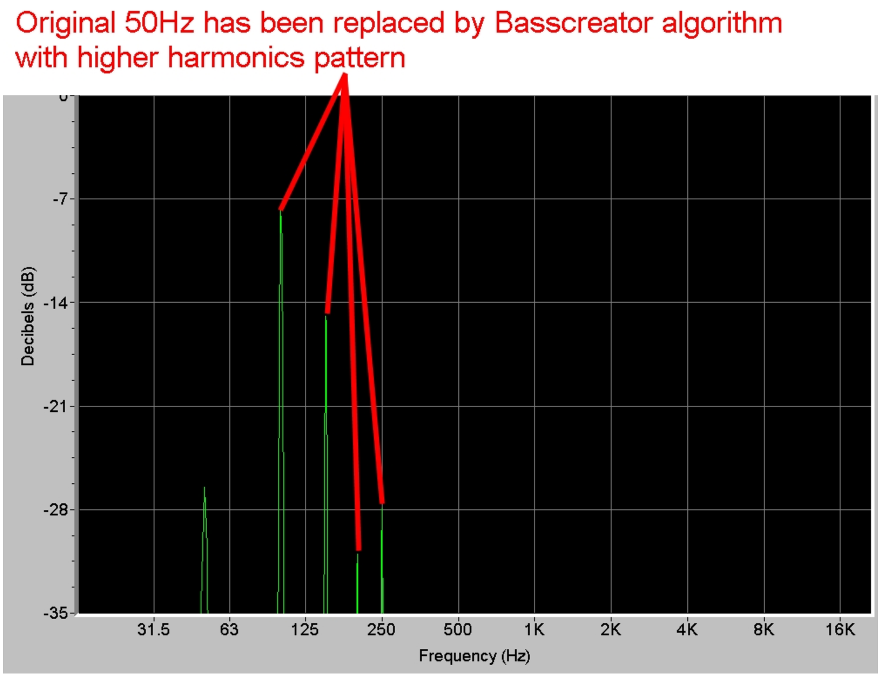
- Reduce the amount of sub-woofers needed to save space and money

- 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.

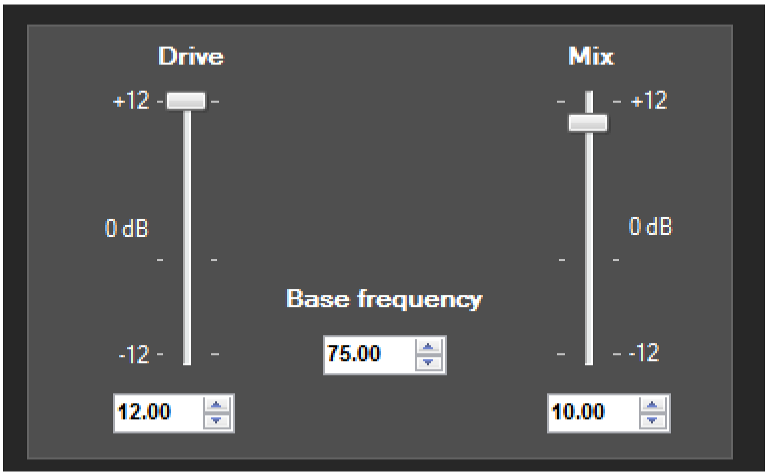
The pictures below give an impression of what happens within the Basscreator algorithm. If we put in a sine wave with a frequency of 50 Hz and we put the output of the device in an analyzer, we see the sine wave as a single spike.

Now if we put that same 50 Hz sine wave through the Basscreator algorithm and analyse the output of that, than we see the picture below. We see that the original 50Hz 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.


Original 50Hz has been replaced by Basscreator algorithm with higher harmonics pattern
Decibels (dB)
0
-7
-14
-21
-28
-35
31.5
63
125
250
500
1K
2K
4K
8K
16K
Frequency (Hz)

Controlling the Basscreator

To start using the basscreator you should first route the inputs you want to use it on toward the Basscreator module in the Routing tab. Then select the Basscreator tab to control its settings.



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 you speaker system.

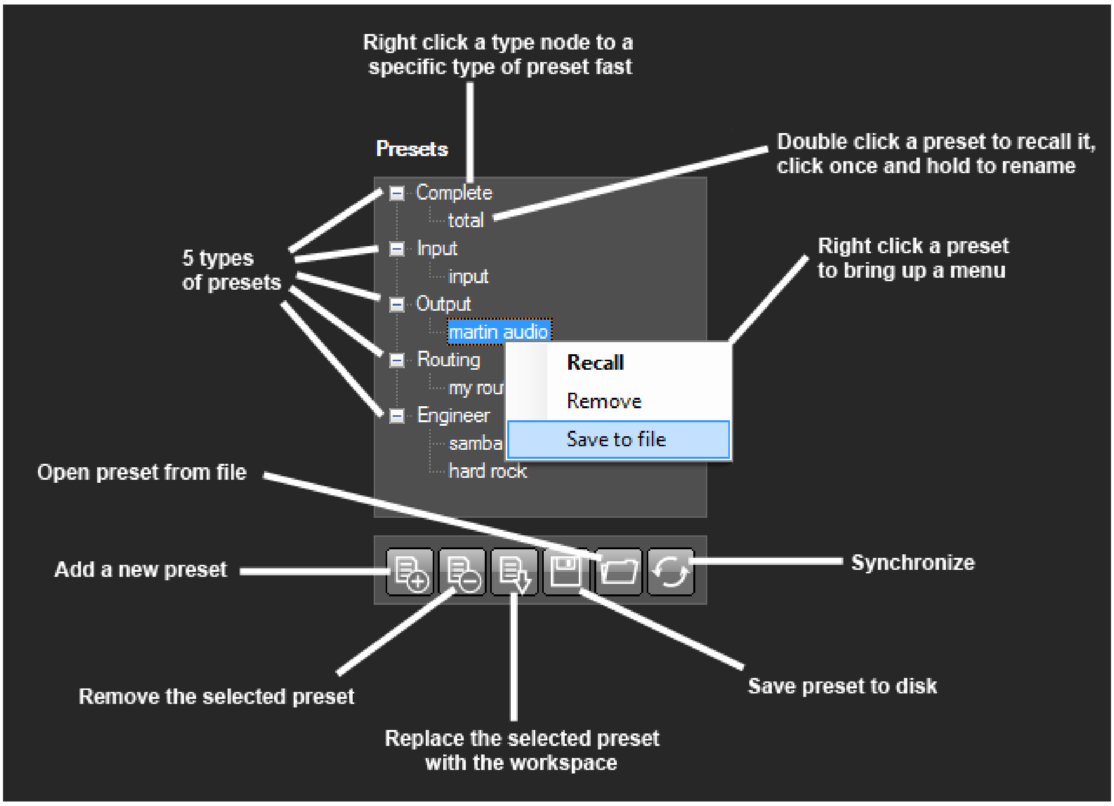
A rule of thumb for this is to choose the frequency about one octave below the low cut-off of your speaker. This means that if you have a speaker which is capable of producing 100 Hz, you should use the 50HZ setting one the Basscreator as a starting point.

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).

Presets and projects

If you've experimented with the software already, you've probably noticed the preset overview fixed in the top right of the window. This allows you to store and retrieve up to 28 presets of five different types on the Engineer.

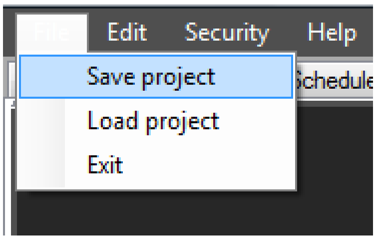
The preset type you'll probably use most is the "complete" type. This stores all settings you made in the device into a preset. This means all settings can be recalled at will.



The other presets are partial presets:

- The input preset stores all input eq settings, input gains, input mutes and simple engineer settings
- The output eq preset stores all output EQ's, output crossovers, output limiters, delays, limiters and phase settings.
- The routing preset stores the way the device is routed as well as the basscreator settings.
- The Engineer preset stores all settings in the Engineer advanced tab.

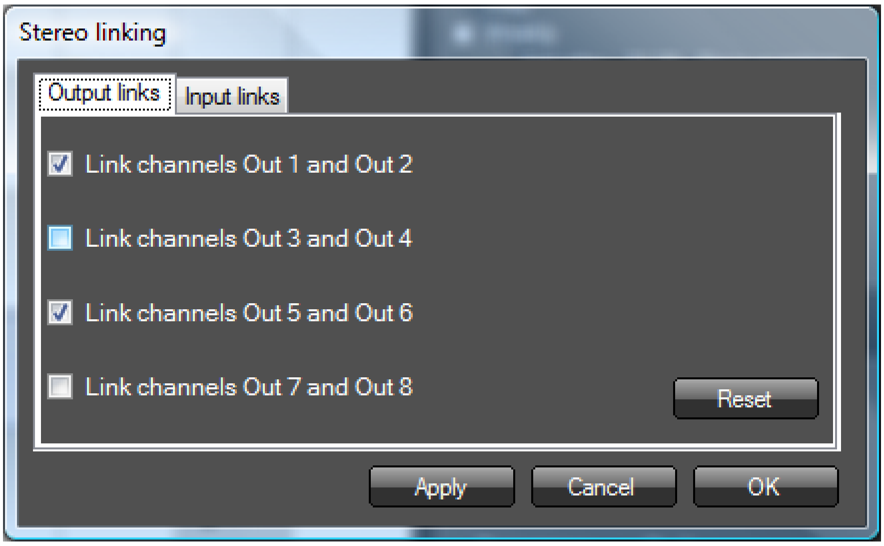
Separate presets can be saved to file, or imported from them. A combination of presets (everything stored in a device at one time) is called a project. A project can also be saved or loaded from file. This is done from the file menu.



When you load a project from file, you'll be asked to synchronize. This is highly recommended. If you want to know why, please see the concepts chapter for more information about synchronization.

Linking channels

Each in- and output can be linked into a stereo pair. This means channels 1 and 2, 3 and 4, and so forth can be controlled as one. To use this option select the “Stereo link” tool from the edit menu in the program’s toolbar.



From here, you can select each channel you want to link. As soon as you hit “Apply” or “OK” each second (even) channel will start using its masters settings. This can result in sudden changes in outbound signal levels, so take care not to damage any equipment.

Once linked, each channel’s tab page will reflect its linked status; the names of both channels will appear as the title, and all settings made will reflect on both channels. Only the output mutes can be controlled separately.

Zoning output channels

The Engineer allows you to assign output channels to up to eight zones. For each zone the volume can be adjusted, affecting every channel assigned to it. This way, you can separate different regions of your venue and assign volumes accordingly, using either the software or the remote control (see hardware manual for further information about the remote controls).

Zones can only be assigned when logged in at security level 3. The zoning manager can be accessed from the zoning tab or direct from each output tab, where a small panel in the bottom left indicates the current output's zone.

By default, output channels aren't assigned to any zone and are shown in a line under "Unassigned Channels". To assign an output to a particular zone, simply drag and drop the output box to the zone of your choice. You can make changes by dragging and dropping the boxes between zones or back down to the Unassigned channels area.

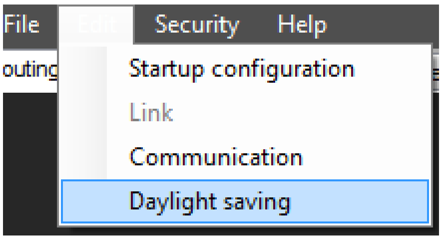
To give each zone a name, click on the zone names down the left side.

To manage each zone's volume, adjust the zone master gain value or the remote control can be used.

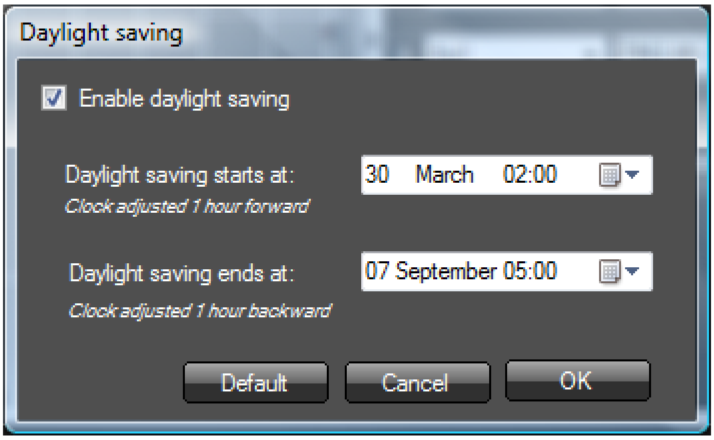
Daylight saving

The Engineer offers the possibility of automatically adapting to daylight saving time (British Summer Time in the UK). The time and date daylight saving starts and stops differs per country, so you will need to let the device know when update its time accordingly.

This can be done from the "edit" menu in the application's top menu bar available from the highest security level (3) only.



When clicked, a window will open up allowing you to adjust the daylight savings setting.

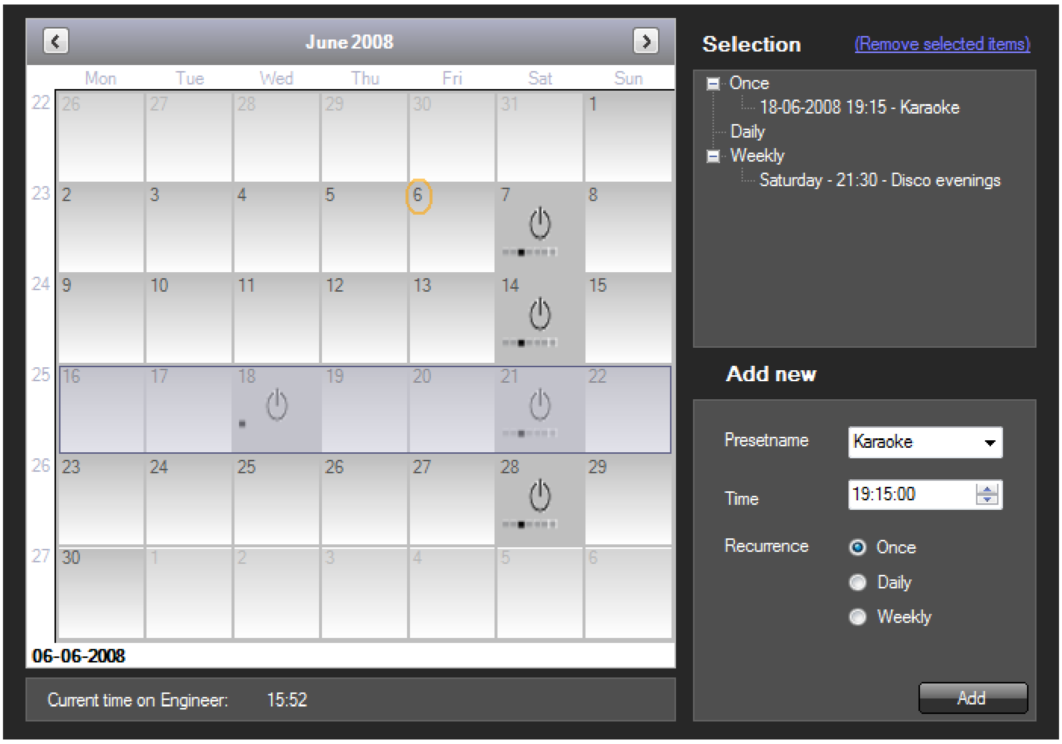


By default, daylight saving is disabled. To enable it, check the box at the top of the window and enter the dates and times for daylight saving begin and end. The calendar icon to the right of each input field pops up a calendar that allows an easy way to select the date.

When daylight saving starts, the clock on the engineer will be adjusted forward by one hour, and when it ends the time will adjusted back.

The dates and times can be set to their default for the current year according to the locale used in Windows. Use the "Default" button to load these settings. Pressing "Cancel" will do just that on any changes made and close the window, pressing "Ok" will apply your new settings and close the window.

Scheduler



The Engineer features an advanced scheduling system, allowing you to adjust your venues sound set-up to recurring events, or plan one-time happenings in advance.

The system is based on preconfigured presets that will be triggered at any point in time you define. This can be daily, weekly, or one time only. To start using this feature, create the preset you require for each event, and select the scheduler tab.

From the calendar on your left, you can select one or multiple dates on which to add a new scheduled event. If you're planning to use a daily preset, you only need to select one day.

To add an event to the Engineer's agenda, all you need to do is specify the preset to use, the time to use it, and the recurrence.

When you press add you'll notice an icon appearing on the selected day. This indicates an event is scheduled for that day. There are different icons for each recurrence setting, which should be quite easy to understand.

To view planned events for a specific amount of time, just select the date range you wish to show. All planned events will appear in the panel to the top right corner. To delete a single event, right click it and press "*Remove*". You can also delete all selected events by pressing "*Remove selected items*".

Limiter overview

So, you've got everything set up the way you want it? All there's left to do is disconnect your computer and sit back and relax. If set up properly your automated sound engineer will be doing all the work for you.

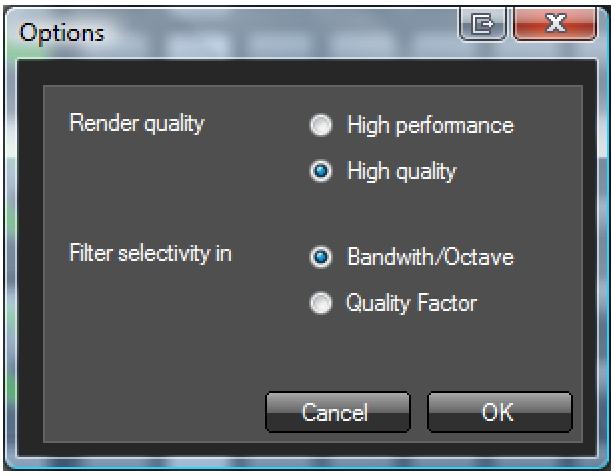
Just in case you want to use the software to keep an eye on things, there's a tab page available showing you the limiter status of all eight output channels. Together with the in- and output level indicators, which are always visible, this is a handy overview to show you when one of the limiters kicks in.

From access level 2, all limiters can be controlled from this page, on lower levels it just shows the status of each. Linked limiters show up in faint red, indicating they can't be adjusted.

In the example below, output four has been linked to output three, and output one's limiter is active.

Preferences

The Engineer Control software offers two user-configurable settings, which can be changed from the preferences dialog. You can open this dialog from the “edit” menu in the program’s toolbar.



The first option you can select in this dialog is the render quality. This affects the quality in which the program’s graphics will be drawn to your screen, most notably the graphic representation of the equalizers and crossovers.

By default, this option is set to “High quality”. Most modern day computers are more than apt to handle high quality graphics without sacrificing much performance. If, however, you experience any lag in using the software, you might want to try setting this option to “high performance”.

Secondly, the filter selectivity can be switched between bandwidth per octave (BW) or quality factor (Q). This affects the units that will be used to express each filter’s steepness. By default, this is set to bandwidth per octave which is the standard used on all Martin Audio processors.

Usage tips

Here are some general tips on using the software:

- Double click any of the numeric values to have a slider bar pop up that let's you adjust values fast

- Right click the routing nodes to apply a gain adjustment at that point. This also works with the large stereo nodes on the inputs.

- Double click any of the graphical equalizers to have a separate version of it pop up in a new window. This way, you can keep the equalizer handy while accessing other parameters

- Use the "Stereo linker" option from the "edit" menu to link in- and output channels

- All parameters in a specific output channel can be copied and pasted using the copy/paste buttons beside the channel's name on the tab page. EQ's can be copy from the right side of their controls.

- To use a single preset in multiple projects, use the import and export function below the preset panel.

Frequently asked question

Q: What are the default passwords?

A: By default, only one password is used, which offers access to all functionality. Using this password you can change the passwords for all access levels. The default password is: "**MA**" (not case sensitive).

Q: I forgot my password! Please give me the master password.

A: There is no master password. If there was, it would spread amongst users in no time. A publicly known master password defeats the purpose of using passwords for security.

So write down your password(s) somewhere and put them in a safe place. If you lose them, contact your dealer.

Q: My computer won't connect to the Engineer, how do I solve this?

A: First, check to see if the com port you selected in the software isn't used by any other application on your computer and make sure you close all of them. Don't forget some of these programs can be working silently in the background. If you're not sure about this, (temporarily) uninstall the program.

Also, make sure you're using the serial cable provided within the package. Check to see that the port you selected in the software is the physical port the cable is connected to.

If this didn't help try running the software from a different computer. If it's working here chances are the serial port on your computer is broken. If you're sure it's not, please contact us.

Q: Will version X of the software work with version X of the Engineer hardware?

A: The Engineer control software will attempt to detect the version of the hardware you're using. In most cases it will let you know if it's incompatible with the hardware you're using. If you're using an older version of the Engineer 418 you can either use version 1 of the Engineer control software or update the firmware so you can use the Mk 2 software. Download both versions and the firmware upgrade from: http://www.martin-audio.com

Q: The software is performing badly on my computer. Can this be fixed?

A: For older computers, you might want to take a look at the render quality setting that can be found in the preferences dialog (see page 41 for more information).

Generally speaking, Microsoft Windows installations that have been in use for a long time or are clogged with unused installed software tend to perform badly. A fresh Windows installation can work miracles in this case.

Q: I'm getting pop-ups with error messages; the software hangs or even crashes. What to do?

A: The first thing to do is to check if you have the Microsoft .Net framework version 2.0 or above installed. You can get it from the Microsoft.com website. If this doesn't resolve the issue, please reinstall the latest version of the Engineer software.

If the problem still remains, please let us know, so that we can fix the issue. Please include as much information as you can: The version and language of your Windows installation, how the error can be reproduced and – if relevant – a screenshot of the error. We'll do our bests to correct the problem.

Notes

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