

Qmod 24.1 Manual

First Install

Create a Qmod folder in your home folder and unzip the files of the Qmod zipfile into the Qmod folder.

Check your audiocard settings in Windows i.e. samplerate, input- and output device. Start Qmod and go to Config. Select the device tab

When Qmod starts the first time the default samplerate is 48000.

For the best quality select the highest samplerate, preferable 192000 or else 96000 but for a lower computing power choose 44100 or 48000. However set the samplerate the same as your windows setting.

If you want to use the built-in stereo encoder, you'll need to set it to 192000 otherwise the stereo encoder is not available.

Choose the appropriate input- and output device or use the default input and output by selecting default.

If your audiocard supports ASIO select ASIO. This will give you the lowest latency - a samplerate of 192000 and a buffer size of 256 gives a latency of +/- 10 ms. However Qmod itself has some delay depending on the processing chain.

Set the buffer size in Qmod the same as the ASIO buffer size in your audio card setup. If you change the ASIO buffer size in your audiocard setup, you need to change the buffer size in Qmod.

Press **activate** and the settings are being initialized.

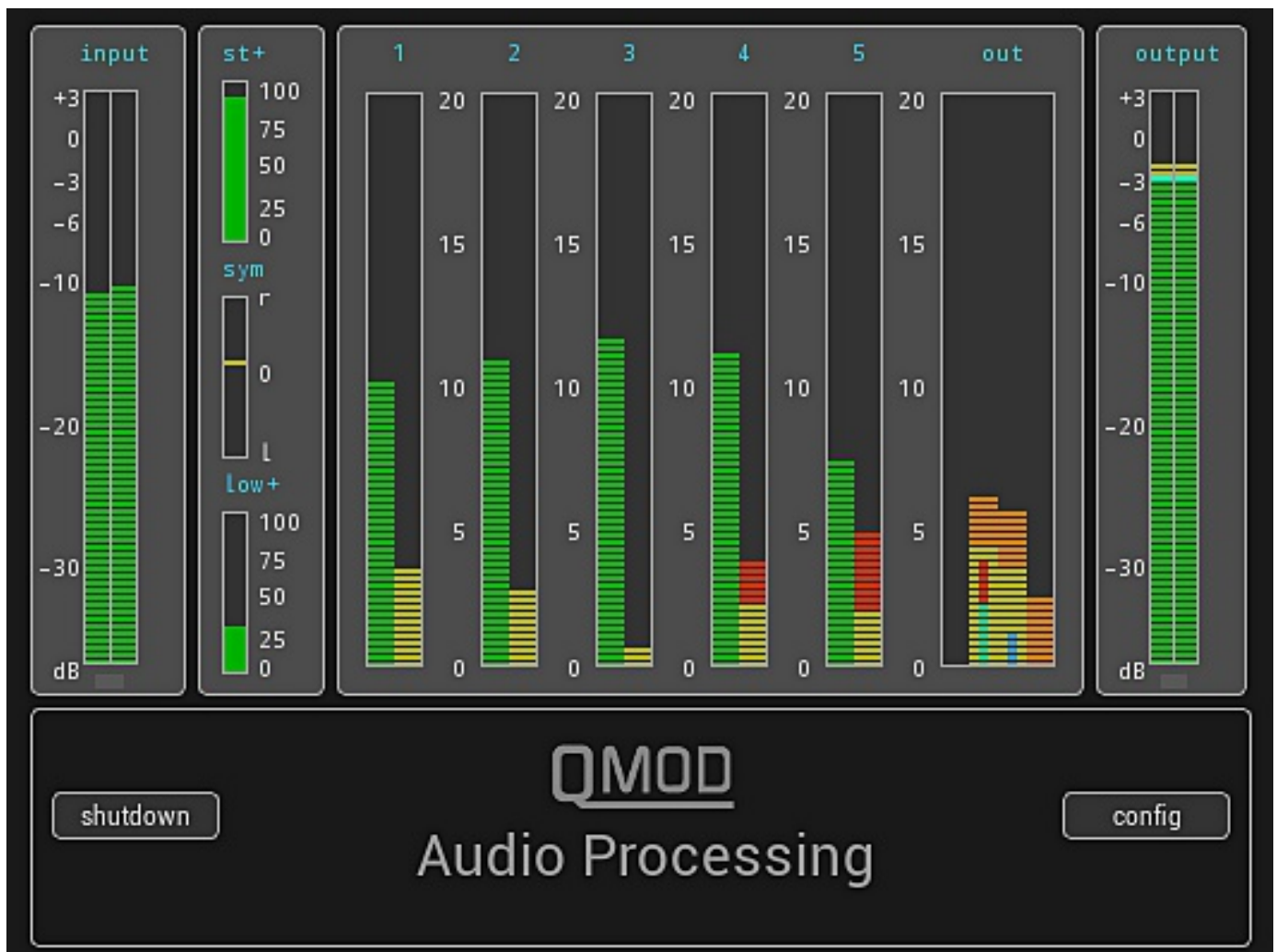
Qmod is ready to go.

Update to a newer version

If necessary rename the Qmod folder. Create a new Qmod folder and unzip the files of the new Qmod zipfile into the new folder. Copy the **Settings** folder from the old folder into the new one. Keep the old Qmod folder as back-up when the update is not ok.

If you want to run multiple instances of Qmod you'll need two or more audio cards and give every Qmod folder a different name. Mind that you also need copy the audio folder as you cannot select the audio folder.

Main Window



The left part of the main window gives you information about the input of the audio signal. You should keep the input signal to a maximum of -3dBFS to give you some headroom and avoid clipping in the ADC of the audio card. Equally the output signal should not reach 0dBFS to prevent intersample distortion. Qmod is setup in a way that the output is not at it's maximum but you should take care.

Beside the input display are three bars for stereo+, symmetry and low+.

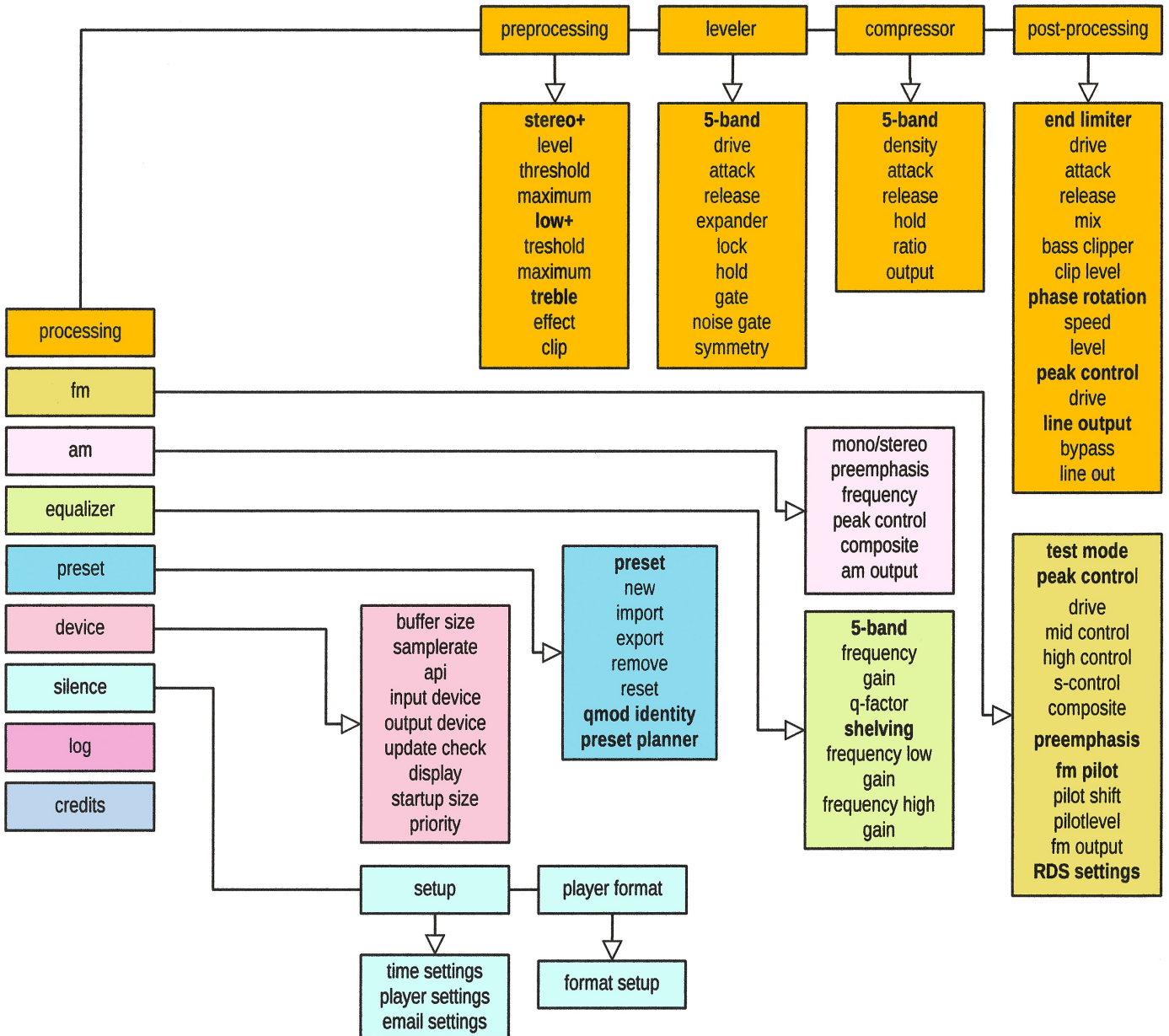
The band 1-5 displays show the leveling in green and the compression in yellow. The red on top of the yellow compression bar indicates the amount of overshoot clipping. The cyan bar indicates the noise muting.

The out display shows four signals. In order from left to right:

1. Expander.
2. End limiter (green/red for bass/clipping and yellow/orange for mid-high/clipping).
3. Peak control in yellow, (orange for fm peak control on top of yellow, blue for stereo clipping in fm mode).
4. Composite clipping.

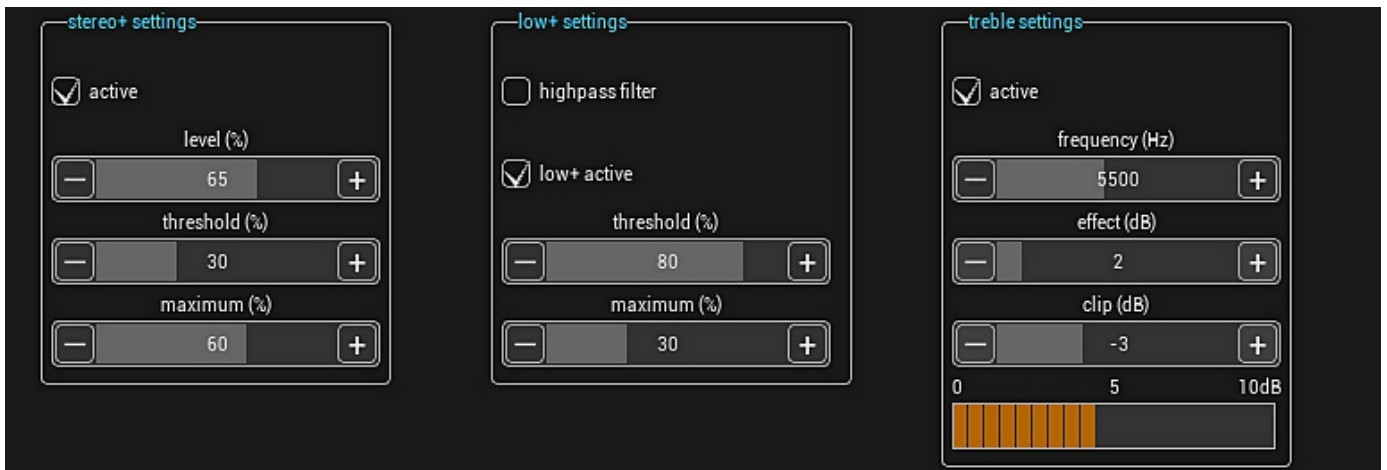
The most right is the L/R output. It's not the output of the audiocard because that one has most likely a separate audio control panel.

The picture below shows the configuration menu setup



A quick description of the panels and functions

Preprocessing





1. Stereo+ Settings

The checkbox **active** is to disable the stereo+ section
level controls the amount of added stereo.
Threshold controls the lowest working point.
Maximum controls the maximum amount added stereo.

The current values are shown in the main window.

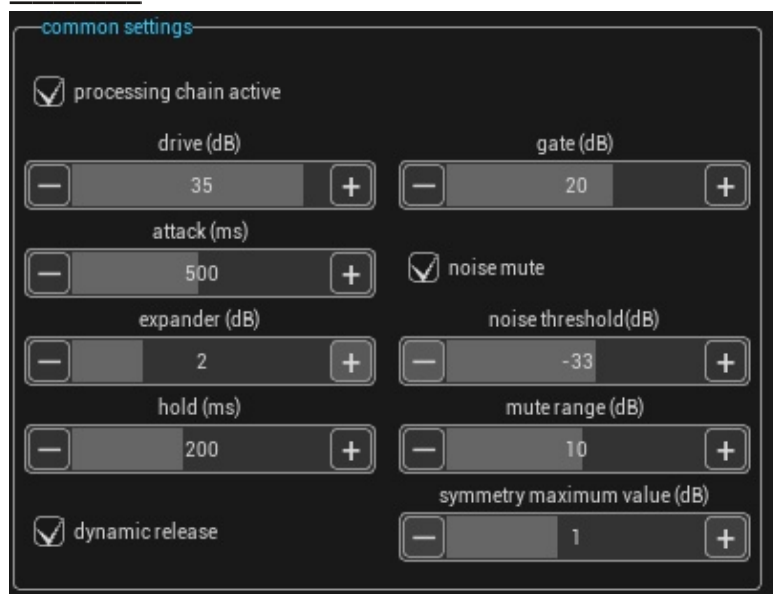
2. Low+ Settings

You can set the cutoff frequency of the highpass input filter. When highpass is checked the frequency is set to 40Hz otherwise it's 20Hz.
Checkbox enables the low+ algorithm. It enhances the very low frequencies.
Threshold button controls the amount of added low.
Maximum sets the maximum amount of low+ signal.

3. Treble+ Settings

Use **active** to enable this function.
 You can set the **frequency** and **gain** to the appropriate setting.
 The clip setting cuts off high overshoots above 5 khz. The level of clipping is shown on the display to the right.

Leveler



The checkbox **processing chain active** enables/disables the processing section. This option is not saved.

The Leveler Controls

Drive controls the amount signal that drives the five band leveler. The leveling is indicated in the main window by the green display.
Attack sets the attack time
Expander controls the amount of expanding the signal after it's processing by the compressors, to reduce the effect of gain riding somewhat.
Hold sets the time the leveler release

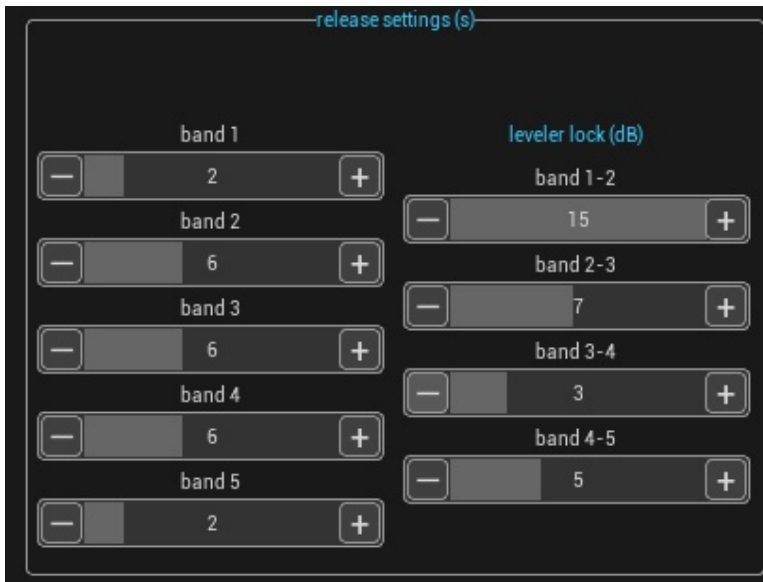
waits before releasing its value.

Gate sets the value of the level at which the input signal needs to drop in fixed time period to freeze the release of the leveler.

When you activate the noise mute section you set the threshold with **noise threshold**. at which the mute starts operating and the amount of muting with **noise mute** that is done.

Symmetry maximum value lets you set the maximum symmetry correction between left and right.

Dynamic release sets the release time depends on the signal.



With the **release** settings you can set the release time of each leveler, the release time measured with a drop of 10dB. **Leveler lock** controls the independancy of levelers. 15 means the maximum difference between two levelers is 15dB, 0 means no difference.

Compressor



Each band has its own control section.

Density controls the amount of compression of the compressors that follow the levelers. The compression is indicated by the yellow display in the main window.

The compressor works in two stages. The first stage is the compressor part and is indicated in bright yellow, the second stage is the limiter and is dark yellow.

The ratio setting sets the relation between the input and output of the compressor. A ratio of 8 means 8dB in is 1dB out.

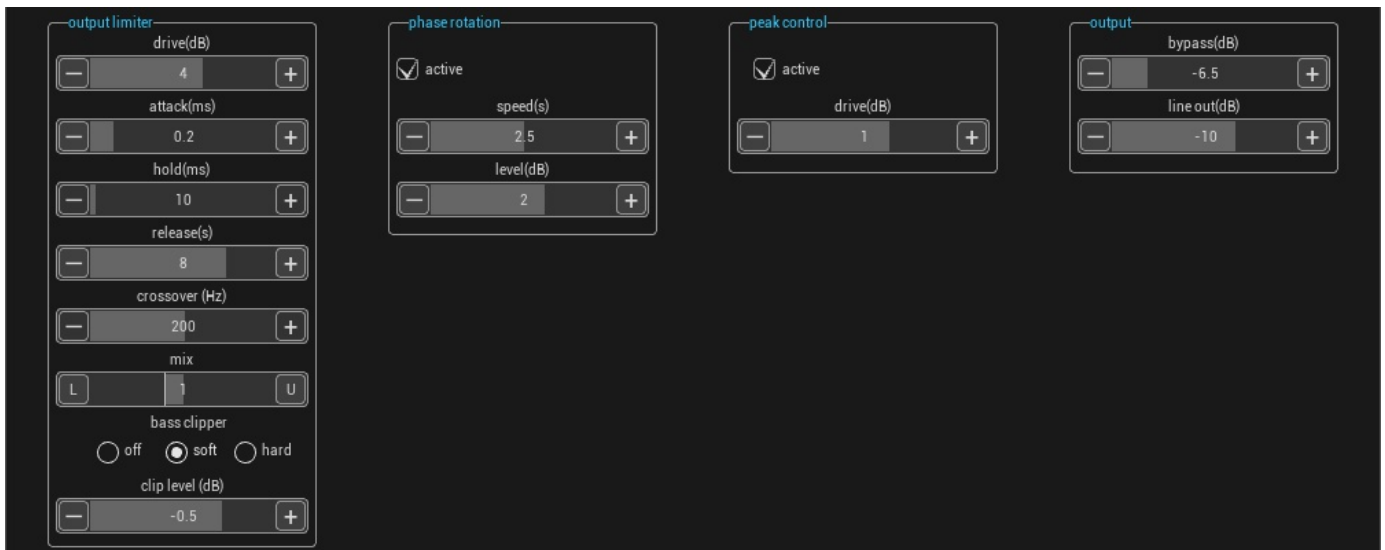
Attack sets the attack time of the compressor.

Release sets the rease time of the compressor. A release value of 0 activates the auto release function.

Hold sets time ahter the release starts.

Output sets the output of the compressor. It can be set from +10 to -10db.

Post-processing



The **output limiter** is a two band limiter that comes after the mixing of five output signals. You can set the **drive**, **attack**, **release** and **mix** of the limiter. L signals low band, U is the upper band of the **mix** button. The **crossover** frequency can be changed with the intended slider. In the main window the second bar of the out display indicates the limiting values, yellow is the upper band value and light green is the low band value.

The **Phase Rotator** can be used to minimize unbalanced sound parts. Especially in AM and FM broadcast when you need to rely on the clipper to prevent over modulated signals you want very little unbalanced clipping which causes second harmonic clipping. The **speed** button controls the speed with which the Phase Rotator reacts and the **level** control sets the sensitivity. When the Phase Rotator is active you'll see it in the main window in the symmetry display. The green bar lights up yellow.

The **peak control** is the last stage in the processing chain. The third bar of the out display in the main window shows the peak control value in yellow. The peak control has no attack and release. It eliminates excessive peaks in the audio signal. It can be reasonably used up to 6dB of overshoots without much distortion. It creates an extra delay of around +/-24 ms. When in FM or AM mode this control is forced on.

The **line out** button controls the overall output of the signal when in line mode. The AM or FM signal out has a separate control on the AM or FM page. Be careful if you set it to 0dB because there's a chance of intersample distortion.

FM

The fm tab is divided in two sections:

1. FM
2. RDS

1. FM

The screenshot shows a software interface for FM and RDS settings. The top section is titled 'fm' and contains several controls: a checked checkbox for 'fm processing active', three radio buttons for 'mono', 'st. mpx' (selected), and 'st. line', a 'cutoff freq. (Hz)' slider set to 15000, a 'peak control' section with a checked 'peak control active' checkbox and sliders for 'drive (dB)', 'mid control (dB)', 'high control (dB)', and 's control (dB)', a 'preemphasis' section with radio buttons for 'off', '50 µsec' (selected), and '75 µsec', and sliders for 'pilot shift (Hz)', 'pilot level (%)', and 'fm output (dB)'. The bottom section is titled 'rds settings' and includes checkboxes for 'rds active', 'rbds', and 'time sync', input fields for 'ps text (max. 8)', 'pi code (0-9/A-F)', 'alternative frequencies: fr1;fr2;fr3;fr4;fr5', 'program type', and 'radio text (max. 64 characters only a-z/A-Z/0-9)', a 'send data' button, and a 'close' button at the bottom.

The checkbox **fm processing active** activates the fm encoder. The default setting is unchecked. The three buttons **mono**, **st. mpx**, and **st. line** give you a choice which output option you want. **St. mpx** gives you a complete MPX signal, **st. line** has only the fm preprocessing when you want to use an external MPX encoder and produce a deemphasized line signal. The **cutoff frequency** sets the fm processing filter. It can be set between 10000 - 15000Hz in steps of 500Hz.

The radiobutton **preemphasis** gives you the possibility to select the desired preemphasis. It is only active when fm is checked.

The **fm peak control** section has five controls. **Mid-ctrl** and **high-ctrl** keep the mid and high frequencies in check, the **s-ctrl** limits the stereo part of the multiplex and the **composite** covers the combined audio with exclusion of the 19kHz pilot and RDS signal. The values are in dB.

To run some tests you can enable the built-in **tone generator**. It can produce the following frequencies 50-100-200-400-800-1600-3200-6400-12800-15000Hz: at 100%(high) and 23.5% (low) modulation.

The pilot tone is derived from the cpu clock and is very precise. But in case the 19kHz pilot tone isn't correct you can adjust it with the **pilot shift** button.

The level of the pilot tone is set to be 7% of MPX signal. In case you want to change this use the **pilot level** button.

composite sets the MPX clipper.

s-control controls the clipper on the L-R signal and thus reduces the stereo part in the MPX signal.

fm output sets the MPX signal output.

2. RDS/RBDS

RDS/RBDS functions only when the samplerate is set to 192000kHz and an appropriate sound card.

RDS/RBDS supports the basic features:

- Program Service (PS)
- Program Identity (PI)
- Program Type (PTY)
- Alternative Frequencies
- Radiotext (RT)
- Time/Date Setting

The checkbox **active** enables/disables RDS/RBDS.

The **Program Service** (PS) can be max. 8 characters long.

The **Program Identity** (PI) is a hex code of 4 characters.

The **Alternative Frequency** value is noted with a point as decimal fraction and in steps of 50kHz.

The **Program Type** can be selected from a list called by click on the button beside the displayed name.

If you want additional information fill in the **Radiotext** in alphanumeric digits with a maximum of 64 digits.

When all information has been put in then press **accept** and it will be send to the encoder.

AM



The checkbox **am processing active** enables/disables them am processing mode. You can choose if the processing is in mono or stereo by selecting **am mono** checkbox.

If you want to use the preemphasis check the **am preemphasis 75 µsec** box.

Choose the appropriate maximum frequency being transmitted by selecting the **range** radiobox. In Europe it's 4.5 kHz, the USA uses 5kHz.

If you select 4.5 or 5kHz only 4 bands are active, in 9 or 10kHz mode all 5 bands are active.

The peak control and the composite can be used to keep the AM signal within limits.

Equalizer

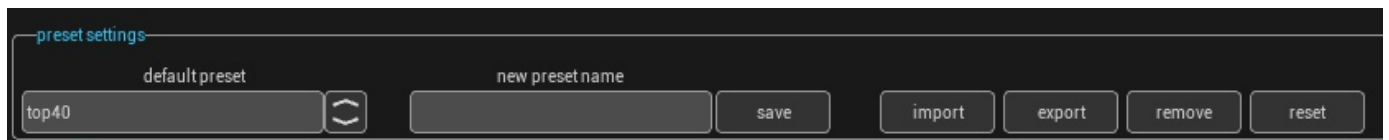


Each equalizer section has the possibility to set the **frequency**, **gain** and **q-factor**. The frequencies partly overlap so you can combine like the shown setting.

Every equalizer section can be enabled/disabled with the **active** checkbox.

You can select what position the equalizer has in the processing chain. Pre means before leveling and post means after the five band processing is done to correct some flatness in the signal.

Presets



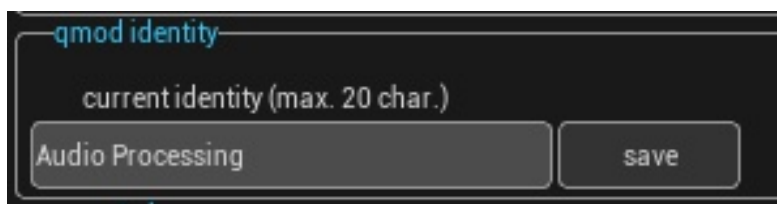
You can choose one of the preinstalled presets or make one yourself by changing a preset and give it a new name. The new settings will be saved by that name and selected the next time Qmod starts up.

If you upgrade from a version older than 17.4, your old setting will be saved as **"custom"** preset. You can also **import**, **export** or **remove** a preset.

The **reset** button resets the preset to its factory setting in case you want a fresh start.

You can select one of the included presets, modify it and save your settings with a new name as your favorite.

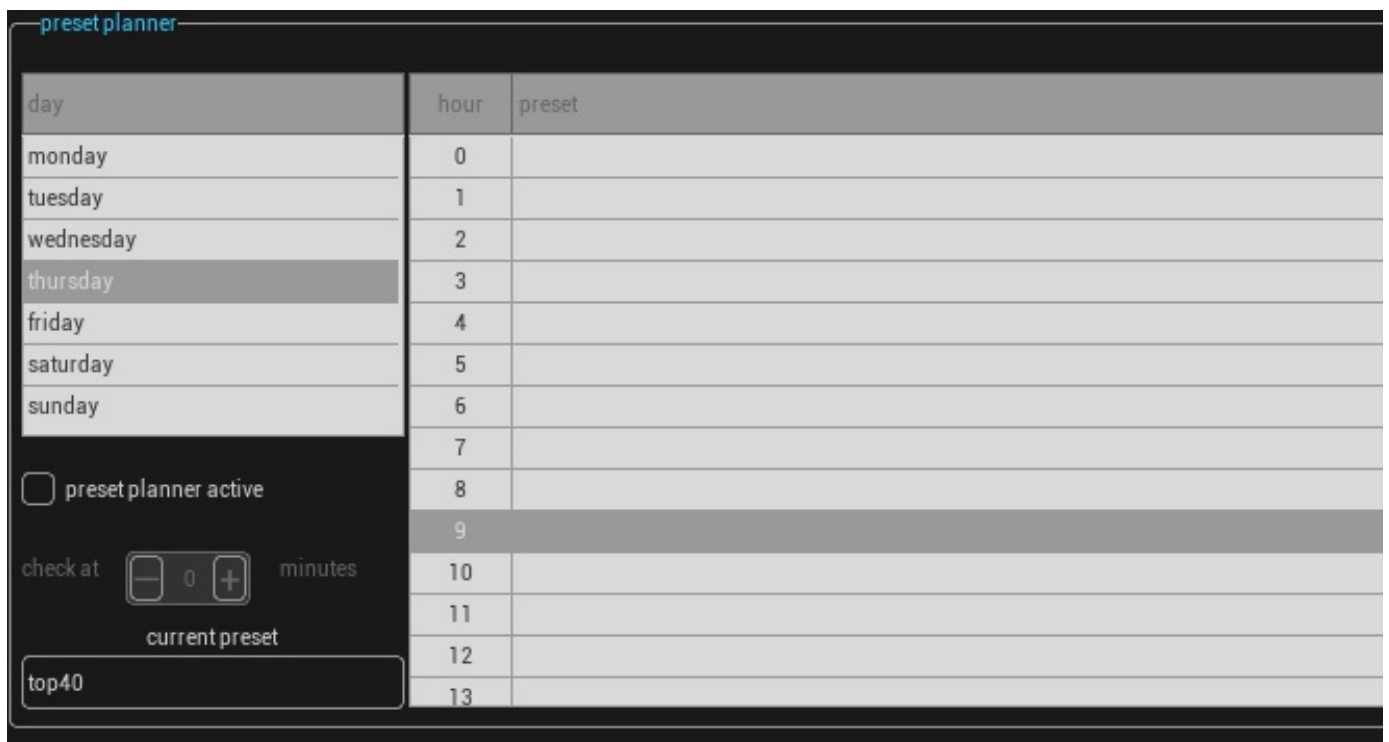
Identity



You can give your Qmod application a name. It can be handy if you're running two Qmod instances.

The name will be displayed in the main window.

Preset Planner



With the **preset planner** you can set a different preset for every hour or a range of hours of the day. When no preset is set for a specific hour the previous hour setting will be used. If no preset is set for the day the original preset setting is used.

Config Device

The screenshot shows the Qmod configuration window with two sections: 'audiocard settings' and 'common settings'. In the 'audiocard settings' section, there are three dropdown menus for 'buffer size' (set to 2048), 'samplerate' (set to 48000), and 'selected api' (set to WASAPI). Below these is a checkbox for 'use default soundcard' which is unchecked. There are two more dropdown menus for 'input device' (set to Line 3 (Breakaway Pipeline)) and 'output device' (set to Lijn (Steinberg UR22mkII)). An 'activate' button is located to the right of the input device dropdown. The 'common settings' section contains four checkboxes: 'check for updates' (unchecked), 'display top down' (unchecked), 'use last size at startup' (checked), and 'high priority active' (checked).

The default samplerate is 48000. You can change the setting by selecting a different value, especially if you like to use the built-in stereo encoder. The values to select from are **44100**, **48000**, **96000** and **192000**.

Select the desired buffer size and api. A larger buffer size uses less processor capacity when running with higher samplerates. Select the same size in the OS device setting.

The checkbox **use default soundcard** lets you select the default soundcard of the system you're running Qmod on.

After you've made a change press **activate** to reinitialize the audio card.

Common settings

When you activate **check for updates**, Qmod will check at startup for an update and once every 24 hours for the availability of an update. When successful it will show in the main window.

The option **display top down** gives you the possibility to select between showing the leveler and compressor meter for top to bottom or bottom to top which is default.

If you want to remember the last size when shutting down Qmod check the option **use last size at startup**. Next time when you start Qmod it will start at the last remembered size.

You can select **high priority active** when you want to run Qmod with a High Priority setting. In that case the OS will give Qmod a higher priority over other applications.

Player

The player panel contains five sub panels.

1. The setup of the backup player.
2. The format planner.
3. The container planner.
4. The message planner.
5. The playlist log viewer.

setup the player



When you want to use the backup player you'll need additional music have in place for this to work. When you already have a directory music available you can set that directory as target for the player. For more information see below in the format section.

When you want to use the silence detection for the first time, activate the player by checking **active**. After that you'll need to set the correct time when the backup player should start. Adjust the minutes and seconds to the appropriate time.

You can set the time after which the backup player starts in minutes and seconds. Also change the detection level when needed by setting **detection at**. The default setting is -40dB.

The player has some settings to make it run smoothly. You can change the detection level at which the current playing track wil end to start a new song by changing **song ends at**.

The player plays random and by setting **min. tracks before repeat** you can prevent it from playing the same track again to soon.

The option **gain reduction before containers** sets the fade out level before the container runs.

When the player is in action a display on the main window wil show the playlist.

To show the artist and title of the track correctly you'll have to name them the correct way like:

Artist - Title.wav (the sign between the artist and title should be the minus key on your keyboard).

The output of the player can be set with the player volume control.

The player operates with four playback channels. If you want to increase the number of channels you can do this by changing the setting in the qmod.ini, find the line NrOfChan:=X and change the value X. The minimum value is internally limited to 4.



Email notification

You can set an email notification when the player starts running. Enable **active**.

The **username** and **password** are optional when using a **NONE** authentication port setting like port 25 or 2525.

Mailserver, **send from** and **subject** are required, at least one **send to** field needs to be filled out.

When you leave the **message** field blank, the **subject** description will be used.

Use the **test** button to send a test email to check your setup.

Player format



Setting up a format is easy. In the **<QMOD DIR>/Audio/Music** and **<QMOD DIR>/Audio/Jingles** directory create sub directories. The names of the sub directories are the formats e.g. **<QMOD DIR>/Audio/Music/classics**, **<QMOD DIR>/Audio/Music/pop**, **<QMOD DIR>/Audio/Music/dance** or **<QMOD DIR>/Audio/Music/hits**. Put the audio tracks in the appropriate directories. Use an external program like Explorer to do so.

Qmod plays **WAV, FLAC, TTA and MP3 files in 2 channel 16bit stereo, 44100 or 48000 kHz**.

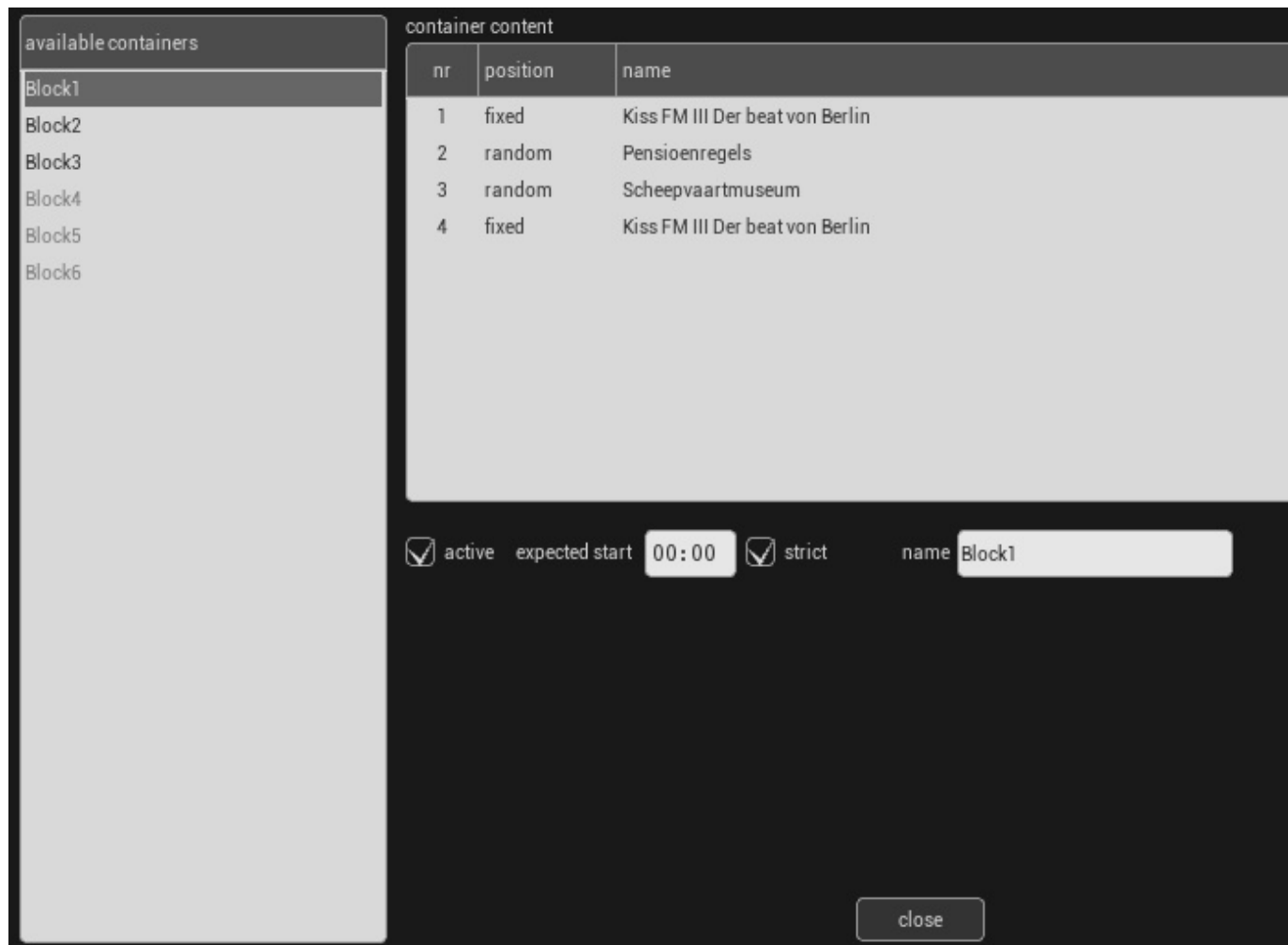
The player format panel has two report controls. the left one shows the formats that are currently available. Select music or jingle formats with the radio button above the left report control. Create the format by dragging a format item from the left to the right report control. When you want to remove a format item from the right report control drag it from right to left.

For moving an item move it up or down the report control.

The format list is scanned every few minutes so changes can take some time.

The option check double artist when checked scans the playlist for multiple entrees of an artist. It is important that the name is identical. **The check is case sensitive.**

Containers

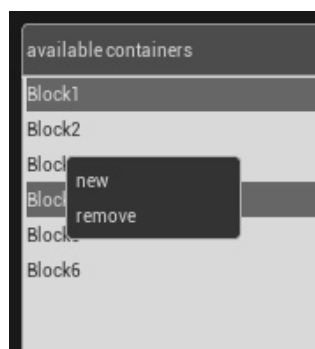


A container is used to place messages into the format. You give a new container a name, a approximated time in minutes when to run and the messages that will run.

If you want to start the container at the exact time check the **strict** option.

A message can be **fixed** in its place or **random** to swap places with other random messages.

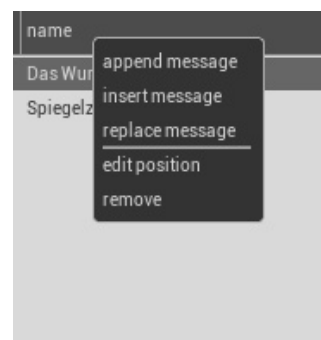
Uncheck the **active** option to disable the container from the scedule, It will be grayed out in the list.



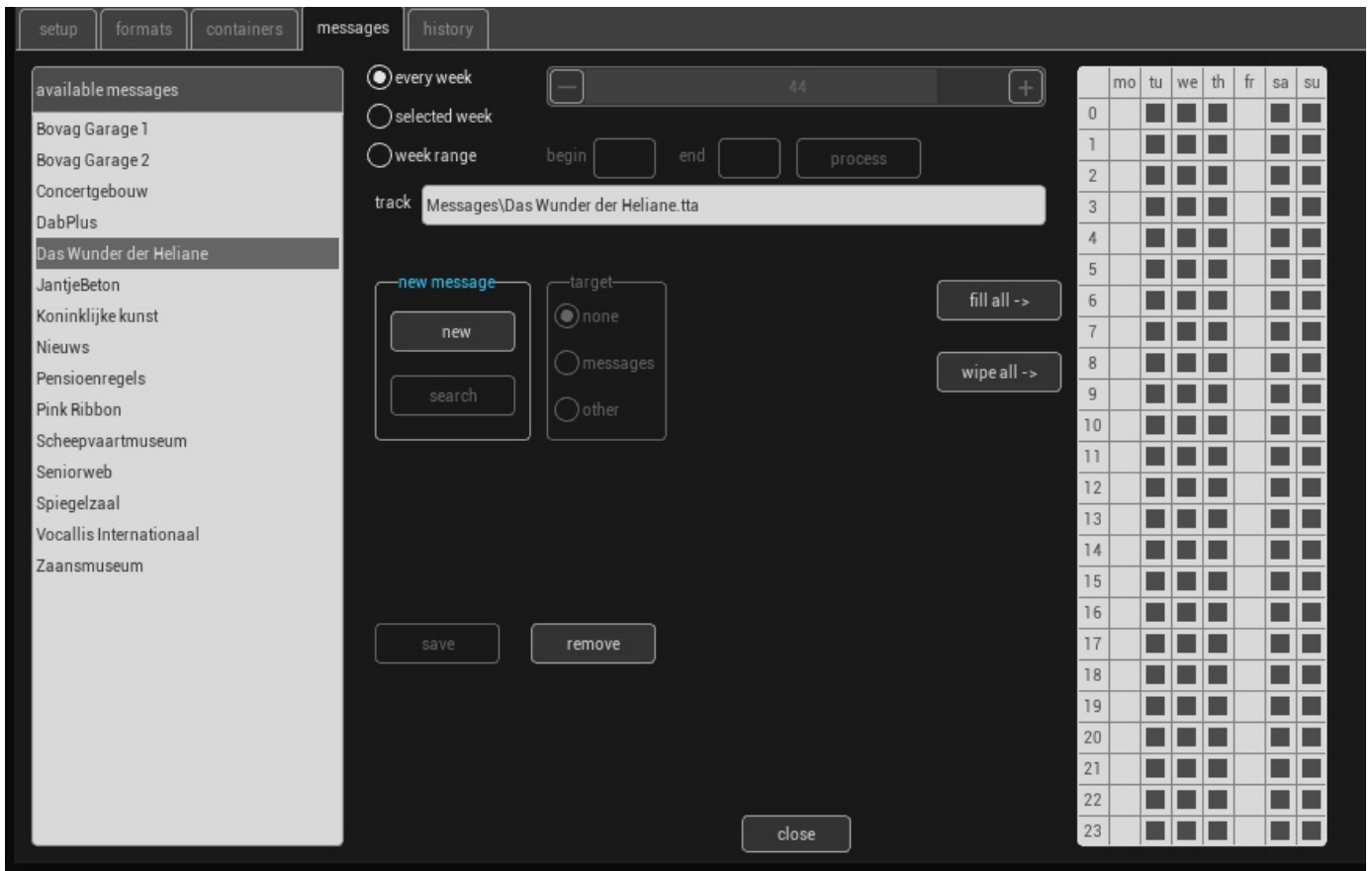
To create a new container right-click in the container list window.

To add a new message right-click in the message window and select append or insert a message from the list that will appear when messages are available.

See page messages how to setup a message.



Messages



A new message

To create a new message click new. Select which type it is: message or other in order to place it in correct folder.

Select if you want to run it every week or just (a) selected week(s), when choosing a selected week select the week by moving the slider, when choosing a week range og mutiple weeks fill in the begin and end week.

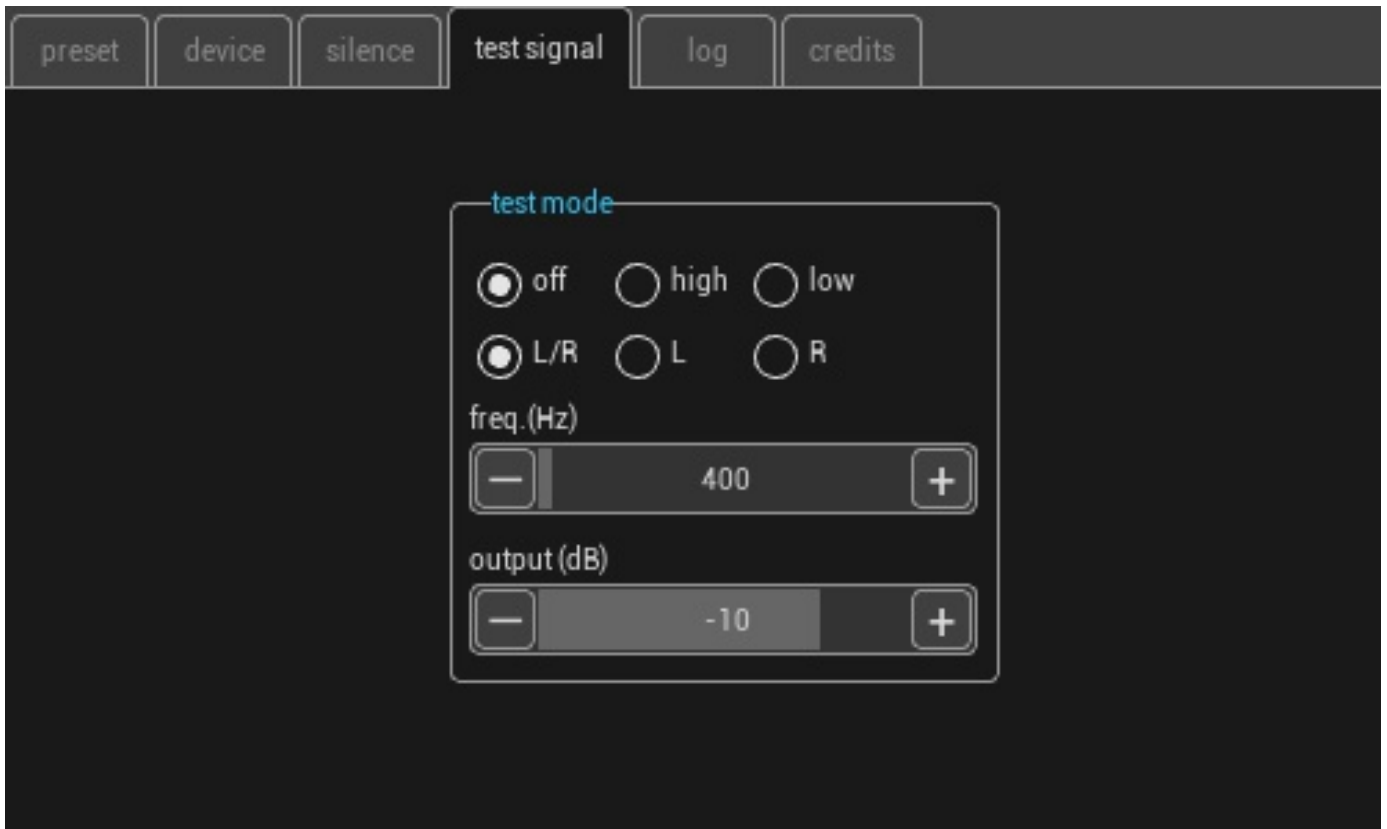
Choose the designated time slots by clicking the day(s) and hour(s).

If want to activate whole day click on the day(s) or when you want to run it on a certain hour slot click on the hour(s).

After that click process and the information is saved.

If you want to remove a message from the list your also beeing asked if you want to remove the sound file from disk.

Test signal



For setting up AM or FM processing you can use it to setup a signal.
The high mode refers to a value of 75kHz deviation for FM and 100% AM modulation.
After that you can the output to the modulator.
Fine tuning should don with a spectrum analyzer.

Log

The log panel gives you an overview of the errors or messages that occur during use.
After reading or copying you can delete all with the delete button