Qmod 26.2 Manual

Installation

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.

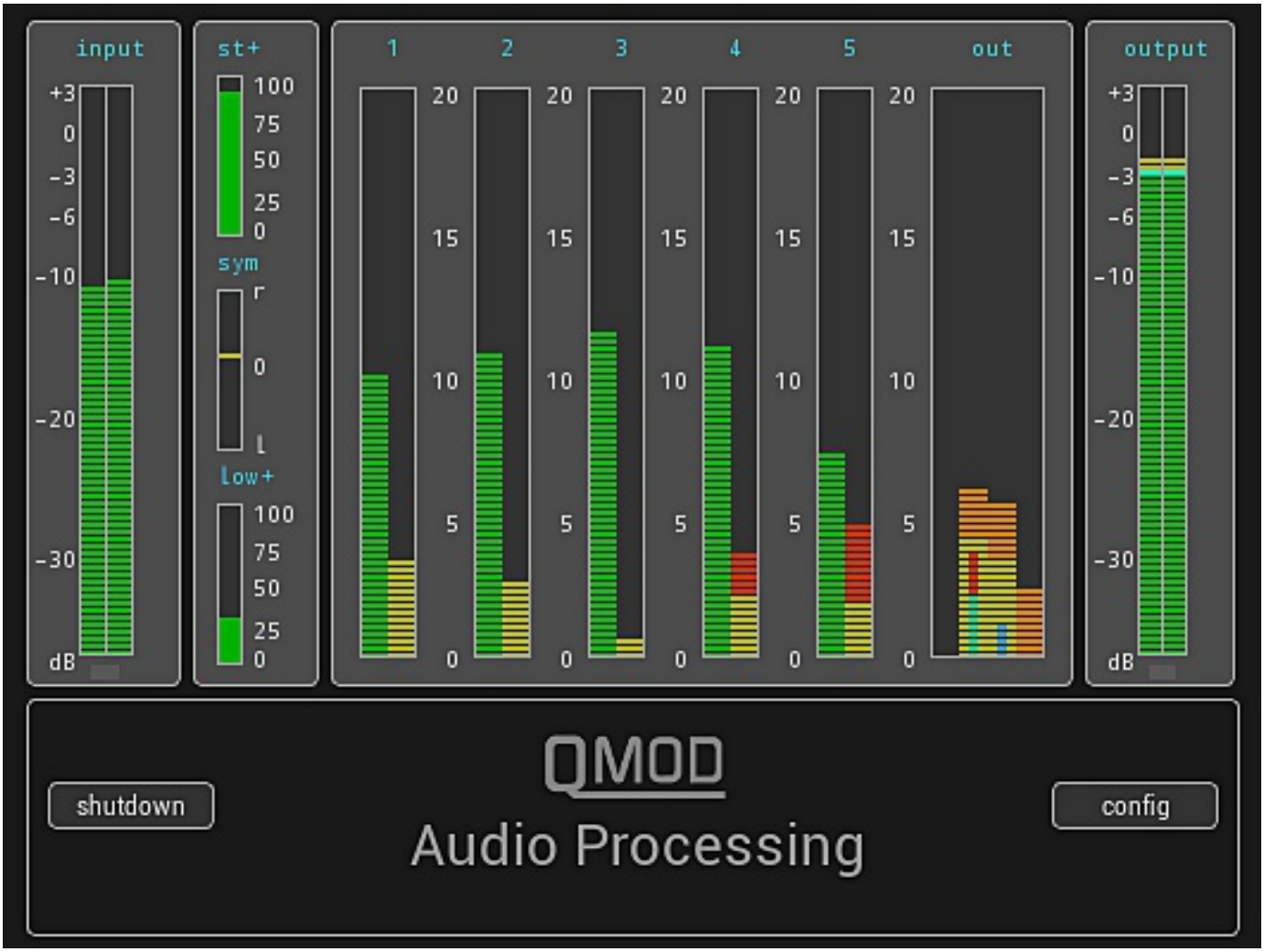
Qmod is ready to go.

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.

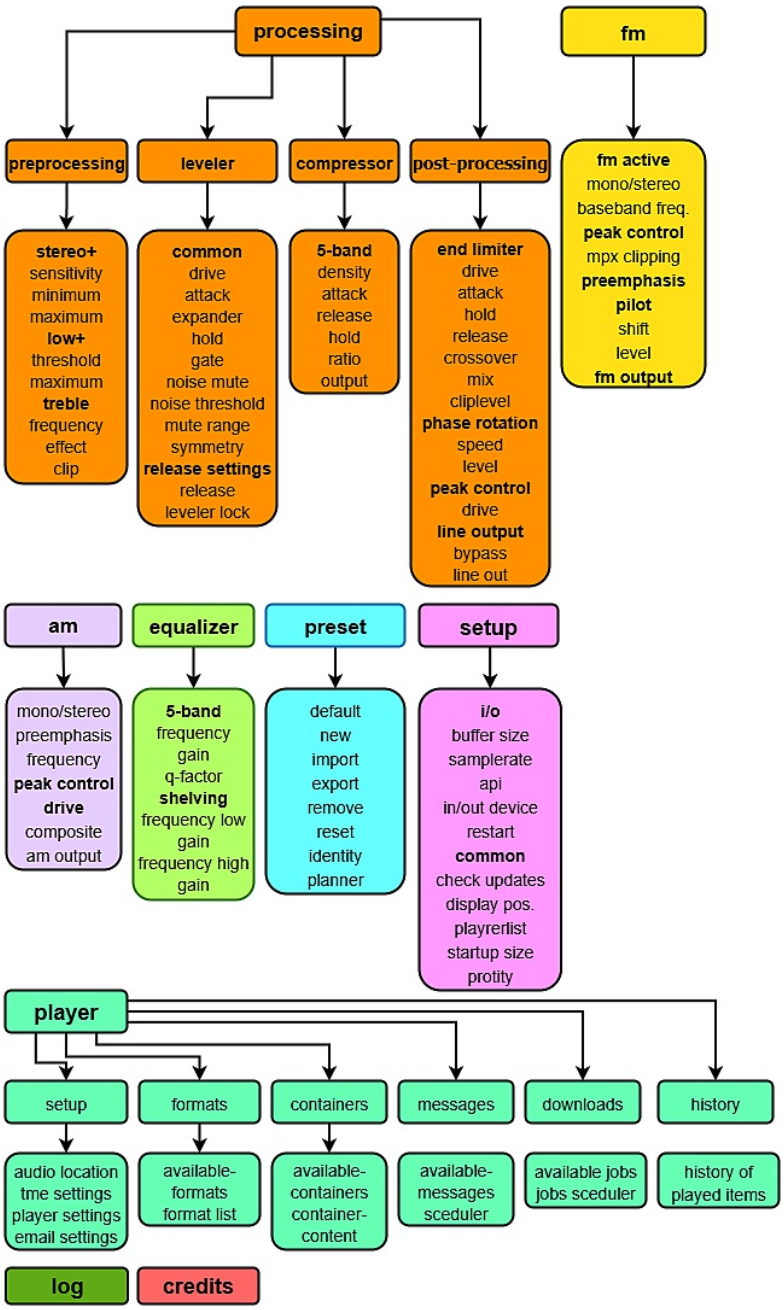
Beside the input display are three bars for stereo+, symmetry and low+.

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.

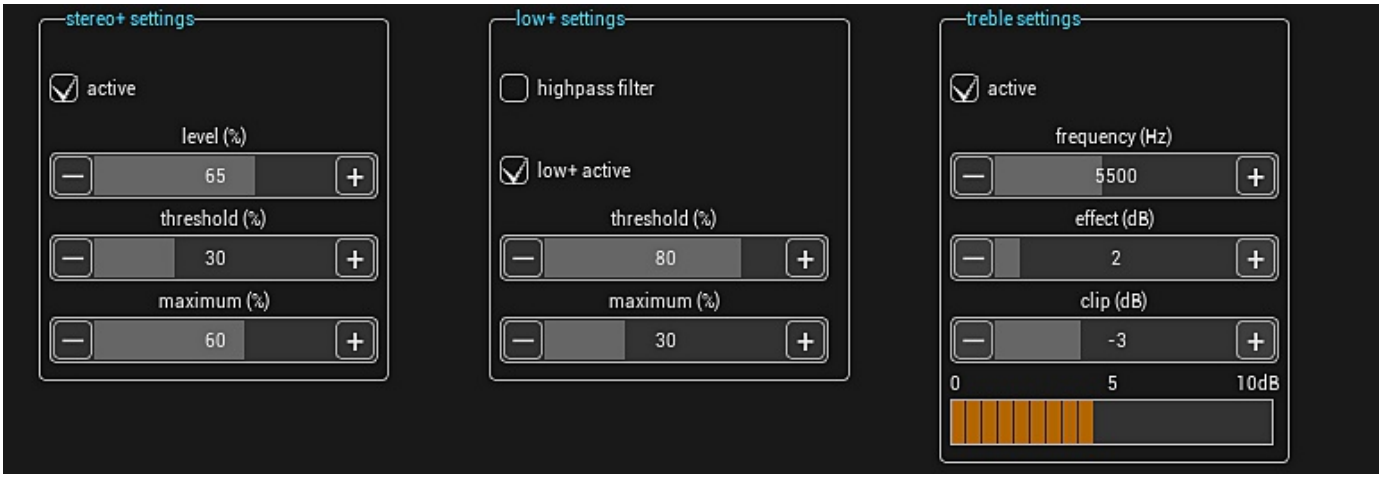
The picture below shows the configutation menu setup



A quick description of the panels and functions

Processing

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 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.

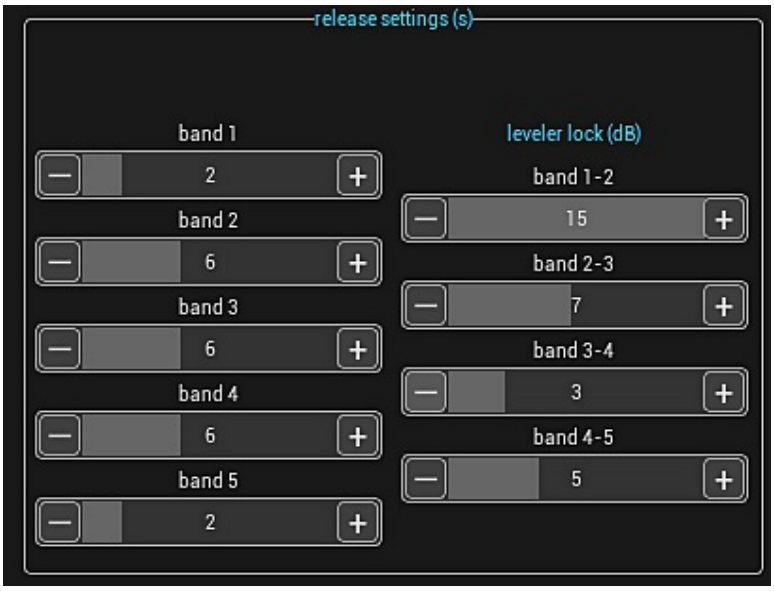
Leveler



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 secion 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.



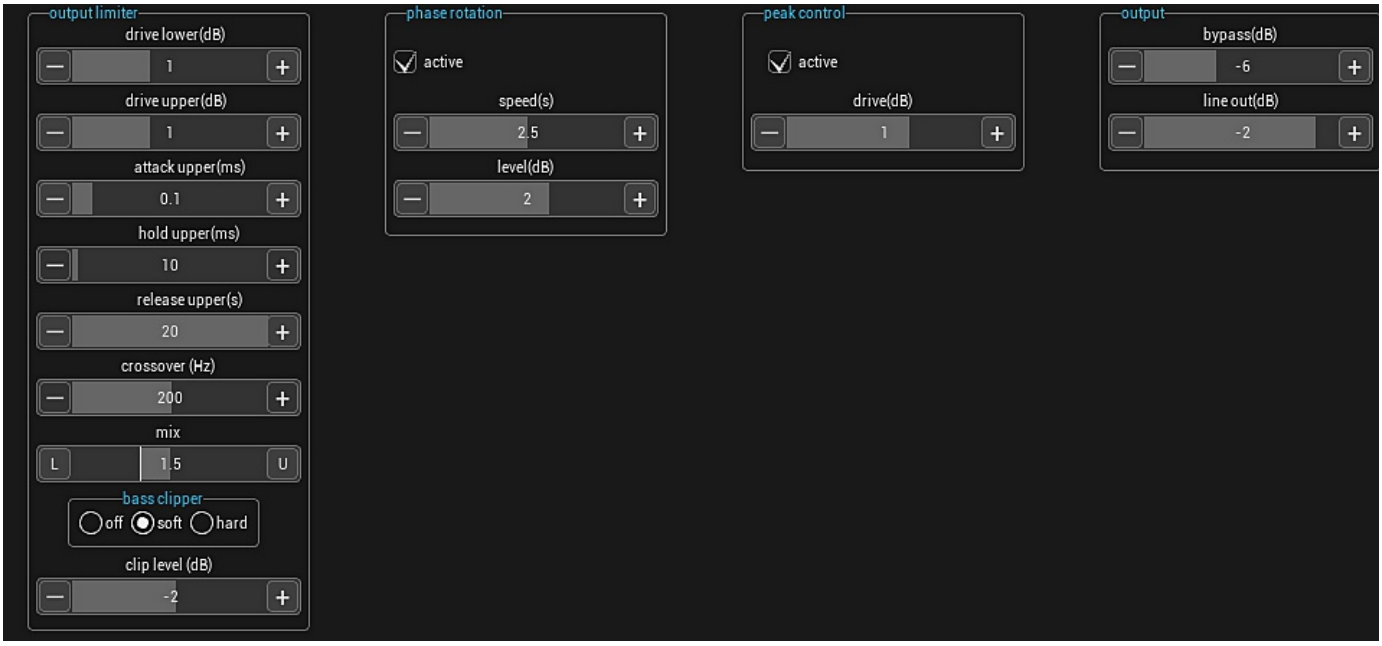
With the **release** settings you can set the release time of each leveler, the release time measured with a drop of 10dB.
Leveler lock controls the independancy of levelers. 15 means the maximum difference between two levelers is 15dB, 0 means no difference.

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 **crossove**r 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 the minimalize 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 set the sensititivity. When the Phase Rotator is active you'll see it in the main window in the symmetry display. The green bar lights up yellow.

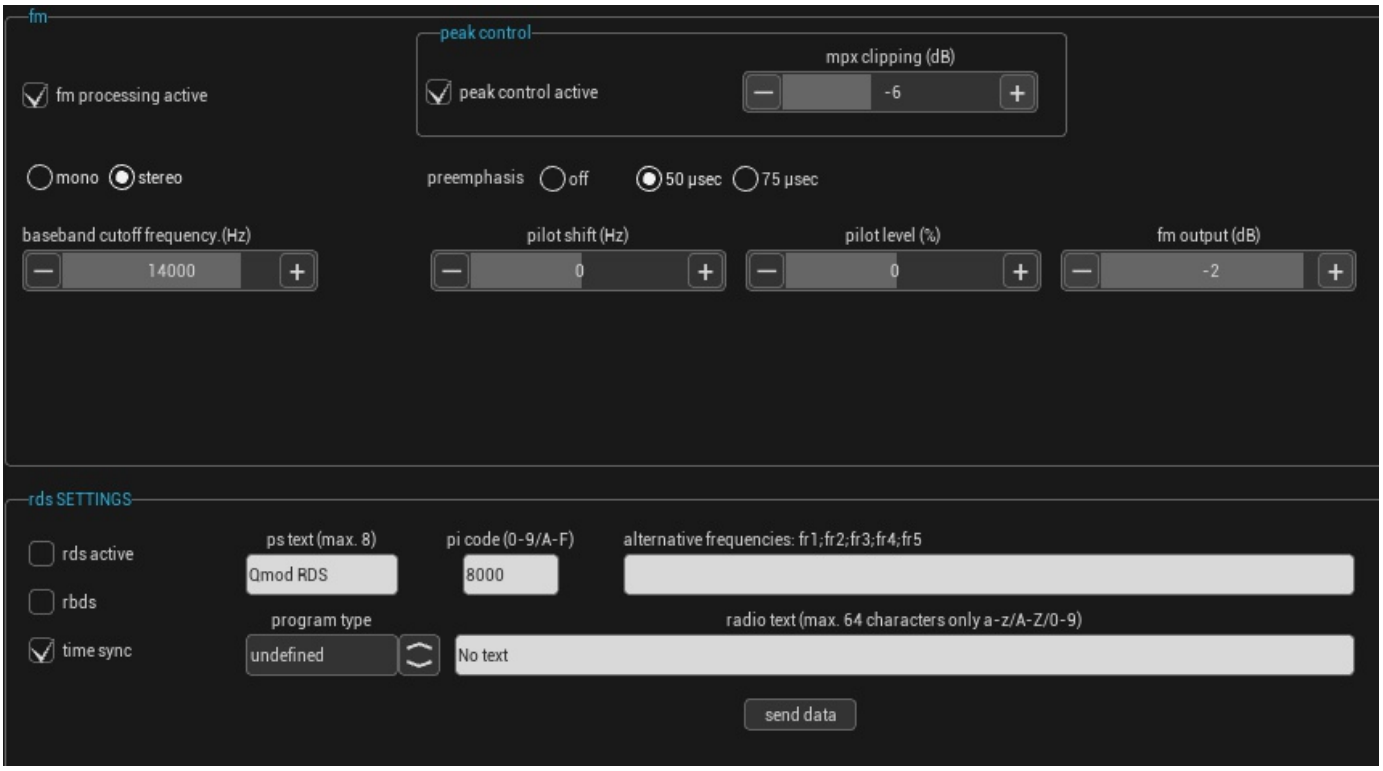
The **peak control** is the last stage in the processing chain. The third bar of the out display in the main window show the peak control value in yellow. The peak control has no attack and release. It eliminates excessive peaks in the audio signal. It can be reasonably used up to 6dB of overshoots without 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 checkbox **fm processing active** activates the fm encoder. The default setting is unchecked. The two buttons **mono and stereo** gives you a choice which output option you want. **Stereo** gives you a complete MPX 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** lets you set the clipping of the MPX signal. The value is 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.

fm output sets the MPX signal output.

2. RDS/RBDS

RDS/RBDS operates 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)
- Progtam Type (PTY)
- Alternative Frequencies
- Radiotext (RT)
- Time/Date Setting

De checkbox **active** enables/disables RDS/RBDS.
The **Program Service** (PS) can be max. 8 characters long.
The **Program Indentity** (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 besite the displayed name.
If you want aditional 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 yoy 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 en 10kHz mode all 5 bands are active.
The peak control and the composite can be used the keep the am signal witin limits.

Equalizer

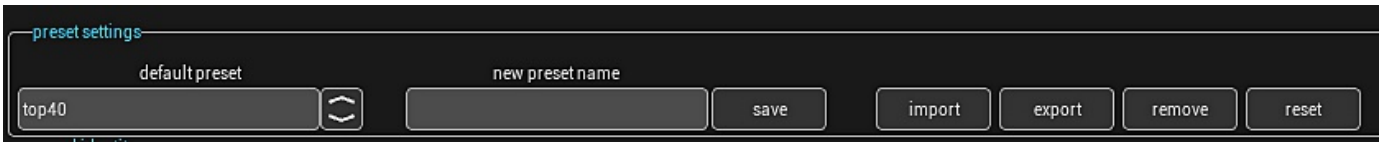


Each equalizer section has the possibility to set the **frequency**, **gain** and **q-factor**. The frequecies partly overlap so you can combine like the shown setting.

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

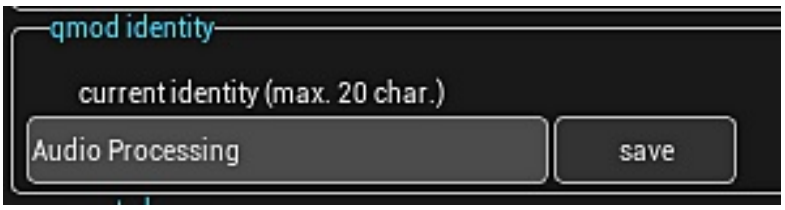
You can select what position the equalizer has in the processing chain. Pre means before leveling and post means after the five band processing is done to correct some flatness in the signal.

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 wil 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 wil 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



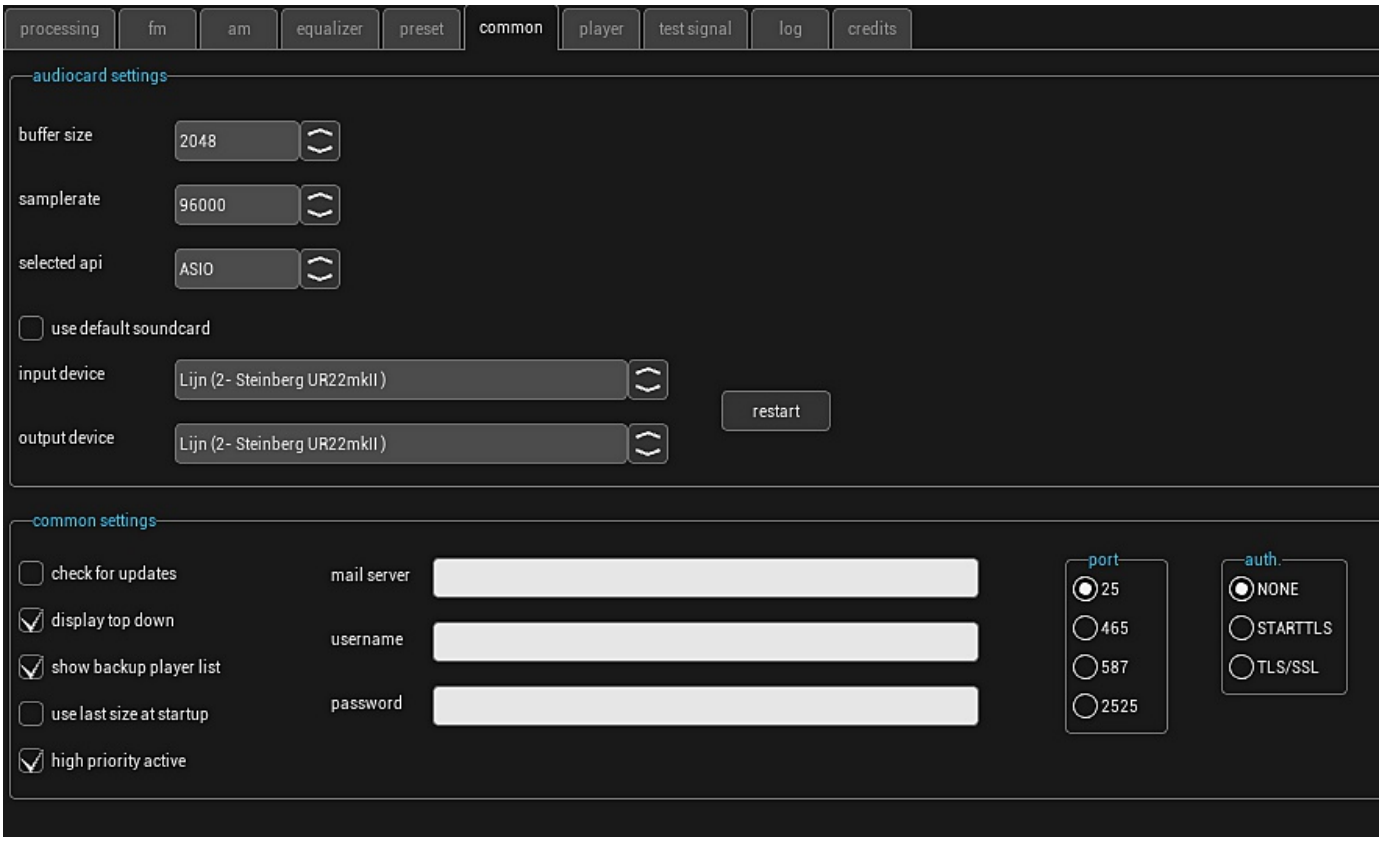
You can give your Qmod application a name. It can be handy if you're running two Qmod instances.
De name will be displayed in the main window.

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 orignal preset setting is used.

Setup



In / out settings

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 de OS device settting.
The checkbox **use default soundcard** lets you select the default soundcard of system you're running Qmod on.
After you've made a change press **activate** to reinitialize the audio card.

Common settings

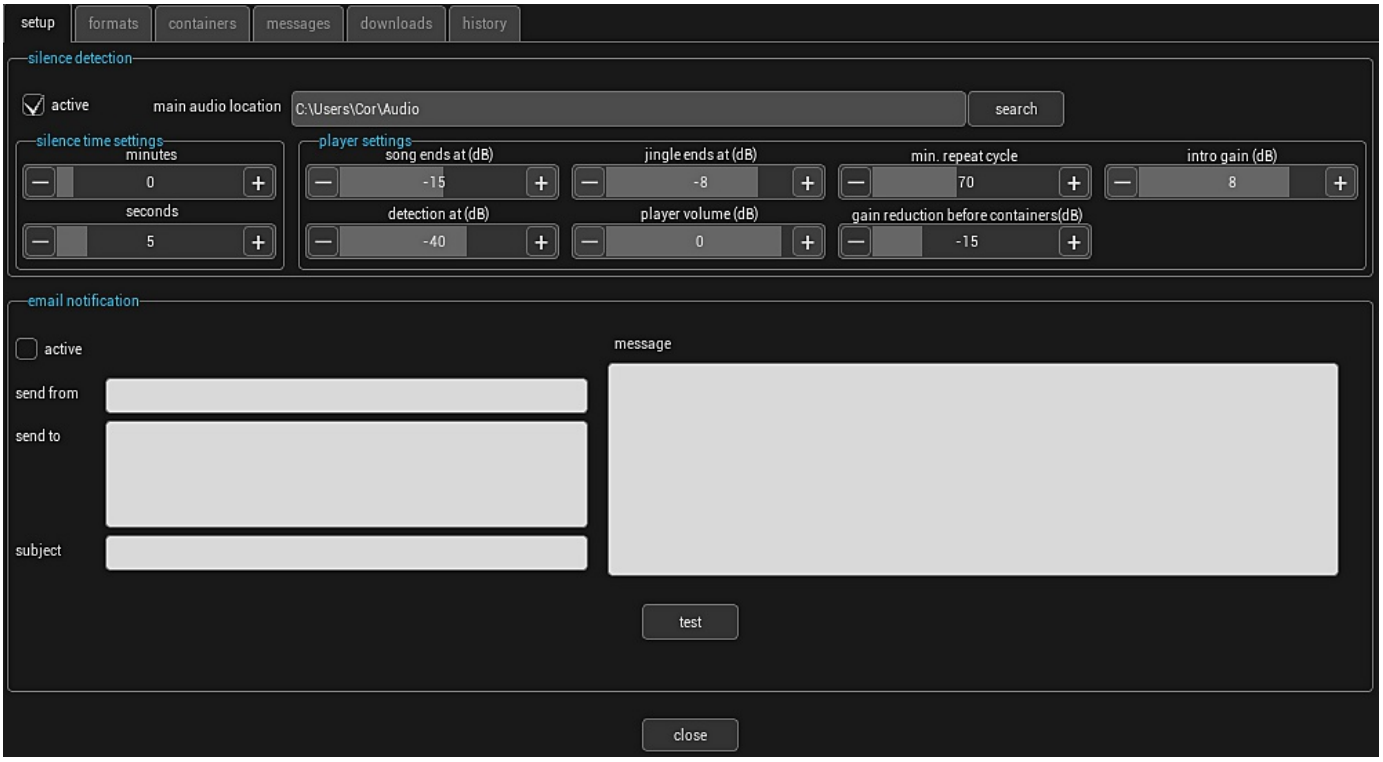
When you activate **check for updates**, Qmod will check at startup for update and once every 24 hours for the availability of an update. When successful it will show in the main window.
The option **display top down** gives you the possibility to select between showing the leveler and compressor meter 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 wil give Qmod a higher priority over other applications.

For sending emails in the player section you will need to setup the SMTP server details.
Fill-in server address, username, password, port and authentication protocol.

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 availlable 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.
The **intro gain** value is used to elevate to intro especially with very low level intros.
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 outputput 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 you can set an email notification when the player starts running because the input signal is lost. Enable **active**.

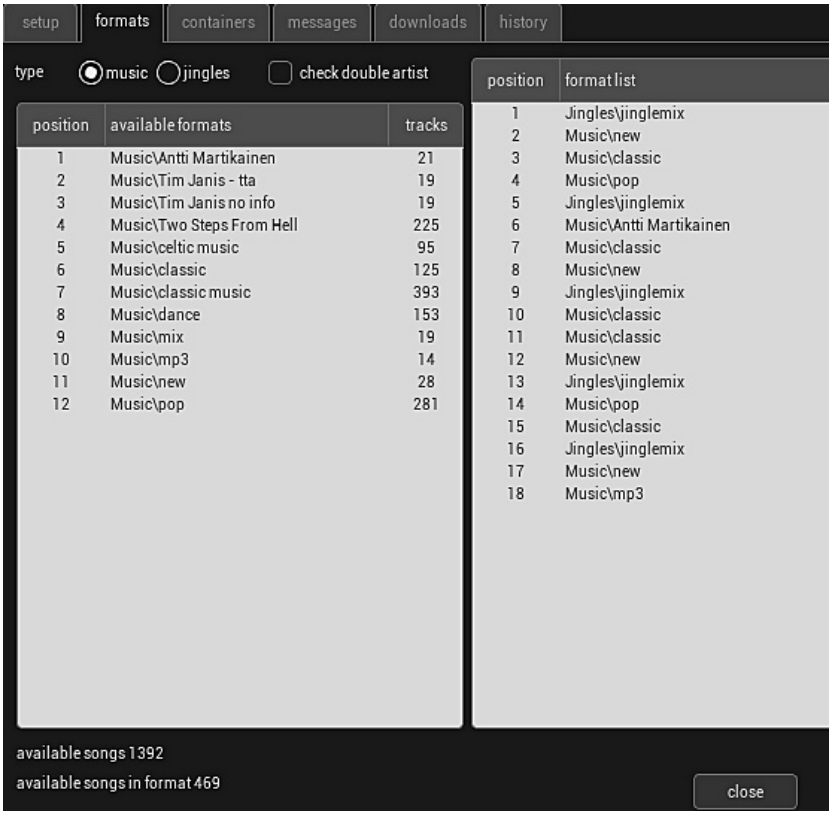
To send any email you'll first needs to setup the mailserver. Fill-in **username** and **password** when necessary (port 25 or 2525 when using a **NONE** authentication)

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

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

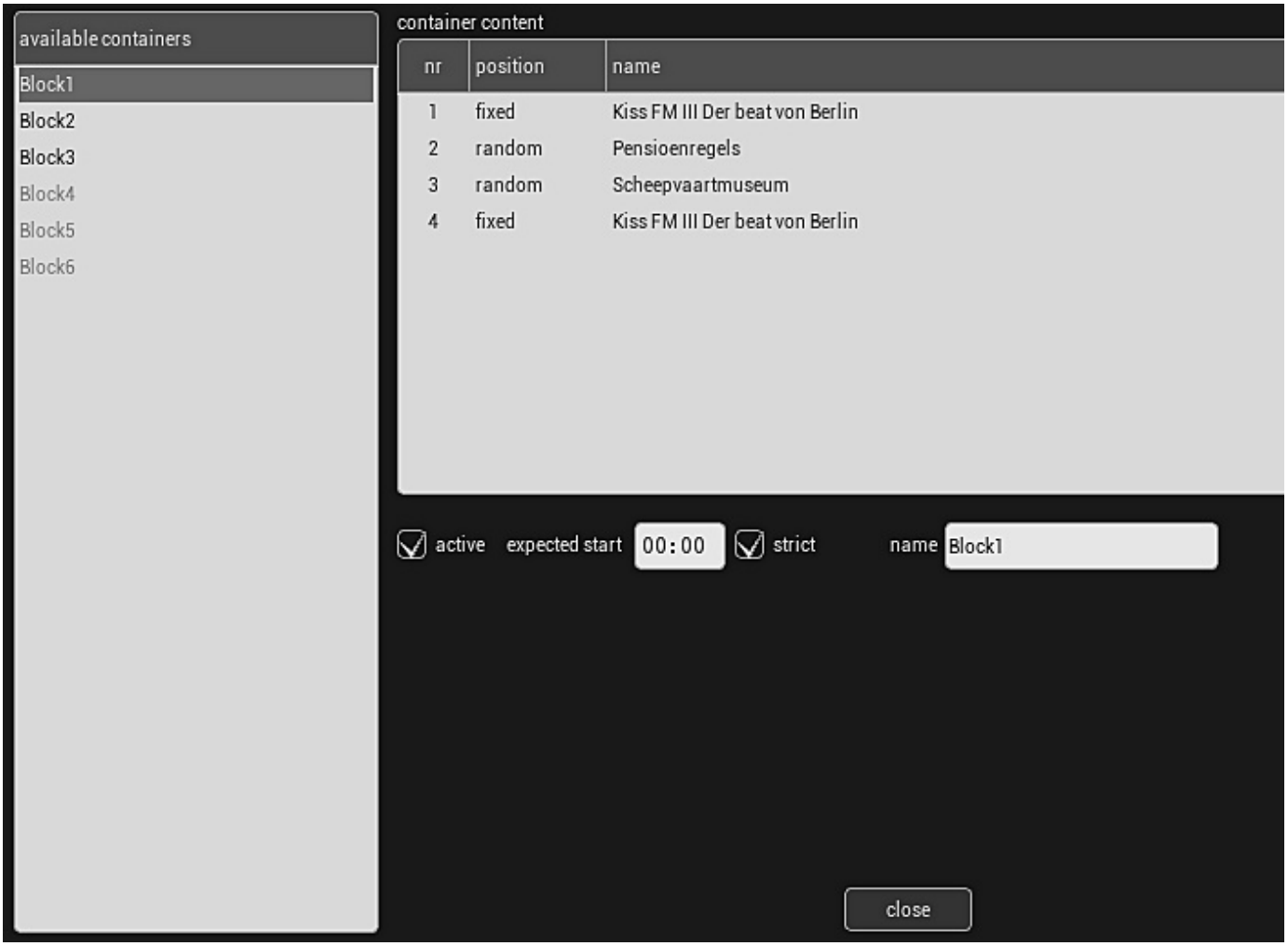
Use the **test** button to send a test email to check your setup.

Format

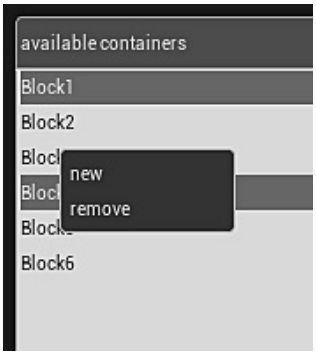


More information about using the format planner see the section **Audio library setup** and **Creating a format.**

Containers



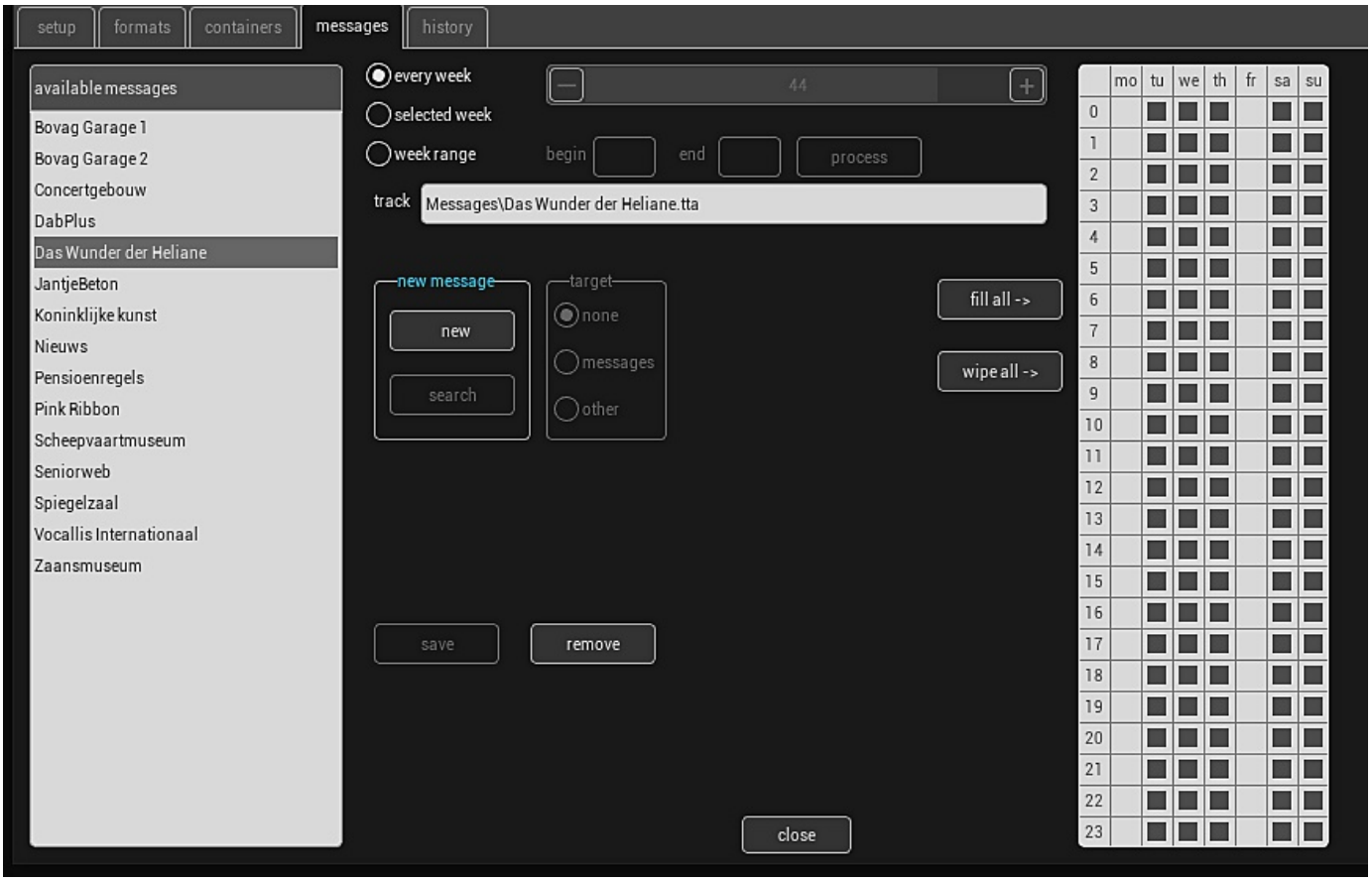
A container is used to place messages into the format. You give a new container a name, a approximated time in minutes when to run and the messages that will run.
If you want to start the container at the exact time check the **strict** option.
A message can be **fixed** in its place or **random** to 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 availlable.
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 chosing a selected week select the week by moving the slider, when chosing 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).

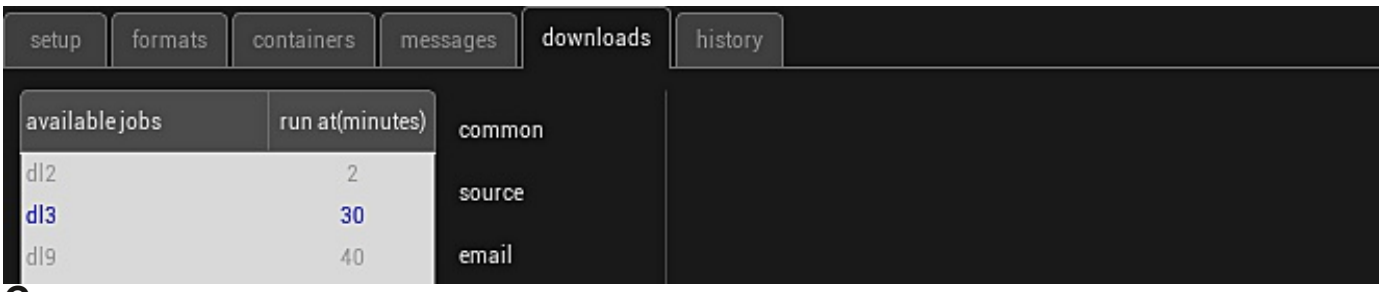
If want toactivate whole day click on the day(s) or when you want to run it on a certain hour slot click on the hour(s).

After that click process and the information is saved.

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

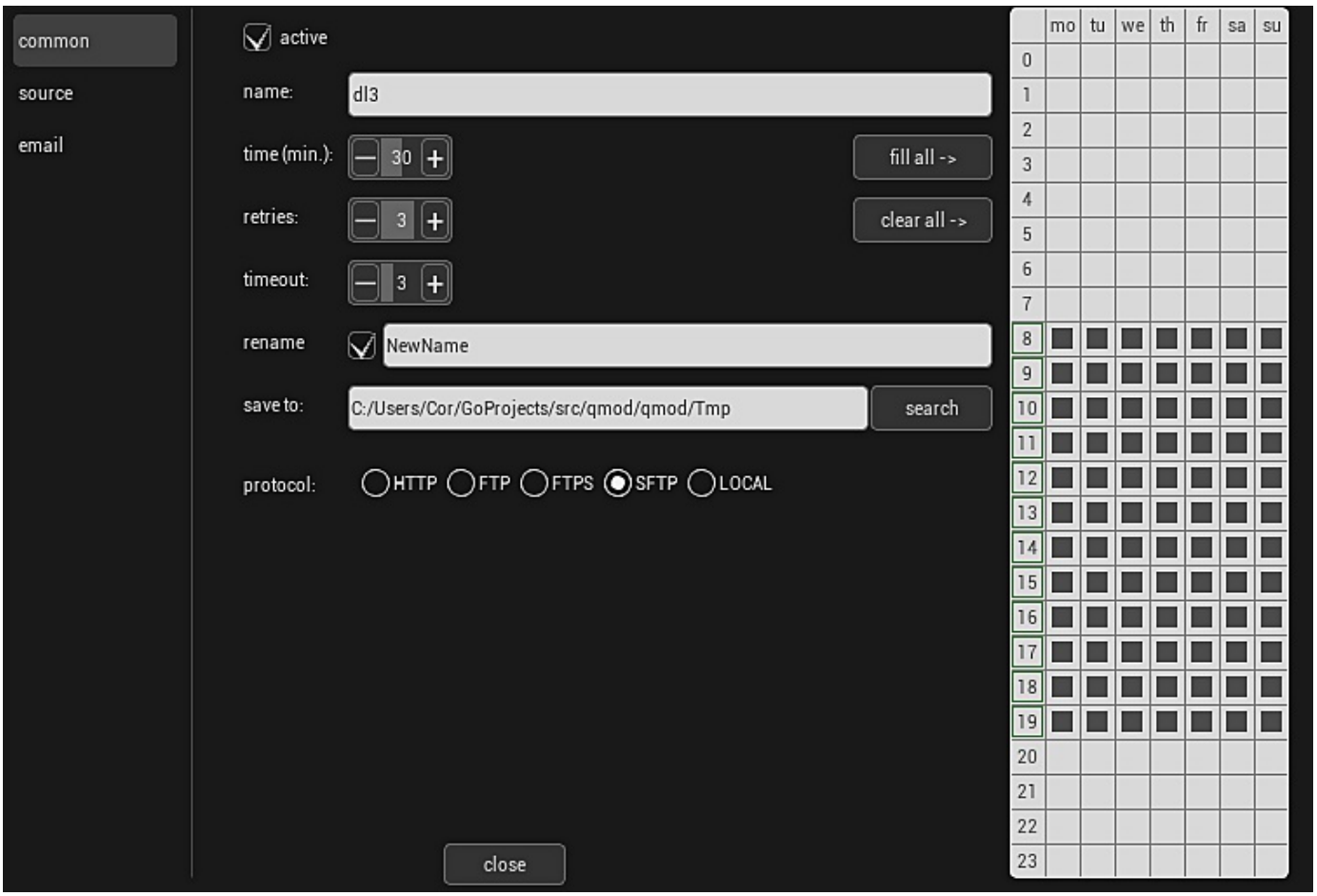
Downloads

In the download section you can setup commands to download from external servers.
That can be a a local server or NAS, or a server somewhere in the cloud.
The setup contains three sections: **common, source and email.**



Common

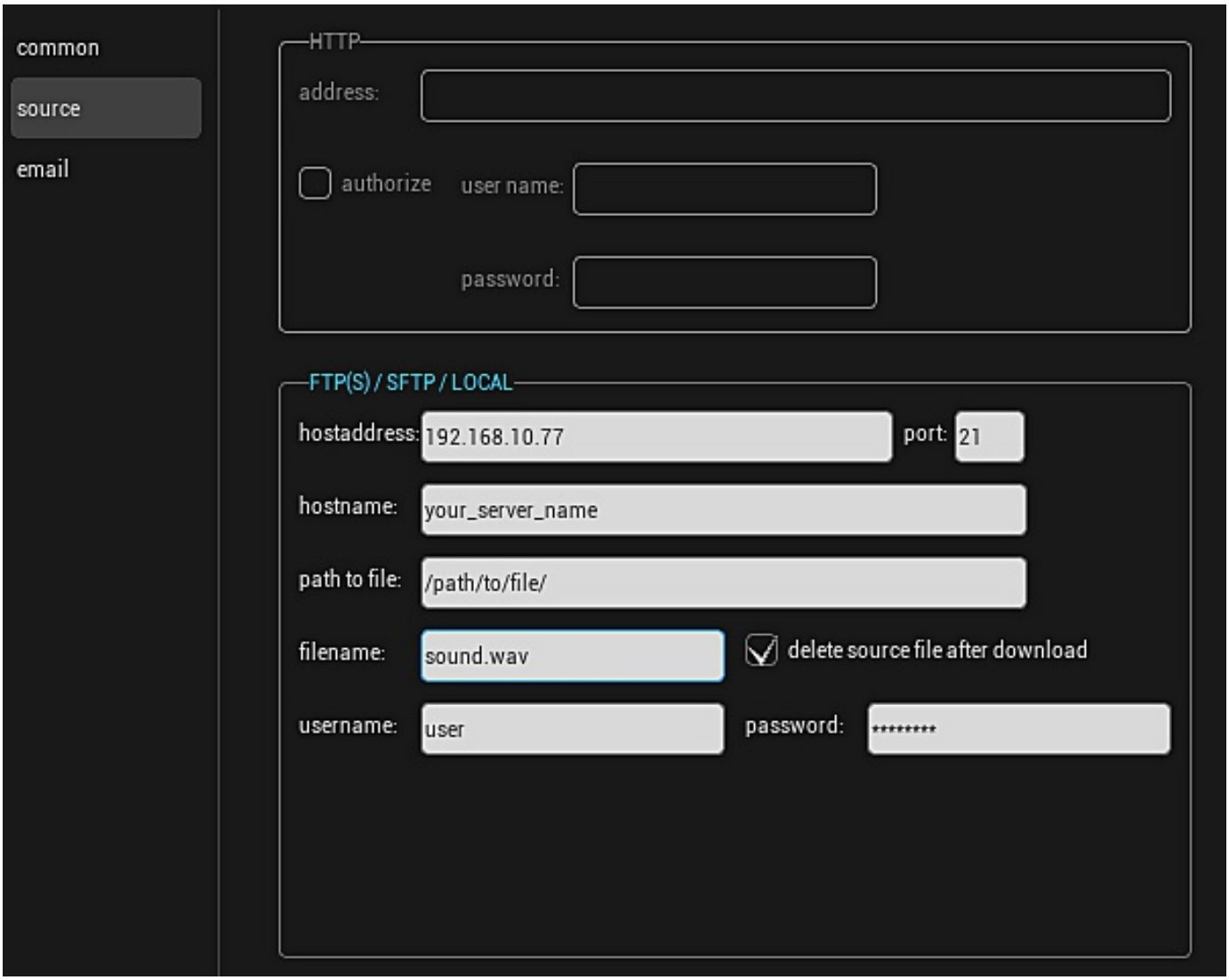
When you create a new job you have to give it a name and its being displayed here.
Set time, retries of download attempts and a timeout.
Check rename to rename the downloaded file to your choice and folder where to save the file when needed. Plan the hours and days when the download should take place and the protocol to use which depends on what the supplier of the download directs you to use.
Local is used for downloading from your own network or computer.



Source

The next step is setting up the source of the download.
Depending on the protocol selection either the HTTP section or the FTP(S) is active with LOCAL there is no IP address and port available.
Fill-in the IP address and port, the path to the file(it needs to begin and end with a slash), the filename, username and password.
if you are using the FTPS option and the hostname is necessary fill-in the hostname of the server.

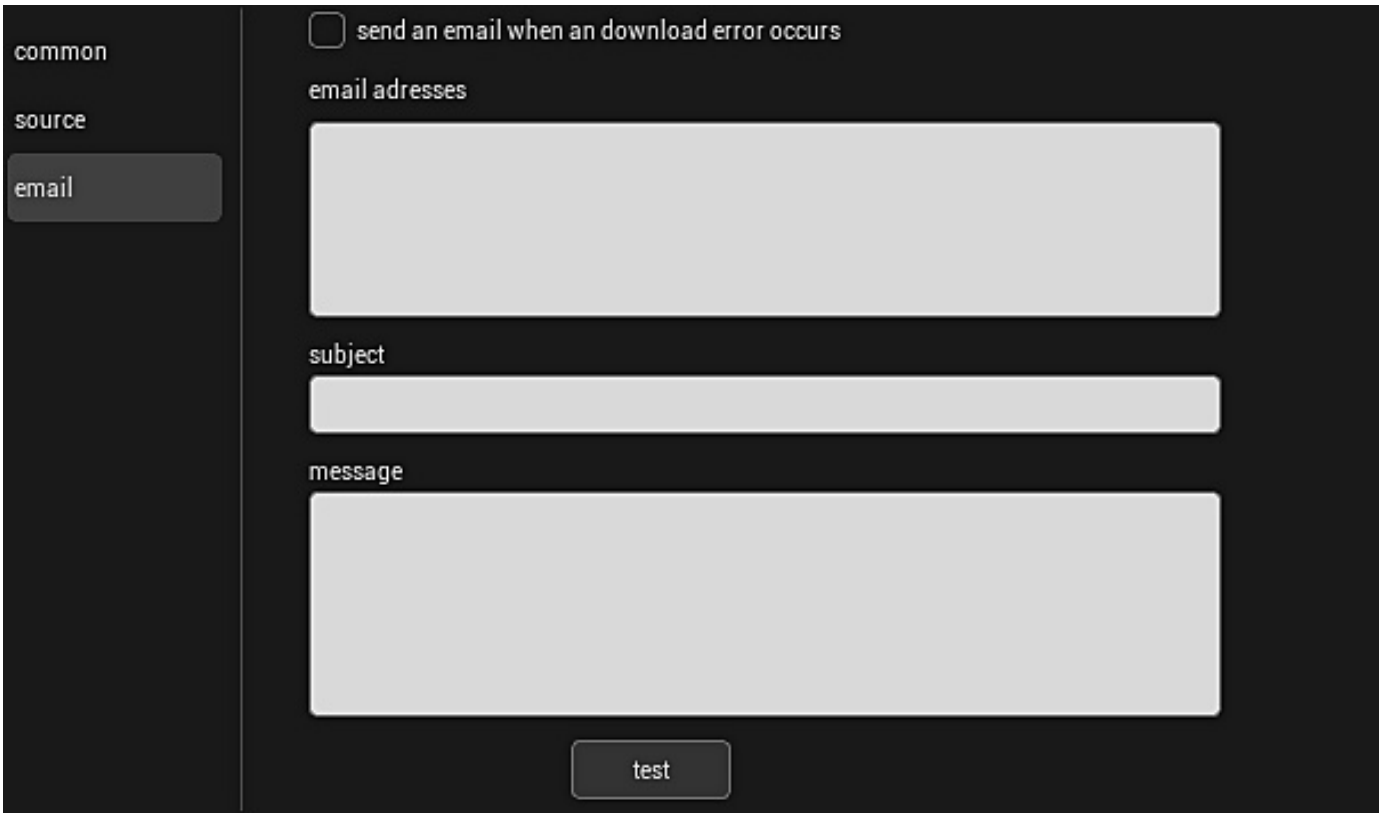
If you are using the SFTP protocol fill-in the path to the private SSH certificate file and when required the passphrase for the certificate file.



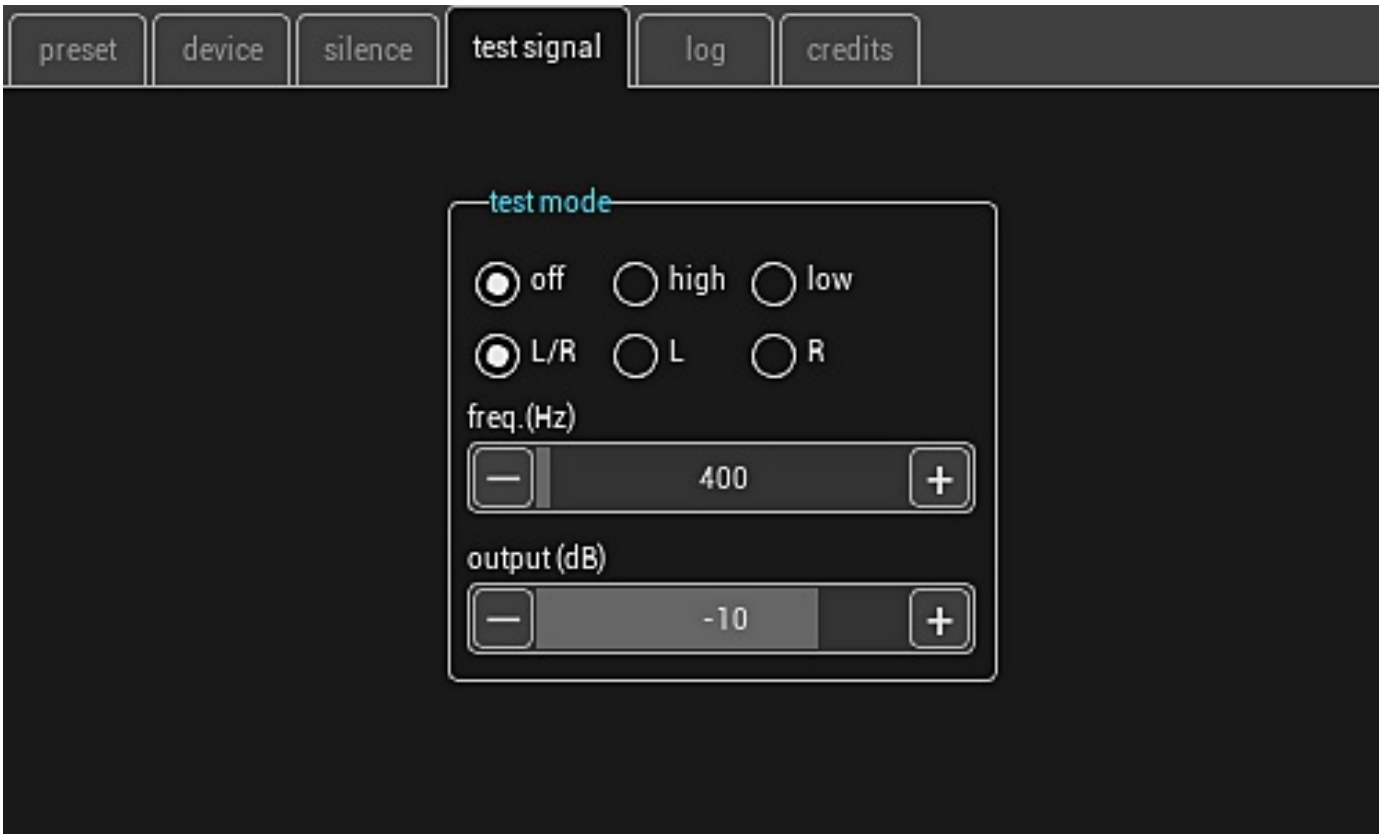
It is important to know when the download protocol is HTTP and username and password is required in the download address you could place ?u for the user and ?p for the password location in the address.
(Example: http://www.website.ext/path/?u_?p_file. Underscores is just for separation purpose. The supplier should give you the correct text string.)

Email

Should an error occur in the download process you can setup an email message to warn the person reponsible for it.
Check the the box and fill-in the email (addresses), Subject and possible message. Included below your own text will be an error message from the system.
If you want to use and test don't forget to setup the server in the common section.



Test signal



For setting up AM or FM processsing you can use it to setup a signal.
The high mode refers to a value of 75kHz devation for FM and 100% AM modulation.
After that you caan 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

Audio library setup

If you want to use the player you'll need to set up an audio library to use.
This is the structure you'll need:
Somewhere in your home folder on your harddrive is the main folder:

Audio/

With sub folders

Music/

And these are some formats you name, where you put in your audio files.

old/

new/

classics/

Jinlges/

Create your jingle formats here.

Messages/

Others/

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

The next step is to activate the player by checking activte and let the player know where the main folder is.
You do this in the setup of the player.



Creating a format

The next step is setting up a format. The left list is showing the formats you've created. The right is the format list.
Drag and drop a format item from left to right and built up a format. Above the left list you can select between music and jingles The option check double artist scans the playlist for multiple entrees of an artist. It is important to know that the name needs to be identical. The name check of the artist is case- sensitive.
To move an item up or down click and drag it to the correct position.
The format list is scanned every few minutes so changes can take some time.

