

# Qmod 19.1 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 and input- output device.  
Start Qmod and go to Config. Select the device tab

When Qmod starts the first time the default samplerate is 44100.

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.

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.

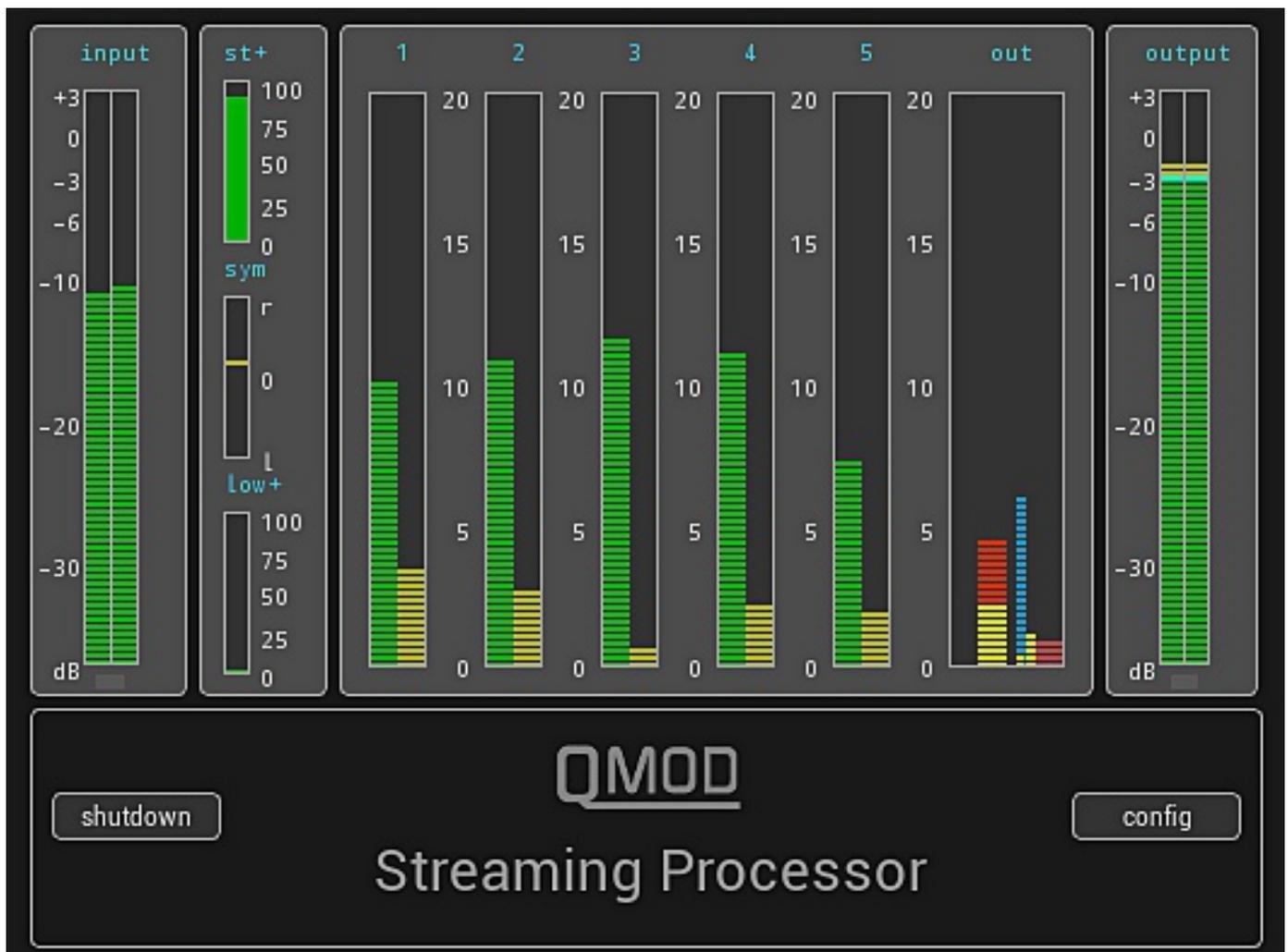
Qmod is ready to go.

## 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.

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.

## Main Screen



The left part of the main screen 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 inter-sample 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: expander, end limiter, peak control in yellow and the red indicates the composite clipping, dark blue is the amount of stereo reduction when in fm or am stereo, depending on the output selection.

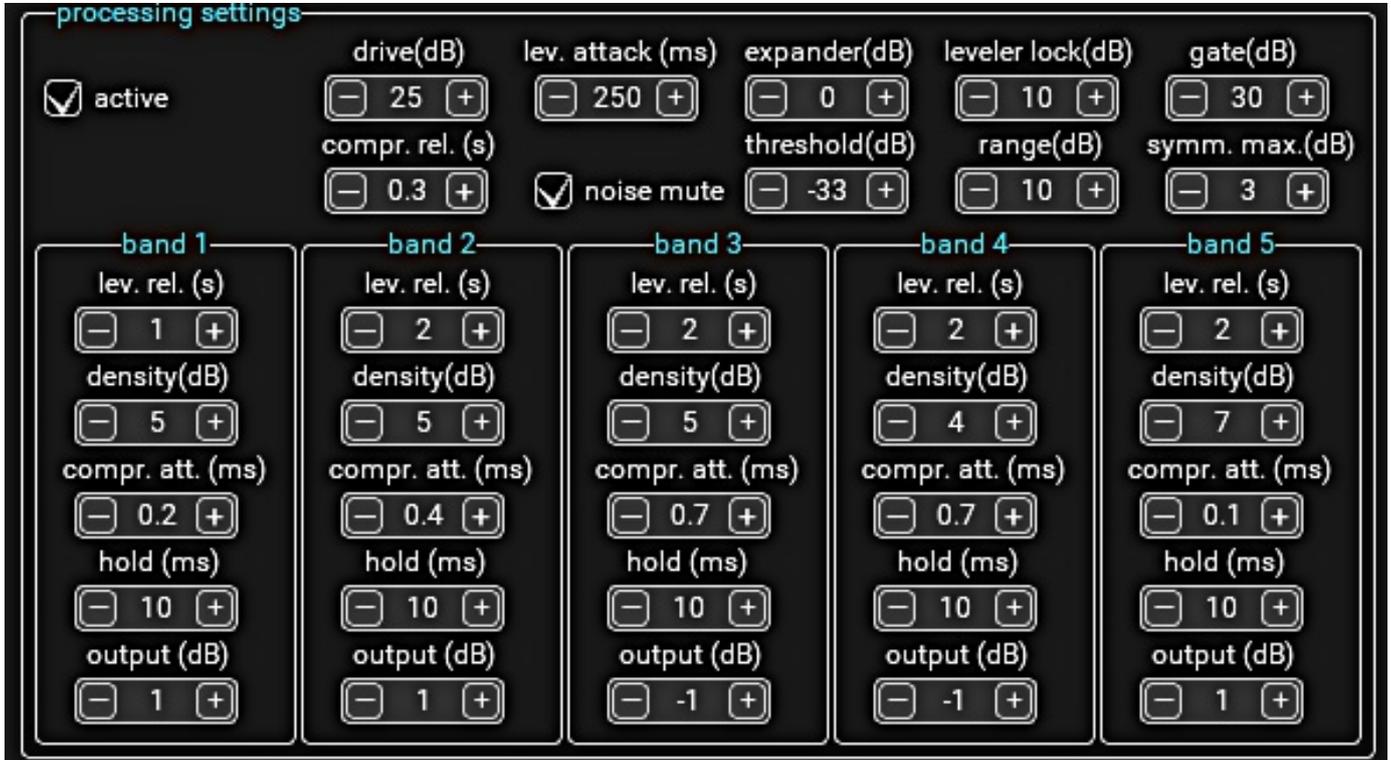
The most right is the output. It's not the output of the audiocard because that one has most likely a separate audio control.

## Config Processing

The processing screen is divided in two sections.

1. Multiband settings
2. Output settings

### 1. Processing Settings



The checkbox **active** disables the processing section. This option is never being saved.

### The Common Controls

The button **drive** controls the amount signal that drives the five band leveler. The leveling is indicated on the main screen by the green display.

The button **density** controls the amount of compression of the compressors that follow the levelers. The compression is indicated by the yellow display.

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

The button **expander** controls the amount of expanding the signal after it's processing by the compressors, to reduce the effect of gain riding somewhat.

The **leveler lock** button locks the independancy of the five levelers. 10 means the maximum difference between two levelers is 10dB, 0 means 0dB.

The **gate** button 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.

The **symm. max.** ctrl lets you set the max symmetry correction between left and right.

The 5 compressors have one release time in common to be set by the **compr.rel.** control

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

## The 5-band Controls

As the audio signal is divided in five sections.

Each section represents the leveler, compressor and output controls.

The **lev. rel.** sets the release time measured with drop of 5dB.

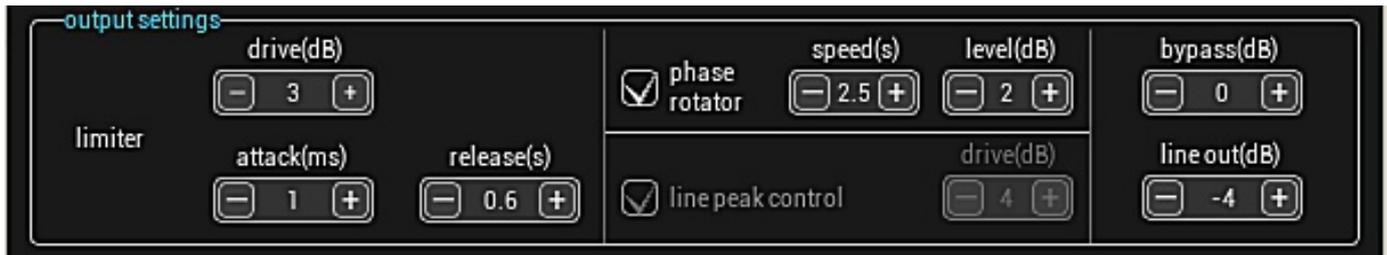
The density sets the amount of compression for that specific band.

The **comp. att.** sets the attack time.

The **hold** delays the start of the release.

The **output** can be set from +10 to -10db

## 3. Output



The **output limiter** button controls end stage limiter that comes after the mixing of five output signals. You can set the drive, attack and release of the limiter.

If the clipper box is checked the clipper is active. When the fm panel is active the preemphasis clipper is included in the process.

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 on the main screen in the symmetry display. The green bar lights up yellow.

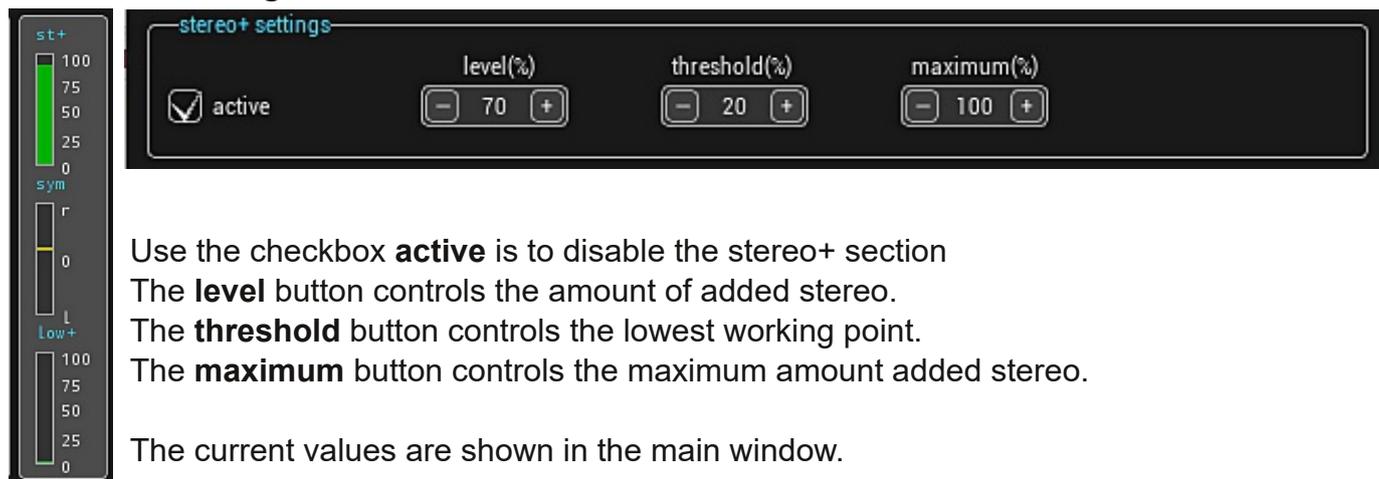
When the **peak control** is active you can set the control level and the buffer size of the control, the value is in dB. For optimum result set the buffersize at least at a value of 15ms. The peak control is always active in FM or AM mode and the control setting is on the FM and Am panel.

The **line out** button sets 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 unwanted clipping. If that's the case you'll see the clip light flash red on the main screen.

# Config Pre Processing

## 1. Stereo+ Settings



Use the checkbox **active** is to disable the stereo+ section  
The **level** button controls the amount of added stereo.  
The **threshold** button controls the lowest working point.  
The **maximum** button controls the maximum amount added stereo.  
The current values are shown in the main window.

## 2. Low+ Settings



The **active checkbox** activates the low+ algorithm.  
The **theshold button** controls the amount of added low.  
The **max button** sets the amount of low+ signal.

## 3. Treble+ Settings



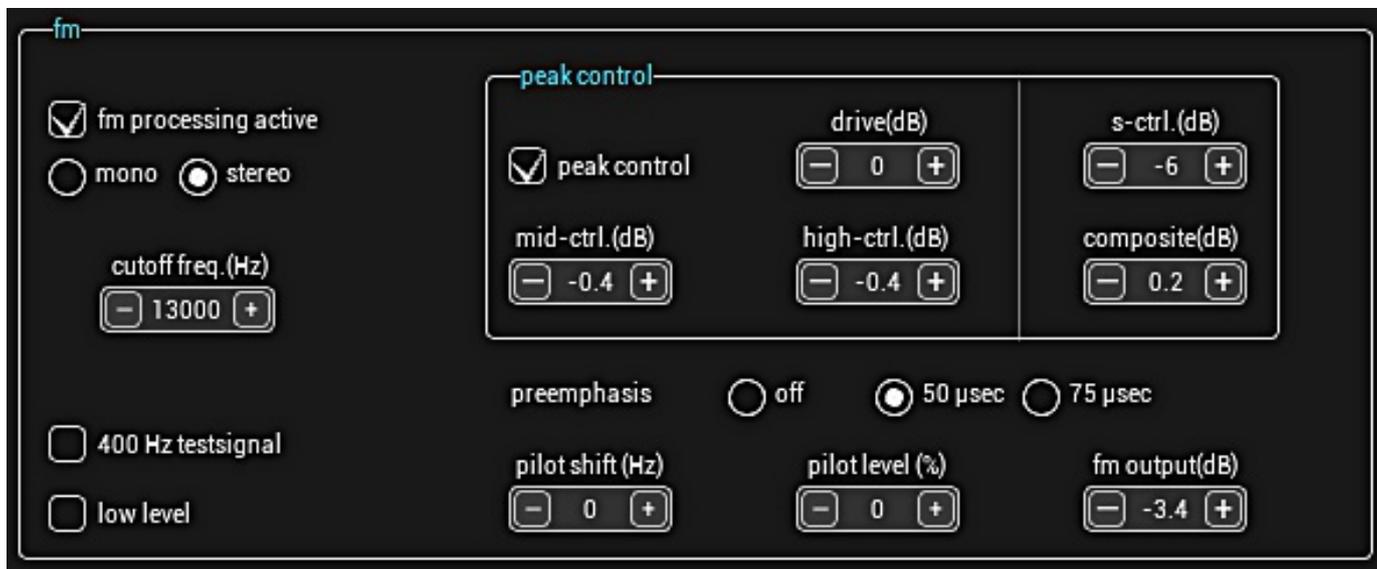
Use **active** to activate this function.  
When active you can set the **frequency** and **gain** to the appropriate setting.

## Config FM

The fm tab is divided in two sections:

1. FM
2. RDS

### 1. FM



The checkbox **fm active** activates the fm encoder. The default setting is unchecked. The two buttons **mono** and **stereo** give you a choice which option you want.

The **cutoff frequency** sets the baseband 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 and RDS signal. The values are in dB.

To run some tests you can enable the built-in **tone generator of 400Hz**. It can produce two levels: full and 23.5% modulation. Activate the desired checkbox.

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.

The total output of the MPX signal is set with the **fm output** button.

The **stereodepth limit** button controls the limiter on the stereo signal and thus reduces the s-part in the MPX signal.

### 3. RDS/RBDS

rds settings

rds active      ps text (max. 8)      pi code (0-9/A-F)      altern. freq.1      altern. freq.2

rbds      Hobby fm      8000      0.00      0.00

time sync      program type

pop music      ↕

accept

radio text (max. 64 characters only a-z/A-Z/0-9)

FM stereo 101.7 The best music, all day

**RDS/RBDS functions only if the audiocard supports a samplerate of 192000kHz.**

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.

## Config AM

The screenshot shows a configuration window titled "am settings" with a dark background. On the left, there are three checked checkboxes: "am processing active", "am mono", and "am preemphasis 75 μsec". In the center, there are two numeric spinners: "peak control" with a value of 2 and "am output" with a value of 0. On the right, there is a "range" section with four radio buttons: "4.5 kHz" (selected), "5 kHz", "9 kHz", and "10 kHz".

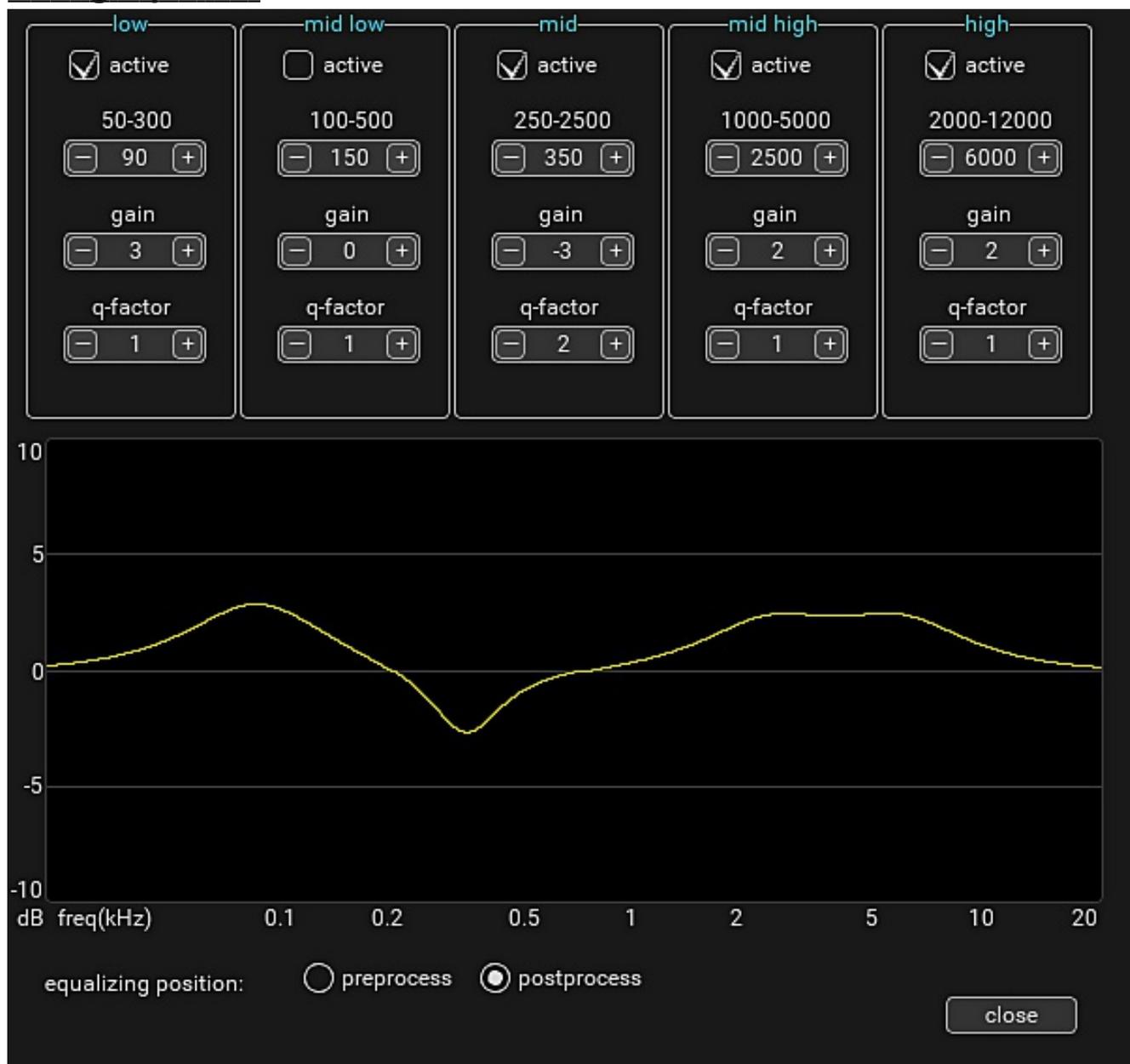
The checkbox **am processing active** enables/disables the AM processing mode. You can choose if the processing is in mono or stereo by selecting **am mono** checkbox.

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 5 kHz.

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

## Config Equalizer

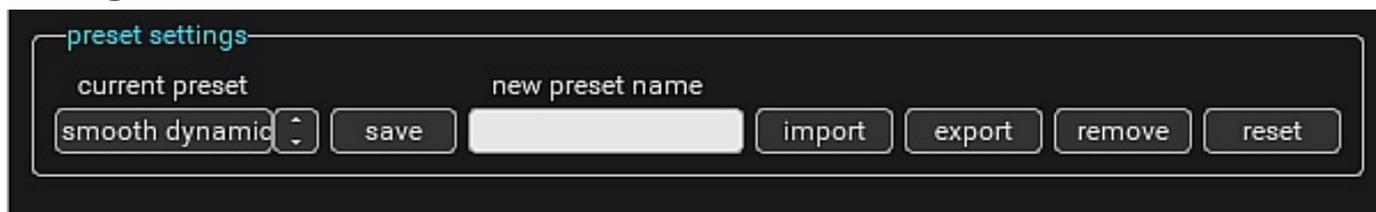


Each equalizer section has the possibility to set the **frequency**, **gain** and **q-factor**. The frequencies partly overlap so you can combine like the shown setting.

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 all processing is done to correct some flatness in the signal.

## Config Presets



preset settings

current preset: smooth dynamic

new preset name: [input field]

Buttons: save, import, export, remove, reset

As of version 17.4 Omod has 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.

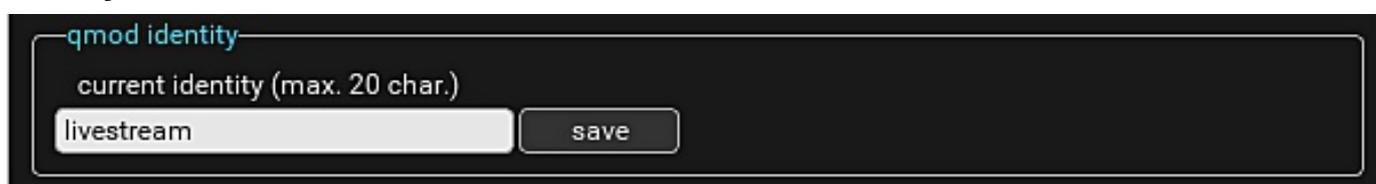
If you upgrade from a version older than 17.4, your old setting will be saved as **"custom"** preset.

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



qmod identity

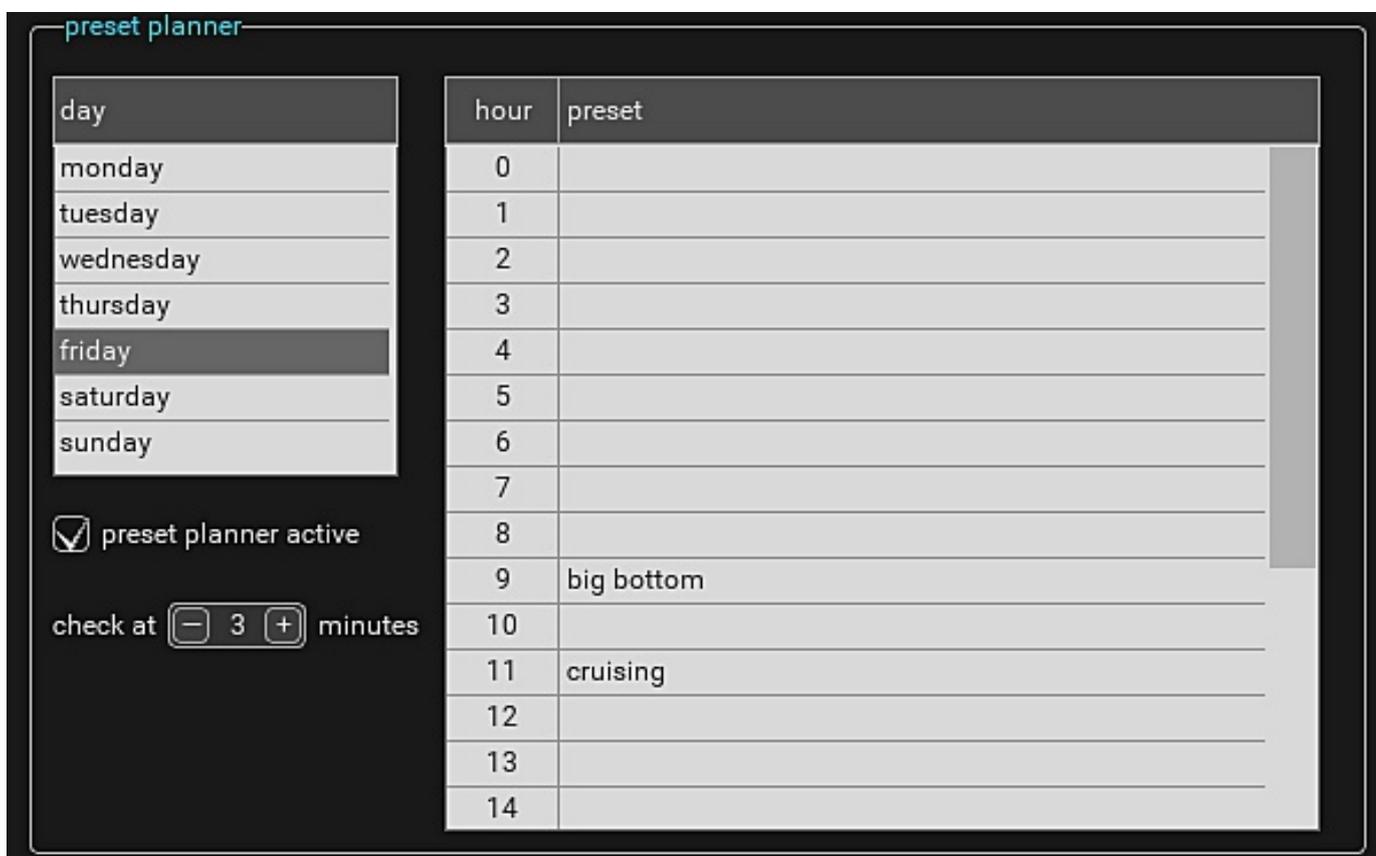
current identity (max. 20 char.): livestream

Button: save

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



preset planner

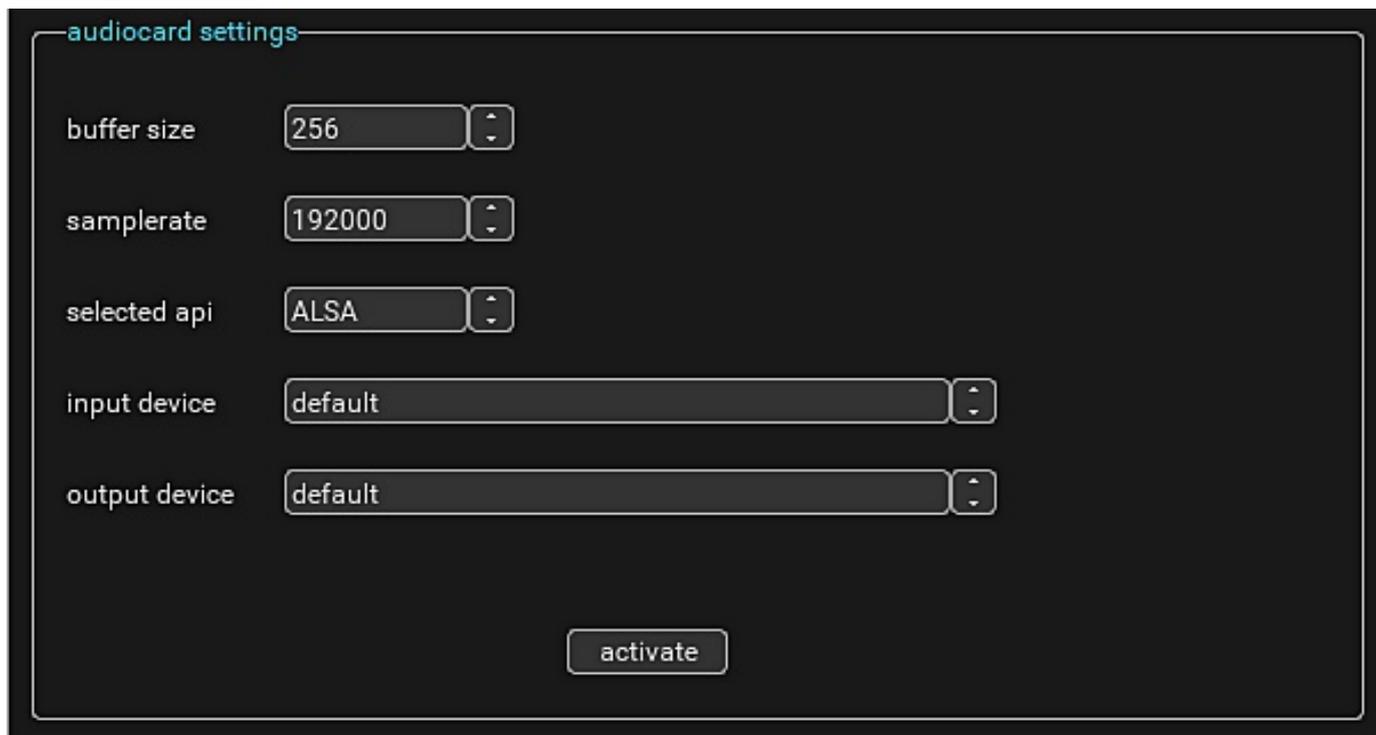
day	hour	preset
monday	0	
tuesday	1	
wednesday	2	
thursday	3	
friday	4	
saturday	5	
sunday	6	
	7	
	8	
	9	big bottom
	10	
	11	cruising
	12	
	13	
	14	

preset planner active

check at: [ - 3 + ] minutes

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



audiocard settings

buffer size 256

samplerate 192000

selected api ALSA

input device default

output device default

activate

The default samplerate is 44100. 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**.

If you've selected the ASIO api make sure the buffer size has the same value as that of the ASIO setting.

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

### \*\*\* Note \*\*\*

When you want to connect to a virtual cable use the MME api, otherwise you'll get an initialization error.

## Update check



update check settings

active

When you activate the update check, Qmod will check at startup and once every 24 hours for the availability of an update. When successful it will show on the main window in the lower section above the logo.