LASTAR V1.9

Logiciel **A**utomatisé **S**imple de **T**raitement **A**udio **R**apide

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Content

1. What is it ?

LASTAR is a batch (non-interactive) audio processor allowing, in a few clics, loudness adjustement and file splitting of a batch of audio recordings.

At the opposite of usual available software, loudness normalisation is done on signal power, which leads to a louder and more homogeneous result than a usual "peak" normalisation, in particular on live recordings.

Its purposes are :

- automatically master recordings,
- to split, equalize and normalise digitized analog tapes or vinyls,
- to split, equalize and normalise live recordings from microphones (ex rehearsals recorded with ZOOM recorder),
- fast and homogeneous normalization and equalization of a group of files (for a compilation for example),
- automatic gain control and automatic equalization for processing for listening in a noisy environment (car...)
- and so on...

The aim of this software is to be very fast and easy to use : the most efficient computing techniques have been implemented (without worsening audio quality), and there are very few parameters to set (most of them are automatically adjusted by analysing the file).

For interactive audio edition, instead please use an audio editor like Audacity (http://audacity.sourceforge.net).

Notice that, to optimize final result for different kinds of music, the software uses a dedicated 3 bands amplification algorithm.

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3. Requirements

Operating system : Windows (TM) 10 32 bits or above. Not tested on older (f.eg. XP or W7) versions.

RAM : 1 Go recommanded. Works on 512 Mo though.

Processor : At least SSE3 compatible CPU (AthlonTM 64, PentiumTM 4), dual core or more recommended.

4. What's new

4.1. *In version 1.9*

- Automatic equalizer profiles can now be added, edited and learned from sound files, see §13.
- New analysis window which displays graphically sound relative spectrum and temporal RMS level, see §12.
- Command line (batch) processing, see §14.
- Default ID tags (including cover image) can now be different for each preset
- Quality of cover image can be adjusted
- Analysis pass are redone only if necessary
- Merge all input files to process them as it was only one big file
- Minor GUI improvements

4.2. *In version 1.8*

- Artwork is now read from input file (if in mp3 format) and stored in the processed file (if mp3 format)
- It is now possible to select a jpeg artwork file in the default tags (for mp3 output only).
- Noise estimation can be disabled.
- Minor GUI improvements.
- Italian translation.

4.3. *In version 1.7*

About automatic frequency equalization

Often, live recordings with basic means have a rather atypical frequency response and can make listening quite difficult. The files of a compilation of the same style can also have large disparities that make listening unpleasant (very high for some, too low for others ...), making gain control and RMS normalization insufficiently effective.

LASTAR now offers a new automated frequency adjustment function that allows you to get closer to typical curves of different styles of music.

What else ?

Various fixes and improvements to the gain control algorithm have been made, plus some minor bugfixes. Automated version update check.

4.4. *In version 1.6*

- New flac and ogg vorbis output formats
- Loudness computation improved thanks to a weighting filter
- Improved I/O error indication

4.5. In version 1.5

- Trimming function to cut silences at the beginning and the end of the tracks processed (see chapter 10.2)
- A progress bar when processing a batch of files
- Several bugs in the output file management options

4.6. In version 1.4

About gain control

When you listen to complex or dynamic music in a noisy environment (bus, underground, car...) with your smartphone or your mp3 player, you only got those choices :

- choose a comfortable loudness and miss all the soft parts of the songs,
- ear everything but smash your ears and bother your neighbours,
- or wreck the volume knob of your device by changing the level every 20 seconds (and being, in between, in the 1st or second case).

That's why I decided to develop an automated gain controller that would change the dynamics of a sound file to make the softer parts louder (as if you were changing the volume by hand), adjusting the output each time it detects a big overall change in the loudness. It was so-call "compression" in the previous releases, and is called now, with some improvements in the algorithm, the **Gain Control** feature. It tries to smartly limit the volume changes to the strict necessary, in order to avoid unnatural effect. Just push the **Gain Control** cursor to the left and listen.

Directory tree processing

Now with this feature, the automated RMS amplifier (which acts more or less like an mp3gain) and the clipping limiter, i've got the perfect tool to process all my music collection.

As many people, my music is stored in a directory tree, I so needed a tool that would allow me to parse and process all the files there, and then save them in an other location (so I can keep the originals as-is), with the same directory structure, with a feature to easily update the output when I buy new albums.

That's the main purpose of this new 1.4.0 release.

How to use it ?

Just follow those steps :

- drop the root directory of your collection in the tool from Windows explorer,
- choose a destination folder,
- select your settings between **Mobile pop/rock** or **Mobile Classical/jazz** collection, and launch processing (note : you can use the same settings for both, but the result will be better if you process separately).

then you'll have in the destination all the files processed with the same names and the same sub-folder organisation. An option will allow you to update the destination with the new source files without reencoding everything, **in order to keep your processed collection up-to-date**.

Notice that this function doesn't copy others files in the folders (covers.jpg, playlists), so you'll have to manage it with Windows explorer if necessary (launch a global copy and ask to ignore existing files).

See chapter 7 page 12.

Is there a limit in the number of files I can process at once ?

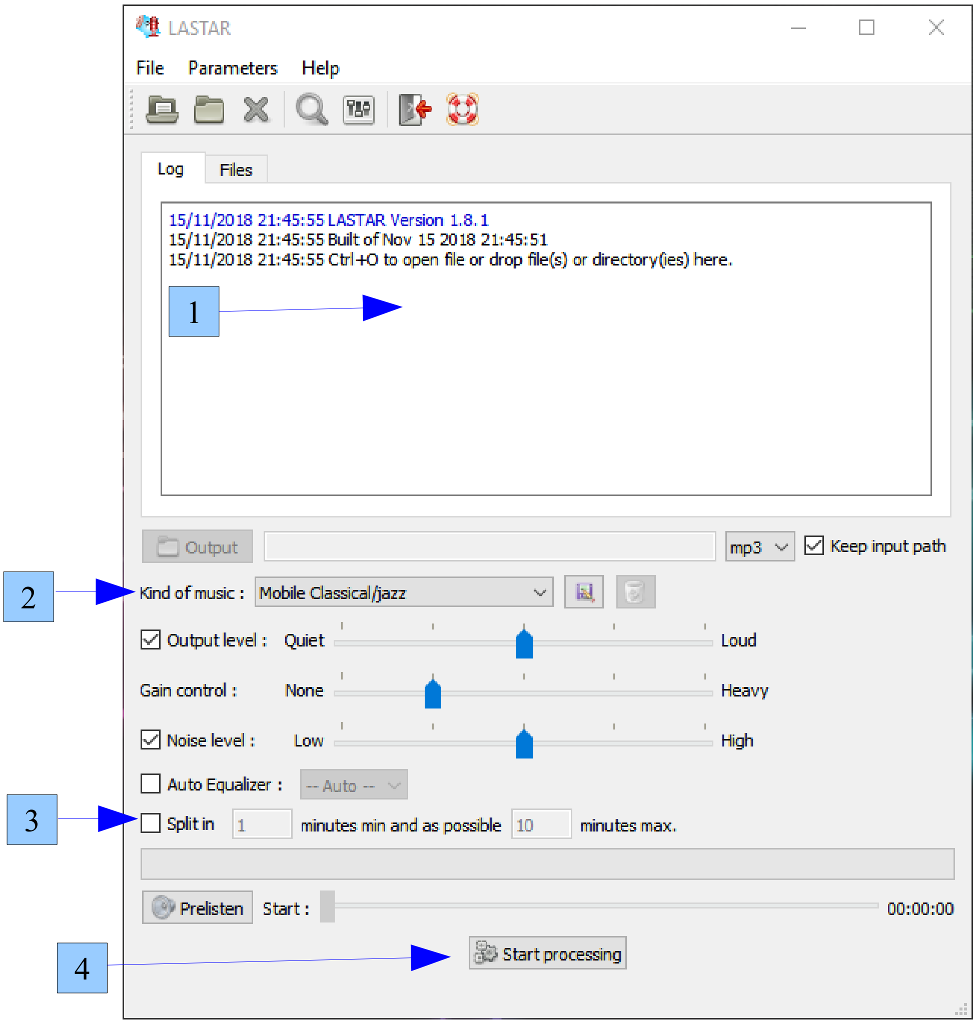
As far as I know, no. I processed in my entire collection (several thousands of files) in a single batch, and after 1 or 2 days of un-interrupted processing, the job was done. And know I can enjoy this "mobile" version of my favourite music everywhere !

What else ?

A new **Close all** menu to empy the file list, and some minor bug fixes.

5. Quick start

The simplest way to use LASTAR just needs 4 clics :



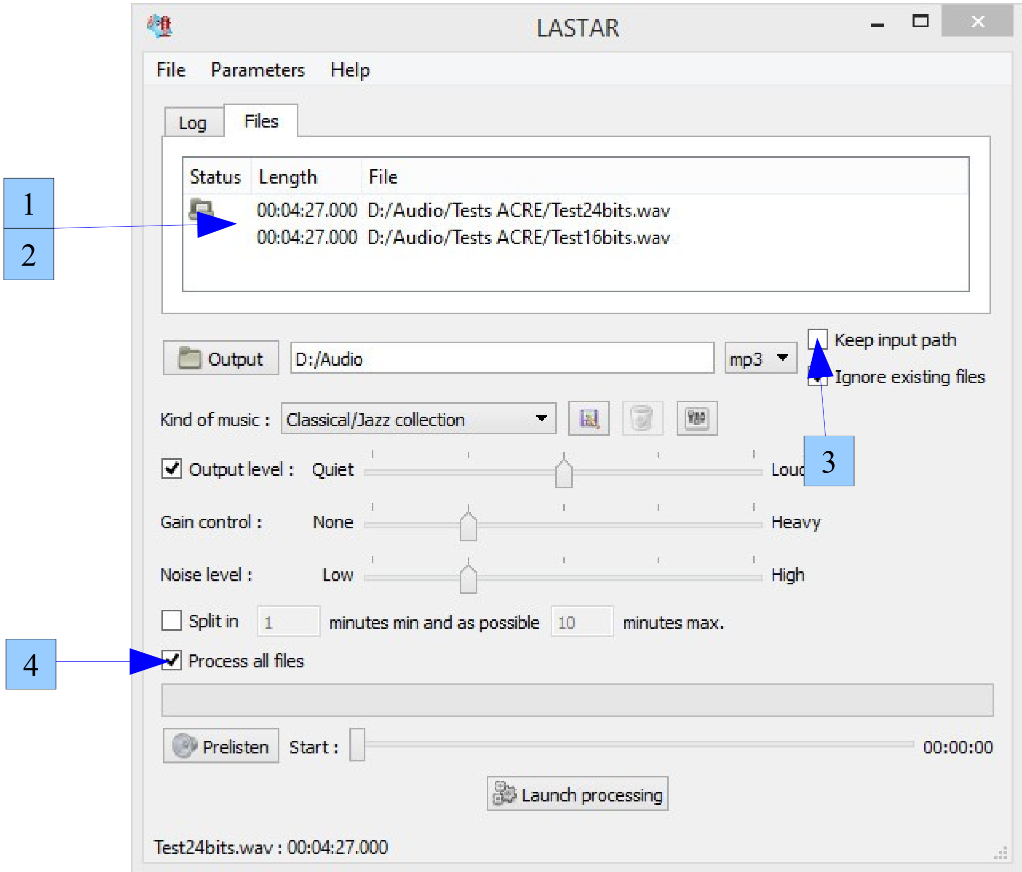
1. Drag and drop one or several files from a Windows explorer. You can also type **Ctrl+O** to open several files at once. If you drop more files, they will be added to the list of files to process.
2. Choose the nearest kind of music
3. Check this to split the file(s) if it contains several music pieces
4. Start processing. LASTAR will first proceed with a first analysis pass, and then will produce the output file. Notice that if you launch processing again after it has finished (after changing some parameters for example), it won't do the analysis pass again.

LASTAR will automatically add a suffix to files and will store split files in folders with the same name as input.

Note : It is possible to open files directly from windows explorer : just right click on the files, select “**Open with...**” and choose LASTAR.exe from its installation directory.

6. Files management

When one or several files are opened, the “*Files*” tab is then displayed :



1. Drop some more files in this frame to add them to the list

The first column in the list show which file is currently opened or processed (it is always the first one !)

2. Select a file and press « **Del** » key to remove it from the list. If you delete the first file, it is closed and the next one is opened (if any).

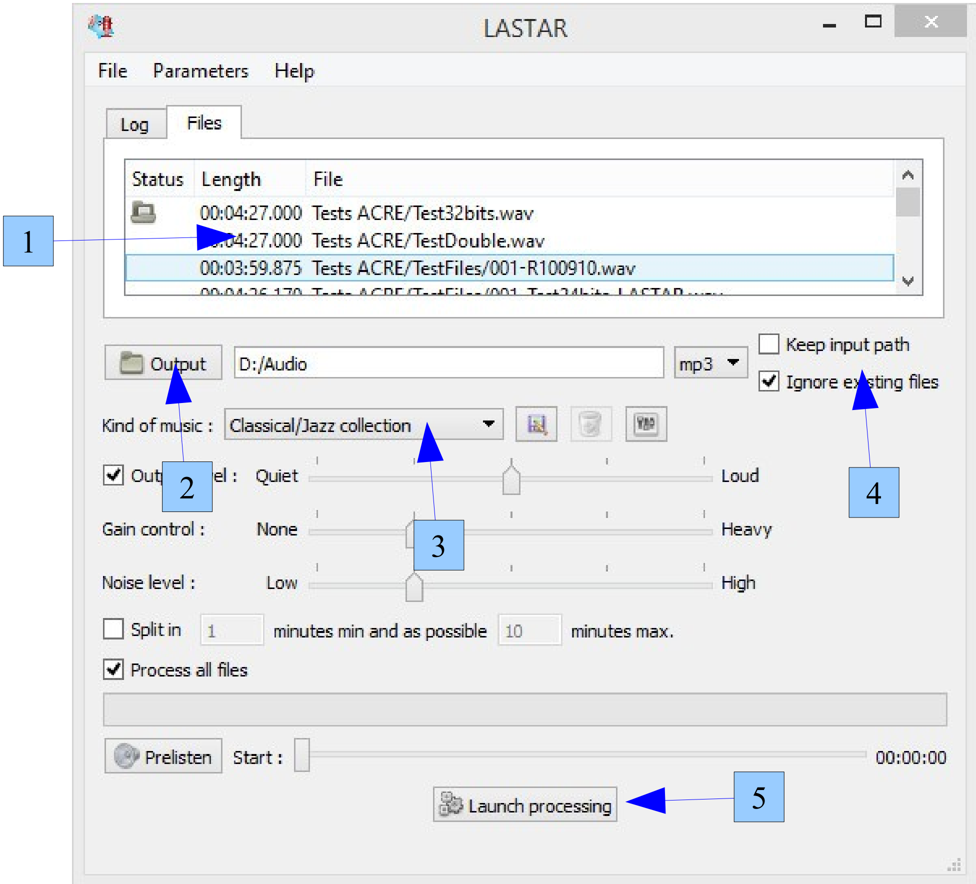
Alternatively, you can use the menu **File>Close all** to empty the list.

3. Adjust output format and directory with these controls. Uncheck “*keep input path*” to be able to change output file name or directory.

4. Uncheck this button if you want processing to stop after the first file. When this button is check, a second option is displayed to allow to process together all input files as if it was a big file (<u>does work only if all files have the same sample rate</u>).

Note : In the case there is only one file to process, the last processed file remains opened. This allows to restart processing with new parameters without having to re-analyse the file content. When **Process all files** is unchecked, the same occurs with the first file.

7. Processing a directory tree



This feature (new from the 1.4) allows you to process a music collection organized in a folder tree.

Follow those steps :

1. drop the root directory of a collection to process (or several sub-directories) from the window explorer to the file list,
2. choose a destination where to copy the new files,
3. choose the main music style,
4. check ignore existing files just to update the destination with the new files,
5. and then launch processing.

You'll notice that in the file list, only the relative path to the root directory is displayed. This shows you exactly what will be the path of the output file in the destination.

In the example above, the first three files will be stored in D:\Audio\Test ACRE\Test32bits.mp3, D:\Audio\Test ACRE\TestDouble.mp3 and D:\Audio\Test ACRE\TestFiles\01-R100910.mp3 respectively.

All non-existing directories in the destination will be created in order to follow the input structure.

8. Basic parameters

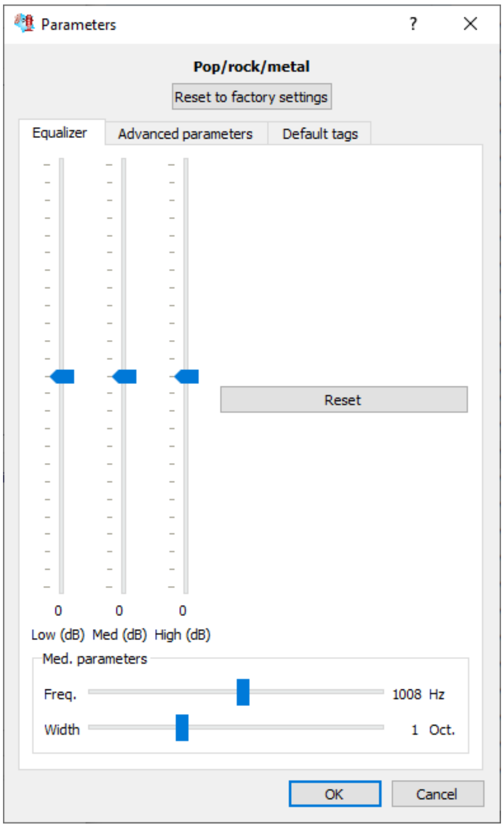
8.1. Output level

For each kind of music, a default relevant output level has been pre-setted. You can adjust locally the output level around the default value with this slider. If you uncheck the *Output level*, there is no amplification, no gain control, no L/R channel balancing, nor equalization in processing. This can be used if you just want to split the file without modifying its content.

Note : while pre-listening is activated, you can adjust this parameter in real-time and directly hear the result

8.2. Classic equalizer

When this button is pressed, you have access to a 3 bands semi-parametric equalizer :

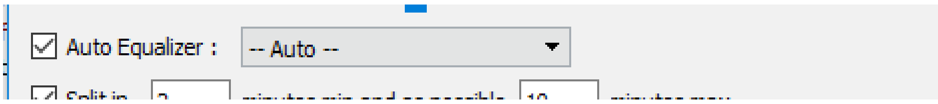


You can adjust bass, medium and treble levels with the 3 vertical sliders. The button *Reset* resets the corrections to 0. The central frequency and width of the medium filter can be adjusted with the *Med. Parameters*.

The equalization is processed before amplification so that no clipping can occur.

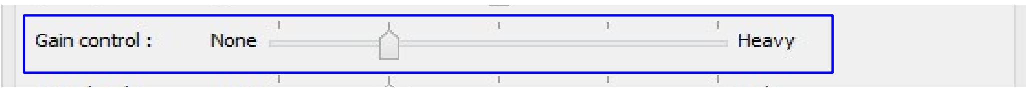
Note : while pre-listening is activated, you can adjust this parameter in real-time and directly hear the result.

8.3. Automatic equalizer



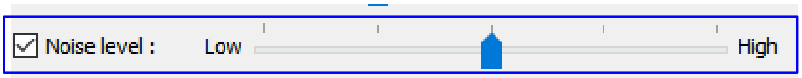
This button is used to apply an automatic frequency equalization on 10 bands in order to bring the response of the track similar to a predefined curve. In - *Auto* - mode, the nearest curve is automatically selected. Otherwise, it is possible to choose a predefined curve in the list. You can see in the *Analysis* tab how the track is corrected.

8.4. Gain control



With some gain control, the softer parts of the music will be made louder, as if you were turning the volume knob of your player. This is specially useful to listen music in a mobile device outdoor, in the bus or the subway... See explanations in chapter 4.6.

8.5. Noise level



The noise level has several effects :

- when splitting is on, it affects the way the background noise is estimated to separate music pieces in the track
- the gain control has no effect on the signal under the threshold driven by this slider, in order not to over-amplify pure noise,
- the *Trim* function (see advanced parameters) relies on the noise level to adjust the beginning and the end of the track.

When the option is checked, the noise threshold used is estimated by taking low level segments from the input with a sensibility given by the slider position. When the option is not checked, the threshold is not estimated, and only computed using the *Max SNR ratio* in advanced parameters and the maximum level of the track.

Note : Noise threshold setting is relative to the track being split. If the track is too long for the max length entered, the absolute threshold used is re-estimated. This point is to consider if some low parts are too drastically cut in splitting : this might be that the max track length setting is too small (and not that the noise threshold is too high !).

Note 2 : This setting has no effect on the Noise gate processing (when enabled). The noise gate has its own threshold.

8.6. Tags

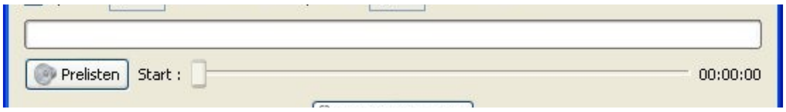
When you save your file to mp3, if there are no tags in input file (because it's wav or tag-less mp3), a window appears to fill in the tags when processing starts :

The *Title* is always deduced from the filename.

You can adjust picture JPEG quality using the slider.

Other tags filled in are stored with the current settings for later use (each settings has its own tags). You can change them by calling the *Parameters* menu and selecting *Default tags* in the window.

9. Prelistening function



The *pre listening* function allows to hear exactly what will be saved on file. To activate it, just define the starting point with the *Start* slider and press *Prelisten*.

If the file has not been analysed yet, it is analysed before playing (so playing starts only when the progress bar is at 50% or 66% according the number of pass necessary). If analysis is done, playing starts immediately.

- Press the *Stop* button to cancel pre-listening
- To adjust the speaker volume or the speaker device, use the Windows controls. Remember that the *Output* slider only adjusts the level of the file content, not the output speaker volume !
- While listening, you can adjust the output level and the equalizer settings in real time
- If you change some other parameters, playing restarts from the defined starting point
- You can drag the *Start* slider while listening, playing will then restart from the new point as soon as you release the slider.

10. Advanced parameters

All those parameters can be accessed from the window opened by the *Parameters* menu.

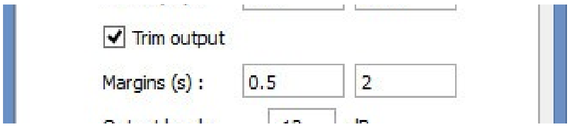
10.1. Multiband amplifier

The multi-band processing amplifies separately the bass, medium and treble of the input signal, and then recombines them. It is very useful when the music contains instruments with very distinct frequency responses (for example music with big bass and bass drums, guitars and bright cymbals). In this case, processing 3 bands is more transparent because limiting (for example) the bass will not affect the other bands, and thus it will prevent audible “pumping effect”. But this might lead to some distortions on single acoustic instruments, so it is sometimes useful to disable it. The factory preset “*Pop/rock/metal*” uses by default this feature, the others don't.

The two fields below the button allow to adjust the frequencies where bands are separated.

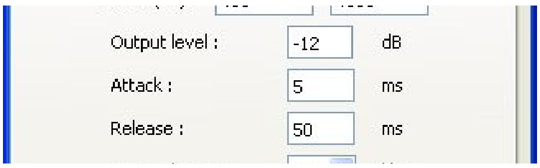
Note that if the two frequencies separating the bands are equal, the processing will be done only on 2 bands (the band below the frequency selected and the one above).

10.2. Trim output



The trimming cuts the start and end silences of processed files. It has no effect if the splitting feature is active. You just have to check the parameter and select the amount of silence that is kept at the beginning and the end to activate it. This function is active by default in factory settings.

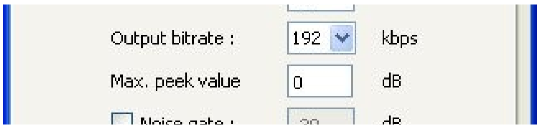
10.3. Amplifier/limiter



Theses parameters control the behaviour of the amplifier/limiter (or the 3 amplifiers/limiters in the case of a multiband processing).

- *Output level* is the target RMS output level used when the slider *Output* is at the middle position. The *Output* slider modifies this value from 0.5 dB per tick from the center.
- Attack is the time is the time the limiter starts to decrease the output level before it detects a clipping (as it is a lookahead limiter).
- Release time is the time the limiter takes to come back to a 0 attenuation if there's no more clipping. If release parameter is set to 0, release time is automatically adjusted from signal content (since version 1.9.2).

10.4. Output settings



- Output bitrate is the bitrate of files saved in mp3 format
- Max. peek value is the Output normalisation value (i.e. the maximum peek level allowed in output file)

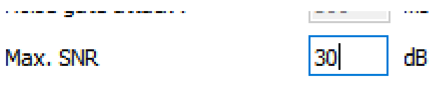
10.5. Weighting filter

The weighting filter is applied before RMS level estimation. It’s very useful when one wants to improve a track audibility (typically when applying automatic gain control, like in mobile factory presets). It should be deactivated when mastering a single track with automated equalizer because it could lead to over-amplification.

10.6. Noise gate

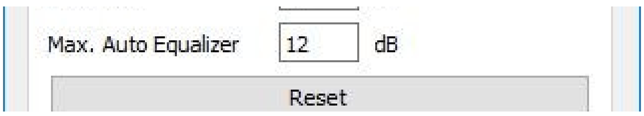
To activate the *Noise gate* on output, just check the button *Noise gate*. When active, all signals below the threshold will be considered as noise and cut to 0. The *attack* time is the time the noise gate takes to cut completely the output to 0 when it detects the level is under the threshold.

10.7. Max SNR



This parameter sets the maximum amplitude level between the loudest segments and the quietest. It thresholds the noise level estimation when activated, and is taken directly as the noise level when estimation is not activated.

10.8. Automatic equalization



It is possible to limit the amplitude of the correction applied for each band by entering a maximum amplitude value in this field.

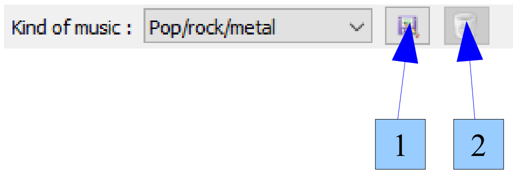
11. Parameters preset management

By default, LASTAR has 5 factory presets in the "**Kind of music**" menu :

- *Pop/rock/metal* : best suited for amplified modern music and percussive instruments. Uses multi-band amplification combined with a high output level and a very aggressive, reactive limiter behaviour
- *Mobile pop/rock/metal* : it is basically the same preset, but with some gain control
- *Classical/jazz* : best suited for melodic acoustic instruments ; non multiband processing here, and a lower output level with a smooth, progressive limiter.
- *Mobile classical/jazz :* The same with some gain control to make a mobile-usable collection
- *Voice/speech* : similar to *Classical/jazz*, but with a more reactive limiter.

Every change in the parameters you can make (even the output file type) is stored independently with each factory preset, and saved when you quit the application.

You can add/remove your own parameters presets using respectively the buttons noted 1 and 2 below :



11.1. *Adding a new preset*

- Press the button 1
- Enter the name
- Validate

The current parameters are stored to the new preset. All the changes you make from now will be stored to the new selected preset.

11.2. *Delete preset*

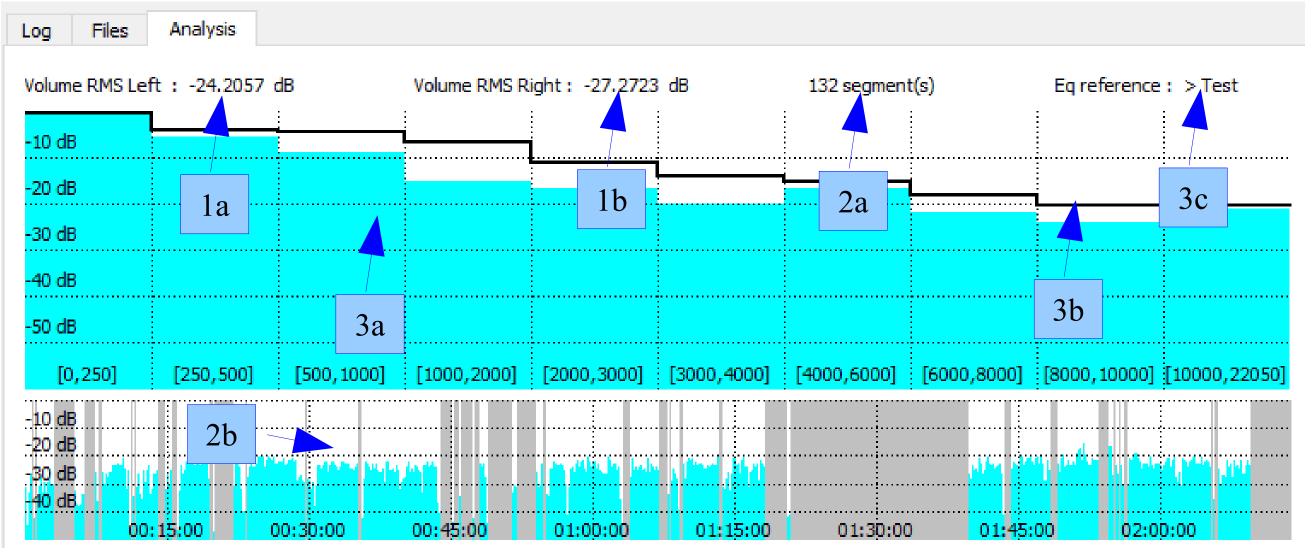
If you press the button 2, current user preset will be deleted. Note : Factory presets cannot be deleted.

11.3. *Reset a factory preset*

In the *Advanced parameters* window, the *Reset* button allows you to reset the factory preset to what it was initially defined.

12. Analysis tab

When processing is complete, or if you only ask for an analysis by pressing the  tool button, you can see the results in the *Analysis* tab :



– **1a** and **1b** give the RMS level of each channel of the input file,

– **2a** gives the number of segments (i.e. pieces of the track that are not considered as noise), which are displayed in the temporal graph **2b** in blue. The grey blocks in **2b** are noise. Notice that if you change the noise estimation parameters, see 8.5, **2a** and **2b** will refresh in real-time.

– **3a** displays, in blue, the input file frequency response. The black line **3b** is the equalizer reference profile, whose name is given in **3c**. Notice that if you choose – *Auto* – as eq reference profile, see 8.3, **3b** and **3c** will show the actual profile that will be applied during processing, chosen as being the best fit to the input file response. Those parameters are also refreshed when eq profile changes.

13. Automatic equalizer profiles

13.1. About automatic equalizer profiles

Since LASTAR 1.9, it is now possible to edit, add and delete automatic equalizer user profiles.

Such a profile represents the desired frequency response of the track to be processed, divided in 10 bands.

Unlike a classical graphic equalizer, the 10 values are not the gains to apply to each band, but the actual level of each band relative to the loudest one (which is always 0, as a consequence).

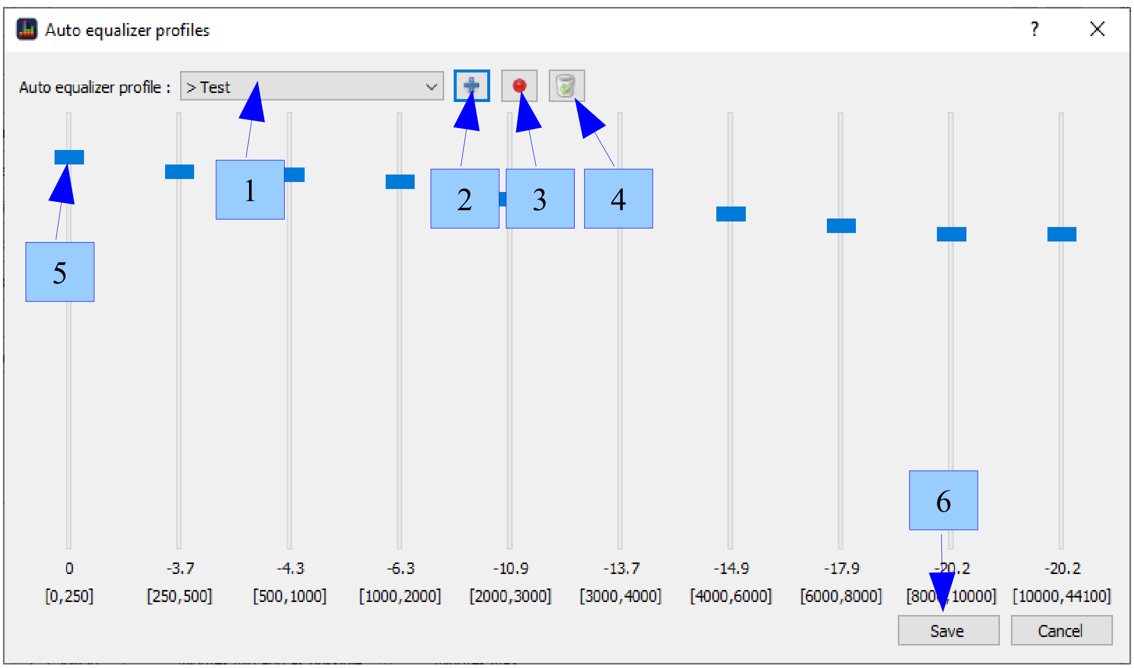
Notes :

- The factory profiles cannot be edited.
- The user profiles always appear in the list with a '>' before their name.
- The user profiles are stored in C:\Users\<userName>\AppData\Roaming\Arthelion\LASTAR\AutoEqCatalog.json. If you want to remove all profiles, stop LASTAR, remove this file, and restart.

13.2. Equalizer user profiles management

To open the equalizer user profiles management, clic on the button in the toolbar.

Then the profiles management window appears :

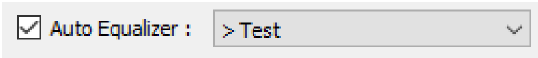


1. Use this menu to select the user profile to edit.
2. Creates a new user profile.
3. Learn user profile from the current input file list.
4. Delete current user profiles.
5. Each slider allows to adjust the band relative level. Max is always 0 as explained above, so if you

move a slider above 0, all sliders will go down.

6. Don’t forget to clic on the *Save* button to keep changes. If you click *Cancel* or close the window, all changes will be lost after a confirmation.

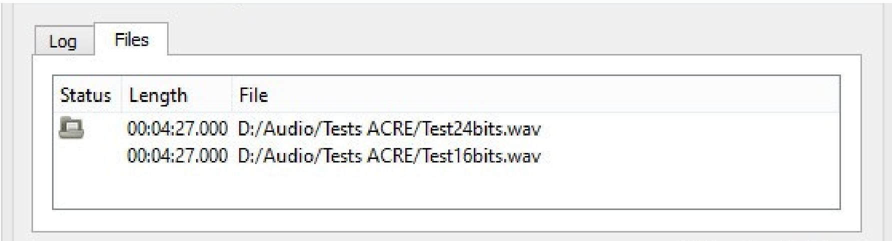
If you create new user profiles, they will appear in the main panel with a ‘>’ before their name :



13.3. Learning an equalizer profile from files

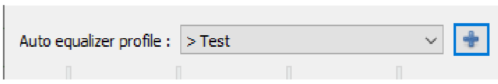
If you want to compute a new profile from existing audio files, in order for example to make your new recording sound alike, follow the following steps :

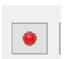
– Put in the input file list the files you want to use to learn your new profile (and only those files). Remember that if you drop a folder in this list, all files under the folder and sub-folder will be loaded recursively.



– Open the Equalizer user profile management

– Select a profile or create a new one :



– Clic on the Learn button 

– Once processing is complete, the sliders will move to reflect input files mean frequency response.

– Clic on *Save* to keep your new profile.

– Now you can empty the input file list (use the *“File > Close all”* menu) and process your recording with the new profile.

14. Command line use

LASTAR, from V1.9, can be launched in a cmd terminal or .bat file to allow batch processing without using GUI.

In this mode, all messages are written to terminal and there is no window displayed.

Basically, it works by applying to the input files specified on the command line an existing parameter preset as described in §11.

The arguments are given in the following table :

Parameter	*Argument value*	*Status*	*Description*
-p, --process	Parameter preset name	Mandatory	This arguments triggers the command line mode. You must specify a preset name after this argument. Add quotes around the preset name if it contains spaces.
-f, --format	Output format	Optional	Specifies the output format (mp3/wav/ogg/flac). By default, it's mp3 with bitrate as specified in preset.
-o, --output	Output file path	Optional	Output file path, including file name or just a directory. If ommited, the input file name is taken.
--artist	Artist name	Optional	Overrides preset default artist ID Tag. Surround with quotes if it contains spaces.
--album	Album name	Optional	Overrides preset default album ID Tag. Surround with quotes if it contains spaces.
--title	Track Title	Optional	Overrides preset default title ID Tag. Surround with quotes if it contains spaces.
--year	Track year	Optional	Overrides preset default year ID Tag.
--track	Track number	Optional	Overrides preset default track number ID Tag
--genre	Track genre	Optional	Overrides preset default genre ID Tag. Surround with quotes if it contains spaces.
--cover	JPEG Cover file	Optional	Overrides preset default cover file. Surround with quotes if it contains spaces.
<file1> <file2>		Mandatory	The command line must end with a list of files to process. Surround each path with quotes if it contains spaces.

Here are some examples :

“C:\Program Files (x86)\LASTAR\LASTAR.exe” -p “Pop/rock/metal” --artist “Dr Gang” --cover “D:/Images/Gang.jpg” “D:/Music/Test.mp3”

Processes “D:/Music/Test.mp3” with Pop/rock/metal” preset, overrides the artist and cover ID tag, and save it in mp3 format as “D:/Music/Test_LASTAR.mp3”.

“C:\Program Files (x86)\LASTAR\LASTAR.exe” -p “Mobile Pop/rock/metal” -f wav -o “D:/Music/output” “D:/Music/Test1.mp3” “D:/Music/Test2.mp3”

Processes “D:/Music/Test1.mp3” and “D:/Music/Test2.mp3” with “Mobile Pop/rock/metal” preset, save them in wav in “D:/Music/output”.

"C:\Program Files (x86)\LASTAR\LASTAR.exe" -p Repets --album "Repets 2020" --year 2020 %1

Typical command line inserted in .bat file that processes input of the bat file (%1) with Repets preset, in mp3, overriding album and year ID Tag.

Others possible arguments can be used to display infos and exit :

Parameter	Description
-? or -h or --help	Displays argument list
-v, --version	Displays LASTAR version

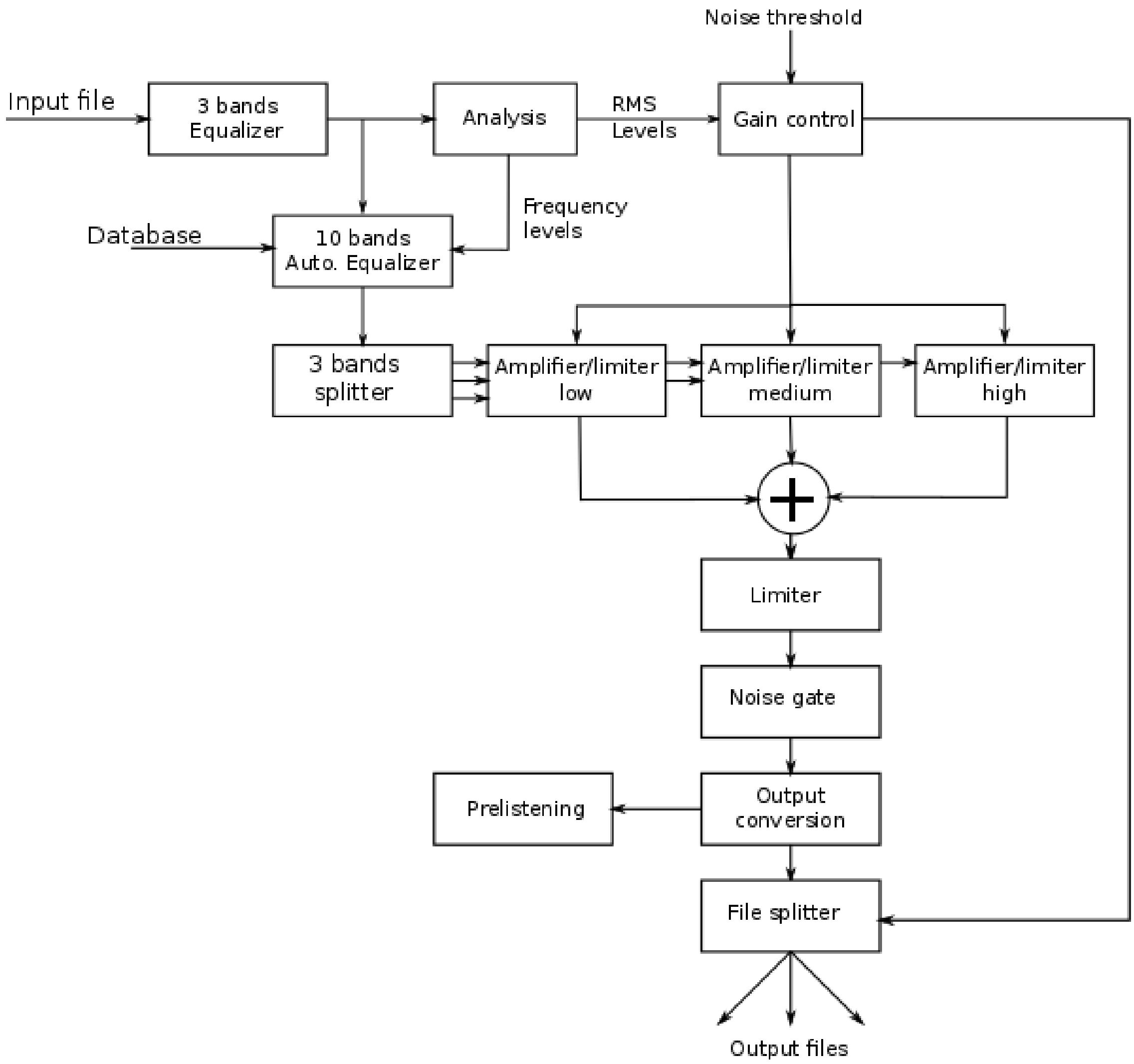
15. Technical features

15.1. Main characteristics

- wav, flac, ogg, vorbis or mp3 input and output,
- Management of IDTags V1 or V2,
- Automatic Left/right channel balancing,
- RMS Amplifier with fast lookahead limiter to prevent clipping,
- 10 bands automatic equalizer with FFT analyser,
- 3 bands processing using Finite Impulse response Filter (FIR), implemented by multi-threaded FFT convolution,
- Automated amplifier gain adjustment and noise estimation for file splitting,
- Smooth automatic gain control (AGC) to compress dynamic without "pumping" effect,
- 3 bands biquad equalizer and noise gate.

15.2. Processing chain description

Here is a brief synopsis of the LASTAR processing chain :



16. To recompile...

You will need following tools :

- MSCV2015 or above
- Qt 5.13.0
- libsndfile
- libmpg123
- boost 1.59
- taglib 1.11.1

The mp3lame library is only used at run-time (the software can run without) and is not necessary for compilation process.

17. Troubleshooting

In case of trouble, please go to http://www.arthelion.com and select *Contact* at the top of the page to fill in a request for information.