Resize Audio v0.5 User handbook

# Index

[0. Index 1](#_Toc429862610)

[1. Author 1](#_Toc429862611)

[2. Introduction 1](#_Toc429862612)

[3. License 2](#_Toc429862613)

[3.1. Download the source code 2](#_Toc429862614)

[4. Quick start 2](#_Toc429862615)

[5. Input parameters 2](#_Toc429862616)

[5.1. Input\_file y Output\_file 3](#_Toc429862617)

[5.2. [-factor \_factor\_] 3](#_Toc429862618)

[5.3. [-lowQuality|-highQuality] 3](#_Toc429862619)

[5.4. [-tb integer\_inverseOfTheTotalTransitionBand] 4](#_Toc429862620)

[5.5. [-br output\_bitrate\_in\_kbps] 5](#_Toc429862621)

[6. Theory 5](#_Toc429862622)

[7. Acknowledgements 5](#_Toc429862623)

[8. Links 6](#_Toc429862624)

# Author

The application has been developed by Francisco Javier Rojas during August of 2015.

Contact e-mail: frojasg1@hotmail.com

# Introduction

With the Resize Audio application you can transform an input audio file to an output audio file whose duration is the original one multiplied by a factor which can be either greater or lesser than one.

The application is run through the command line interface.

The following input and output formats are supported:

Input: wav, mp3, ac3, dts

Output: mp3, ac3

Currently, only the Windows version of the application is available, and it is not probable that other platforms will be supported.

# License

This application is made available under GPL license.

This means that its source code is open for everyone to see, use or modify, provided that any new applications using it are distributed under GPL license.

For more information about this type of licenses, you can check out this link:

http://www.gnu.org/licenses/gpl-3.0.en.html

## Downloading the source code

You can download the source code of the application and the libraries used by it in the following link:

https://sourceforge.net/projects/audio-synchonization/

# Quick start

Follow the next steps to start using this application:

1. Open a command window (*Windows key + R*and type *"cmd" + return*).
2. Go to the folder where the Resize Audio binary is (in the subfolder \_binary), by running: *cd* path...*/\_binary*
3. Run: *PATH=%PATH%;lib*
4. Now you can run the application. The application can accept some input parameters, which will be explained later on in this handbook.

An easy way to run the application could be this:

***resizeAudioFile.exe* "Input\_file" "Output\_file" *-factor \_factor\_***

# Input parameters

The use of the input parameters will be explained in this chapter, as well as the available values.

The pattern of the command used to launch the application is the following:

***resizeAudioFile.exe*****"Input\_File" "Output\_File" *[-factor \_factor\_] [-lowQuality|-highQuality] [-tb integer\_inverseOfTheTotalTransitionBand] [-br output\_bitrate\_in\_kbps]***

The first two parameters are mandatory and have to appear in exactly the same order as in the line above.

The braces [ ] indicate that the parameter is optional.

The pipe | indicates that either the lefthand side or the righthand side option appear, but not both.

The optional parameters don’t need to be given in exactly the same order as in the sample command line.

## Input\_file and Output\_file

These parameters are mandatory and have to appear in the order shown above.

*Input\_file* and *Output\_file* can be double-quoted to allow for the possibility of having spaces in the file names or their paths.

They may include absolute or relative paths to the files.

The completion option of the command line interface can be used by writing part of the file name and then pressing one or more times the tab key. This can be repeated for every folder in the path.

Normally, the quotation mark is included in case it is needed when the tab key is used for completion.

The input formats which are supported by the application are some of those supplied by the ffmpeg library.

For each input format, the application assigns an output format, and this assignation cannot be changed. The input and output formats match as follows:

|  |  |
| --- | --- |
| **Input format** | **Output format** |
| .wav | .mp3 |
| .mp3 | .mp3 |
| .ac3 | .ac3 |
| .dts (only some versions are supported) | .ac3 |

## [-factor \_factor\_]

This parameter is optional. If it is omitted, the default value will be taken (1, the audio is not transformed).

*\_factor\_* is to be replaced by the factor which you want to apply to the duration of the input audio file.

The number must be in English format, that is, using the . as the separator between the integer and the decimal part.

For example, to apply a factor of 101%, you have to set *\_factor\_* to 1.01

The command line to run the application would thus be:

***resizeAudioFile.exe* "Input\_file" "Output\_file" *-factor 1.01***

The application offers a maximum resolution of 5 decimal digits and *\_factor\_* will be approximated to this resolution (typically, the factor applied will be slightly different than the input *\_factor\_*, but within the resolution of 5 decimal digits).

## [-lowQuality|-highQuality]

This parameter is optional. If it is not present, the application will behave as in *-highQuality* mode.

To choose the quality, one of these parameters must be included.

*-lowQuality*. When this parameter is used, the interpolation quality is lower for high frequencies.

If this value is set, the interpolation filter used is an ideal low-pass filter windowed with a rectangular window. It provides less quality than the *–highQuality* option, but it is twice as fast.

*-highQuality*. When this parameter is used, the interpolation quality is higher for high frequencies.

If this value is set, the interpolation filter used is an ideal low-pass filter windowed with a Hamming window. It provides more quality than the *–lowQuality* option, but it is twice as slow.

## [-tb integer\_inverseOfTheTotalTransitionBand]

This parameter is optional.

If this parameter is omitted, the following default values are taken (depending on the sample rate):

|  |  |  |
| --- | --- | --- |
| **Sample rate****(Sr)** | **Supposed bandwidth of the input audio** | **integer\_inverseOfTheTotalTransitionBand** |
| 44100 Hz | 20000 Hz | 6 |
| 48000 Hz | 20000 Hz | 3 |
| A different one | ?? | 6 |

If the sample rate is different from 44100 Hz or 48000 Hz, a default value of 6 is taken. It might then be the case that this value is not suitable for the transformation.

Should the sample rate be different from 44100 Hz or 48000 Hz, a value for this parameter should be calculated and set through the command line.

The higher the values for this parameter are, the lower the transition bandwidth for the low-pass filters will be, and thus better quality will be achieved.

Nevertheless, the time used to process the input audio samples is proportional to the value of this parameter. That is, if the value of this parameter is doubled, the processing time is doubled too.

A suitable value for this parameter can be calculated through the following formula:

**integer\_inverseOfTheTotalTransitionBand = Sr / 4 / (Sr / 2 - InputAudioBandWidth )**

Where Sr is the sample rate.

Remember that the application expects a positive integer value for this parameter.

For instance:

 Audio with a sample rate of 8000 Hz

 Bandwidth of the audio,3800 Hz

Then:

integer\_inverseOfTheTotalTransitionBand = 8000 / 4 / ( 8000 / 2 - 3800 ) = 2000 / 200 = 10

For this example, the command line to run the application will be:

***resizeAudioFile.exe* "Input\_file" "Output\_file" *-factor 1.01 -tb 10***

## [-br output\_bitrate\_in\_kbps]

This parameter is optional. If it is not specified, a default value will be used which depends on the input audio format:

|  |  |  |
| --- | --- | --- |
| **Input audio format** | **Output audio format** | **Default output bitrate** |
| .wav | .mp3 | 320 kbps |
| .mp3 | .mp3 | Input bitrate |
| .ac3 | .ac3 | Input bitrate |
| .dts (only some versions are supported) | .ac3 | 640 kbps |

As it can be seen in the table, when the output audio format is the same as in the input, the default output bitrate is the same input bitrate.

When the output format is different from that of the input, the default output bitrate is the maximum bitrate allowed for that particular output audio format.

If you need the output to have a different bitrate from the default, you will have to use this parameter.

For instance, if you want to transform an *input.dts* file to an *output.ac3* file with a bitrate of 384 kpbs, you could use the following command:

***resizeAudioFile.exe "input.dts" "output.ac3" -br 384***

# Theory

To learn more about the theory this application is based on, see the following folder:

...\\_docs\theory

# Acknowledgements

- To the developers of the ffmpeg library.

- To the developers of the libraries ffmpeg is based on.

- To the developer of the FOBS4JMF library ([José San Pedro Wandelmer](http://www.jsanpedro.es)), a wrapper of the ffmpeg library which greatly simplified its use.

Without these libraries (a fenomenal work), the development of the Resize Audio application could not have been possible.

# Links

http://frojasg1.com

http://fobs.sourceforge.net/index.html

https://www.ffmpeg.org/