



AudioBot

# User Documentation

*Version 1.0*

tsugi

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## **Introduction**

AudioBot is the ultimate audio batch processor. Fast, reliable, and extensible, with an intuitive interface and a plethora of features that will improve the workflow of anybody dealing with audio assets. Sound artist, musician, web designer or game developer? AudioBot has some extremely innovative features just for you! You will be wondering how you have been working without it.

### ***Smart input***

Import samples in the most common file formats (e.g. Wave, Aiff, Mp3, Ogg Vorbis, Flac), with any bit depth, number of channels and sample rates. Need support for another format or for your proprietary tool? You can very easily write your own importer plug-in! Put watchers on folders and whenever a new file is added, it will get automatically processed! Import / export lists of files to other tools and find duplicates in your sound database.

### ***Powerful processing***

In addition to typical processing features (normalization, removing quiet parts, fading in/out, automatic padding or truncation...), AudioBot is a VST host: use any of your favorite plug-ins to process sounds! AudioBot's multithreaded processing adapts itself to the number of cores in your machine and crunch numbers faster than any other batch processor. Full reports can be generated in PDF or Excel format, notifications can be sent by email or by tweet etc... Batch processing can be stopped at any point and long sessions can be restored and continued. A command line version is available, making it possible to integrate AudioBot with your audio pipeline.

### ***Output versatility***

Go further than simple file conversion and renaming: automatically create HTML5 audio web sites, PDF sample catalogues, Excel or XML reports! Working in the game industry? AudioBot can generate work units for Audiokinetic's Wwise as well as projects for Firelight Technologies' FMOD and CRI Middleware's ADX2. And if that was not enough yet, AudioBot's plug-in system allows you to development your own exporter!

Want us to develop a specific import, processing or export plug-in for your game studio technology? Don't hesitate to contact us at [consulting@tsugi-studio.com](mailto:consulting@tsugi-studio.com).

## **Main Window**

In the top left corner of the main window, buttons can switch between three tabs. The first one is the input tab, dedicated to the selection of the files. The second one is the process tab, where the process chain can be defined. Finally the third one is the output tab, where export settings will be set. These tabs will be described in details below.

At the bottom of the window the main commands are available:

### *New*

This command creates a new AudioBot project. If a project is already open and has been modified, a confirmation message will appear asking if it should be saved first.

### *Load*

This command opens a previously saved AudioBot project. AudioBot projects have a “.bpp” extension, which stands for Batch Processor Project.

### *Save*

This command saves the current AudioBot project. If it has never been saved before, a dialog box will appear to let the user select a name (equivalent of a “Save as” function). Otherwise, the project file will be overwritten.

### *Start*

This command initiates the batch processing. A new window appears, displaying the progress of the operations.

### *Settings*

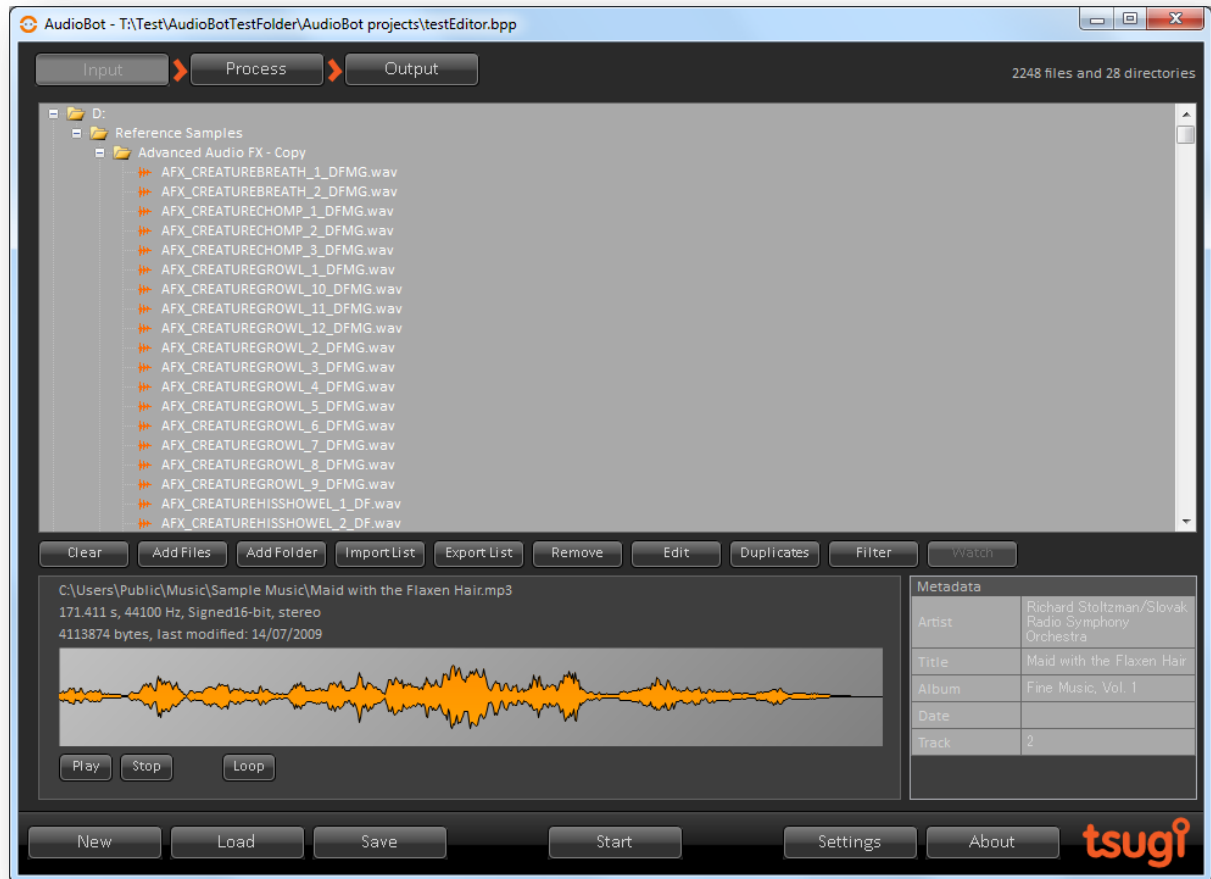
This command opens a new window, allowing the user to specify general settings (related the program and not to a particular project). These settings are not stored inside the .bpp file like input, process, and output settings. Instead, they are saved in the Windows registry.

### *About*

This command displays information about the program, its version, and what assemblies are used.

## Input Tab

When starting AudioBot, the input tab is the one shown by default. It allows you to specify what files you want to process with the tool.



## Managing the files list

There are several ways to add files to the input list:

- Use the “Add Files” button: this will open a file dialog box allowing the selection of multiple files at the same time. The following file formats are supported: Wave, Aiff, Mp3, Ogg Vorbis, and Flac.
- Use the “Add Folder” buttons: this will open a folder selection dialog box allowing for the addition of full folders. Files Sub folders are added as well.
- Drag & drop of sounds or folders directly from Windows to the input list
- Import list of sounds with the Import button. They can be in simple .txt file format (one file name per line) or in the Excel XML file format (one file name per row in the first column).

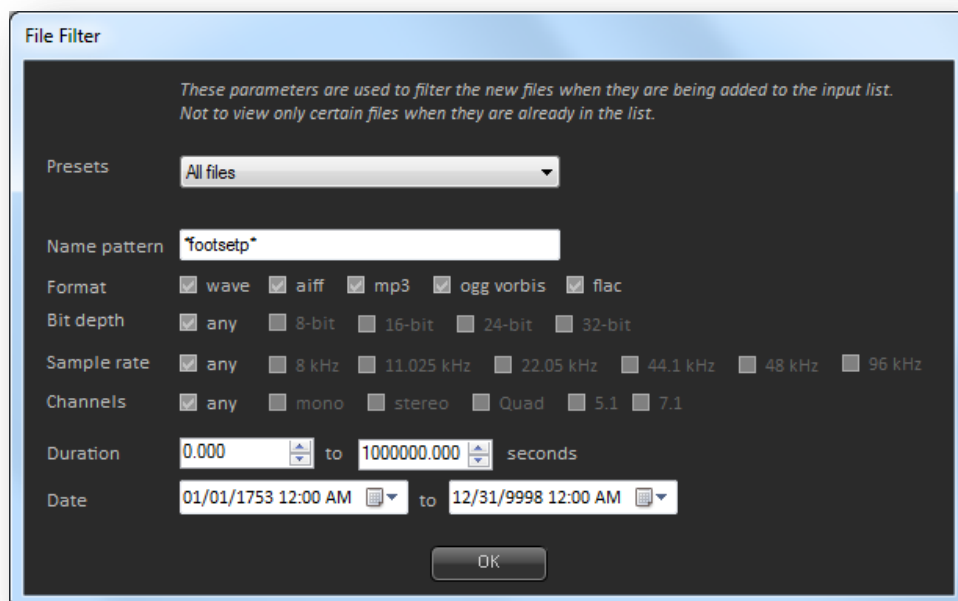
AudioBot will also accept playlists from Windows Media Player (.wpl format) or Winamp (.m3u format). Like for samples and folders, these files can be dropped directly onto the input list. List of sounds can be typed manually, generated by another tool or even by AudioBot by using the Export button. Like for import .txt, Excel .xml, .m3u, and .wpl are supported.

In all cases, files which are already in the input list will not be added a second time.

It is possible to manually remove samples or full folders by selecting one or more items from the input list and pressing the Remove button (or pressing the “Delete” key). If the “Confirm before removing files from input list” option has been selected in the Settings window, a confirmation message will appear first.

## **File Filter**

When adding files, a filter can be used to add only the ones of interest. The filtering parameters can be set in the “File Filter” window accessible by pressing the “Filter” button. A number of filter presets are available, for example for files at CD quality, or files created or modified during the last week etc... However, custom filters can be set as well. By default, the “All files” preset is selected, which means that no filtering will happen and that all the files you selected will be added.



### ***Name Pattern***

This describes the pattern (arrangement of characters) that the name of a file must follow. This does not take into account the extension of the file, nor the full path (i.e. it excludes folders). It should not contain any of the invalid characters for a file’s name. However the following wildcard specifiers are permitted:

\* : represents zero or more characters

? : represents exactly zero or one character

#### *Format*

This allows you to specify the allowed formats. Please note that at this time, files with another extension than the ones displayed and imported using custom plug-ins don't have their own checkbox.

#### *Bit depth*

The bit depth can be limited to one or more of the preset values (8-bit, 16-bit, 24-bit and 32-bit) or can be any number of bits if "Any" is selected.

#### *Sample rate*

Similarly, the sample rate of the sounds can be limited to one or more of the preset values (8 kHz, 11.025 kHz, 22.050 kHz, 44.1 kHz, 48 kHz and 96 kHz) or can be any frequency if "Any" is selected.

#### *Channels*

Again, the number of channels in the sound file can be limited to one or more of the preset values (mono, stereo, quad, 5.1 and 7.1) or can be any number if "Any" is selected.

#### *Duration*

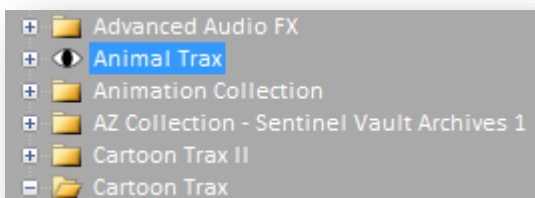
This allows you to specify the minimum and maximum durations (in seconds) a sound can have to be added to the input list.

#### *Date*

It is possible to choose a time interval in which the file must have been created / modified to be added to the input list. This is especially convenient to add only old files which were not processed, or the latest files added to a project.

### **Watchers**

You can also put a watch on some folders, so that whenever a new file is added, it gets automatically processed. For example, if you know you will get some voice lines recorded and delivered regularly in the same folder, you can just put a watch on it. Just select one or more folders and press the Watch button, the icon will switch to an eye. Press the "Watch" button again to remove the watch.



Watches can only be set on folders, not on individual files. Once you put a watch on a folder, it will detect changes in that folder and in all its subfolders as well. Therefore, if you try to put another

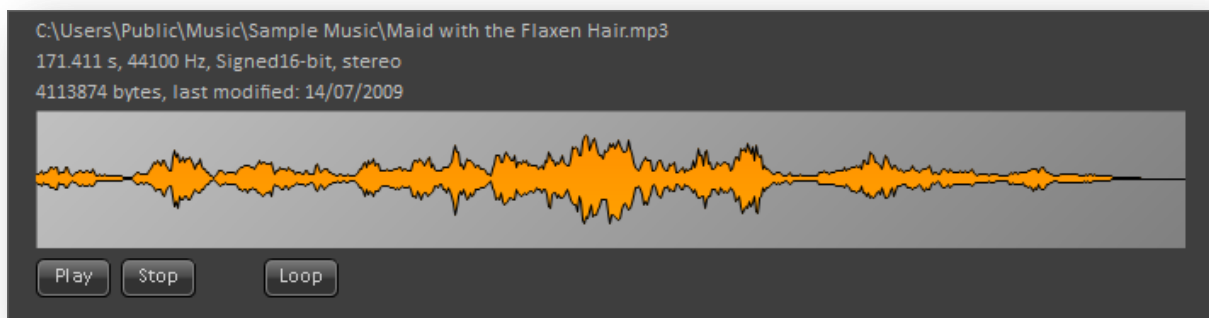


watch on one of the subfolders, it will not be added, as it is already under watch. Similarly, if you put a watch on a parent folder, any watches on the subfolders will be removed.

## **File visualization**

When a file from the input list is selected, its waveform will be displayed, along with some of the sound's characteristics such as sample rate in Hz, duration in seconds, bit depth and number of channels. The size of the file in bytes as well as the last time it was modified are indicated too.

If the *"Play sounds from input list when selected"* option has been selected in the Settings window, the sound will also be played back automatically. Otherwise, the *"Play"* button can be used. For long sounds, the *"Stop"* button can abort the playback. A *"Loop"* mode is also available.



When the *"Play folder contents when selected"* option is activated and if a folder is selected in the input list, each time the *"Play"* button is pressed a sound from that folder will be played back (the waveform of the signal will appear in gray to differentiate it from the regular playing mode). Which sound is played next depends on the chosen mode: sequential, shuffle, or random. The sequential mode will play each sound in the order in which they appear in the input list. The shuffle mode will play the sounds in the folder randomly while making sure that each of them is played once before repeating them. The random mode is truly random which means that the same sound could be played several times in a row. These playback modes are especially useful for game sound designers to test the randomization of assets. Please note that the *"Play sounds from input list when selected"* option also has to be selected for the folder contents to be played back.

## **Metadata**

When a sound is selected and its waveform displayed, information about the metadata contained in the file is also retrieved.

Metadata	
Artist	Richard Stoltzman/Slovak Radio Symphony Orchestra
Title	Maid with the Flaxen Hair
Album	Fine Music, Vol. 1
Date	
Track	2

### **Sound editing**

If a sample editor has been specified in the “*Settings*” window, it is possible to launch it (and to load a sample automatically) by double-clicking on a file in the input list, by pressing the “*Edit*” button, or by hitting the “Enter” key. If several samples are selected and the target application allows it, they will be opened simultaneously.

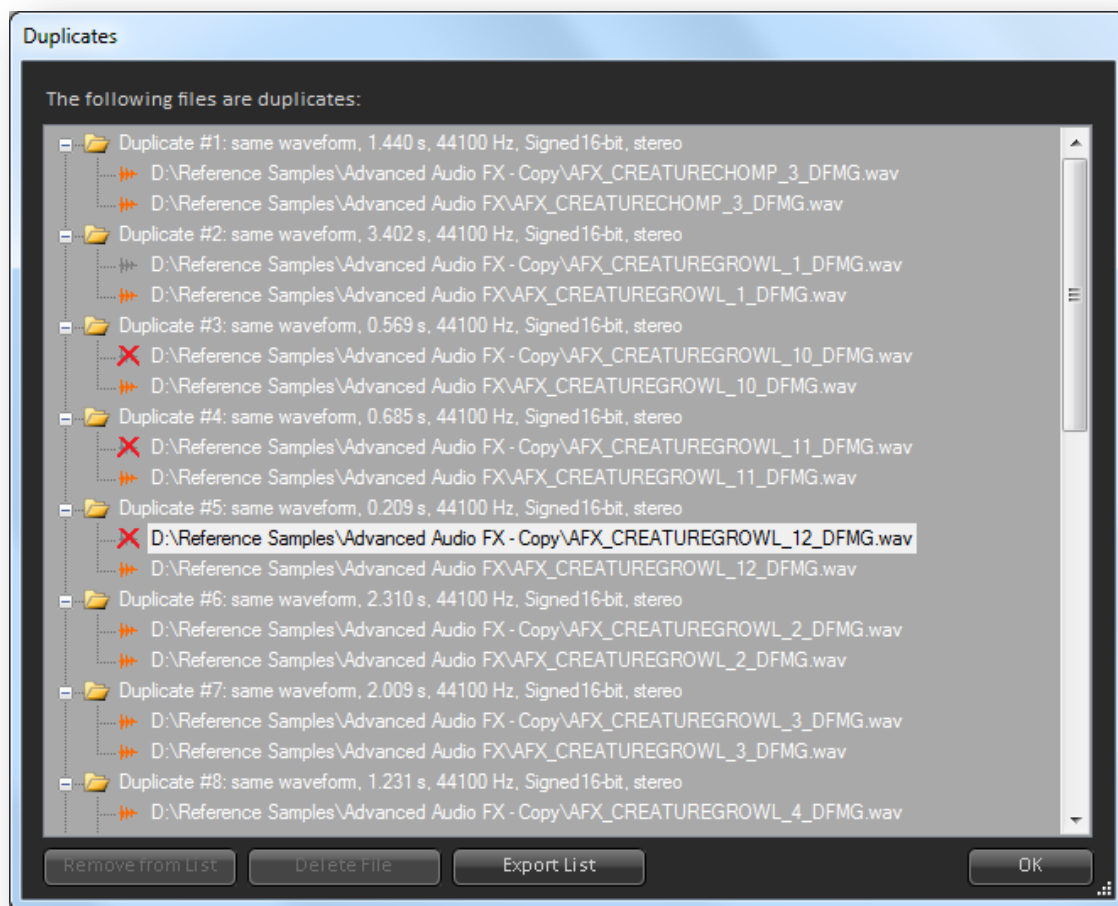
### **Duplicates**

From the input tab, it is also possible to find duplicates (i.e. identical sounds). Duplicate files could be stored in a different location, with a different name, or with different metadata, but still contain the exact same sample data.

AudioBot first compare the audio characteristics of the files from the input list (such as sample rate, duration, bit depth or number of channels). If two files have exactly the same characteristics, then the sample data is loaded and compared, sample by sample.

Depending on the number of files in the input list and their duration, this operation can take more or less time. A progress bar is displayed and the operation can be aborted at any time by the user.

If the search goes all the way and AudioBot found duplicates, a new window appears displaying all the identical files. Their characteristics are also indicated, as pictured below:



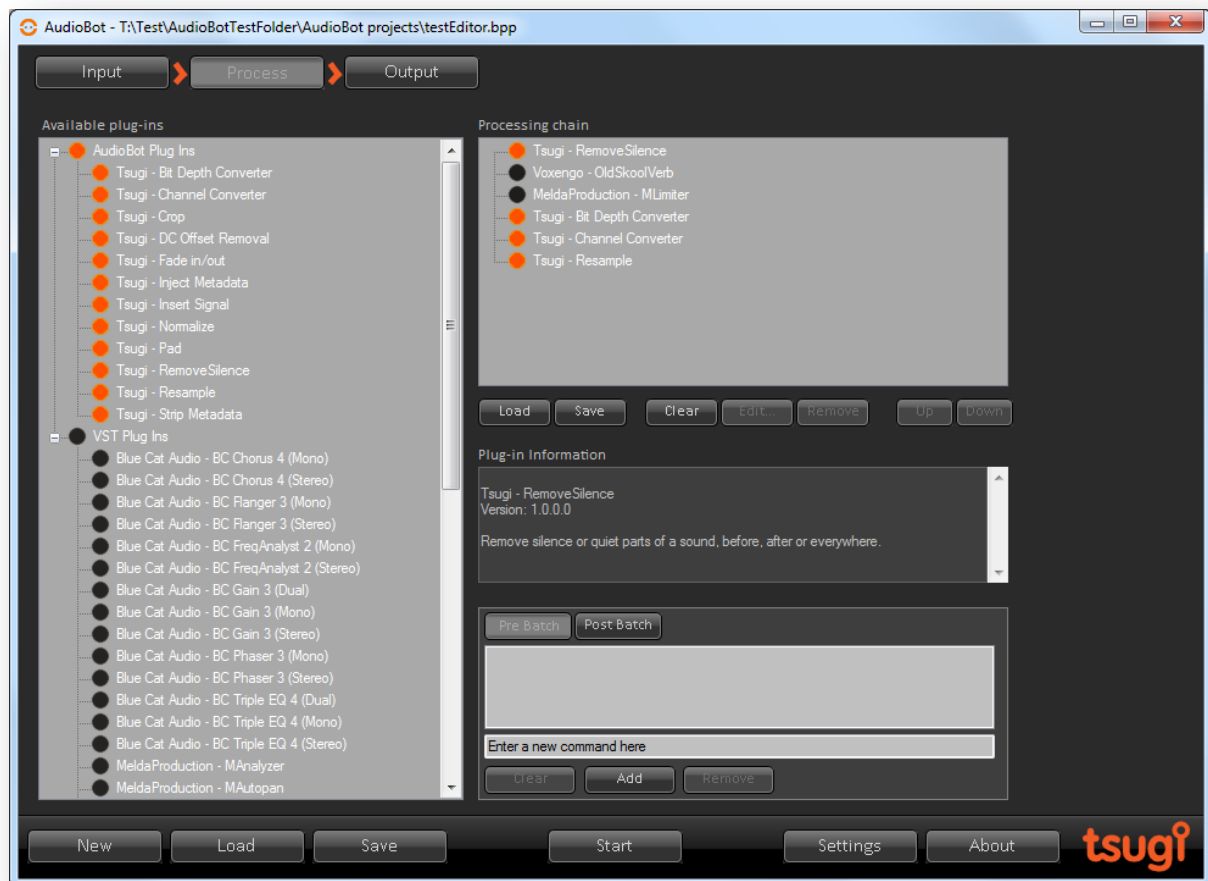
Once the list of duplicates is displayed, you can select any of the files and remove it from the input list with the “*Remove from List*” button. Its icon will appear grayed, and when you go back to the input list after pressing OK, the files you removed will not be included anymore. It is also possible to delete a file (i.e. remove it from your hard drive) by pressing the “*Delete File*” button. In this case, its icon will also appear as a gray sample (since it will be removed from the input list) but under red mark to indicate that it has been deleted. If you would rather deal with the duplicates at another time or with another tool, you can export the list of sounds either as a text file or an Excel-compatible XML file with the “*Export List*” button.

This function to find duplicates is particularly useful and can be used even if you are not planning on using the batch processor itself: simply drag and drop your sample folder onto the input list, click on the “*Duplicates*” button, and start cleaning-up your sample database!

## Process Tab

### Processing chain

The processing tab allows you to define the plug-in chain that will be used to process each sample from the input list. Each element of the processing chain can either be an AudioBot, a VST or a Vamp plug-in.



Adding items to the processing chain is done by dragging them from the plug-ins tree on the left and dropping them onto the chain list. If a new item is dropped over an existing item, it will be inserted just before it. It is also possible to double-click on a plug-in in the tree to add it to the processing chain. Again, if an item is already selected in the processing chain, the new item will be inserted just before it otherwise it will be added at the end of the chain. The order of the plug-ins in the chain can further be modified with the *Up/Down* buttons or by drag and drop.

Once a plug-in is in the list, it becomes possible to adjust its parameters by double-clicking on it, or selecting it and pressing the “*Edit...*” button (or hitting the “*Enter*” key). Of course it is possible to have several instances of the same plug-in in the processing chain, each having its own set of parameters. Plug-ins can be removed with the “*Remove*” button (or by using the “*Delete*” key). If the “*Confirm before removing a plug-in from processing chain*” option has been selected in the “*Settings*” window, a confirmation window will appear before the plug-in is actually removed. Whole

chains can be loaded, saved, or cleared thanks to the buttons of the same name. Plug-ins chains are saved in XML format and could technically be written manually or generated by a third party software.

### **Types of plug-ins**

This section describes the three types of plug-ins supported by AudioBot. They can be mixed inside the same processing chain and appear in any order.

#### **AudioBot plug-ins**

AudioBot plug-ins have been developed by Tsugi but you can also build your own by using the AudioBot plug-in SDK. They are typically easier to write than VST plug-ins and you can either define your own GUI or let AudioBot generate it automatically for you. Please refer to the AudioBot plug-in SDK documentation to learn more. You can also contact tsugi for consulting services if your studio is in a hurry or does not have the bandwidth / knowledge to develop a plug-in.

#### **VST plug-ins**

VST stands for Virtual Studio Technology and is a registered trademark from Steinberg Media Technologies GmbH. VST plug-ins are very popular in the world of computer music and a lot of them are available, either as freeware, shareware or commercial products. A good source of information on VST plug-ins is KVR (<http://www.kvraudio.com>). The VST plug-ins listed are the ones present in the *VST plug-ins folders* specified in the *Settings* window. Only 64-bit VST plug-ins are recognized by AudioBot. At this time, the tool follows the VST 2.4 specifications. Moreover, only VST plug-ins with a maximum of 2 inputs and 2 outputs are currently supported.

VST plug-in sometimes come with presets. In AudioBot, they are available through the combo box in the top-left corner of the VST plug-in window (which appears when you double-click on a VST plug-in in the processing chain). AudioBot also allows you to save and load your own VST presets in XML format from the same window.

#### **Vamp plug-ins**

Vamp is an audio processing plugin system for plugins that extract descriptive information from audio data — typically referred to as audio analysis plugins or audio feature extraction plugins. As Vamp plug-ins usually calculate features or indicators from the samples instead of modifying them, the user has to specify output files where the results will be stored for each of the analyses.

The Vamp system has been developed at the Centre for Digital Music (CDM), Queen Mary, University of London. The CDM and other research centres offer Vamp plug-ins. They can be found here: <http://www.vamp-plugins.org/download.html>. Like for the VST plug-ins, the Vamp plug-ins listed in this window are the ones present in the *Vamp plug-ins folders* specified in the *Settings* window.

### **Note on plug-ins changing the duration of sounds**

Some plug-ins (e.g. reverberations, delays) will change the duration of the input signals. There is no easy way for the host - in this case AudioBot - to determine what the final duration of the sound will be as it depends on the plug-in algorithm. A common solution is for the host to use a gate with a threshold level and a long release. The end of the processed signal is then set when the output has been under the threshold for a given time.

Another solution, which is usable with AudioBot, is to first add enough silence at the end of the sound to process, either with the "Pad" or the "Insert signal" plug-ins (before the plug-in that modifies the duration of the sound). Then, after the duration-changing plug-in, remove the quietest or silent part at the end of the processed sound with a "Remove silence" plug-in.

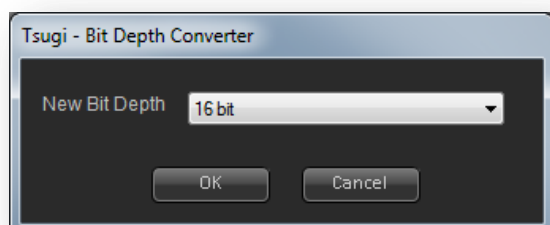
### **Available processing units**

The application does not ship with any VST or Vamp plug-ins but you will be able to use any VST plug-ins following the 2.4 specifications of the VST format and any Vamp plug-in. It also comes with a number of tsugi AudioBot plug-ins, which are described below.

#### **Bit Depth Converter**

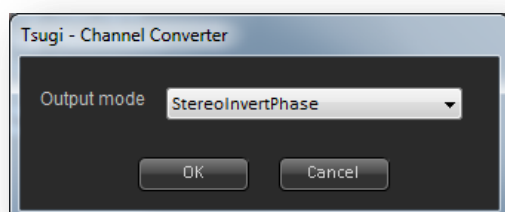
This plug-in changes the bit depth of a signal.

*New Bit Depth:* this is the target bit depth. Choices are 8-bit, 16-bit, 24-bit, 32-bit (integer), 32-bit (floating point number), 64-bit (floating point number with double precision). Please note that changing the bit-depth impacts the sound quality of the sound files.



#### **Channel Converter**

This plug-in can convert mono signals into stereo, stereo signal into mono, or change the balance of stereo files. Here is the list of possible transformations:



*Mono*: if the original signal is mono, it is left untouched. If it is stereo, the two channels are mixed into one to create a mono signal.

*MonoLeftOnly*: if the original signal is mono, it is left untouched. If it is stereo, only the left channel is kept to create a mono signal.

*MonoRightOnly*: if the original signal is mono, it is left untouched. If it is stereo, only the right channel is kept to create a mono signal.

*Stereo*: if the original signal is mono, it is copied into the left and right channels of a new stereo signal. If the original is stereo, it is left untouched.

*StereoSwap*: if the original signal is mono, it is copied into the left and right channels of a new stereo signal. If the original is stereo, the left and right channels are swapped.

*StereoInvertPhase*: if the original signal is mono, its phase is inverted before being copied into the left and right channels of a new stereo signal. If the original is stereo, the left and right channels both have their phase inverted.

*StereoInvertPhaseWiden*: if the original signal is mono, it is copied into the left and right channels of a new stereo signal, with the phase of the right channel being inverted. If the original signal is stereo, the phase of the right channel is inverted.

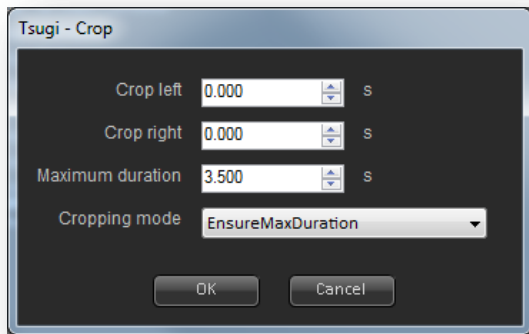
*StereoPannedCentre*: if the original signal is mono, it is copied into the left and right channels of a new stereo signal. If the original signal is stereo, left and right channels are combined so the sound appears panned in the centre.

*StereoLeftOnly*: if the original signal is mono, it is copied into the left and right channels of a new stereo signal. If the original is stereo, the left channel is copied into the right channel as well.

*StereoRightOnly*: if the original signal is mono, it is copied into the left and right channels of a new stereo signal. If the original is stereo, the right channel is copied into the left channel as well.

### **Crop**

This plug-in truncates a sound by removing a fixed amount of data (at the beginning, at the end, or both) or by ensuring it has a given duration.



### *Cropping mode*

There are two cropping modes: “Remove fixed amounts” and “Ensure max duration”. In the first mode, a given amount of signal (expressed in seconds and specified by the user) can be removed either from the beginning of the sound, the end or both. In the second mode, if the sound is longer than the maximum duration authorized, it will be truncated (both at the beginning and at the end, in equal proportions).

### *Crop left*

This is the amount of signal (expressed in seconds) to be removed from the beginning. It is only used when the cropping mode is set to “Remove fixed amounts”.

### *Crop right*

This is the amount of signal (expressed in seconds) to be removed from the end. It is only used when the mode is set to “Remove fixed amounts”.

### *Maximum duration*

This is the duration to which the signal must be truncated. It is used when the mode is set to “Ensure given duration”.

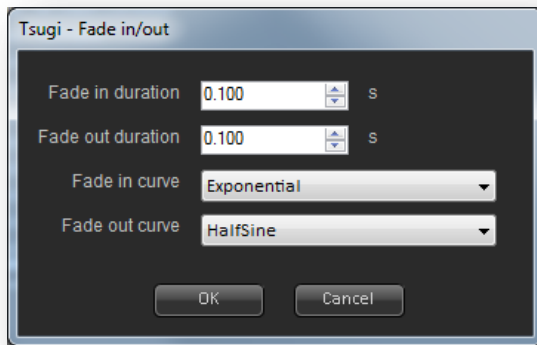
## **D.C. Offset Removal**

The plug-in removes the D.C. component of a signal by calculating its mean and subtracting it from the signal. The DC offset is an amplitude imbalance of the waveform which in some cases can cause clicks at the start and at the end of a sound during playback, as well as other artifacts. This plug-in has no parameters.

## **Fade in/out**

This plug-in applies a fade-in and/or fade-out to the sound.





#### *Fade in duration*

This is the duration of the fade-in, expressed in seconds.

#### *Fade out duration*

This is the duration of the fade-out, expressed in seconds.

#### *Fade-in curve*

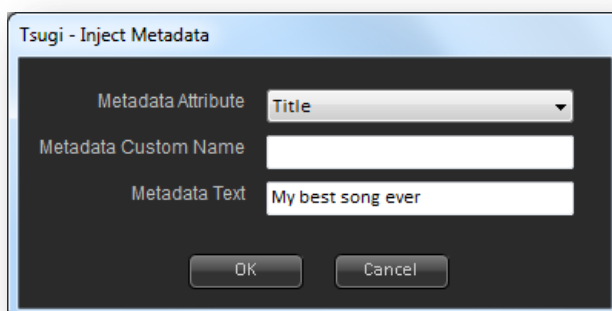
This is the volume curve that will be used to fade the audio in. Available curves are: linear, exponential, inverse exponential and half-sine.

#### *Fade-out curve*

This is the volume curve that will be used to fade the audio out. Available curves are: linear, exponential, inverse exponential and half-sine.

### **Inject Metadata**

The “Inject Metadata” plug-in allows the user to add metadata to all the sounds processed.



#### *Metadata Attribute*

This combo box allows the selection of the name of the metadata that will be added. Common names such as “Artist”, “Title”, “Album” etc... are proposed. If the metadata to add does not correspond to any of these, “Custom” can be selected. In this case, the user will be able to enter a name of his /her choice for the metadata (in the textbox just below).

#### *Metadata Custom Name*

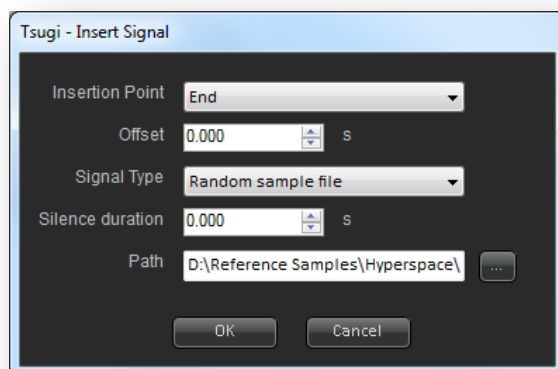
This is a custom name for the metadata. It will only be taken into account if the Metadata Attribute combo box is set to “Custom”.

#### *Metadata Text*

This is the metadata itself, represented as a text. If “Artist” was selected for the attribute, “Pink Floyd” could be entered here for example.

### **Insert Signal**

This plug-in inserts an audio file or some silence into the processed sounds.



#### *Insertion Point*

This parameter defines at what point in the sound the new signal or silence will be added. It can be at the “Start”, at the “End”, or at a given “Offset”, in which case the “Offset” parameter described below will be used.

#### *Offset*

The time at which the audio file or the silence will be inserted, expressed as a number of seconds.

#### *Signal Type*

This describes what type of data should be inserted: “Silence”, a specific “Sample file”, or an audio file randomly chosen from within a folder. Audio files can be in any of these formats: Wav, Aiff, Mp3, Ogg Vorbis, or Flac.

#### *Silence duration*

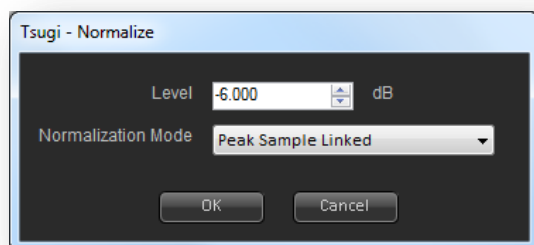
If we are inserting silence, this parameter describes the duration of that silence, expressed in seconds.

#### *Path*

If we are inserting a specific audio file, this is the path of the file to be added to the sounds. In the case where we are adding a random audio file from within a folder, simply select one of them and its parent folder will be taken into account.

### **Normalize**

This plug-in normalizes the sounds to a given level. The unit used for the level is dependent on the selected mode.



#### *Mode*

When this combo box is set to "Peak Sample Unlinked", each channel is normalized individually. When it is set to "Peak Sample Linked", the maximum amplitude of the whole signal (including all channels) is detected and the signal is normalized accordingly. In these modes, its peak amplitude is considered to be the individual sample value with the greatest magnitude. When the combo box is set to "LUFS (Integrated)", the amplitude of the signal is modified so that the loudness of the whole sound is equal to the level specified. With the mode set to "dBTP (True Peak)", the amplitude of the sound is modified so that the "True Peak" of the sound is equal to the desired level.

#### *Level*

This must be a value between -96 and +6. The unit is dictated by the selected mode as follows:

"Peak Sample Linked": dBFS

"Peak Sample Unlinked": dBFS

"LUFS (Integrated)": LUFS

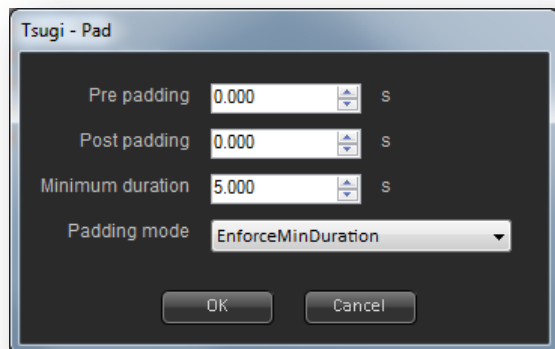
"dBTP (True Peak)": dBTP

#### **Additional note**

In "LUFS (Integrated)" and "dBTP (True Peak)" modes, AudioBot uses the EBU R128 recommendations provided by the European Broadcasting Union to calculate loudness and true peak. Further details on this specification can be found at: <http://tech.ebu.ch/loudness>

## **Pad**

This plug-in adds a fixed amount of silence before and/or after a sound, or ensure it has a minimum duration by padding it.



### *Padding mode*

There are two padding modes available. In the “Enforce minimum duration” mode, if the signal is shorter than the specified duration, silence will be added after the sound so that it reaches that minimum duration. In the “Add fixed silence” mode, the tool adds a given amount of silence (specified by the user) before and/or after the signal (the amounts can be different).

### *Minimum duration*

This is the minimum duration we want the signal to have (in seconds). This parameter is only used when the mode is set to “Enforce minimum duration”.

### *Pre padding*

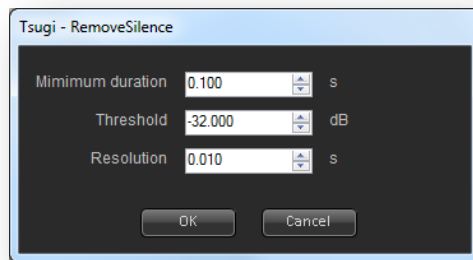
This is the amount of silence to be added before the signal. This parameter is only used if the mode is set to “Add fixed silence”.

### *Post padding*

This is the amount of silence to be added after the signal. This parameter is only used if the mode is set to “Add fixed silence”.

## **Remove silence**

This plug-in removes the silence or the quietest parts of a sound.



### *Threshold*

The threshold is the level in dB under which the signal is considered as quiet.

### *Minimum duration*

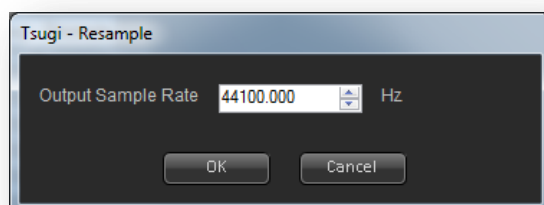
The quiet part must be at least that long to be removed (in seconds).

### *Resolution*

This is the duration in seconds of the analysis window on which the RMS is calculated.

## **Resample**

This plug-in modifies the sample rate of a signal by downsampling or upsampling it.

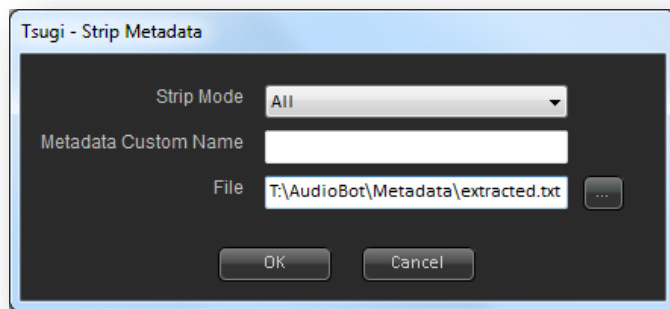


### *Output Sample Rate*

This is the new sample rate of the signal. It can be set to any value between 4000 Hz and 96000 Hz. Please note that file formats have different limitations concerning the valid range of sample rates.

## **Strip Metadata**

This plug-in removes all the metadata of a given type associated with the sounds.



### *Strip Mode*

The strip mode determines the type of metadata that will be removed from the sound files. A combo box allows the selection of the most common metadata fields, such as “Author”, “Title” or “Album” for example. It is also possible to remove a type of metadata that is not proposed by the combo box (for example a proprietary type) by selecting “Custom”, in which case the name of the custom metadata will have to be entered in the textbox just below. Finally, it is possible to strip all the metadata from the sound files by selecting “all”.

### *Metadata Custom Name*

This is a textual field that describes the type of the metadata to be removed. It is only taken into account if the strip mode is set to “custom”.

### *File*

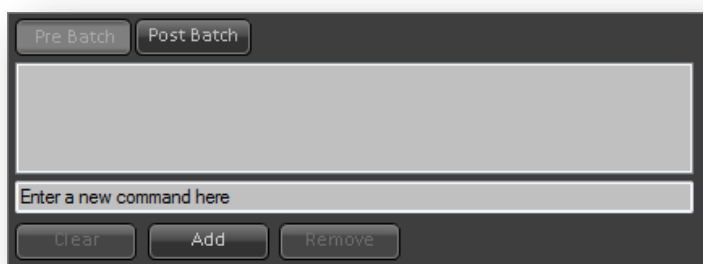
This is an optional text file which will contain all the metadata removed from the sounds.

## Pre / Post Batch Commands

In addition to configuring the plug-in chain, the processing pane allows the user to set pre and post-batch commands that will be executed before the batch is started and after it has completed respectively.

It is possible to switch between the editing of the pre and post-batch commands by pressing the “*Pre Batch*” and “*Post Batch*” buttons. In both cases, to add a command, simply click on the text box, type your command and press the “*Add*” button. You can also remove the selected command or clear them all with the buttons of the same names.

Commands are executed in sequential order, from the top of the list to the bottom.



## Output Tab

The output tab allows the specification of AudioBot exporting options. Processed files can be renamed, converted into a different format, or grouped into banks for popular game audio middleware such as AudioKinetic's Wwise, Firelight Technologies' Fmod or ADX 2 from CRI Middleware. In addition, statistical reports can be generated, as well as full sample catalogues in PDF format or HTML5-compatible web sites. All this can be done simultaneously by selecting all the exporters you want to use (click on the LED on their tab to activate or deactivate them).



We describe below the options available for each of the exporters.

### Single File exporter

A configuration window for the 'Single File exporter'. It is divided into two main sections. The top section, titled 'File Name', contains a 'File output' dropdown menu set to 'write in new folder with new file names', a 'New path' text field with 'C:\Users\Nicolas\Documents' and a browse button (...), and a 'New name' section with a text field containing 'Use the buttons below to specify how to build new file names...' and four buttons: 'File name', 'Extension', 'Text', and 'Index', along with a 'Clear' button. The bottom section, titled 'File Format', contains an 'Output format' dropdown menu set to 'original format' and two checked checkboxes: 'Retain format changes made by the processing chain' and 'Clip out of range samples'.

When audio files are processed they can be exported according to different schemes:

- *Replace files with processed ones*

The new files will have the same name and the original files will be overwritten. This is not recommended as the original data will be lost. However, in some circumstances (files are not the only copies and are coming from a depot, too much data to duplicate etc...) it can be a valid choice. Please note that this option requires the user to have rights to delete / create files in the original folder.

- *Write in same folder with new names*



The new files will be saved along the original files, in the same folder, but with different names. The new names are created from a set of rules specified below.

- *Write in new folder with new names*

The whole folder structure is duplicated in another location. Therefore the root folder has a new name. The names of the files are also different; they will be generated from a set of rules specified below.

- *Write in a new folder with same names*

The new files are saved with the same name than the original files, but in another folder, specified by the user. If there were any subfolders, the entire subfolder hierarchy will be duplicated under the destination folder.

- *Write in a flat folder with unique names*

The new files are saved in a new folder, specified by the user. The original subfolders hierarchy is not duplicated and all files are saved at the same level. New names are created from a set of rules specified below. Files which end up having the same names because of the disappearance of the subfolders have an extra index added to their name.

- *Get new names from a reference file*

A reference file is used to specify the new paths of the filenames. This reference file contains pairs of paths (always the original one first, and then new one). Three types of files can be used as a reference. First it can be an Excel file with the original names in the first column and the new names in the second column. This type of file must be saved in the "XML Spreadsheet 2003" format in Excel. A text file (having the .txt extension) can also be used as a reference. Each line will have both the old and the new paths of a sound file, separated by a ">". Finally, a reference file can also be a regular XML file. Usually this type of reference will be generated by another tool. The overall format is free as long as it is a valid XML file: the file can contain other information but entries related to the renaming should have the form `<file originalName="xxx" newName="yyy"/>`.

- *Write in one single file*

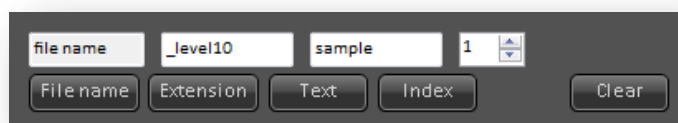
All the data from the processed sounds will be saved in one big single file. An optional silence segment whose duration is specified in milliseconds can be added between the processed sounds.

### **Building new file names**

For some of the export modes described above, generation of new filenames is required. AudioBot offers you a powerful yet intuitive way to specify how to build these new filenames, by concatenating several blocks of text together. Up to five text blocks can be added to form a new name, and four types of blocks are available:

- Text: a regular text. It can be as simple as “\_” to separate to other blocks, or “MyGameProject” to indicate what the files are.
- Filename: the original name of the file that has been processed (without the path and extension).
- Extension: the original extension of the file. Note: this does not correspond to the extension of the file that will be generated, as this one depends on the output format you will choose.
- Index: a number increasing with each file processed and for which you can select the starting value.

AudioBot will make sure that the filenames built by concatenating the text blocks are unique (you should at least add a Filename or an Index block). If you want to start again, simply press on the “Clear” button.



The screenshot shows a dark-themed interface with four input fields in a row: 'file name' (containing '\_level10'), 'Extension' (containing 'sample'), 'Text' (containing '1'), and 'Index' (containing '1'). Below these fields are four buttons labeled 'File name', 'Extension', 'Text', and 'Index'. To the right of these buttons is a 'Clear' button.

### **File Format**

In addition to their new location and name, the processed files can be saved in a different format. The “*Output format*” combo box allows you to select one of the file formats supported by AudioBot. For example, if you specify “*Mp3*”, all the files will be saved in Mp3, regardless of their original format. If on the contrary you want to keep the files’ format untouched, simply select “*original format*” from the combo box.

### ***Retain changes made by the processing chain***

When this option is selected (which is the default), any modifications made to the number of channels, sample rate, or bit depth of the original sample are retained when generating the output sample. If the option is not selected, then the number of channels, sample rate, and bit depth of the output sample will match that of the input sample, irrespective of the plug-ins used in the processing chain.

### ***Clip out of range samples***

If for some reason the processing chain generated sample values outside of the valid range, this option will allow you to clip them so that they stay in the valid range.

### **Report exporter**

The processing output can be stored as a report. It can be an Excel file, a HTML web page, or even a PDF document. Statistics about the data processed and the results of the batch processing are included in the reports, such as:

- The total number of files
- The number of files successfully processed

- The number of files whose processing failed
- The number of AIFF files
- The number of FLAC files
- The number of MP3 files
- The number of OGG files
- The number of WAVE files
- The various sample rates and the number of files having these sample rates
- The various bit depths and the number of files having these bit depths
- The various channel numbers and the number of files having these channel numbers
- The number of files having metadata
- The number of files without metadata
- The total batch processing time
- The average processing time per file
- The shortest processing time for a file
- The longest processing time for a file
- The total data processed
- The average data processed per file
- The smallest file processed
- The largest file processed
- The oldest file processed
- The most recent file processed

Path: T:\Test\AudioBotTestFolder\myreport.xls

Options

Type: Excel File

☐ Include list of files

☐ Open report once generated

#### *Path*

This is the path of the report file that will be generated.

#### *Type*

This is the type of report generated. It can be an Excel file, an HTML file, or a PDF file. In the case of the PDF reports, pie charts will also be included.

#### *Include list of file*

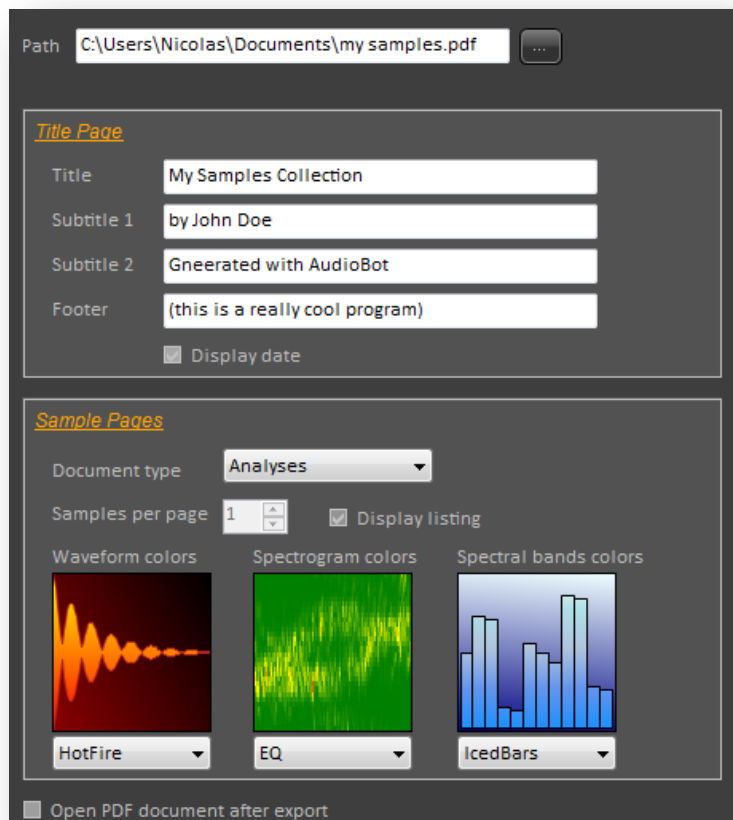
If this checkbox is marked, the list of all the samples processed will be included in the report, in addition to the statistics.

#### *Open report once generated*

If selected, the report will be opened after being generated: in your default browser for HTML reports, in a PDF viewer (such as Acrobat Reader) or in Excel.

## **PDF exporter**

This exporter allows the generation of PDF documents that will display the waveforms of the samples from the input list, as well as some information about their audio characteristics such as bit depth, number of channels or sample rate. In addition spectra and spectrograms can be displayed. An optional listing page can be added to these sample catalogues.



The screenshot shows a configuration window for the PDF exporter. At the top, there is a 'Path' field with the value 'C:\Users\Nicolas\Documents\my samples.pdf' and a browse button. Below this, the window is divided into two main sections: 'Title Page' and 'Sample Pages'. The 'Title Page' section contains four text input fields: 'Title' (filled with 'My Samples Collection'), 'Subtitle 1' (filled with 'by John Doe'), 'Subtitle 2' (filled with 'Gneerated with AudioBot'), and 'Footer' (filled with '(this is a really cool program)'). There is also a checked checkbox for 'Display date'. The 'Sample Pages' section features a 'Document type' dropdown menu set to 'Analyses', a 'Samples per page' spinner set to '1', and a checked checkbox for 'Display listing'. Below these are three color selection options: 'Waveform colors' with a 'HotFire' color swatch, 'Spectrogram colors' with an 'EQ' color swatch, and 'Spectral bands colors' with an 'IcedBars' color swatch. At the bottom, there is an unchecked checkbox for 'Open PDF document after export'.

### *Export Path*

This is the path of the PDF file which will be generated.

### *Title*

This is the title of the document, as displayed on the first page.

### *Subtitle1*

This is an optional subtitle, displayed on the first page of the document.

### *Subtitle2*

This is a second optional subtitle, displayed on the first page of the document. It could be used to display the author of the samples for example.

#### *Footer*

Some text can be displayed at the bottom of the first page, for example copyright information.

#### *Display date*

If selected, the date and time of generation of the PDF document will be added to the front page.

#### *Display listing*

If selected, a listing of all the samples represented in the document will be included at the beginning of the PDF.

#### *Samples per page*

The number of samples per page can be set between one and nine. This parameter is not taken into account if the document type is Analyses as in this case each page contains only one sample and its various representations.

#### *Document type*

The document type can be either waveform only or full analysis, in which case there will be one sample per page only.

#### *Waveform colors*

This combo box lets you select the color scheme of the waveforms that will appear in the PDF document.

#### *Spectrogram colors*

This combo box lets you select the color scheme of the spectrograms that will appear in the PDF document.

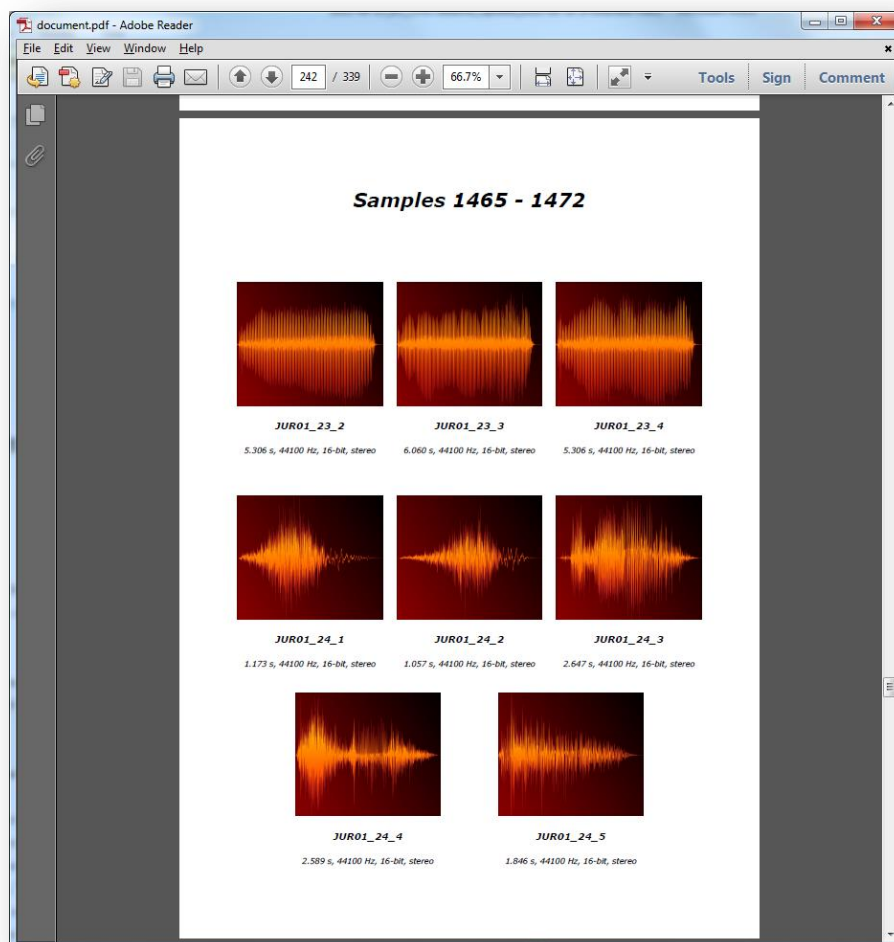
#### *Spectral bars colors*

This combo box lets you select the color scheme of the spectra that will appear in the PDF document.

#### *Open PDF document after export*

If checked, the PDF file generated will be open once the export finished.

Here is an example of result obtained with this exporter:



## Web exporter

The web exporter generates HTML pages that can be consulted offline or uploaded to a web site. More specifically, it creates an index.htm page and as many pageXXX.htm files as necessary to browse all the samples from the input list.

The index page has a title and a couple of subtitles and direct links to all the sample pages. Each of them displays a given number of samples, organized in columns. For each sample, a waveform is visible, as well as some information about the audio characteristics of the file and its size or last modification date. The name can be set as a download link and optional navigation links can be added to each page.

This exporter also uses the HTML5 audio tag to insert playback controls for each sample. At the time of the release of this version, the level of support of the audio tag still varies greatly from one browser to the other, with only Google Chrome offering a full implementation. It is therefore the only browser we support concerning this feature. Moreover, only Wave, Mp3 and Ogg files are usually supported for playback in the browsers.

The screenshot shows the AudioBot configuration window. At the top, the 'Path' field is set to 'C:\Users\JohnDoe\Documents'. Below this, the 'Index Page' section contains four text input fields: 'Title' (My best samples), 'Subtitle 1' (A web site by John Doe), 'Subtitle 2' (Created by AudioBot), and 'Footer' ((a very cool program)). There is a checkbox for 'Display date'. The 'Sample Pages' section includes two spinners for 'Samples per page' (20) and 'Columns per page' (2). It has four checkboxes: 'Allow download' (checked), 'Copy files locally' (checked), 'Add navigation links on top' (checked), and 'Add navigation links at bottom' (checked). A 'Waveform colors' preview shows a blue and red waveform. Below the preview is a dropdown menu currently set to 'CoralReef'. At the bottom, there is a checkbox for 'Open HTML pages in default browser after export'.

### *Export Path*

This is the path where all the HMTL files will be copied, and possibly the audio files as well (if the *Copy files locally* option is selected).

### *Title*

The title of the web site as it appears on the index page.

### *Subtitle1*

This is the first optional subtitle for the index page.

### *Subtitle2*

This is the second optional subtitle for the index page.

### *Footer*

This is the footer for the index page. It could be a copyright notice for example.

### *Display date*

If selected, the date and time of generation of the web site will be added to the index page.

### *Allow download*

If selected, the name of the sample will actually be a link to the audio file, thus allowing its download.

### *Samples per page*

This is the number of samples (from 1 to 500) listed per page.

*Columns per page*

This is the number of columns (from 1 to 10) in which the samples are organized on a page.

*Copy files locally*

This option can be used to copy all files, wherever they are from, into a flat folder structure (an “Audio” subfolder located at the root of the export path). This is especially useful to upload everything on a web site, in conjunction with the *Allow download* option, so that it can be browsed online.

*Add navigation links on top*

This self-explanatory option adds navigation links (top, first, previous, next and last) at the top of each page.

*Add navigation links at bottom*

This option does the same as the one above but for at bottom of the page.

*Waveform colors*

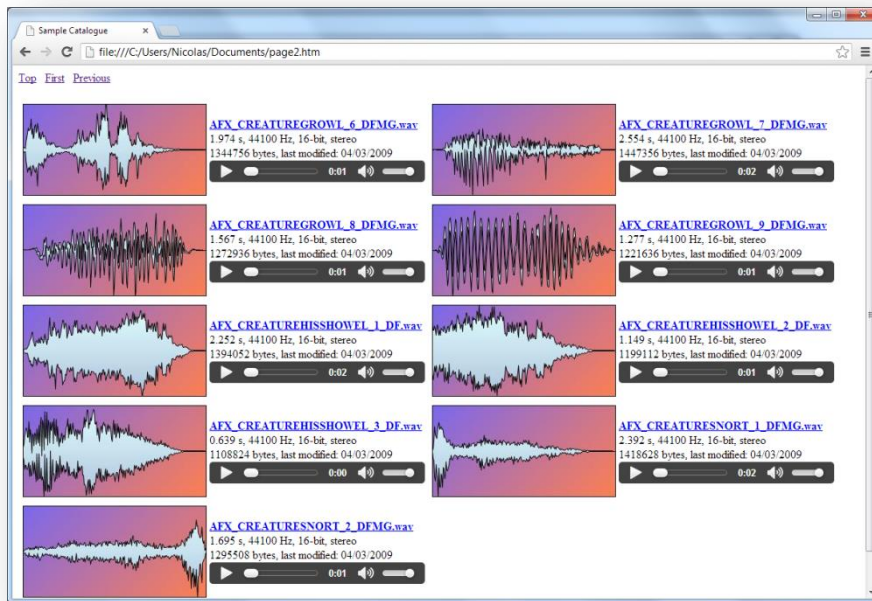
This combo box lets you select the color scheme of the waveforms that will appear on the web pages.

*Open in default browser after export*

If checked, the “index.htm” file generated will be open in your default browser once the export finished, and from there you will be able to browse the various HTML pages created, looking at waveforms and playing samples.

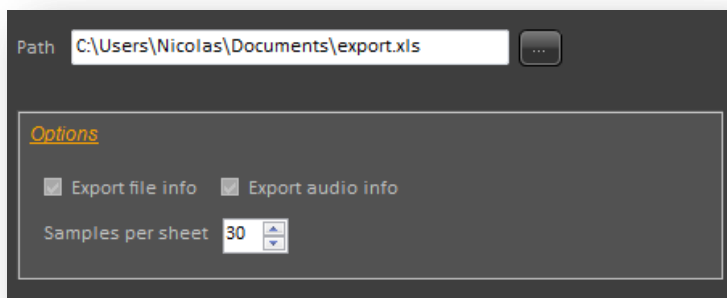
Here is an example of result obtained with this exporter:





## Excel exporter

This exporter generates an XML document compatible with Excel. This document contains one or more sheets including the list of the samples in the input list and their characteristics.



### *Export Path*

This is the path of the Excel file that will be generated.

### *Samples per sheet*

For a better readability, it is possible to limit the number of samples per Excel sheet.

### *Export audio info*

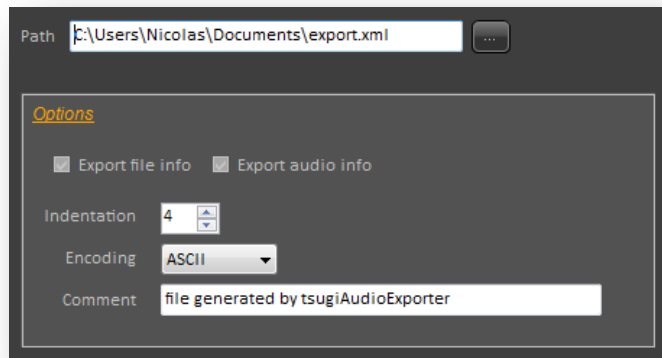
If selected, columns for the audio characteristics of the file will be added to sheet (e.g. bit depth, sample rate, duration, channels...).

### *Export file info*

If selected, columns for the file details will be added to sheet (e.g. size in bytes, last modification date).

## **XML exporter**

XML is a good choice to generate data that can be loaded by another tool later while staying readable by a programmer / designer.



### *Export Path*

This is the path of the XML file that will be generated.

### *Indentation*

This is the indentation of the file (typically set to 4).

### *Encoding*

This is the encoding of the file (ASCII or Unicode).

### *Comment*

This is the comment line which is at the top of the file, usually something about the version of the format or what tool has generated it.

### *Export audio info*

If selected, extra XML tags will be added containing the audio characteristics of the file (e.g. bit depth, sample rate, duration, channels...).

### *Export file info*

If selected, extra XML tags will be added containing the file details (e.g. size in bytes, last modification date).

## **Wwise exporter**

Wwise (WaveWorks Interactive Sound Engine) is a game audio middleware developed by AudioKinetic. This exporter generates work units, the basic objects handled by a Wwise project.

Path: T:\Test\AudioBotTestFolder\Wwise project\Test P...

**Work Unit**

Work Unit Name: Explosions

Container Name: Explosions container    Type: RandomContainer

**Sound Properties**

Default Voice Volume: -6.00 dB

Default Voice Pitch: 1200 cents

Default Voice Lowpass: 10 %

☒ Run Wwise authoring tool after export

#### *Export Path*

This is the path to the Wwise project we want to add work units to. The audio files from the input list will also be copied there, in the `\Originals\sfx\` subfolder.

#### *Work unit name*

This is the name of the work unit that will be created and added to the project. The file itself will have a `.wwu` extension.

#### *Container name*

This is the name of the container that will be created in the work unit, and which will reference all the files from the input list.

#### *Container type*

This is the type of the container: ActorMixer, Switch, Blend, Sequence and Random containers are supported, as well as simple sounds and sounds in a folder.

#### *Default voice volume*

This is the volume that will automatically be assigned to all the sounds in the work unit (between -96 dB and +12 dB).

#### *Default voice pitch*

This is the pitch variation that will automatically be assigned to all the sounds in the work unit (between -2400 cents and +2400 cents).

#### *Default voice lowpass*

This is the lowpass value that will automatically be assigned to all the sounds in the work unit. It represents the percentage of low-pass filtering that will be applied, where 0 means no low-pass filtering (signal unaffected) and 100 means maximal attenuation.

#### *Run Wwise authoring tool after export*

If selected, the Wwise authoring tool will be launched once the data has been exported. The path of this executable can be specified in the *Settings* window.

### **FMOD Designer exporter**

FMOD from Firelight Technologies is another game audio middleware. FMOD Designer (the authoring tool) uses a monolithic project file format, unlike Wwise. However, it has a function to merge two projects together. Therefore this exporter generates a full FMOD Designer- compatible project from the files in the input list and that project can then be merged with whatever project we want the files to be added to.

The screenshot shows the FMOD Designer exporter settings window. At the top, there is a 'Path' field with the value 'C:\FMOD projects\my project.fdp' and a browse button. Below this, the 'Sound Events' section contains three fields: 'Event Group name' with the value 'tsugi event group', 'Sound Bank name' with the value 'tsugi bank', and 'Objects created' with a dropdown menu set to 'Simple Events'. The 'Simple Events' section contains two columns of settings. The first column includes 'Spatial Mode' (2D), 'Volume' (-3.00 dB), 'Volume Rnd' (-2.00 dB), 'Pitch' (0.00 octaves), 'Pitch Rnd' (1.00 octaves), 'Priority' (128), 'Steal Priority' (10000), and 'Max Playback' (1). The second column includes 'Play Behaviour' (Steal oldest), 'Reverb Dry Level' (0 dB), 'Reverb Wet Level' (0 dB), 'Fade In Time' (1000 ms), 'Fade Out Time' (1000 ms), 'Spawn Intensity' (1), and 'Spawn Intensity Rnd' (1). At the bottom, there is a checkbox labeled 'Run FMOD Designer after export' which is checked.

#### *Export Path*

This is the path of the FMOD Designer project that will be generated.

#### *Event group name*

This is the name of the event group that will be created. It will contain the events that trigger the sounds coming from the input list.

#### *Sound bank name*

This is the name of the sound bank that will reference all the audio files from the input list.

*Objects created:*

This allows the user to select the type of objects created by the exporter.

If *"Simple Events"* is selected, we create FMOD "simple events" and the sound definitions are hidden. Default settings that can be specified in AudioBot for simple events include: 2D/3D mode, voice allocation / playback behaviour, volume and pitch values and randomization, reverb levels, as well as fade in and fade out times. For more detailed editing, or parameters which typically vary from one sound to the other, please use FMOD Designer.

If *"Create sound definitions"* is selected, we generate FMOD "sound definitions" - which will be visible - to use later to create multi track events (in that case the events themselves are not created automatically because there is no way to know what the user wants to do). The default settings that can be set in AudioBot for sound definitions include: play mode, volume and pitch values and their randomization, minimum and maximum spawn times, minimum and maximum delays, and randomization of the 3D position. Again, for more detailed editing, please use FMOD Designer.

Path

---

Sound Events

Event Group name

Sound Bank name

Objects created

---

Sound Definitions

Play Mode

Pitch Rnd Mode

Volume	<input type="text" value="0.00"/>	<input type="button" value="dB"/>	Min Spawn Time	<input type="text" value="0"/>	<input type="button" value="ms"/>
Volume Rnd	<input type="text" value="0.00"/>	<input type="button" value="dB"/>	Max Spawn Time	<input type="text" value="0"/>	<input type="button" value="ms"/>
Pitch	<input type="text" value="0.00"/>	<input type="button" value="octaves"/>	Min Delay Time	<input type="text" value="1000"/>	<input type="button" value="ms"/>
Pitch Rnd	<input type="text" value="2.00"/>	<input type="button" value="octaves"/>	Max Delay Time	<input type="text" value="2000"/>	<input type="button" value="ms"/>
Play count	<input type="text" value="0"/>		3D Pos Min Rnd	<input type="text" value="0.00"/>	<input type="button" value=""/>
Max Polyphony	<input type="text" value="1"/>		3D Pos Max Rnd	<input type="text" value="0.00"/>	<input type="button" value=""/>

☒ Run FMOD Designer after export

### *Run FMOD Designer after export*

If selected, the FMOD Designer tool will be launched once the data has been exported. The path of this executable can be specified in the *Settings* window.

### **ADX 2 exporter**

ADX 2 from CRI Middleware is a game audio middleware used by many companies in Japan. AudioBot can create brand new projects for ADX 2 or insert data in existing projects. The type of data generated can be selected, and various default settings specified.

Path: C:\Games\Platformer.atmcproj

**General**

Mode: Insert      Objects created: One Cue Per Sound

**Cues**

Cue sheet name: tsugi cue sheet      Cue type: Shuffle

Voice behaviour: Normal

Default Volume: 1.00      Default Pitch: 0 cents

Volume Random Range: 0.20      Pitch Random Range: 120 cents

**Tracks**

Default Volume: 1.00      Default Pitch: 0 cents

Volume Random Range: 0.80      Pitch Random Range: 0 cents

☒ Run AtomCraft tool once exported

### Export Path

This is the path of the ADX 2 - compatible project file (with an .atmcproj extension).

### Mode

This is the mode of operation of the exporter. It can either create a new project from scratch or insert data in an existing one. Please note that if the export path does not exist, a new project will be created even if the “Insert” mode is selected. Also, if the “New” mode is selected and a project already exists at the specified path, this project will be overwritten.

### Objects created

This combo box lets the user select what type of objects should be created. 3 modes are available:

- “Materials only”: in this mode, the waveforms are copied into the project and will appear in its Materials root folder, but no high level objects are created.
- “One cue per sound”: in this mode, in addition to adding the waveforms to the materials, a new cue sheet is created. For each sound file, a new cue that contains a single track is added to the cue sheet.
- “One cue for all sounds”: this mode is similar to the previous one, except that only one cue is added to the new cue sheet, and a new track is added to that cue for each sound file.

#### *Cue sheet name*

This is the name of the new cue sheet that will be added to the project.

#### *Cue type*

This parameter describes how the tracks contained in a cue will be played by the audio engine. All the options offered by ADX 2 are available: *Polyphonic, Sequential, Shuffle, Random, Random no repeat, Switch* and *Combo sequential*.

#### *Voice behavior*

This parameter describes the behavior of a voice when it becomes silent. All the options offered by ADX 2 are available: *Normal, Voice stop, Virtual voice, Virtual voice retrigger*.

#### *Cue default volume*

This is the default volume of a cue, from 0.0 to 1.0.

#### *Cue volume random range*

This is the range of random variation for the volume of a cue, from 0.0 to 1.0.

#### *Cue default pitch*

This is the default pitch of a cue, from -1200 to + 1200 cents.

#### *Cue pitch random range*

This is the range of random variation for the pitch of a cue, from 0 to 1200 cents.

#### *Track default volume*

This is the default volume of a track, from 0.0 to 1.0.

#### *Track volume random range*

This is the range of random variation for the volume of a track, from 0.0 to 1.0.

#### *Track default pitch*

This is the default pitch of a track, from -1200 to + 1200 cents.

#### *Track pitch random range*

This is the range of random variation for the pitch of a track, from 0 to 1200 cents.

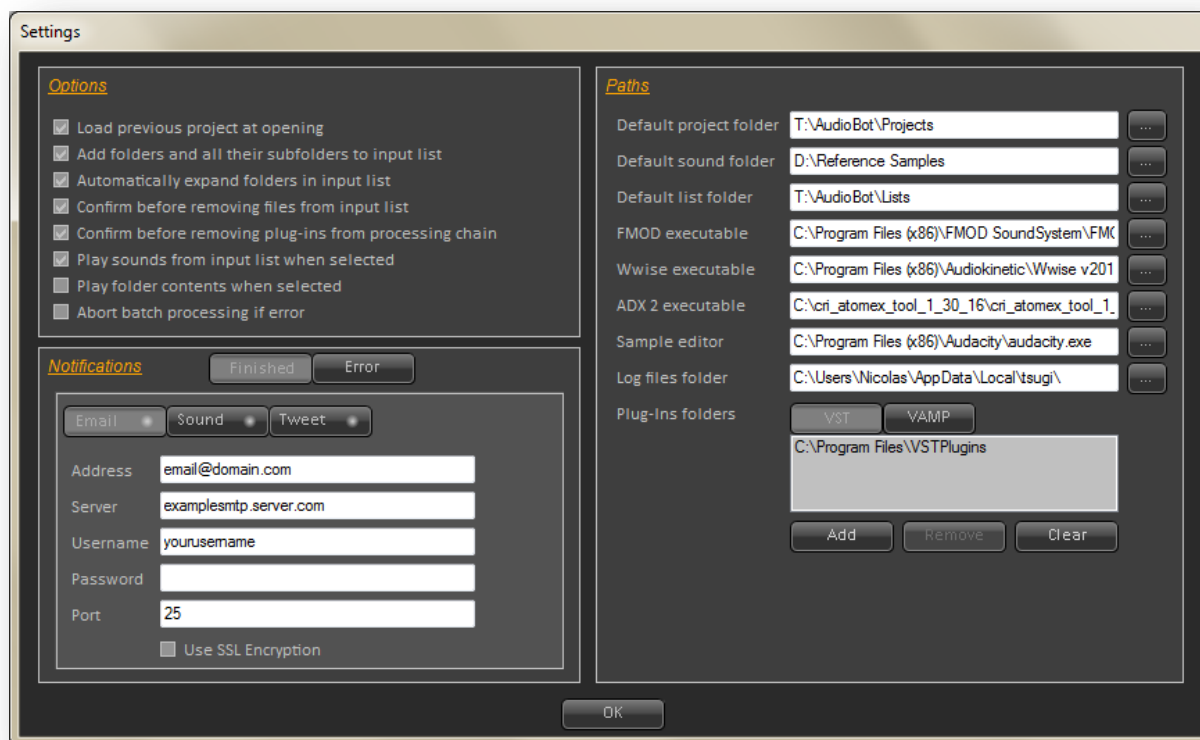
#### *Run AtomCraft tool once exported*

If selected, the AtomCraft tool will be launched once the data has been exported. The path of this executable can be specified in the *Settings* window.



## General Settings

These settings are project-independent and are saved in the Windows registry. Therefore any change applied will be remembered next time the application is started.



## Toggles

### *Load previous project at opening*

If selected, the last project open or saved will be automatically loaded when the application starts.

### *Add folders and all their subfolders to input list*

If selected, when adding a folder to the input list, all the samples contained in its subfolders will be added as well. Otherwise, only the samples contained in the top folder are included.

### *Automatically expand folders in input list*

If selected, any folder added to the input list (by loading a project, dropping a folder on the list etc...) will be automatically expanded and all its files will become visible.

### *Confirm before removing files from input list*

If selected, a confirmation request will appear when trying to remove files or folders from the input list, either with the *Remove* button or the “Delete” key shortcut.

### *Confirm before removing plug-ins from processing chain*

If selected, a confirmation request will appear when trying to remove plug-ins from the processing chain, either with the *Remove* button or the “Delete” key shortcut.

#### *Play sounds from input list when selected*

If selected, sample files will be played automatically when they are selected. To stop them, simply press on the space bar.

#### *Play folder contents when selected*

If selected, the contents of a folder will be automatically played when that folder is selected. Each time the play button is pressed, a new sample from that folder will be played. The order in which the samples are played depends on the playing mode selected: sequential, shuffle or random.

#### *Abort batch processing if error*

If selected, the batch processing will stop when an error occurs, for example if a file cannot be read, or an external plug-in cannot process the data.

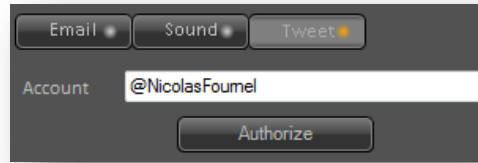
### **Notifications**

Notifications can be generated when the batch processing has ended (“Finished” notification) or when an error occurred (“Error” notification). To notify the user, the application can do three things:

- Send an email. In this case, the email address and the server must be specified by the user as well as a username, password, port and if SSL encryption must be used. The application will check that a valid email is entered. This is the most informative way to get a notification as the body of the email will contain a detailed description of the error.

A screenshot of a notification settings dialog box. At the top, there are three buttons: "Email" (highlighted with a yellow star), "Sound", and "Tweet". Below these are several input fields: "Address" with the value "john.doe@domain.com", "Server" with "smtpout.domain.com", "Username" with "JohnDoe", "Password" with a masked field of asterisks, and "Port" with "25". At the bottom, there is a checkbox labeled "Use SSL Encryption" which is checked.

- Send a tweet to an account specified by the user (the application will check that a valid account name is given). The first time, the *OAuth2* authorization process will have to be followed to allow the application to access your twitter account. This is done by pressing “*Authorize*” button. Because of the limited length of a tweet (140 characters), no details about the error will be given.



- Play a sound file. By default the Windows sound “Media\Windows Notify.wav” is selected, but any other wave file can be chosen.

One or several of these actions can be triggered simultaneously (e.g. the application could send an email and play a sound at the same time when an error occurs). To activate or deactivate notification actions, simply click on the little light on their button. When it is orange, the action is activated, when it is gray, it is off.

## **Paths**

### *Default project folder*

This is the folder where the application will first look for its project files (extension .bpp as Batch Processor Projects). By default, it is set to the “My Documents” folder.

### *Default sound folder*

This is the folder where the application will first look to add new sound files (e.g. wave, aiff, mp3, Ogg Vorbis) to the input list. By default, it is set to the “My Documents” folder.

### *Default list folder*

This is the folder where the application will first look to load or save file lists, including playlists such as .m3u and .wpl files. By default, it is set to the “My Documents” folder.

### *Wwise executable*

This is the path to the Wwise authoring tool. It will be used while exporting data towards Wwise. The application will try to locate and select the most recently installed version of Wwise present on the machine and set this path accordingly.

### *FMOD executable*

This is the path to the FMOD Designer tool. It will be used when exporting data towards FMOD. By default, the application will check if FMOD designer is present at the usual path: c:\Program Files (x86)\FMOD SoundSystem\FMOD Designer\fmod\_designer.exe

### *ADX 2 executable*

This is the path to the AtomCraft tool for the ADX 2 game audio middleware from CRI Middleware. This tool will be started automatically if you select the ADX 2 exporter as an output and check the "Run AtomCraft tool once exported" option.

### *Sample editor*

This is the path to your favorite sample editor, such as Sound Forge, Audacity etc... By default, no path is set here. The sample editor will be started when the user double-clicks on a sound file in the input list, or presses the “Edit” button, or hits the “Enter” key.

### *Log files folder*

This is the path to the folder where the log files generated by AudioBot will be saved. The name of a log file itself is built by concatenating the name of the project and a time stamp, followed by the “.log” extension. By specifying a folder rather than a specific file name, this system makes it possible to keep up a history of log files for a project. It is also possible for example to integrate AudioBot in a game audio pipeline and to run it on different types of assets (dialogue, music, sound effects...) with different settings. In that case, a separate log file will be generated for each type of assets (or each game level etc...).

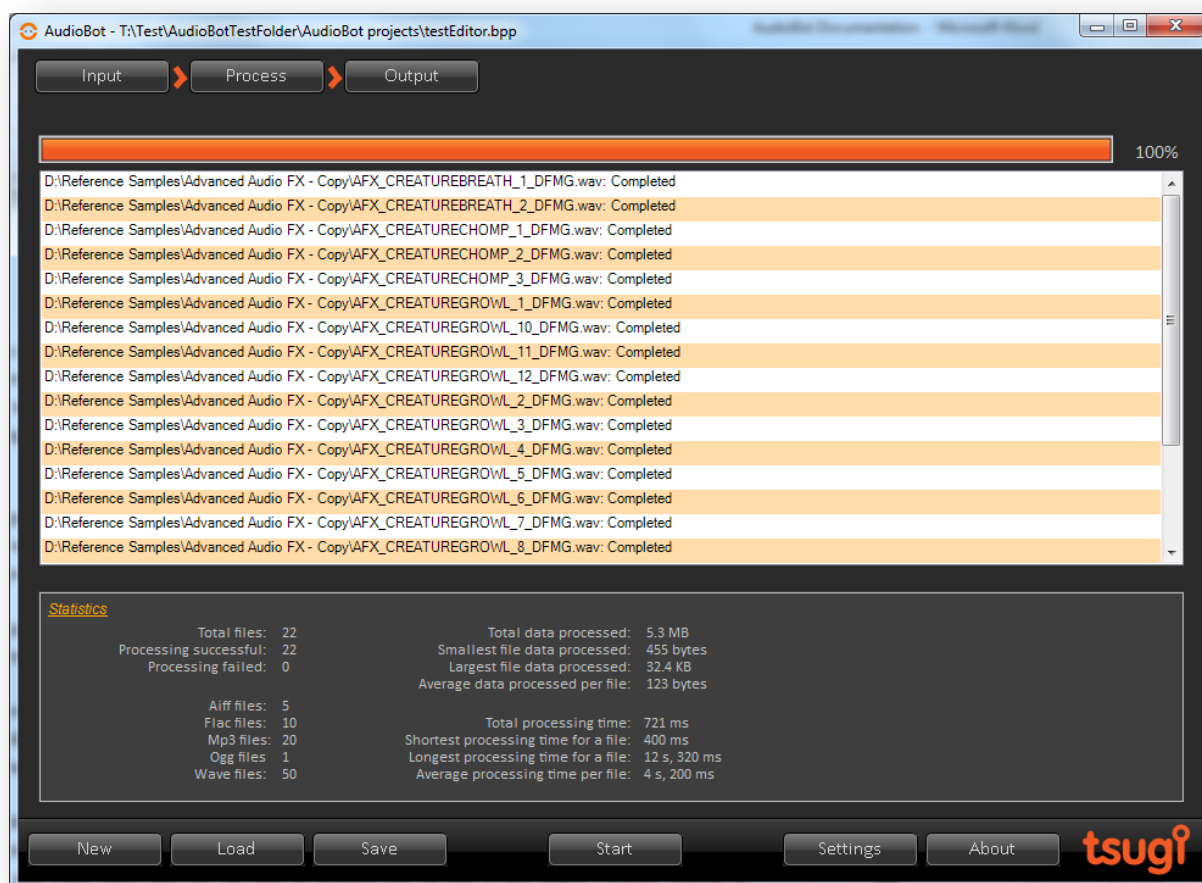
### *Plug-ins folders*

This section allows you to set the folders where the VST and Vamp plug-ins can be found. To add a folder, simply click on the “Add” button and browse to it. Folders can be removed by selecting them and pressing the “Remove” button. The whole list of folders can also be removed by pressing “Clear”. There are two lists of folders, one for VST plug-ins and one for Vamp plug-ins. You can switch between the two by pressing on the tab buttons at the top of the folders list. The first time it is run, the application will try to locate appropriate folders by testing some common paths such as C:\VST, C:\Program Files\VST plugins etc... If it does not succeed, the lists will be left blank. Once the folders have been selected, the application will look for plug-ins in the specified folders but also in any of their subfolders. However this will only happen the next time the program is started. Please note that AudioBot is a 64-bit tool and therefore needs to use plug-ins compiled for 64-bit.

## Batch Processing

The batch processing operation is triggered by pressing on the “Start” button. It is multithreaded and several files will be processed simultaneously. The number of threads to use is not set by the user as the program will use some heuristics to choose the best and most optimized multithreading settings.

A progress bar is displayed and the progress of each sound is displayed. The batch processing can be interrupted at any moment by pressing “Stop”. Otherwise, it will stop by itself when all files from the input list have been processed. If some folders in the input list are watched, the batch processing operation never ends by itself. Once all the files are processed accordingly to the user settings, the batch processor will be in standby mode, waiting for anything to happen in the folders it is watching. The user has to explicitly press the “Stop” button to end the operation.



Once the batch processing is finished a report (PDF, HTML or Excel) can be generated that will contain statistics about the data processed, the errors encountered etc... Some of this data is also displayed during the batch processing, in particular:

- The total number of files
- The number of files successfully processed
- The number of files whose processing failed

- The number of AIFF files
- The number of FLAC files
- The number of MP3 files
- The number of OGG files
- The number of WAVE files
- The total batch processing time
- The average processing time per file
- The shortest processing time for a file
- The longest processing time for a file
- The total data processed
- The average data processed per file
- The smallest file processed
- The largest file processed

### **Pausing and resuming a batch processing session**

When you stop a batch whilst it is in progress, AudioBot generates a record of the current state of the batch. The next time you press the “*Start*” button, you will be given the option to pick up the batch from its previous position.

Please note that:

- If you have closed and re-opened the application since aborting a batch, the batch resume functionality is still available.
- If you have opened and run other batch projects completely since aborting a batch, the batch resume function is still available for the aborted batch.
- If the project settings have changed since aborting a batch, the saved state for the batch is invalidated and the resume option will no longer be offered.
- The batch 'state' is stored as a '.lastbatch' file in the following folder:  
`<WindowsDrive>:\Users\<YourUsername>\AppData\Local\tsugi\AudioBot"`

### **Command line version**

A command line version of AudioBot is available. It is named AudioBotCLI.exe and is located in the same folder than the GUI-based executable (that is, if you didn't change the folder during installation, in C:\Program Files\Tsugi\AudioBot). It makes it possible to integrate AudioBot with your audio production pipeline, for example when preparing and packaging sound assets for a video game.

The command line version can read the same .bpp project files than the GUI version and also share the same general program settings, stored in the registry.

The following line will start the batch processing accordingly to settings (list of files, processing chain etc...) in the project file named Game.bpp.

AudioBotCLI.exe D:\Data\Batch\Game.bpp

## **Support**

Support is available via email at [audiobot.support@tsugi-studio.com](mailto:audiobot.support@tsugi-studio.com).

Information about new versions and updates will be posted on our web site: [www.tsugi-studio.com](http://www.tsugi-studio.com).

Alternatively you can also follow us on Twitter to get the latest news:

@tsugistudio for news about our company in English.

@tsugistudioJP for news about our company in Japanese.

@tsugiAudioBot for news about AudioBot.