

AVS4YOU Programs Help



AVS Audio Editor

www.avs4you.com

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Resources

Documentation for your AVS4YOU software is available in a variety of formats:

In-product (.chm-file) and Online Help

To reduce the size of the downloaded software installation files the in-product help was excluded from the installation although you can always download it from our web-site for your convenience. Please, visit AVS4YOU web-site at <http://onlinehelp.avs4you.com/index.aspx> to download the latest available version of the help executable, run it and install into the AVS4YOU programs folder. After that you will be able to use it through the **Help** menu of the installed AVS4YOU software.

Online Help include all the content from the In-product help file and updates and links to additional instructional content available on the web. You can find the **Online Help** at our web-site - <http://onlinehelp.avs4you.com/index.aspx>. Please note, that the most complete and up-to-date version of AVS4YOU programs help is always on the web.

PDF Documentation

The offline help is also available as a pdf-file that is optimized for printing. All PDF help files are available for download at the programs pages at AVS4YOU web-site (both <http://www.avs4you.com/index.aspx> and <http://onlinehelp.avs4you.com/index.aspx>). To be able to read and print AVS4YOU PDF help files you will need to have a PDF reading program installed.

User Guides

You have access to a wide variety of resources that help you make the most of your AVS4YOU software. The step-by-step user guides will be of help not only to the novice users but also to the users that face a certain task to be performed and look for a way to do it. Please, visit our **User Guides** section of AVS4YOU web-site at <http://www.avs4you.com/Guides/index.aspx> to read the detailed instructions for various software and tasks

Technical Support

Visit the **AVS4YOU Support** web-site at <http://support.avs4you.com> to ask your questions concerning AVS4YOU software installation, registration and use. Feel free to also use our e-mail address support@avs4you.com.

Downloads

Visit the **Downloads** section - <http://www.avs4you.com/downloads.aspx> - of our web-site to find free updates, tryouts, and other useful software. We constantly update the software, new versions of the most popular programs and new software are also frequently released.

Contact Us

If you have any comments, suggestions or questions regarding **AVS4YOU** programs or if you have a new feature that you feel can be added to improve our product, please feel free to contact us.

When you register your product, you may be entitled to technical support.

General information:	info@avs4you.com
Technical support:	support@avs4you.com
Sales:	sales@avs4you.com
Help and other documentation:	help@avs4you.com

Technical Support

AVS4YOU programs do not require any professional knowledge. If you experience any problem or have a question, please refer to the **AVS4YOU Programs Help**. If you cannot find the solution, please contact our support staff.



Note: only registered users receive technical support.

AVS4YOU staff provides several forms of automated customer support:

- **AVS4YOU Support System**

You can use the **Support Form** on our site to ask your questions.

- **E-mail Support**

You can also submit your technical questions and problems via e-mail to support@avs4you.com.



Note: for more effective and quick resolving of the difficulties we will need the following information:

- Name and e-mail address used for registration
- System parameters (CPU, hard drive space available, etc.)
- Operating System
- The information about the capture, video or audio devices, disc drives connected to your computer (manufacturer and model)
- Detailed step by step describing of your action

Please do **NOT** attach any other files to your e-mail message unless specifically requested by AVS4YOU.com support staff.

Overview

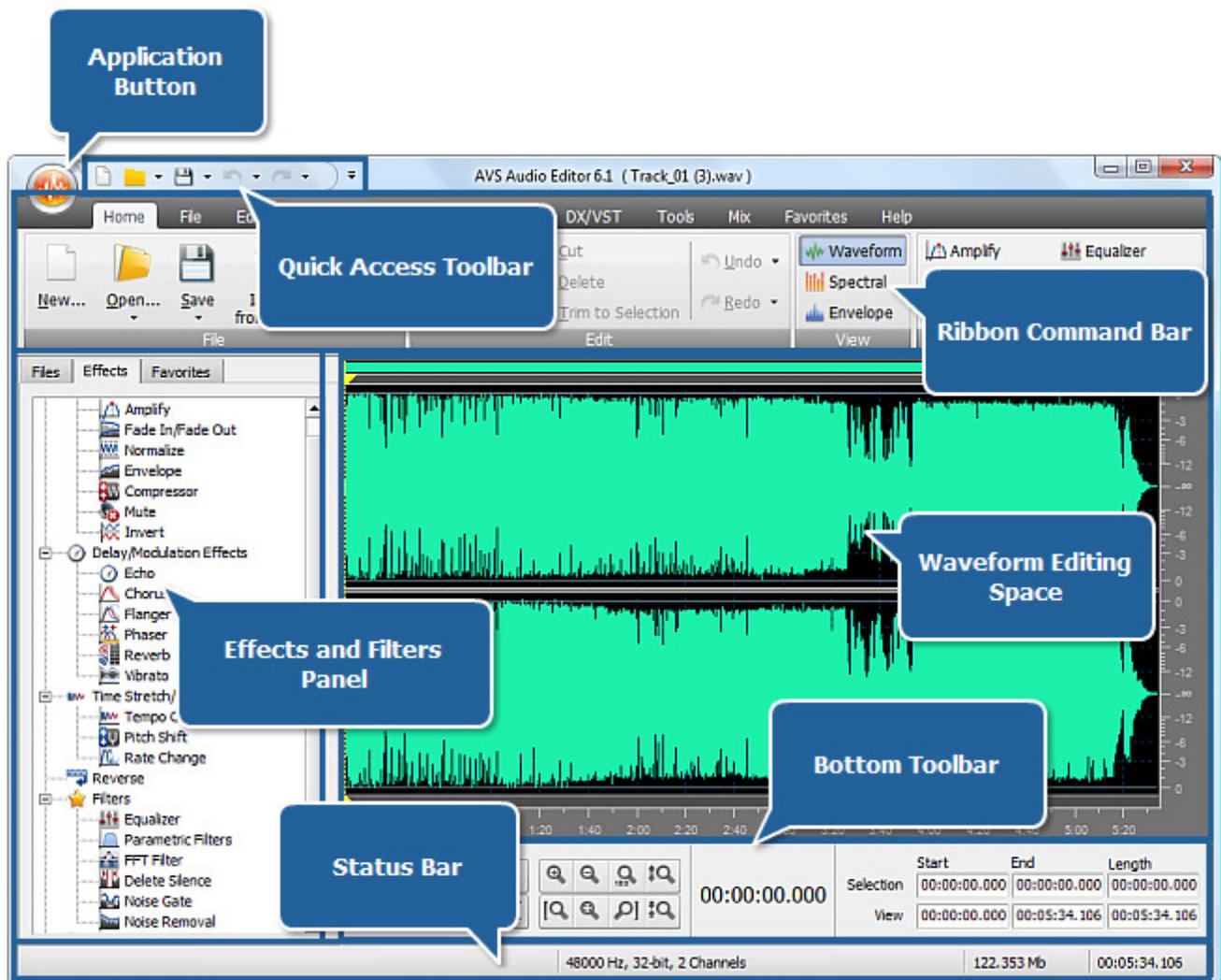
AVS Audio Editor is a feature-rich audio production software that combines advanced audio editing features with an easy-to-use **ribbon-based interface**, which makes it suitable for both professionals and amateurs.

AVS Audio Editor allows you to perform the following operations:

- **record audio** from any input and import it to the program for further editing;
- **mix** several audio tracks in one;
- **apply effects and filters** to your audio - there are over 20 available including delay, chorus, fade in/out effects, low/high /bandpass filters etc.;
- **generate sounds** - the software supports generating chirp, noise, DTMF signals;
- **convert** created audio to a wide range of modern audio formats;
- **customize AVS Audio Editor settings** to meet your specific needs.

To start **AVS Audio Editor** go to **Start** menu and choose **All Programs -> AVS4YOU -> Audio -> AVS Audio Editor**.

Main Window

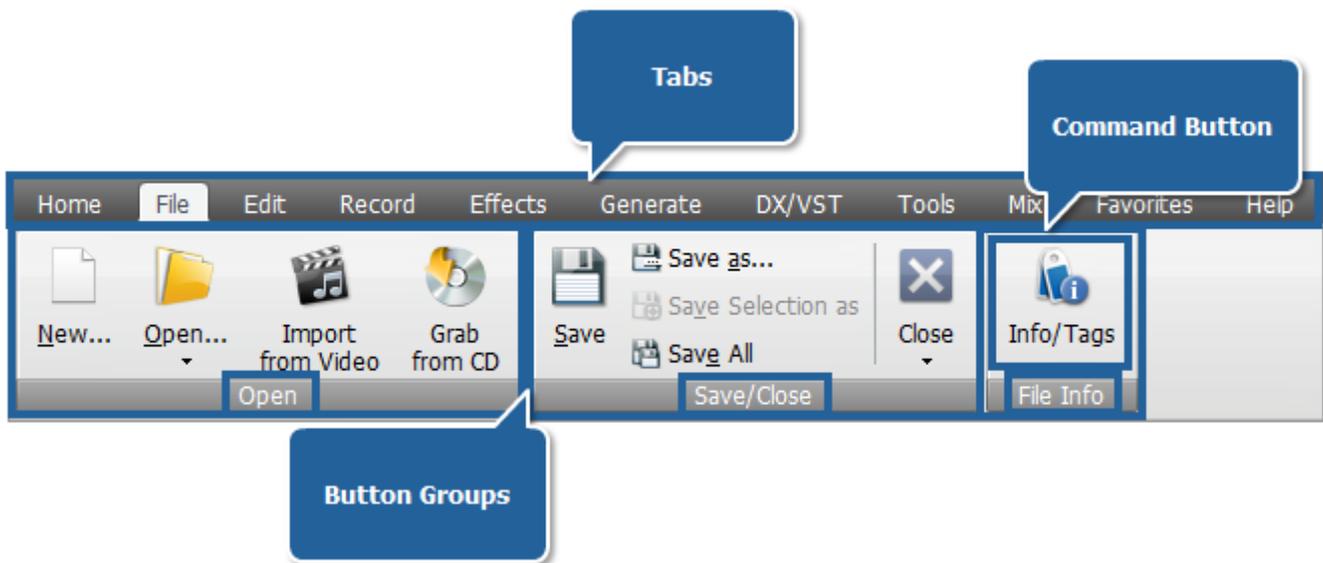


AVS Audio Editor main window comprises the following parts:

- **Ribbon Command Bar**
- **Quick Access Toolbar**
- **Application Button**
- **Effects and Filters Panel**
- **Waveform Editing Space**
- **Bottom Toolbar**
- **Status Bar**

Ribbon Command Bar

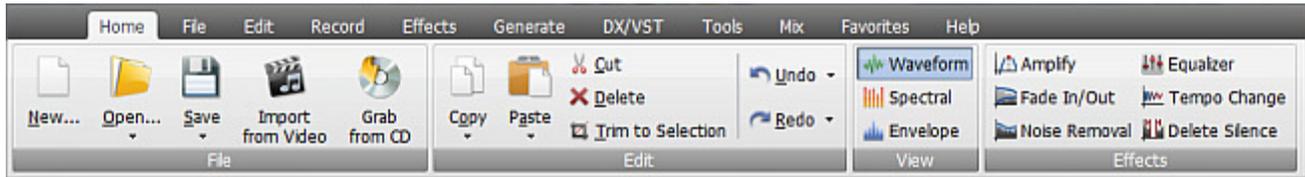
The **Ribbon Command Bar** located at the top of the **AVS Audio Editor Main Window** offers a better visibility of the software functional capabilities which is achieved through sets of intuitive command buttons. Related command buttons are organized in button groups. Related button groups are organized in tabs.



AVS Audio Editor ribbon command bar comprises the following tabs:

- **Home Tab**
- **File Tab**
- **Edit Tab**
- **Effects Tab**
- **Generate Tab**
- **DX/VST Tab**
- **Tools Tab**
- **Mix Tab**
- **Favorites Tab**
- **Help Tab**

Home Tab

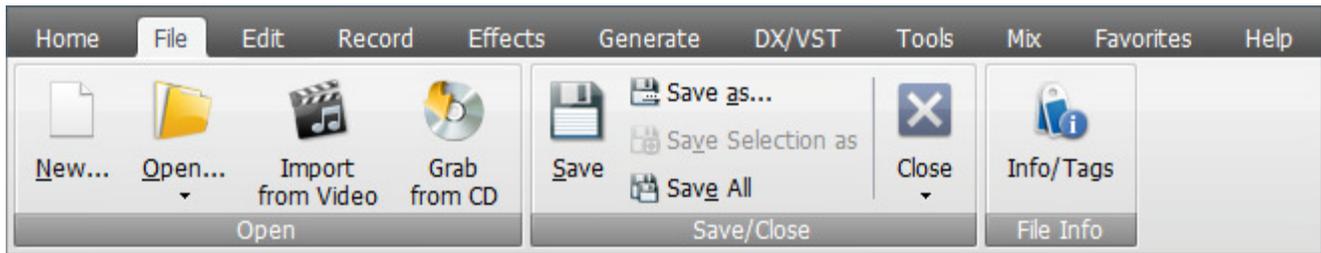


Group	Command	Description
File	 New	Use it to create a new audio file.
	 Open...	Use it to open an existing audio file. When you load a file for editing, AVS Audio Editor converts the audio file type to its own internal temporary file type for faster editing and better handling of larger file sizes. Clicking the arrow below this button will additionally display the Open as... , Open and Append... and Recent Files options.
	 Open as...	Use it to open an existing audio file as if it were a file of some other format. See the Open as.. section for more detail on it.
	 Open and Append...	Use it to open an existing audio file and append it to the end of the previously opened file. The newly created file will consist of two opened files at once, the second one starting where the first one ends.
	Recent Files	Use it to display the list of 5 audio files most recently loaded into AVS Audio Editor .
	 Save	Use it to save the active audio file with its current file name and location. Clicking the arrow below this button will additionally display the Save as... , Save Selection as... and Save All options.
	 Save as...	Use it to save the active audio file with a different file name and/or location, in a different file format.
	 Save Selection as	Use it to save just the highlighted selection to a file.
	 Save All	Use it to save all the changes to all the opened edited files.
	 Import from Video	Use it to import the audio track(s) of a video file into the program.
 Grab from CD	Use it to grab an audio CD and add the grabbed tracks to the program.	
Edit	 Copy	Use it to copy the selection into the program internal clipboard. It is used when you work with one file within one copy of AVS Audio Editor . Clicking the arrow below this button will additionally display the Copy to New... and Copy to Win Clipboard options.

	 Copy to New...	Use it to copy the selection or the whole audio file into a new one. A new file will be automatically created and opened.
	 Copy to Win Clipboard	Use it to copy the selection into the windows clipboard so that it would be available to other opened programs. It is used when you work with two or more copies of AVS Audio Editor and want to copy/paste from one program to another.
	 Paste	Use it to insert the contents of the internal clipboard at the insertion point or to replace any selection. Clicking the arrow below this button will additionally display the Paste from Clipboard... option.
	 Paste from Clipboard...	Use it to insert the contents of the windows clipboard at the insertion point or to replace any selection.
	 Cut	Use it to remove the selection from the active file and place it to the program internal clipboard.
	 Delete	Use it to remove the current selection. The deleted portion is not copied to the clipboard, and can only be retrieved through Undo .
	 Trim to Selection	Use it to delete everything except the selected portion (the exact opposite of Delete).
	 Undo	Use it to reverse the last command, such as deletion, transforms, etc. If this option is not available, it means there is no action to undo. AVS Audio Editor allows virtually unlimited (limited only by hard drive space) levels of Undo .
	 Redo	Use it to repeat your latest undone action.
View	 Waveform	Use it to choose a Waveform View mode for viewing data. It displays audio data in the familiar sound wave format, where y-axis (vertical) represents amplitude and x-axis (horizontal) represents time.
	 Spectral	Use it to choose a Spectral View mode for viewing data. It displays a waveform by its frequency components, where x-axis (horizontal) is time, and y-axis (vertical) is frequency. This allows you to analyze your audio data to see which frequencies are most prevalent.
	 Envelope	Use it to choose a Envelope View mode for viewing data. It displays audio data in the familiar sound wave format, where y-axis (vertical) represents amplitude and x-axis (horizontal) represents time. It is the same as the Waveform View but its boundaries are smoothed.
Effects	 Amplify	Left-click it to apply the Amplify effect to your audio.
	 Fade in/out	Left-click it to apply Fade in or Fade out effect to audio.
	 Noise Removal	Left-click it to apply the Noise Removal filter to your audio.

 Equalizer	Left-click it to apply the Equalizer effect to your audio.
 Tempo Change	Left-click it to apply the Tempo Change effect to your audio.
 Delete Silence	Left-click it to apply the Delete Silence filter to your audio.

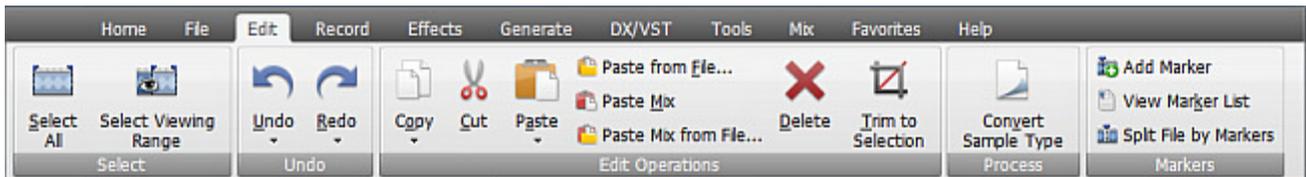
File Tab



Group	Command	Description
Open	 New	Use it to create a new audio file.
	 Open...	Use it to open an existing audio file. When you load a file for editing, AVS Audio Editor converts the audio file type to its own internal temporary file type for faster editing and better handling of larger file sizes. Clicking the arrow below this button will additionally display the Open as... , Open and Append... and Recent Files options.
	 Open as...	Use it to open an existing audio file as if it were a file of some other format. See the Open as.. section for more detail on it.
	 Open and Append...	Use it to open an existing audio file and append it to the end of the previously opened file. The newly created file will consist of two opened files at once, the second one starting where the first one ends.
	Recent Files	Use it to display the list of 5 audio files most recently loaded into AVS Audio Editor .
	 Import from Video	Use it to import the audio track(s) of a video file into the program.
	 Grab from CD	Use it to grab an audio CD and add the grabbed tracks to the program.
Save/Close	 Save	Use it to save the active audio file with its current file name and location.
	 Save as...	Use it to save the active audio file with a different file name and/or location, in a different file format.
	 Save Selection as	Use it to save just the highlighted selection to a file.
	 Save All	Use it to save all the changes to all the opened edited files.
	 Close	Use it to close the currently opened edited file. Clicking the arrow below this button will additionally display the Close All option.

	 Close All	Use it to close all the opened edited files.
File Info	 Info/Tags	Use it to view the available audio file info and modify audio tags.

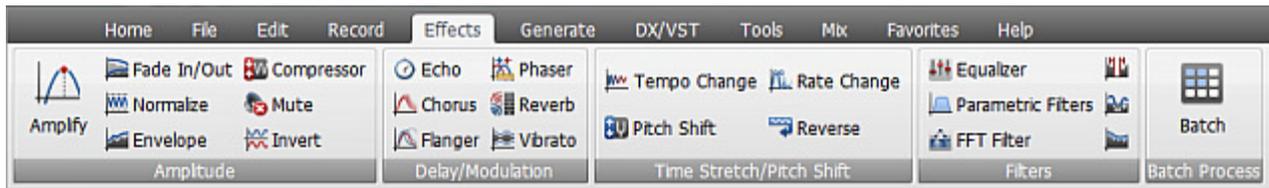
Edit Tab



Group	Command	Description
Select	 Select All	Use it to select the whole track.
	 Select Viewing Range	Use it to make the selection of the visible area of the audio track only (in case the waveform is zoomed in, only the area within the visible area will be selected).
Undo	 Undo	Use it to reverse the last command, such as deletion, transforms, etc. If this option is not available, it means there is no action to undo. AVS Audio Editor allows virtually unlimited (limited only by hard drive space) levels of Undo . Pressing the downward arrow under the Undo button will open the list of the commands and actions available.
	 Redo	Use it to repeat your latest undone action. Pressing the downward arrow under the Redo button will open the list of the commands and actions available.
Edit Operations	 Copy	Use it to copy the selection into the program internal clipboard. It is used when you work with one file within one copy of AVS Audio Editor . Clicking the arrow below this button will additionally display the Copy to New... and Copy to Win Clipboard options.
	 Copy to New...	Use it to copy the selection or the whole audio file into a new one. A new file will be automatically created and opened.
	 Copy to Win Clipboard	Use it to copy the selection into the windows clipboard so that it would be available to other opened programs. It is used when you work with two or more copies of AVS Audio Editor and want to copy/paste from one program to another.
	 Cut	Use it to remove the selection from the active file and place it to the program internal clipboard.
	 Paste	Use it to insert the contents of the internal clipboard at the insertion point or to replace any selection. Clicking the arrow below this button will additionally display the Paste from Clipboard... option.
	 Paste from	Use it to insert the contents of the windows clipboard at the insertion point or to replace any selection.

	Clipboard...	
	 Paste from File...	Use it to insert the contents of an audio file at the insertion point or to replace any selection.
	 Paste Mix...	Use it to insert the contents of the internal clipboard at the insertion point mixing the audio tracks.
	 Paste Mix from File...	Use it to insert the contents of the audio file at the insertion point mixing the audio tracks.
	 Delete	Use it to remove the current selection. The deleted portion is not copied to the clipboard, and can only be retrieved through Undo .
	 Trim to Selection	Use it to delete everything except the selected portion (the exact opposite of Delete).
Process	 Convert Sample Type	Use it to convert your opened audio to another format or change bitrate, sample rate, number of channels and so forth.
Markers	 Add Marker	Use it to add a marker to the currently edited audio.
	 View Marker List	Use it to open the Marker List window.
	 Split File by Markers	Use it to split currently edited audio file by markers.

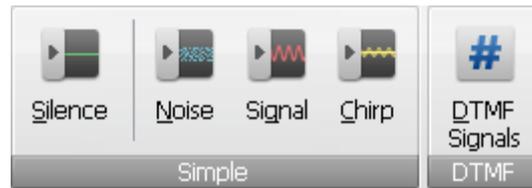
Effects Tab



Group	Command	Description
Amplitude	 Amplify	Click it to apply the Amplify effect to your audio.
	 Fade In/Out	Click it to apply Fade In or Fade Out effect to audio.
	 Normalize	Click it to apply the Normalize effect to your audio.
	 Envelope	Click it to apply the Envelope filter to your audio.
	 Mute	Click it to mute the selected audio portion.
	 Compressor	Click it to apply the Compressor effect to your audio.
	 Invert	Click it to apply the Invert effect to your audio.
Delay/Modulation	 Echo	Click it to apply the Echo effect to your audio.
	 Chorus	Click it to apply the Chorus effect to your audio.
	 Flanger	Click it to apply the Flanger effect to your audio.
	 Phaser	Click it to apply the Phaser effect to your audio.
	 Reverb	Click it to apply the Reverb effect to your audio.
	 Vibrato	Click it to apply the Vibrato effect to your audio.

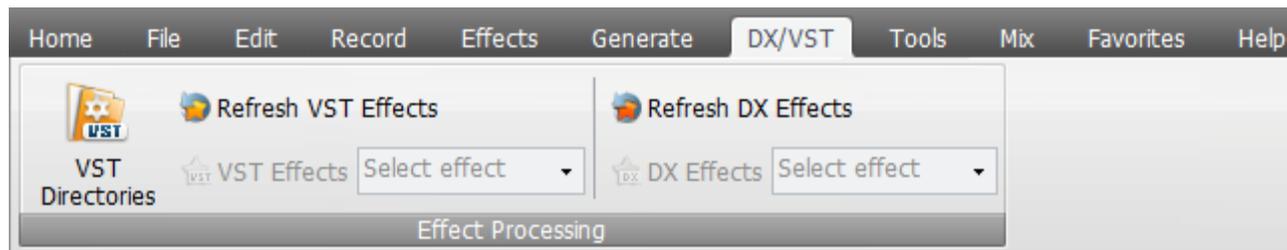
Time Stretch/Pitch Shift	 Tempo Change	Click it to apply the Tempo Change effect to your audio.
	 Pitch Shift	Click it to apply the Pitch Shift effect to your audio.
	 Rate Change	Click it to apply the Rate Change effect to your audio.
	 Reverse	Click it to apply the Reverse effect to your audio.
Filters	 Equalizer	Click it to apply the Equalizer filter to your audio.
	 Parametric Filters	Click it to apply the Parametric filters to your audio.
	 FFT Filter	Click it to apply the FFT filter to your audio.
	 Delete Silence	Click it to apply the Delete Silence filter to your audio.
	 Noise Gate	Click it to apply the Noise Gate filter to your audio.
	 Noise Removal	Click it to apply the Noise Removal filter to your audio.
Batch Process	 Batch	Click it to open the Batch Processing window.

Generate Tab



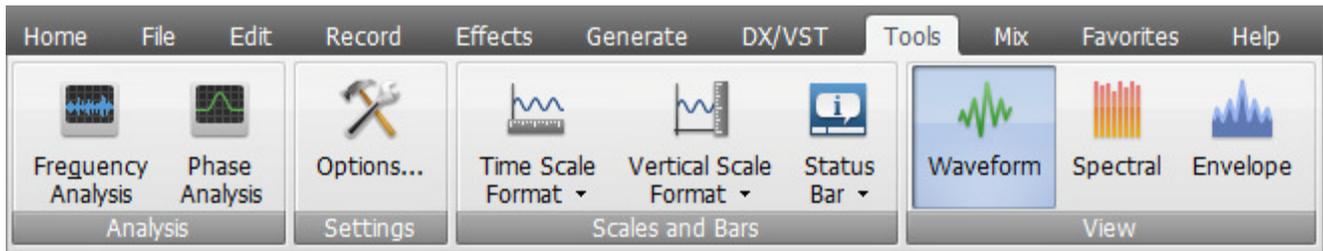
Group	Command	Description
Simple	 Silence	Use it to create a period of silence of a certain length within the audio.
	 Noise	Use it to create a noise signal of a certain type and length within the audio.
	 Signal	Use it to create a signal with certain parameters within the audio.
	 Chirp	Use it to create a chirp signal with certain parameters within the audio.
DTMF	 DTMF Signals	Use it to create DTMF signals with certain parameters within the audio.

DX/VST Tab



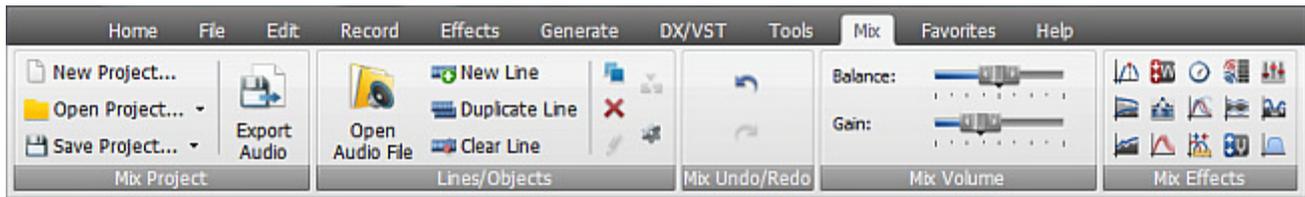
Group	Command	Description
Effect Processing	 VST Directories	Use it to add or remove the VST effects loaded to AVS Audio Editor and available for audio editing.
	 Refresh VST Effects	Use it to refresh the list of VST effects in case some effects became available.
	 VST Effects	Use it to list VST effects installed on your system and apply them to the audio.
	 Refresh DX List	Use it to refresh the list of DirectX filters in case some effects became available.
	 DX Effects	Use it to list DirectX filters installed on your system and apply them to the audio.

Tools Tab



Group	Command	Description
Analysis	 Frequency Analysis	Use it to show or hide Frequency Analysis Window . The Frequency Analysis Window contains a graph of the frequencies at the insertion point or at the center of a selection.
	 Phase Analysis	Use it to show or hide Phase Analysis Window .
Settings	 Options...	Use it to view and edit the program settings.
Scales and Bars	 Time Scale Format	Use it to change the format of the horizontal scale which represents the timeline of the audio file. The possible values include: Decimal (mm:ss.ddd) , Samples , Frames 60 fps , Frames 59,94 fps , Frames 50 fps , Frames 30 fps , Frames 29.97 fps , Frames 25 fps and Frames 24 fps .
	 Vertical Scale Format	Use it to change the format of the vertical scale which represents the changes of the audio in the file within time. The possible values include: Sample Values , Percentage and Decibels .
	 Status Bar	Use it to show or hide the elements of the Status Bar . The possible items include: Sample Format , File Size and Duration (mm:ss.ddd) .
View	 Waveform	Use it to choose a Waveform View mode for viewing data. It displays audio data in the familiar sound wave format, where y-axis (vertical) represents amplitude and x-axis (horizontal) represents time.
	 Spectral	Use it to choose a Spectral View mode for viewing data. It displays a waveform by its frequency components, where x-axis (horizontal) is time, and y-axis (vertical) is frequency. This allows you to analyze your audio data to see which frequencies are most prevalent.
	 Envelope	Use it to choose a Envelope View mode for viewing data. It displays audio data in the familiar sound wave format, where y-axis (vertical) represents amplitude and x-axis (horizontal) represents time. It is the same as the Waveform View but its boundaries are smoothed.

Mix Tab



Group	Command	Description
Mix Project	 New Project	Use it to create a new audio project.
	 Open Project...	Use it to open an existing mix project. Clicking the arrow below this button will additionally display the Recent Projects option.
	Recent Projects	Use it to view a list of the projects that have been recently loaded into AVS Audio Editor..
	 Save Project	Use it to save the currently edited mix project with its current file name and location.
	 Save Project as...	Use it to save the currently edited mix project with a different name and/or location. The project will be saved as a project file with the .mix extension.
	 Export Audio	Use it to export audio from the mix project and save it to one of the available audio formats.
Lines/Objects	 Open Audio File	Use it to load an audio track into the mix working space. The selected file will be placed to the empty line at current cursor position.
	 New Line	Use it to add a new line to the mix working space.
	 Duplicate Line	Use it to duplicate the selected line in the mix working space.
	 Clear Line	Use it to clear the selected line in the mix working space.
	 Duplicate Object	Use it to duplicate the currently selected object in the mix working space. The object will be duplicated in the same line.
	 Delete Object	Use it to delete the currently selected object in the mix working space.

	 Edit Object	Use it to edit the currently selected object in the mix working space.
	 Split Object	Use it to split the currently selected object in the mix working space at current cursor position.
	 Object Properties	Use it to view the currently selected object properties.
Mix Undo/Redo	 Undo	Use it to cancel the last action performed in the mix working space.
	 Redo	Use it to repeat the last undone action performed in the mix working space.
Mix Effects	 Amplify	Left-click it to apply the Amplify effect to your audio.
	 Fade in/out	Left-click it to apply Fade in or Fade out effect to audio.
	 Envelope	Left-click it to apply the Envelope filter to your audio.
	 Compressor	Left-click it to apply the Compressor effect to your audio.
	 FFT Filter	Left-click it to apply the FFT Filter to your audio.
	 Chorus	Left-click it to apply the Chorus filter to your audio.
	 Echo	Left-click it to apply Echo effect to audio.
	 Flanger	Left-click it to apply the Noise Remover filter to your audio.
	 Phaser	Left-click it to apply the Phaser effect to your audio.
	 Reverb	Left-click it to apply the Time Stretch effect to your audio.
	 Vibrato	Left-click it to apply the Delete Silence filter to your audio.

 Pitch Shift	Left-click it to apply PitchShift effect to audio.
 Equalizer	Left-click it to apply the Equalizer filter to your audio.
 Noise Gate	Left-click it to apply the Noise Gate effect to your audio.
 Parametric Filters	Left-click it to apply Parametric Filters to your audio.

Help Tab



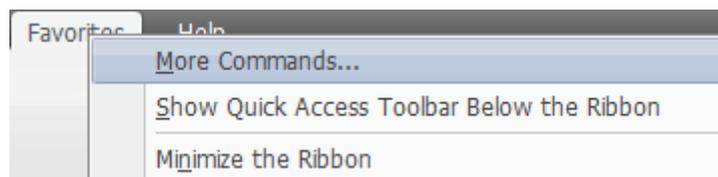
Group	Command	Description
General	 Help	Use it to open the Help File window.
	 About	Use it to open the window that shows the information about AVS Audio Editor .
	 Our Site	Use it to see the links to AVS4YOU.com on the web.

Favorites Tab

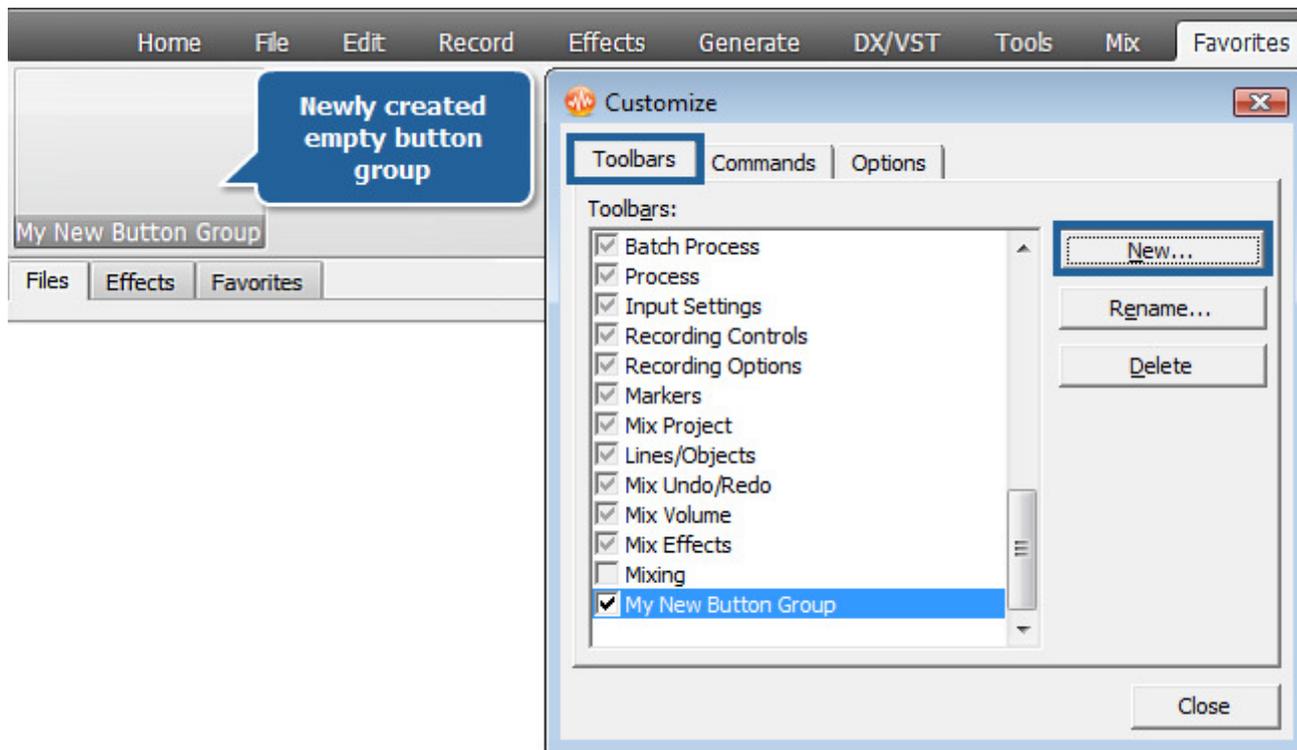
The **Ribbon Command Bar** located at the top of the **AVS Audio Editor** main window contains tabs with a set number of commands which are already grouped in a certain way. The **Favorites** tab has been designed specifically for you to display the most frequently used commands in a single place and group them according to your wish.

The **Favorites** tab is empty by default. To customize it (create **Button Groups** and add **Command Buttons**), do the following:

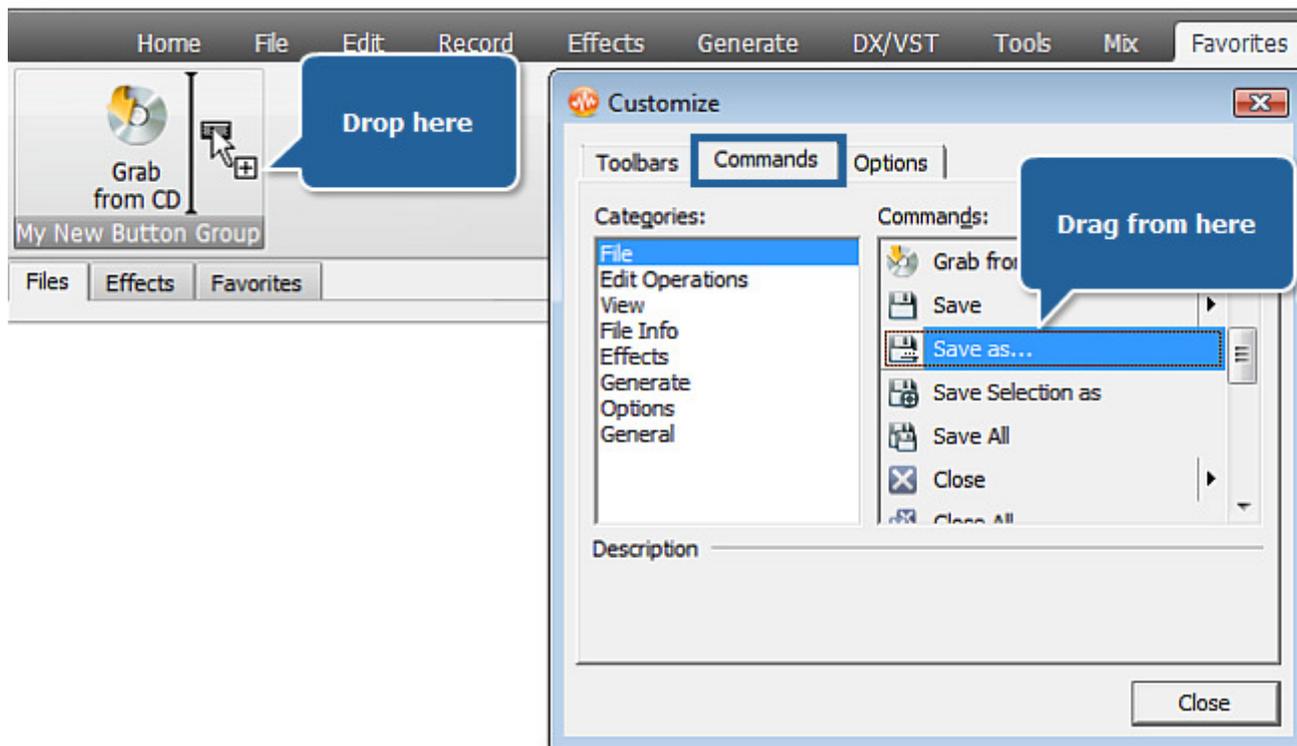
- Right-click the **Favorites** tab and select **More Commands...** to open the **Customize** window.



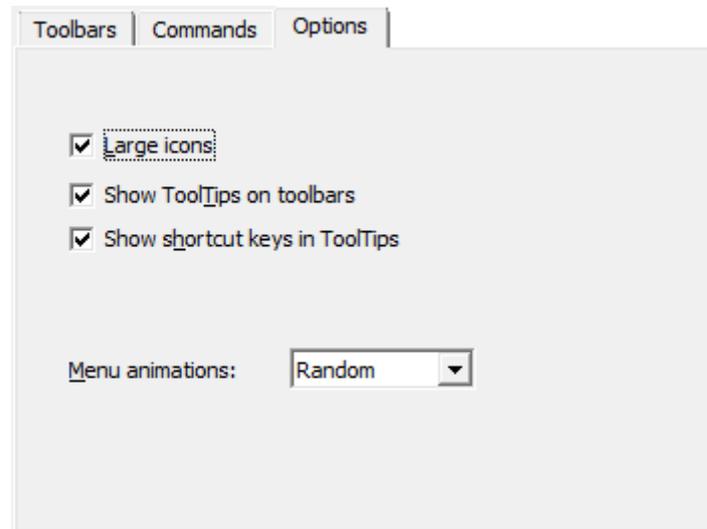
- On the **Toolbars** tab, click **New...** to create a new button group. Enter a name for your new button group and press **OK**. The new empty button group will appear on the **Favorites** tab:



- To add command buttons to your button group (toolbar), switch to the **Commands** tab. Select the desired command button, then drag and drop it within the button group you have just created.



Use the **Options** tab of the **Customize** window to set **Ribbon Command Bar** properties.



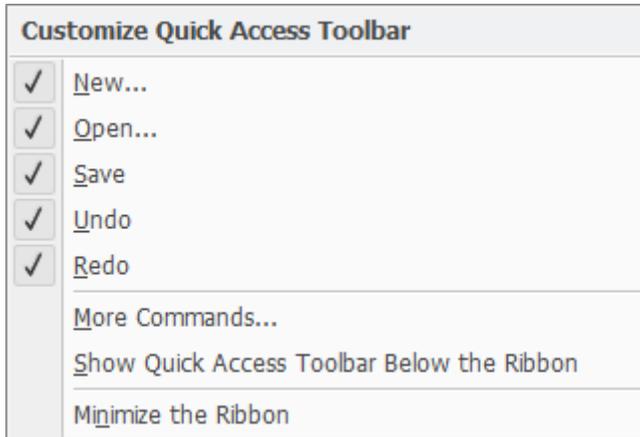
- Tick the **Large icons** option to show large icons on the ribbon command bar.
- Tick **Show tooltips on toolbars** to display tooltips (hints) when you place the cursor over ribbon command bar elements.
- Tick **Show shortcut keys in tooltips** to include shortcuts in tooltips (hints).

Quick Access Toolbar



Quick Access Toolbar located in the top left corner of the **AVS Audio Editor Main Window** is a customizable tool allowing you to get quick access to the most frequently used functions of the software.

Clicking the arrow  to the right of the toolbar will display the **Customize Quick Access Toolbar** menu (see image below).



Tick/untick the **Customize Quick Access Toolbar** menu elements to show/hide them on the toolbar.

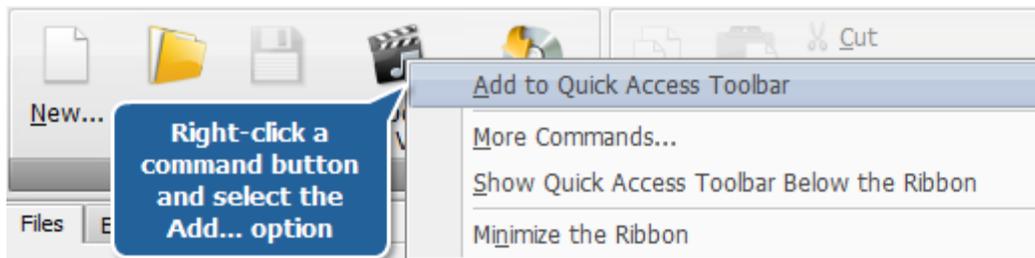
Click **More Commands...** and uncheck the **Quick Access Toolbar** box on the **Toolbars** tab to hide the quick access toolbar.

Tick the **Show Quick Access Toolbar below the Ribbon** option to show the toolbar below the ribbon.

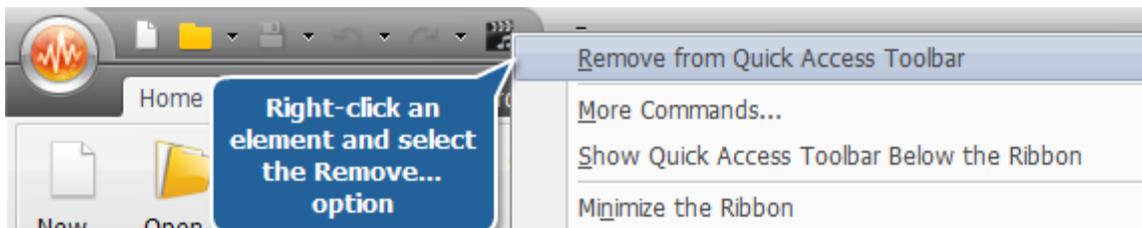
Tick **Minimize the Ribbon** to hide command buttons and button groups of the **Ribbon Command Bar** so that only tabs would be visible. Untick it to make the ribbon command bar fully visible again.

Adding/Removing Elements from Toolbar

You can add/remove elements from **Quick Access** toolbar. To add an element to the toolbar, right-click the desired **Command Button** on the ribbon command bar and select **Add to Quick Access Toolbar**:



To remove an element from **Quick Access** toolbar right-click it and select **Remove from Quick Access Toolbar**:



Application Button



The **Application** button located in the top left corner of the **AVS Audio Editor Main Window** is a part of the program interface that gets you to a special menu containing available operations with files (corresponds more or less to the **File** section of the classic menu).

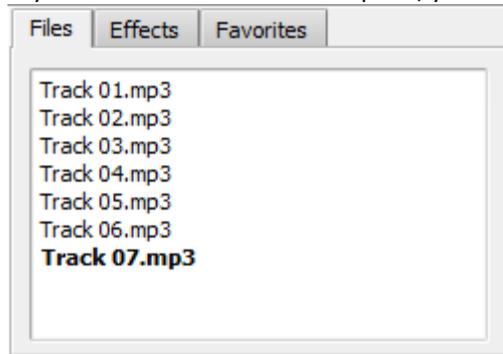
Clicking this button will display a menu with the following elements:

Button/Field	Description
 New...	Use it to create a new audio file.
 Open...	Use it to open an existing audio file. When you load a file for editing, AVS Audio Editor converts the audio file type to its own internal temporary file type for faster editing and better handling of larger file sizes.
 Import from Video	Use it to import the audio track(s) of a video file into the program.
 Grab from CD	Use it to grab an audio CD and add the grabbed tracks to the program.
 Save	Use it to save the active audio file with its current file name and location.
 Save as...	Use it to save the active audio file with a different file name and/or location, in a different file format.
 Save All	Use it to save all the changes to all the opened edited files.
 Info/Tags	Use it to view the available audio file information and modify audio tags.
 Close	Use it to close the currently opened edited file.
 Close All	Use it to close all the opened edited files.
Recent Files	Use it to display the list of five most recently loaded files.
 Preferences	Use it to view and edit the program preferences.
 Exit	Use it to close AVS Audio Editor .

Effects and Filters Panel

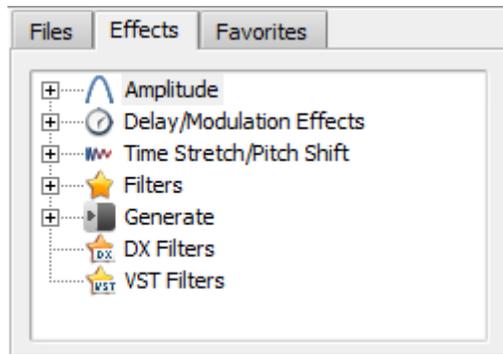
Effects and Filters Panel is a universal easy-to-use tool to perform many different operations with audio files. Using this panel you can add, remove files and apply various effects and filters to the opened files.

If you select the **Files** tab of the panel, you will be able to see the list of the files currently loaded into **AVS Audio Editor**:



All the newly created files will have the **Unknown** name until you save them and assign a name to them. You can add new files to the list, remove the unwanted ones from the list or create new files using the corresponding elements of the right-click menu.

To start editing the files loaded into **AVS Audio Editor** you will need to select the **Effects** tab:

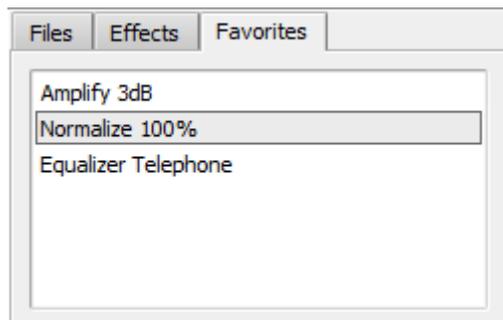


You can create lots of presets for every **Effects and Filters Panel** option or use the already existing ones.

The **Effects** tab menu consists of the following submenus:

- **Amplitude Effects**
- **Delay Effects**
- **Time/Pitch Effects**
- **Invert Effect**
- **Reverse Effect**
- **Filters**
- **Generate Sounds**
- **DX Effects**
- **VST Effects**

The **Favorites** tab lists the most recently used effects and filters:



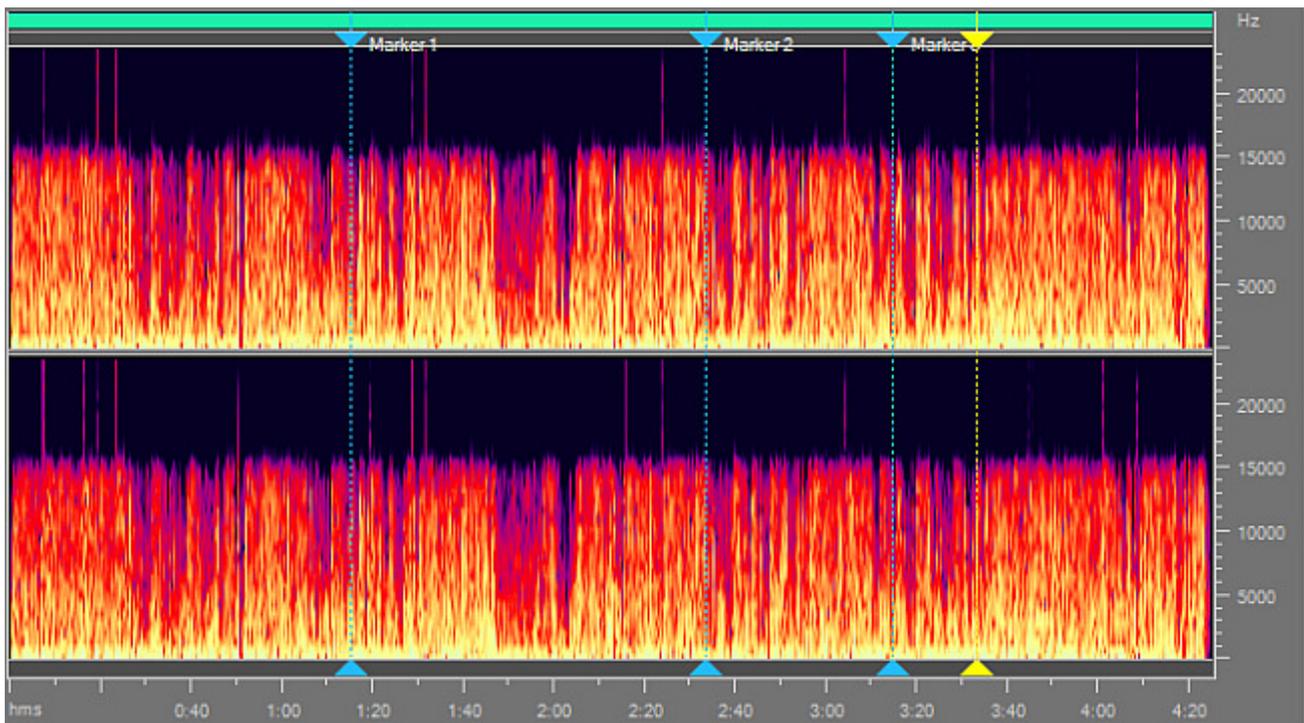
You can use the effects and filters from this list to quickly access and reuse them.

Waveform Editing Space

Waveform Editing Space consists of four main parts:

- **Waveform Display**
- **Display Range Bar**
- **Vertical Scale**
- **Time Scale**

Waveform Display



Waveform Display is the area where you view your audio material. There are many mouse operations available in the **Waveform Display** to select data:

- Left-click anywhere in the waveform and you will change the position of the playback cursor.
- With stereo or multichannel waveforms, use the **Control** key of the keyboard together with the mouse pointer (the cursor will acquire **L** or **R** letter - **L_f**, **R_f**, **C_f**, **L_F**, **L_r**, **R_r**, **L_S** and **R_S** for eight channel audio) and left-click to enable only the necessary channel for editing (the disabled channels will gray-out), another left-click within the disabled channel will re-enable it.
- Left-click and drag in the waveform display to make a selection.
- Double left-click in the waveform display to select the entire visible portion of the waveform (when zoomed in this does not select unseen areas which may be to the right or left).

Display Range Bar



The **Display Range Bar** on the top of the **Waveform Display** window indicates which part of the entire waveform is currently being viewed in the **Waveform Display**. When zooming in or out, this bar will get smaller or larger, as the portion being viewed changes with respect to the entire waveform or session.

Left-click (the mouse cursor should change to a hand) and drag the bar to scroll forward or backward in time:





Display Range Bar with mouse cursor down. Now you can drag the bar to scroll forward or backward in time

As with dragging in the **Time Scale**, the zoom level is retained, only the viewing range is altered.

To change the zoom level, move the mouse cursor to the edge of the **Display Range Bar** in zoom-in mode (it will turn to the double-sided arrow with a magnifying glass) and drag it to zoom out or zoom in:



Display Range Bar zoomed with mouse cursor



Vertical Scale

Vertical Scale measures the changes of audio data at a certain moment of time. You can select the measurement units for the **Vertical Scale** using the **Scales and Bars** button group of the **Tools Tab** - press the **Vertical Scale Format** button and select one of the following values: **Sample Values**, **Percentage** and **Decibels** (you can select the measurement units for the **Vertical Scale** using the right-click mouse menu on it).

Time Scale



Time Scale shows the current location at any point in the waveform display. The time markings are subdivided into more detail when zoomed in, and less detail when zoomed out.

Left-click (the mouse cursor should change to a hand) and drag the ruler to scroll forward or backward in time:



Time Scale with mouse cursor over it



Time Scale with mouse cursor down. Now you can drag the ruler to scroll forward or backward in time

As with dragging in the **Display Range Bar**, the zoom level is retained, only the viewing range is altered.

You can select the measurement units for the **Time Scale** using the **Scales and Bars** button group of the **Tools Tab** - press the **Time Scale Format** button and select one of the following values: **Decimal (mm:ss.ddd)**, **Samples**, **Frames 30 fps**, **Frames 29.97 fps**, **Frames 25 fps** and **Frames 24 fps** (you can select the measurement units for the **Time Scale** using the right-click mouse menu on it).

Bottom Toolbar

Bottom Toolbar consists of the following main parts:

- **Audio player**
- **Zooming toolbar**
- **Current position field**
- **Time display**

Audio Player consists of the following buttons:



In the table below you will find the functions you can perform with the help of each button:

Button	Name	Description
	Play	Starts playback of the current selection, or of the current audio file from either the left edge of the selection, or from the beginning of the file, to either the right edge of the selection or to the end of the file.
	Play Looped	Plays the current audio file or selection repeatedly, looping it until the Stop button is pressed.
	Play to End	Starts playback of the current selection, or of the current audio file from either the left edge of the selection, or from the beginning of the file, to the end of the file.
	Rewind	Rewinds the audio back several thousandths of a second.
	Forward	Steps the audio forward several thousandths of a second.
	Record	Start the recording from the source selected in the AVS Audio Editor Parameters dialog.
	Pause	Pauses playback.
	Stop	Stops playback.
	Go to Previous Marker	Quickly goes to the previous set marker. If there are no markers set in the audio, then this option lets the user quickly go to the beginning of the audio.
	Go to Next Marker	Quickly goes to the next set marker. If there are no markers set in the audio, then this option lets the user quickly go to the end of the audio.

Zooming toolbar consists of the following buttons:



In the table below you will find the functions you can perform with the help of each button:

Button	Name	Description
	Zoom In	Zooms in on the center of the current audio file window. After zooming, use the Time Scale to scroll to the desired location.
	Zoom Out	Zooms out from the current location.
	Full Zoom	Zooms all the way out to fit the entire waveform or session in the display window.
	Vertical Zoom in	Increases the vertical resolution scale of the waveform and Vertical Scale .
	Zoom in to Left Edge of Selection	Zooms in to the left edge of the current selection.
	Zoom to Selection	Zooms to the current selection. If no selection is made, this button zooms in on the cursor location.
	Zoom in to Right Edge of Selection	Zooms in to the right edge of the current selection.
	Vertical Zoom Out	Decreases the vertical resolution scale of the waveform and Vertical Scale .

Current Position Field displays the current position of the active window cursor (measured in **hours:minutes:seconds:thousandths of seconds** by default):

00:01:32.904

When you change the units of the Time Scale to the samples or frames, the units of the **Current Position Field** will be changed as well.

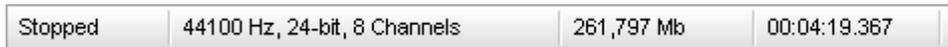
Time Display consists of 6 fields:

	Start	End	Length
Selection	00:01:32.904	00:02:47.743	00:01:14.839
View	00:00:00.000	00:04:58.841	00:04:58.841

These fields display start, end, and length information for the visible portion of the current waveform or session, as well as for the currently selected range. The top row of fields shows the selection time, and the bottom row the viewing range.

Status Bar

Status Bar displays information relating to the file properties and the uncompressed file size.



In the left part of the bar you can see the current file operation status, then the information on the format of the current waveform or session, with sample rate, sample size and the number of channels. In the middle part of the **Status Bar** the file size is displayed. In the right part of the bar you can see the duration of the audio (in **hours:minutes:seconds.thousands of seconds**). When you change the units of the Time Scale to the samples or frames, the units of the duration will be changed as well.

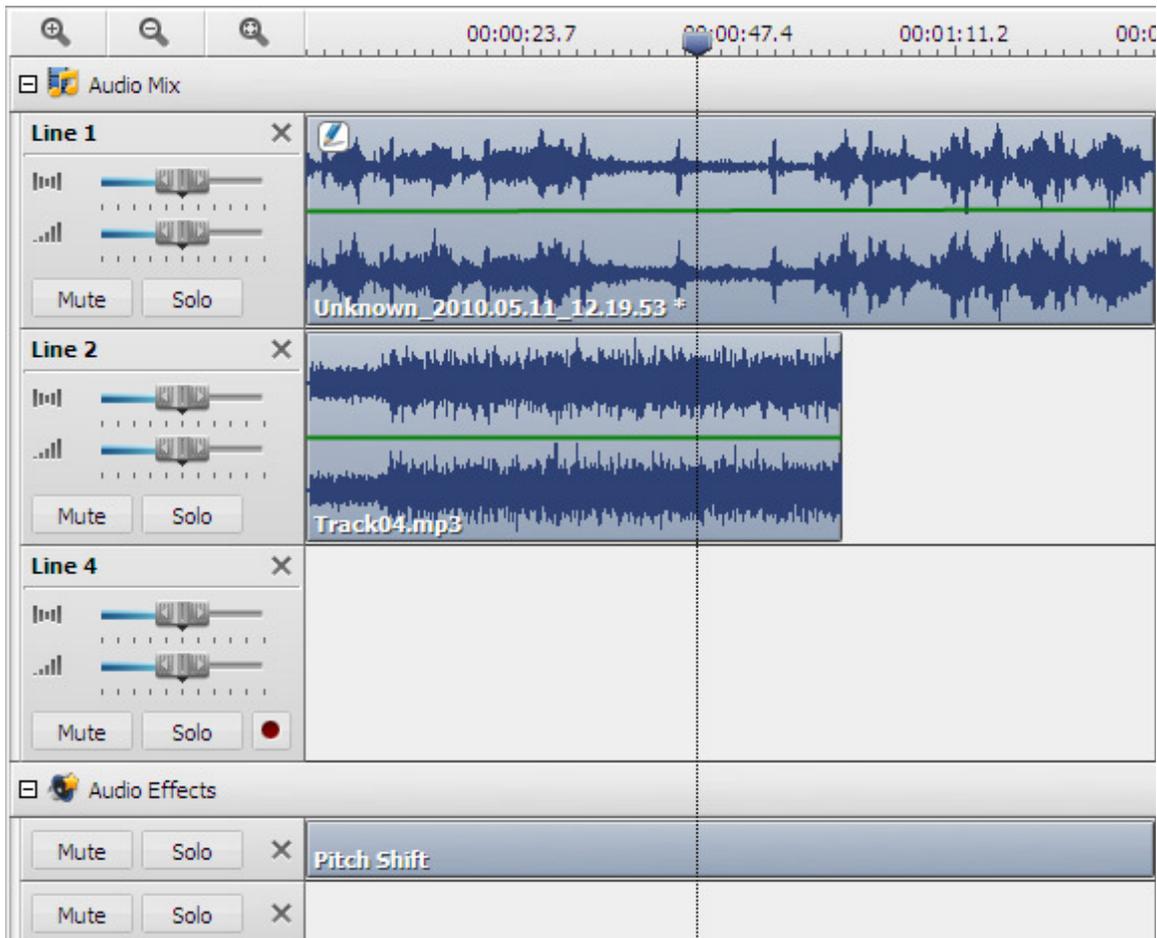


Note: you can select to show or hide the file information in the **Status Bar** using the **Status Bar** button of the **Tools Tab**.

Mix Timeline

The **Mix Timeline** tool is displayed when you switch to the Mix tab of the **AVS Audio Editor** Ribbon Command Bar. It consists of the following parts:

- **Time Scale**
- **Audio Mix/Audio Recording Lines**
- **Audio Effect Lines**



Time Scale

The **Time Scale** located on the top of the **Mix Timeline** area shows the current location at any point in the mix working space. The time markings are subdivided into more detail when zoomed in, and less detail when zoomed out.

To zoom in/out use the buttons on the left of the **Time Scale**.

Button	Name	Description
	Zoom in	Use this button to zoom in the Time Scale to find a certain scene
	Zoom out	Use this button to zoom out the Time Scale till the needed size
	Zoom to fit	Use this button to recover the 100% view

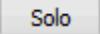
Right-clicking within this scale will activate the **Time Scale Express Menu**. This menu allows you to access to the available operations with markers.

Add Marker
Auto Add Markers
Prev.
Next
Remove Marker
Clear All Markers

Button	Description
Add Marker	Use it to add marker at the current cursor position.
Auto Add Markers	Use it to set markers at the beginning of each audio file added to this audio mix line.
Previous	Use it to go to the previous marker on the Time Scale .
Next	Use it to go to the next marker on the Time Scale .
Delete Marker	Use it to delete the currently selected marker.
Clear All Markers	Use it to delete all the set markers.

Audio Mix/Audio Recording Lines

The **Mix Timeline** is divided in lines depending on the file number you add for mixing. Each line includes the working space where the audio waveform is displayed and the **Toolbar** which consists of the following items:

Element	Description
 Delete	Use it to delete the selected audio file from the mix and clear the audio mix line.
Balance	Use this sliding bar to adjust the selected audio file balance.
Gain	Use this sliding bar to adjust the selected audio file volume.
 Mute	Use it to mute the selected audio mix line.
 Solo	Use it to mute all the lines except the selected one.
 Recording	Use it to start the recording process.

Right-clicking within the **Audio Mix** line will activate the **Express Menu** which allows you to facilitate the access to the main line editing options.

Open and Add File to Mix
New Line
Duplicate Line
Delete Line
Clear Line

If the **Audio Mix Line** is empty the options displayed in the activated right-click menu correspond to the buttons of the **Lines/Objects** group on the **Mix Tab**.

If there is some audio file added to your mix you will be presented with the following **Audio Mix Line Express Menu**:

Loop the Selection Playback
Duplicate Object
Delete Object
Edit Object
Split Object
Object Properties
Reset Envelope

Button	Description
Loop the Selection Playback	Use it to repeatedly play the selected part of the audio file.
Duplicate Object	Use it to duplicate the currently selected object in the mix working space. The audio file will be duplicated and displayed in the same line.
Delete Object	Use it to delete the currently selected object in the mix working space.
Split Object	Use it to split the currently selected object in the mix working space at current cursor position.
Object Properties	Use it to view the currently selected audio file properties.
Reset Envelope	Use it to reset the envelope of currently selected audio file.

Audio Effect Lines

AVS Audio Editor allows you to apply one or several effects to your mix. Each effect you add will be displayed in the separate **Audio Effect** line. The **Toolbar** on the left of each line includes the following buttons:

Element	Description
 Delete	Use it to delete the currently selected effect.
 Mute	Use it to mute the currently selected effect line.
 Solo	Use it to mute all the lines except the selected one.

Right-clicking within this line will activate the **Express Menu** which facilitates the access to the main effect line editing options.

New Line
Duplicate Line
Delete Line
Clear Line

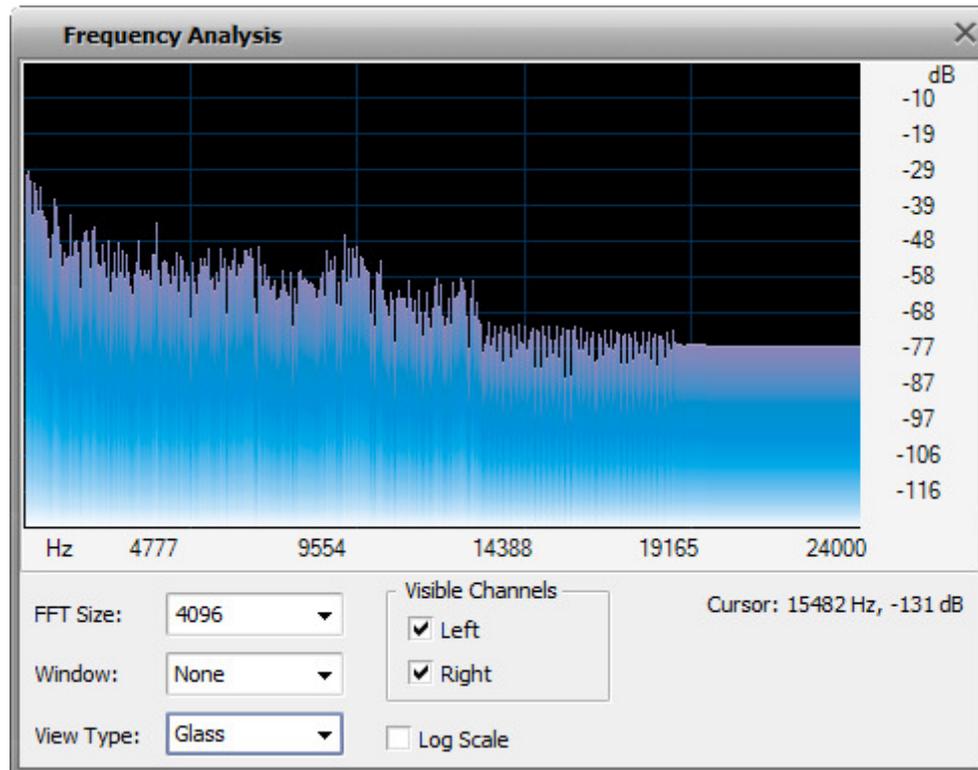
If the **Audio Effect Line** is empty the options displayed in the activated right-click menu correspond to the buttons of the **Lines/Objects** group on the **Mix Tab**.

Edit Effect
Duplicate Object
Delete Object
Split Object

Clicking with the right mouse button within the added audio effect block will display the following **Express Menu**:

Button	Description
Edit Effect	Use it to open the selected Audio Effect properties window and adjust them
Duplicate Object	Use it to duplicate the currently selected effect in the mix working space. The object will be duplicated and displayed in the same line.
Delete Object	Use it to delete the currently selected effect in the mix working space.
Split Object	Use it to split the currently selected effect in the mix working space at current cursor position.

Frequency Analysis Window



Frequency Analysis Window contains a graph of the frequencies at the insertion point (yellow arrow cursor) or at the center of a selection. This window "floats", meaning that you can click in the waveform on the main **AVS Audio Editor** window to update the analysis while the **Frequency Analysis** window is on top.

The information in this dialog is like one "slice" or line in the **Spectral View** of the waveform. The most prominent frequency is interpolated and displayed. You can move the mouse over the graph area to display the frequency and amplitude components of that frequency.

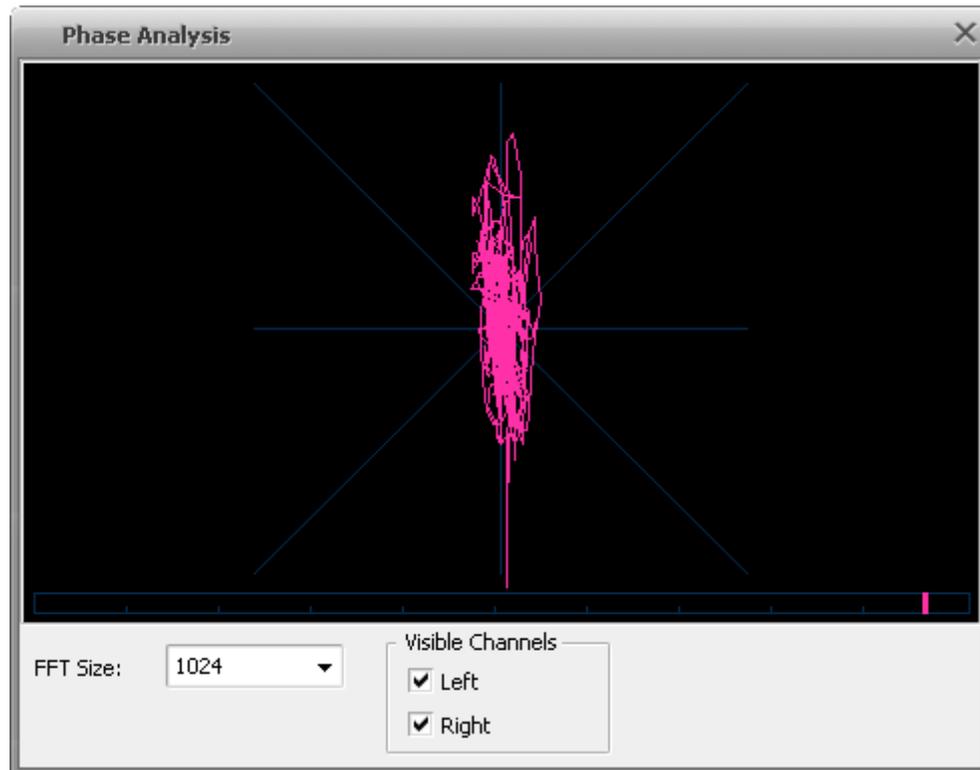
The **Frequency Analysis** window is updated in real time while you play your file.

You can select the properties of the **Frequency Analysis** using the appropriate options:

- **FFT Size** - the size of the FFT filter. You can select between the following values: **64, 256, 1024, 4096** and **16384**.
- **Window** - the amount of transition width and ripple cancellation that occurs during filtering, with each one resulting in a different frequency response curve. You can select between **None, Welch, Bartlett, Hanning** and **Hamming**.
- **View Type** - the appearance of the graph. You can select between **Analog, Normal, Fire, Glass** and **Kryptonite** depending on your needs and preferences.
- **Visible Channels** - the number of channels of the audio used to produce the **Frequency Analysis**. You can either select all the channels or switch off some of them. One of the channels have always to stay on.

To display the **Frequency Analysis** window, switch to the **Tools Tab** and press the **Frequency Analysis** button.

Phase Analysis Window



Phase Analysis Window is used to reveal phase differences between the stereo channels of an audio file. This window "floats", meaning that you can click in the waveform on the main **AVS Audio Editor** window to update the analysis while the **Phase Analysis** window is on top.

Phase analysis may be of help if you plan to convert a stereo file to a single-channel audio. When you convert stereo to mono, the sounds in both channels are summed up. If sounds in both channels are in-phase it will cause no problems. If the channels in the original audio are out-of-phase, the resulting mono file may have significant sound distortions.

To check whether the channels of the audio file you plan to convert to mono are in-phase, start the file playback, then open the **Phase Analysis** window by pressing the corresponding button on the **Tools Tab**. Follow the pink image changes in the window. If it's more vertically stretched (like on the figure above), it means that the left and right channels of your stereo file are in-phase and it can be easily converted to mono. If the image is stretched horizontally, you can correct it by means of **inverting** one of the channels of the original audio. After this operation is done you can convert your file to mono.

The **Phase Analysis** window is updated in real time while you play your file.

You can select the properties of the **Phase Analysis** using the appropriate options:

- **FFT Size** - the size of the FFT filter. You can select between the following values: **64**, **256**, **1024**, **4096** and **16384**. The bigger the value is, the more precise the analysis will be, but overall analysis speed will decrease.
- **Visible Channels** - the number of channels of the audio used to produce the **Phase Analysis**. Both should be activated for stereo signal phase comparison.

Keyboard and Mouse Shortcuts

To speed up your editing **AVS Audio Editor** offers you a wide variety of keyboard shortcuts. Use the table below to learn how to edit audio data in a fast and easy way.

Keyboard

Controls	Shortcut Key	Description
File Controls	Ctrl+N	Creates a new, initially blank file
	Ctrl+O	Opens an existing waveform
	Ctrl+S	Saves the current waveform to disk
	Shift+Ctrl+S	Saves the active audio file with a different file name and/or location, in a different file format.
Editing Controls	Ctrl+A	Selects Entire Waveform
	Ctrl+Z	Undo, reverses the last command or action
	Shift+Ctrl+Z	Redo, repeats the last command or action
	Ctrl+X	Cuts a waveform to internal clipboard
	Ctrl+C	Copies a selection to internal clipboard
	Ctrl+V	Pastes from internal clipboard
	Ctrl+T	Trims to selection
	Del	Deletes selection
Navigation Controls	Left Arrow	Move cursor left on the Timeline
	Right Arrow	Move cursor right on the Timeline
	Home	Move cursor to the beginning of the waveform on the Timeline
	End	Move cursor to the end of the waveform on the Timeline
Selection Controls	Shift+Left Arrow	Increase the left selection boundary
	Shift+Right Arrow	Increase the right selection boundary
	Ctrl+Left Arrow	Decrease the left selection boundary
	Ctrl+Right Arrow	Decrease the right selection boundary
	Shift+Home	Move the left selection boundary to the beginning of the waveform
	Shift+End	Move the right selection boundary to the end of the waveform
Zoom Controls	Ctrl+Numeric Pad Plus	Zooms In

	Ctrl+Numeric Pad Minus	Zooms Out
Miscellaneous	F1	Brings up the current Help File

Mouse Operations

Left click and drag a waveform to make a selection.

Double left-click a waveform to select the entire visible portion of the waveform (when zoomed in this does not select unseen areas which may be to the right or left).

Left-click anywhere in the waveform and you will change the position of the playback cursor.

With stereo or multichannel waveforms, use the **Control** key of the keyboard together with the mouse pointer (the cursor will acquire **L** or **R** letter - **L_f**, **R_f**, **C_f**, **LFE**, **L_r** and **R_r** for six channel audio) and left-click to enable only the necessary channel for editing (the disabled channels will gray-out), another left-click within the disabled channel will re-enable it.

Left-click (the mouse cursor should change to a hand) and drag the **Display Range Bar** to scroll forward or backward in time.

Click and drag on the **Time Scale** to scroll horizontally.

Rest mouse over any toolbar button to get an explanation of the button's function.

Double-click title bar to **Maximize/Restore**.

Wheel Mouse Specific

Roll the mouse wheel forward to zoom in. Roll the mouse wheel backwards to zoom out.

Supported Audio Formats

Most of the commonly used media file formats are supported by **AVS Audio Editor**:

Format	Read	Write
MP3 - MPEG-1 Layer 3 Audio (.mp3)	+	+
WMA - Windows Media Audio Format (.wma)	+	+
WAV, or WAVE - Waveform Audio Format (.wav)	+	+
M4A - MPEG-4 Compressed Audio Format (.m4a)	+	+
M4R - iPhone ringtones (.m4r)	+	+
AAC - Advanced Audio Coding (.aac)	+	+
AMR - Adaptive Multi-Rate Narrowband/Wideband (.amr, .awb)	+	+
OGG - Ogg Vorbis Audio Format (.ogg)	+	+
MP2 - Musicam, or MPEG-2 Level II (.mp2)	+	+
FLAC - Free Lossless Audio Codec (.flac)	+	+
AIFF - Audio Interchange File Format (.aiff)	+	-
AU - Au File Format (.au)	+	-
Musepack File Format (.mpc)	+	-

Supported Video Formats

Most of the commonly used video file formats are supported by **AVS Audio Editor**:

Format	Read*	Write**
AVI Files - Audio Video Interleave (.avi, .ivf, .div, .divx, .mtv, .amv)	+	+ (except .mtv and .amv)
MPEG Video - Motion Picture Experts Group (.mpg, .mpeg, .mod, .mpe, .mvv, .m2v)	+	+ (except .m2v)
Windows Media Formats (.wmv, .asf)	+	-
Microsoft Digital Video Recorder (.dvr-ms)	+	-
CD Video (.dat)	+	-
DVD Video (.ifo)	+	-
VOB Video (.vob, .vro)	+	-
Blu-ray Video (index.bdm, index.bdmv, .m2ts)	+	-
AVCHD, HDV (.m2ts, .m2t, .mts, .tod)	+	+ (.mts and .tod only)
TS Files (.ts, .m2ts, .m2t, .mts, .tp)	+	-
DV Files (.dv)	+	-
Matroska Files (.mkv)	+	-
Ogg Vorbis Movie (.ogm, .ogg, .ogv, .oga, .ogx)	+	-
Real Media (.rm, .rmvb, .ram, .ivr)	+	-
Quick Time (.mov, .qt, .m4v, .mp4)	+	+
Mobile Video (.3gp2, .3gpp, .3gp, .3g2)	+	+
Flash Video (.flv)	+	+
Shockwave Flash (.swf)	+	-

* Supporting a video file for reading means that you can import its audio track into **AVS Audio Editor**, edit and then save it as an

audio file of one of the supported formats.

** Supporting a video file for writing means that you can open it in **AVS Audio Editor** as a video file, edit and then save it as a video file with the edited audio track.

Operations with Files

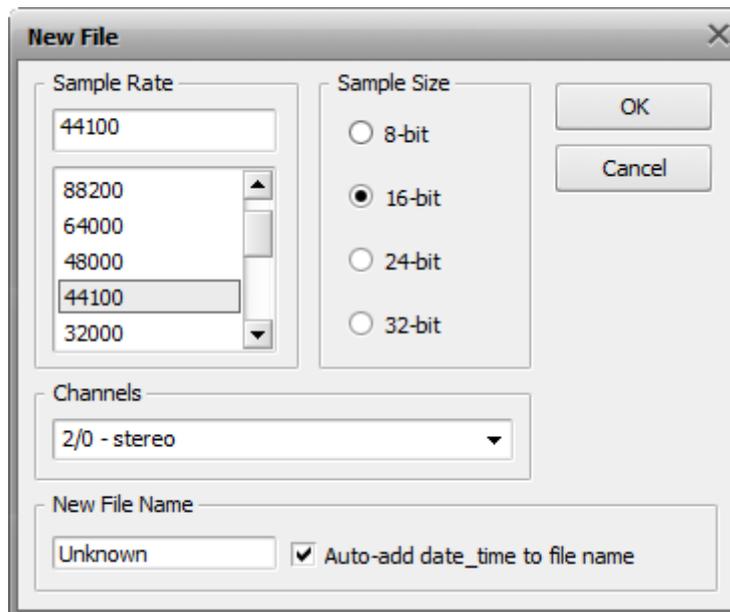
When you use **AVS Audio Editor** you will need to select some files, load them into the program, view their information and save them into different formats.

Using the items of the **File** tab you can quite easily:

- **create a new file;**
- **open an existing file for editing;**
- **view the file information;**
- **save the edited file into one of the available formats.**

Creating a New File

Press the **New...**  command button on the **Home** or **File** tab (the same **New File** option can be accessed from the **Application Button** menu or **Quick Access Toolbar**). The following window will be opened:



Select the **Sample Rate** (8'000 - 96'000 Herz), **Sample Size** (8-bit - 32-bit) and the **Number of Channels** (1/0 - mono -- 3/4+LFE 7.1 surround) and click the **OK** button to quickly create a file of the selected format.

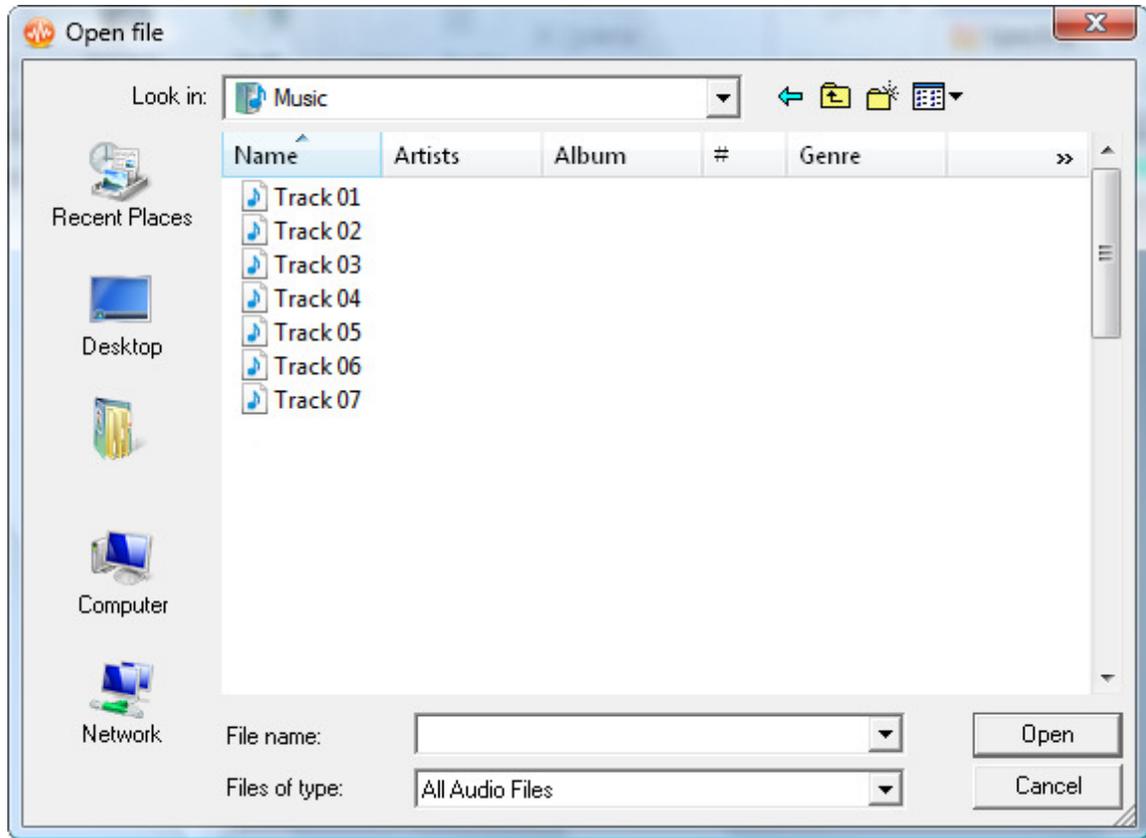
Note! **Note: AVS Audio Editor** creates all files using a temporary audio file format and saves all the changes to the edited file only when you select to **Save** them using one of the ways.

Opening Files

Opening Audio Files

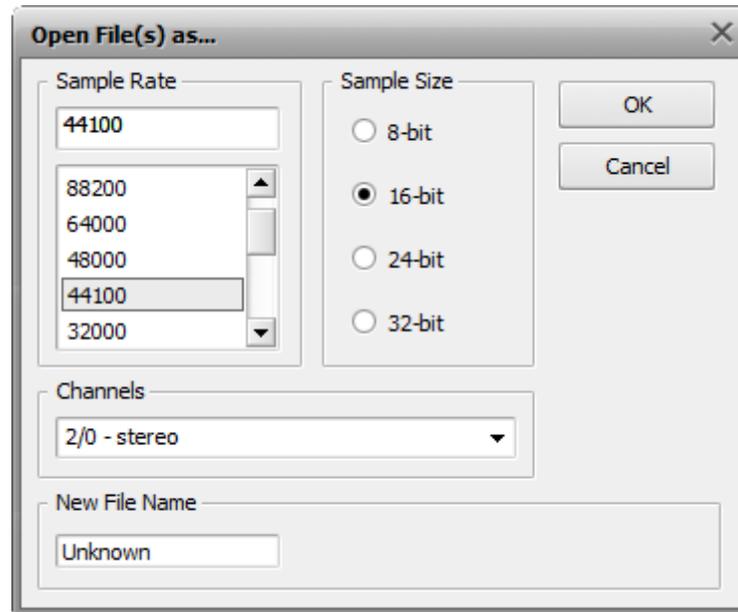
To start editing a file, you need to load it into **AVS Audio Editor**. There are several ways to do it:

1. Select the **Open...**  command button on the **Home** or **File** tab. The following window will be opened:



You can select a file or several files with your mouse or keyboard and click the **Open** button.

2. Select the **Open as...**  command button on the **Home** or **File** tab (the **Open as...** option will be displayed after you press the arrow next to the **Open** button). This option will let you open the file as if it were of a different format than it actually is (so called "quick-and-easy" conversion method). After you select a file or several files in the window above, you will be offered to set the following parameters:



Select the **Sample Rate** (8'000 - 96'000 Herz), **Sample Size** (8-bit - 32-bit) and the **Number of Channels** (1/0 - mono -- 3/4+LFE 7.1 surround) and click the **OK** button to quickly convert the edited file into the selected format.

Note! **Note:** the original file will not be converted unless you decide to save it. **AVS Audio Editor** opens all files into a temporary format and saves all the changes to the original file only when you select to **Save** them using one of the ways.

3. Select the **Open and Append...**  command button on the **Home** or **File** tab (the **Open and Append...** option will be displayed after you press the arrow next to the **Open** button). This option will let you open the file and append it to the already opened one making one file out of two. The files will be merged into one with the second opened file appended to the end of the first opened one.

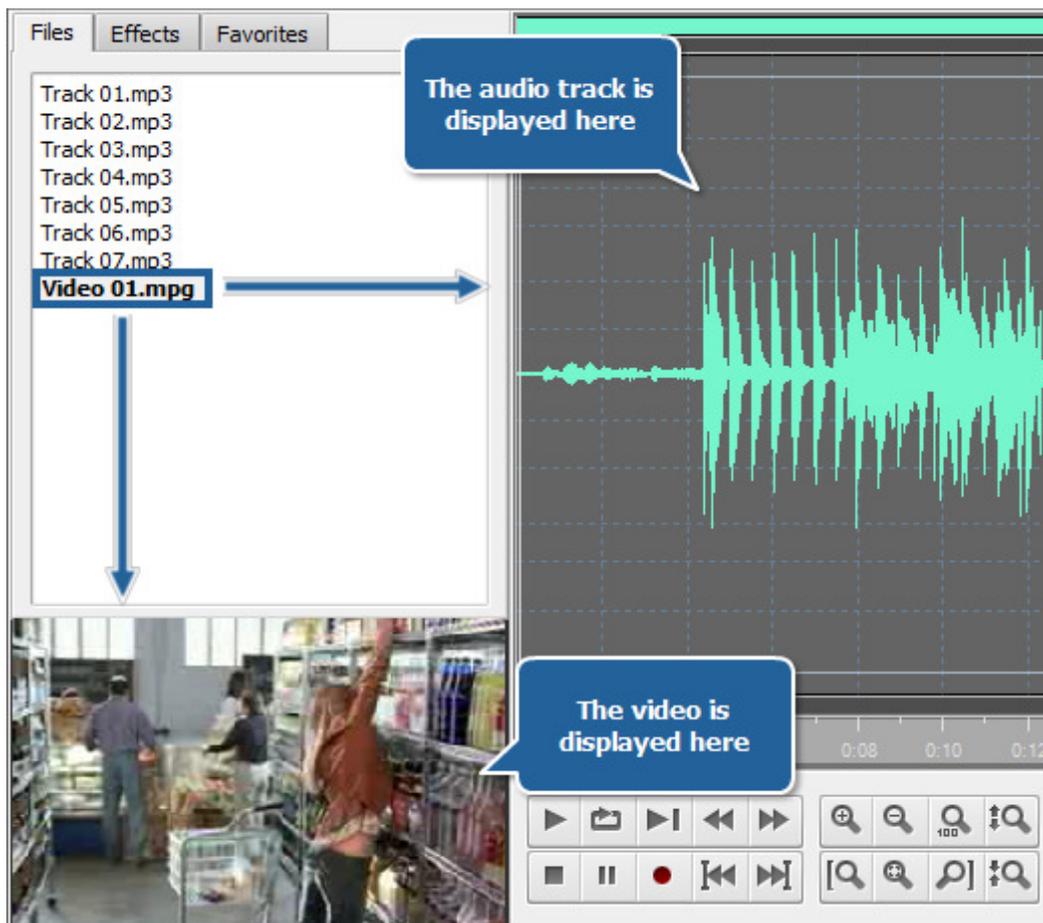
AVS Audio Editor supports all major audio formats. When you load a file for editing, **AVS Audio Editor** converts the waveform type to its own internal temporary file type for faster editing and better handling of larger file sizes.

Opening Video Files

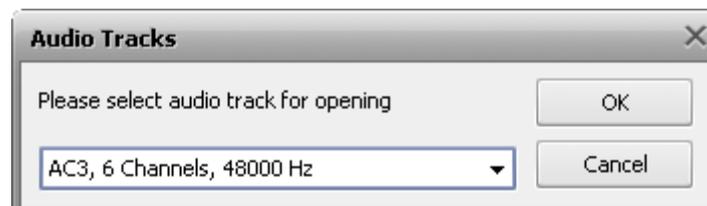
You can import a video file into **AVS Audio Editor** with the purpose of editing its audio track(s). You can open video files of the following formats:

- AVI Files (.avi, .ivf, .div, .divx)
- Mpeg Files (.mpg, .mpeg, .mod, .mpe, .mvv)
- Quick Time Files (.mov, .qt, .m4v, .mp4)
- Mobile Video (.3gp2, .3gpp, .3g2, .3gp)
- AVCHD (.mts, .tod)
- Flash Video (.flv)

To open a video file, select the **Open...**  command button on the **Home** or **File** tab. Browse for the necessary video file (make sure the **All Video Files** option is selected in the opened window) and press **Open**. The selected video file will be added to the **Files** section of the **Effects and Filters Panel**:



In case the video file you are importing contains several audio tracks, you will be presented with the following window:



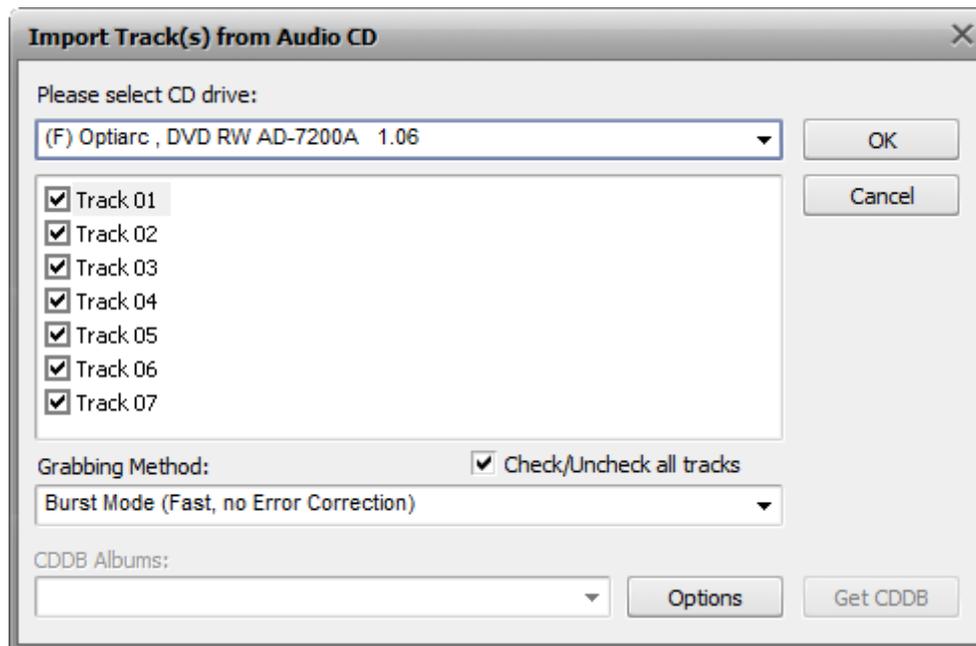
Select the necessary audio track and press **OK**. The video file with the selected audio track will be imported into the program.

You can edit the audio track of your video file just like any audio track loaded into the program. Just note that you can apply a **limited number** of effects to your audio tracks. **AVS Audio Editor** will not allow you to apply effects that change the track duration and generate sounds.

When **saving** your video file, you will be able change its audio track format. The result will be saved as a video file with the edited audio track of the selected audio format.

Grabbing Audio CDs

AVS Audio Editor allows you to grab audio CDs and import the grabbed audio into the program for editing. To load audio files from a CD into **AVS Audio Editor** press the **Add from CD** button on the **Home** or **File** tab. You will be presented with the following window:



Select the **CD Drive** with the disc which contains the tracks you wish to load into the program. As soon as you do so, all the tracks on the disc will be displayed in the area below.

Uncheck the boxes for the tracks you are not planning to import into **AVS Audio Editor**. Use the **Check/Uncheck all tracks** option to quickly select or deselect all tracks in the list.

Specify the **Grabbing Method** in the appropriate window. The following options are available:

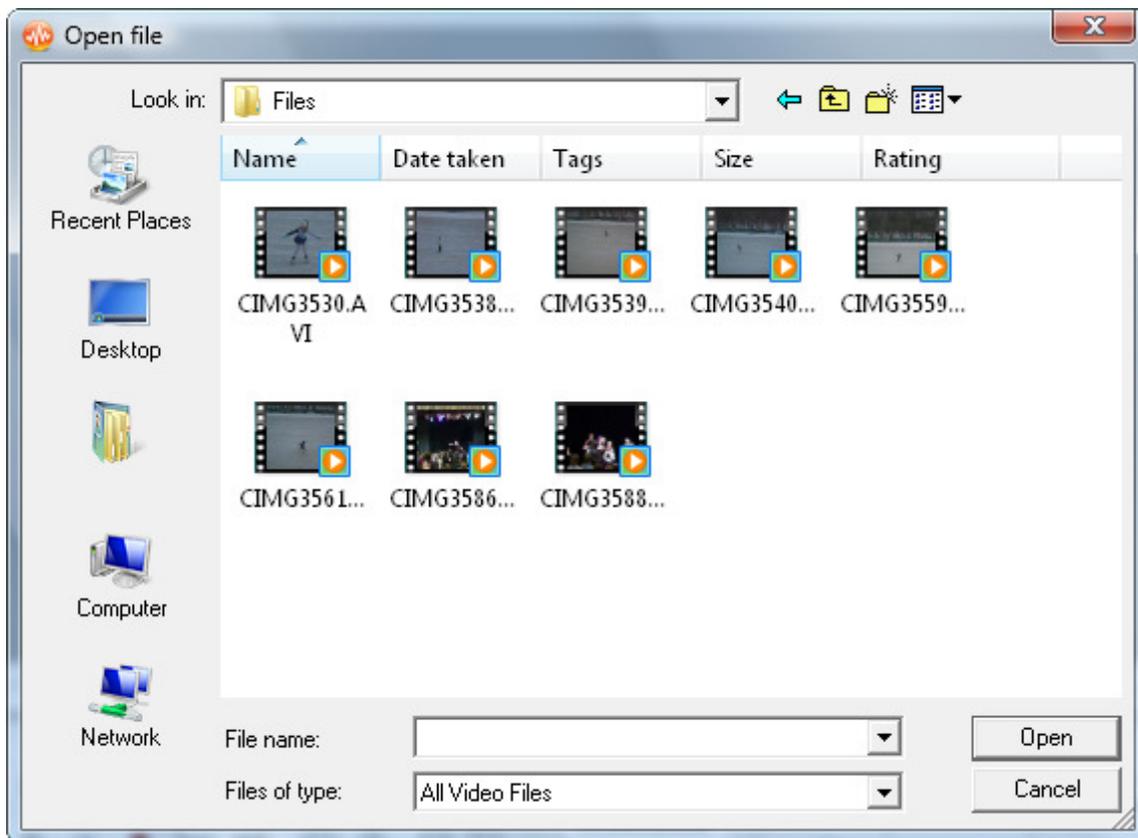
- **Burst Mode** - use it for fast grabbing without error correction. Select it for grabbing discs in perfect condition (with no scratches or other visible damage).
- **Sync Mode** - use it for fast audio track grabbing. Only jitter errors will be corrected if present.
- **Secure Mode** - use it for good-quality grabbing. Errors will be corrected with double precision but overall grabbing speed will decrease.
- **Paranoid Mode** - the slowest and the most accurate grabbing mode available.

Click **OK** to start the grabbing process. Press **Cancel** any time during the grabbing process to cancel it.

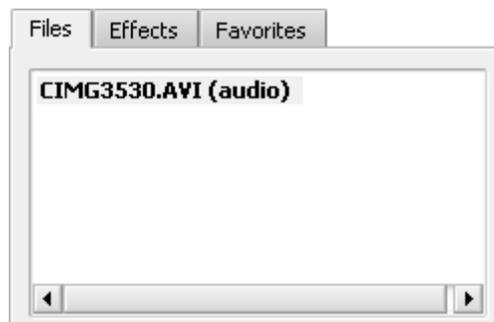
As soon as the grabbing is finished, all selected tracks will appear on the **Files** tab of the **Effects and Filters Panel**.

Importing Audio Tracks from Video

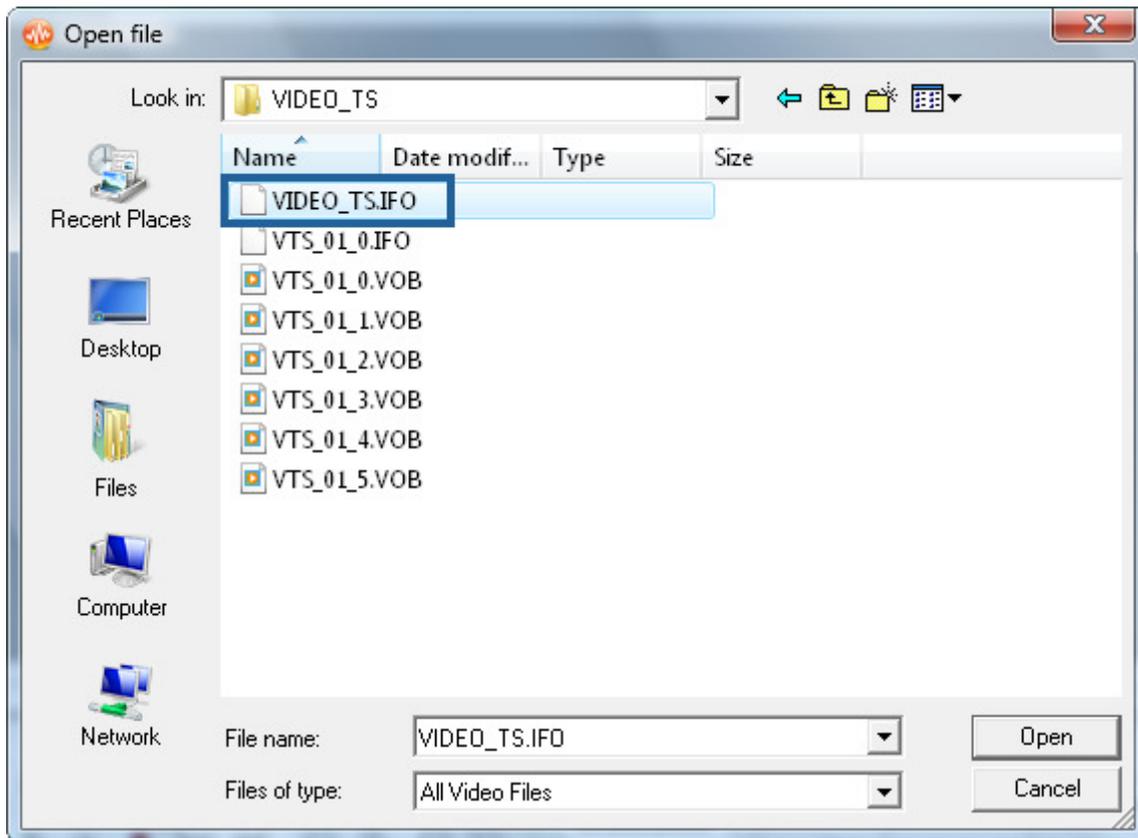
AVS Audio Editor allows you to load audio track(s) of a video file, edit and save it(them) to one of the available audio formats. To import the audio track of a video file, press the **Import from Video** button on the **Home** or **File** tab. The following window will be opened:



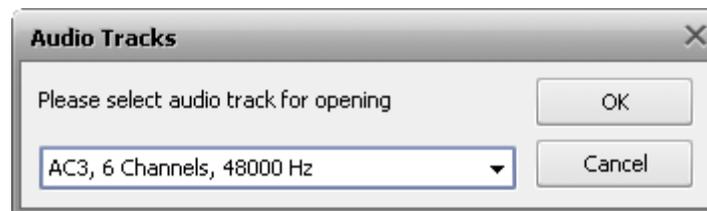
Browse for the video file you wish to import and press **Open**. The audio track of the selected video file will be added to the **Files** section of the **Effects and Filters Panel**:



If you wish to import the audio track of a **DVD Video**, please select the **video_ts.ifo** file when opening your DVD:



In case the video file or DVD movie you are importing contains several audio tracks, you will be presented with the following window:



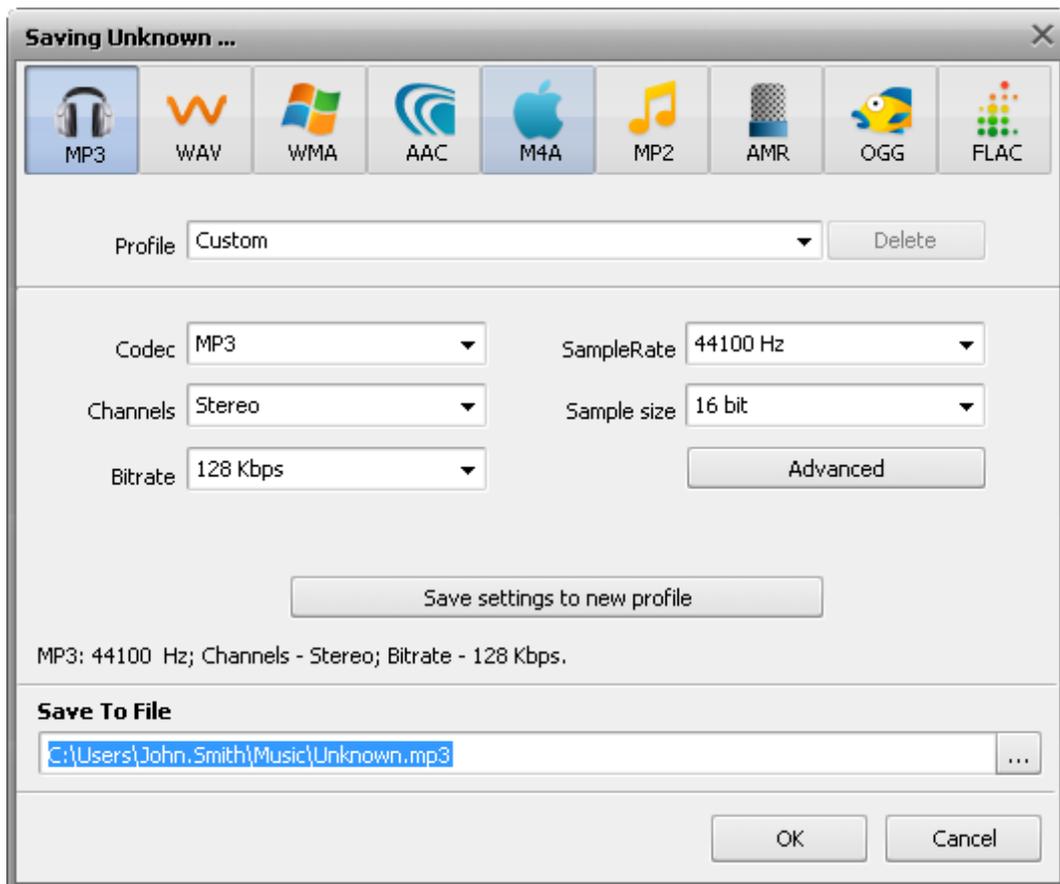
Select the necessary audio track and press **OK**. The selected audio track will be imported into the program.

Saving Files

Saving Audio Files

You can change the format of the edited file when you save it. It can be done to reduce the size of an audio file, to change the number of channels or to make it possible to playback the file using mobile devices, such as mobile phones, portable players, etc.

To change the file format and save it using this selected format, you can press the **Save as...**  command button on the **File** tab (the same **Save as...** option can be accessed from the **Application Button** menu or **Quick Access Toolbar**). The following window will be opened:



Here you can select the output file format and specify the output file format parameters such as **Sample rate (Frequency)**, **Bitrate** and the number of **Channels**.

Note! **Note:** some formats, such as **AAC**, **M4A**, **WMA** and **WAV** allow you to select more than two channels. You can set up to eight channels depending on your desires and the devices that will be used to playback the resulting audio files.



Note: it is possible to specify **Advanced MP3** parameters clicking the appropriate button. You can find the detailed information about these settings in the **Appendix** section. See also:

- **Available MP3 Parameter Combinations**
- **Available MP2 Parameter Combinations**

After you select all the parameters, click the **OK** button to accept the changes and save the audio file or **Cancel** to discard the changes and close this window.

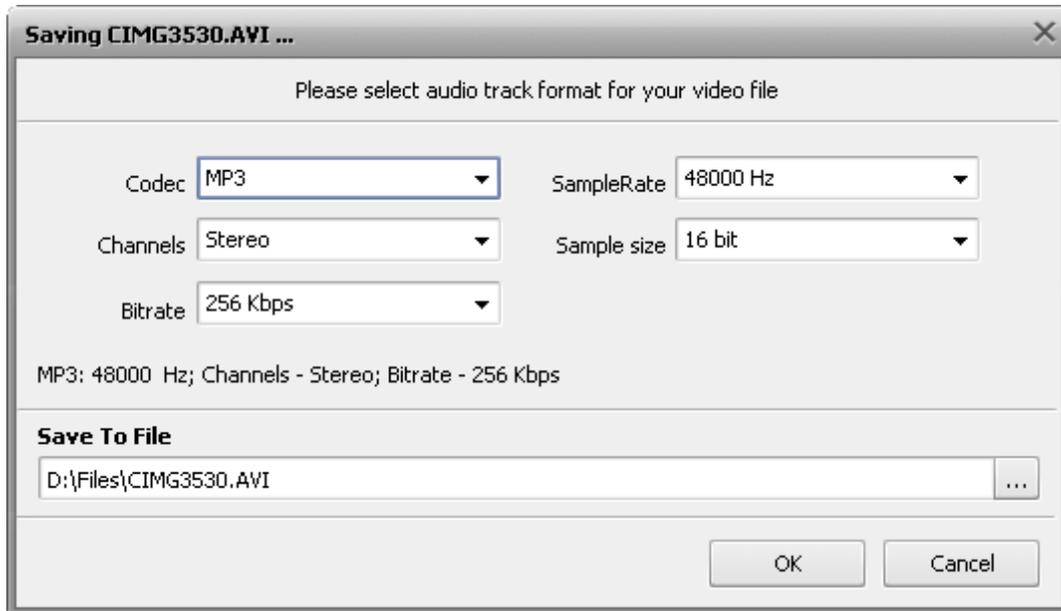
If you loaded several files into **AVS Audio Editor** and would like to save them all, you can use the **Save All**  command button on

the **File** tab. It is also possible to save not the whole file but only the currently selected part of it using the **Save Selection as...** button on the **File** tab.



Saving Video Files

To save a video file with the edited audio track, press the **Save** or the **Save as...** command button on the **File** tab (the same **Save** options can be accessed from the **Application Button** menu or **Quick Access Toolbar**). The following window will be opened:



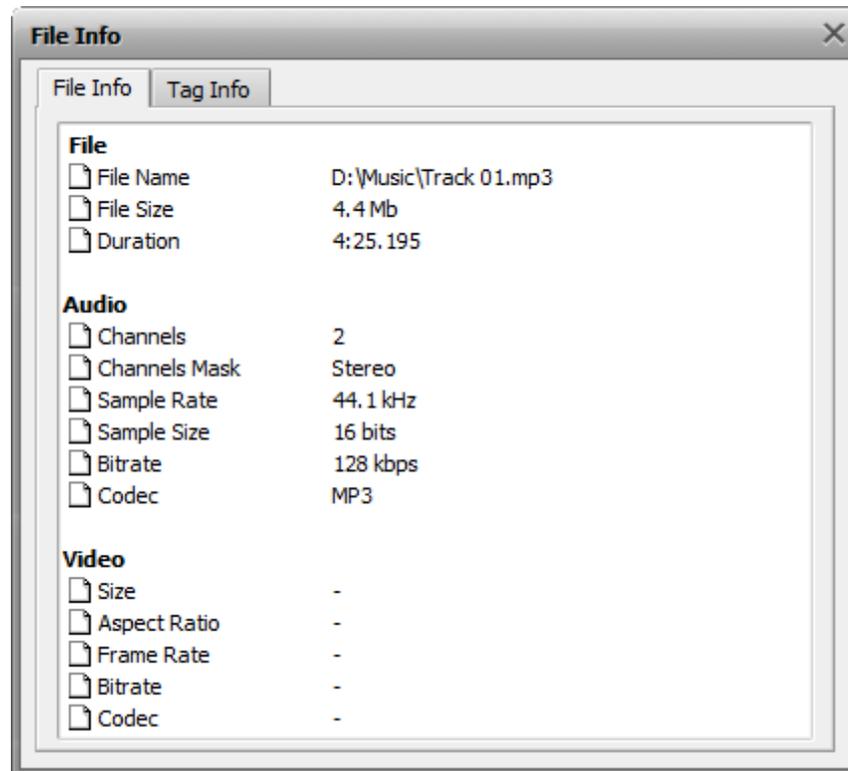
Select audio track format and parameters, then click **OK** to accept the changes and save the video file or **Cancel** to discard the changes and close this window. The result will be saved as a video file with the edited audio track of the selected audio format.



Note: AVS Audio Editor sets the path to the original video file in the **Save to File** field by default. So, if you do not wish the original file to be affected, press the **Browse** button and select another directory for your output video or specify another name for it in the **Save to File** field.

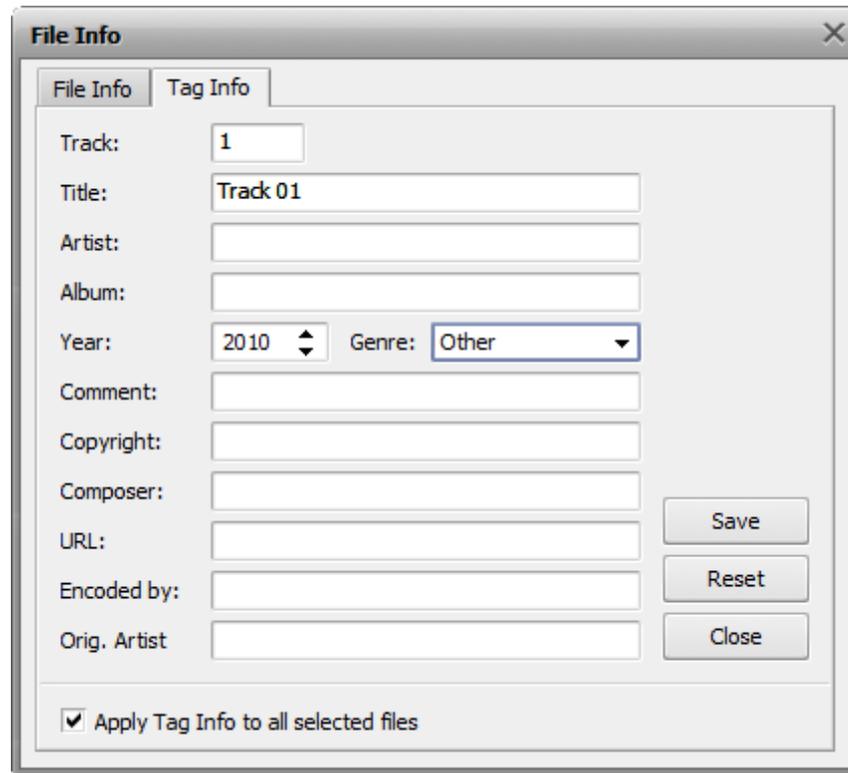
Viewing File Information/Modifying Audio Tags

When you first open a file it might be quite useful to know as much information about it as possible. This can be done pressing the **Info/Tags**  button on the **File** tab:



The **File Info** tab of the opened window contains the following information:

- The **File** section contains information about **File Name**, **File Size** and its **Duration**.
- The **Audio** section contains information about audio track properties including **Sample Rate**, **Sample Size**, the **Number of Channels** and **Codec** used to compress it.
- The **Video** section displays information only in case you **load a video file** into the program. It includes video file **Size**, **Aspect Ratio**, **Frame Rate**, **Bitrate** and the **Codec** used for video file compression.



The **Tag Info** tab contains the list of audio tags that can be edited and saved to output audio file. Audio tags are built-in text information about audio files such as track name and number, author of the song, its release date etc. It might be useful to fill out the tag fields, because many audio players use tags rather than file names when playing back audio files.

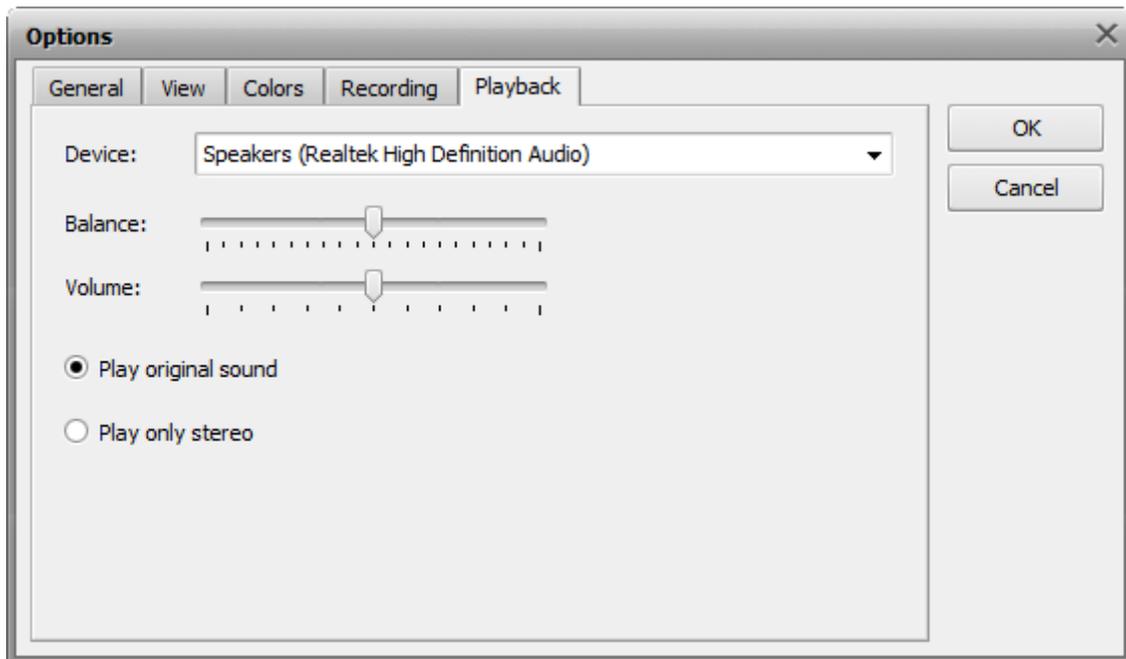
Fill out the tag fields you consider necessary (all fields are optional). Then press the **Save** button to accept the changes and close the **File Info** window, the **Reset** button to return to the original tag information or the **Close** button to discard the changes and close the window.

Playing and Recording

In **AVS Audio Editor** you can operate **Play** and **Record** as well as other transport functions in the same way as in any standard home player or recorder. **Audio Player Toolbar** is situated in the left bottom corner of the **Main Window**.

- ▶ - Click **Play** to play the portion of the waveform that is currently being viewed, or the portion that is highlighted.
- || - Click **Pause** to pause the playback.
- - Click **Stop** to end waveform playback.

To adjust the playback volume and balance, press **Options** on the **Tools Tab**. In the opened window switch to the **Playback** tab:



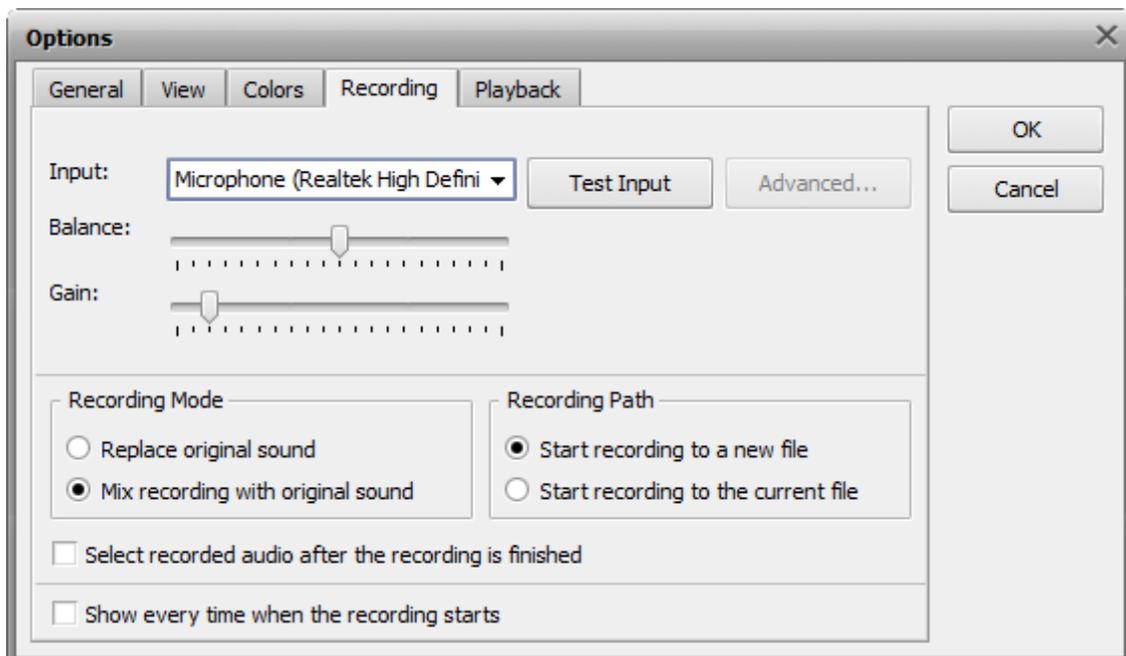
Select the device used for playback in the **Device** drop-down list. Use the scroller of the **Balance** section to adjust balance (move to the left to make the sound play in the left speaker, move to the right to make it play in the right speaker). Use the scroller of the **Volume** section to adjust volume (move to the left to decrease volume, move to the right to increase volume). The **Play original sound** and **Play only stereo** options are used to let the player decide how to playback the audio. This might be useful when you playback a multichannel (5.1 and more channels, for instance) audio on stereo speakers/headphones.

● - Click **Record** on the **Audio Player Toolbar** to start recording at the current insertion point. Any waveform data after that will be recorded over.



Note: the **Record** button will change into **Stop** -  when you start the recording process. Just press it to stop the recording and return to the editing.

You can configure your input devices right from **AVS Audio Editor**, without need to go to the windows **Control Panel**. To do that, press **Preferences...** on the **Tools Tab**. In the opened **Preferences** window switch to the **Recording** tab. The form that will let you change the device settings will appear:



In the upper section of the window you can:

- select the device that will be used for sound recording and the input jack active on the device. Press the drop-down combo-box and select the necessary device from the list, if you have more than one input device installed on your computer;



Note: if you use Windows Vista, you'll be able to select both the device and the input jack in one and the same **Input** field (the device name being indicated in brackets). If you run Windows XP, the input device will be displayed in the **Device** field and the list of available input jacks will be displayed in the area below.

- set the device **Balance** - the difference of the sound volume between the right and the left channels;
- set the input device **Gain** - the loudness of the device input;
- test the device pressing the **Test Input** button to make sure that the device is in working order and the **Balance** and **Gain** are set correctly.

In the **Recording Mode** section you can select between two available options:

- **Replace original sound** - use it to replace any of the audio signal present in the audio file with the recorded audio.
- **Mix recording with the original sound** - use it to mix your recording with the audio signal that's already present in the file.

In the **Recording Path** section you can select between two available options:

- **Start recording to a new file** - use it to start the recording to a new file. A new file will be created each time you press the **Record** button.
- **Start recording to the current file** - use it to record various recording sessions to one and the same file. No new file will be created when you press the **Record** button to resume the recording, the recording will continue to the current file.

Finally, tick/untick the **Select recorded audio after the recording is finished** and **Show every time when the recording starts** options to enable/disable the corresponding features.

After you select all the settings for your input device you can click the **OK** button to accept the changes made and go on recording the sound from the selected and configured device.

Editing Options

AVS Audio Editor lets the user perform simple editing operations - copy, paste and delete - with audio files. Please refer to the appropriate sections to read more on these operations:

- **Copy operations**
- **Paste operations**
- **Delete and trim operations**

You can also add **markers** to your audio file to mark the important moments and simplify the navigation through the file and **convert sample type** of your audio file, changing its **Sample Rate**, **Sample Size** and the **Number of Channels**. See the appropriate chapters for more detail.

Copying

The following **Copying** operations are available in **AVS Audio Editor**:

-  **Cut** - use this option to cut the currently selected part of the edited audio and copy it to the program internal clipboard. This option only allows you to cut the selected part of the audio, thus if no part of the audio file has been selected nothing will be cut and copied. When you cut a part of the audio you can insert it afterwards to some other place of the same audio or to a different audio, opened in **AVS Audio Editor** using one of the **Paste** options.
-  **Copy** - use this option to copy the currently selected part of the edited audio to the program internal clipboard. This option only allows you to copy the selected part of the audio, thus if no part of the audio file has been selected nothing will be copied. When you copy a part of the audio you can insert it afterwards to some other place of the same audio or to a different audio, opened in **AVS Audio Editor** using one of the **Paste** options.
-  **Copy to New** - use this option to copy the currently selected part of the edited audio to a new file. The new file will be created automatically and will have the same parameters (sample rate, sample size and the number of channels) as the currently edited file. This option only allows you to copy the selected part of the audio, thus if no part of the audio file has been selected nothing will be copied.
-  **Copy to Clipboard** - use this option to copy the currently selected part of the edited audio to the common windows clipboard. This option only allows you to copy the selected part of the audio, thus if no part of the audio file has been selected nothing will be copied. When you copy a part of the audio you can insert it afterwards to some other place of the same audio or to a different audio, opened in **AVS Audio Editor** using one of the **Paste** options. This option is particularly useful when there are several instances of **AVS Audio Editor** opened and you would like to copy some audio between them or if you additionally use some other editor and want to copy audio into it.



Note: all the **Copy** options are available on the **Edit Tab**.

Pasting

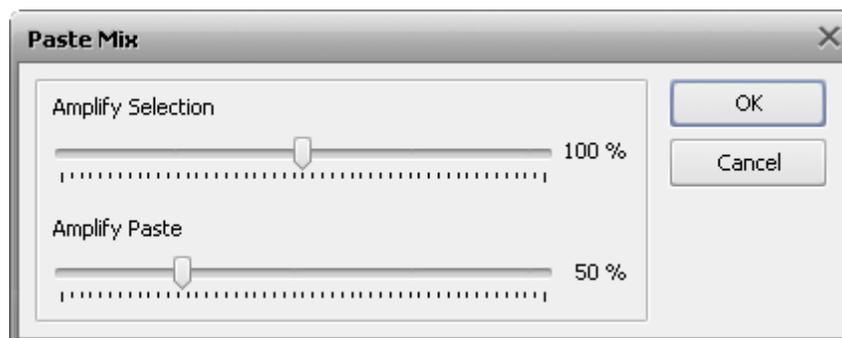
The following **Pasting** operation are available in **AVS Audio Editor**:

- 
Paste - use this option to paste (insert) the previously **copied** audio fragment to the currently edited audio file. It will be inserted right from the cursor position expanding the file to make it fit the inserted fragment. For instance, if you have a three-minute file and paste a copied fragment of 20 seconds, the resulting audio will be 3 minutes 20 seconds. If some part of the file is selected prior to the paste operation, it will be overwritten.
- 
Paste from File - use this option to paste (insert) an audio file to the currently edited audio file. After you select this option the **Open File** window will be opened to let you select an audio file. It will be inserted right from the cursor position expanding the file to make it fit the inserted fragment. For instance, if you have a three-minute file and paste an audio of 20 seconds, the resulting audio will be 3 minutes 20 seconds. If some part of the file is selected prior to the paste operation, it will be overwritten.
- 
Paste from Clipboard - use this option to paste (insert) the previously **copied** audio fragment to the currently edited audio file. This option is particularly useful when there are several instances of **AVS Audio Editor** opened and you would like to copy and paste some audio between them or if you additionally use some other editor and want to paste audio from it. It will be inserted right from the cursor position expanding the file to make it fit the inserted fragment. For instance, if you have a three-minute file and paste a copied fragment of 20 seconds, the resulting audio will be 3 minutes 20 seconds. If some part of the file is selected prior to the paste operation, it will be overwritten.



Note: to use this option with more than one instance of **AVS Audio Editor** you will need to use the **Copy to Clipboard** in one of them and **Paste to Clipboard** in the other, so that the selected fragment could be available to all the instances of the program.

- 
Paste Mix - use this option to paste (insert) the previously **copied** audio fragment to the currently edited audio file. It will be inserted right from the cursor position overwriting the original audio and not changing its duration. For instance, if you have a three-minute file and paste a copied fragment of 20 seconds, the resulting audio will be still 3 minutes. Before you paste some audio, you can change the following parameters:



Set **Amplify Selected** option to change the current audio file or selection amplification in relation to the rest of the audio and **Amplify Paste** to set the loudness of the audio in the inserted mix. If some part of the file is selected prior to the paste operation, the **Paste Mix** operation will be applied to it only.

- 
Paste Mix from File - use this option to paste (insert) an audio file to the currently edited audio file. After you select this option the **Open File** window will be opened to let you select an audio file. It will be inserted right from the cursor position overwriting the original audio and not changing its duration. For instance, if you have a three-minute file and paste an audio of 20 seconds, the resulting audio will be still 3 minutes. If some part of the file is selected prior to the paste operation, the **Paste Mix** operation will be applied to it only.

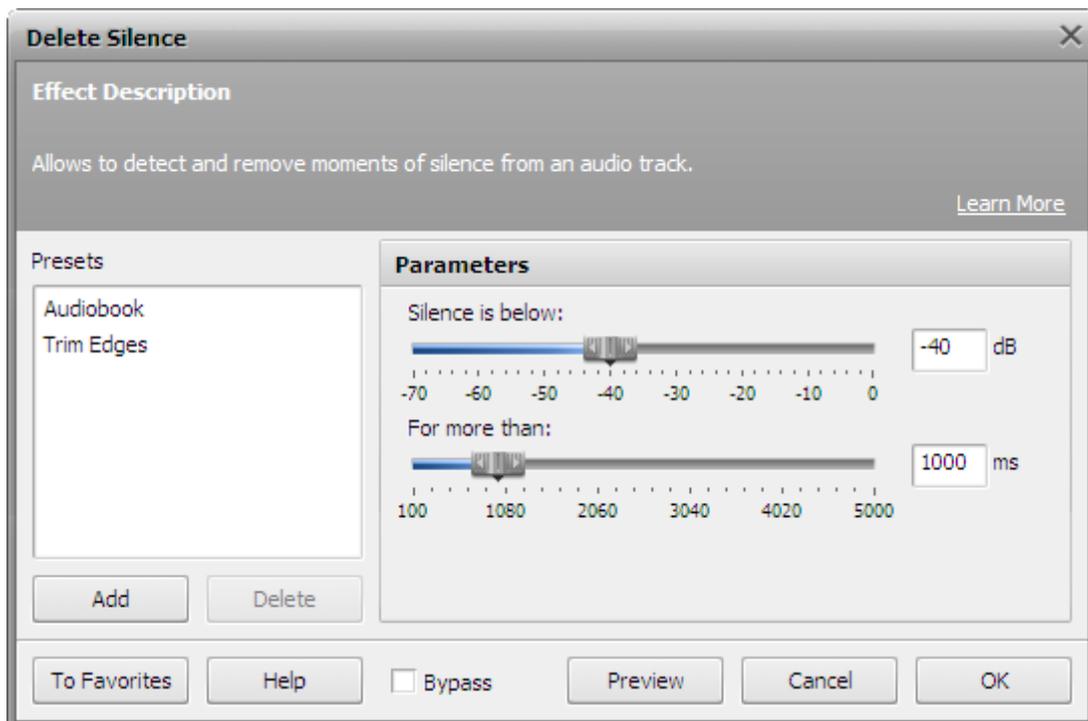


Note: all the **Paste** operations can be accessed from the **Edit Tab**.

Deleting and Trimming

The following **Deleting and Trimming** operations are available in **AVS Audio Editor**:

-  **Delete** - use this option to delete the currently selected fragment of the audio file. This option only allows you to delete the selected part of the audio, thus if no part of the audio file has been selected nothing will be deleted.
-  **Trim to Selection** - use this option to delete the all the audio in the file except the currently selected fragment. It is the opposite to the simple **Delete Selection** option. This option only allows you to delete some part of the audio if there is a selected part, thus if no part of the audio file has been selected nothing will be deleted.
-  **Delete Silence** - use this option to delete all the moments of silence in the audio. When you select this option the following window will be opened:



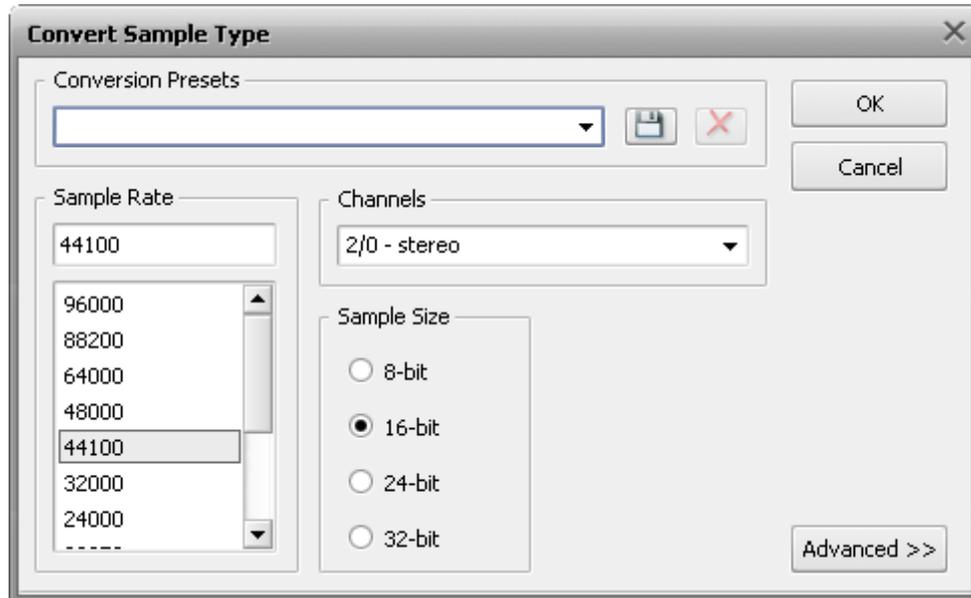
You will need to tell the program what exactly it should consider as moments of silence - you need to set the strength of the signal in decibels (for silence usually negative values are used) and its duration (in milliseconds). If some part of the audio is selected, the instances of silence will be deleted only in this selection.



Note: the **Delete** and **Trim to Selection** options are available on the **Edit Tab**, **Delete Silence** can be found on the **Effects Tab**.

Convert Sample Type

To convert the sample-rate, bit resolution, and channel format of an audio file that is currently loaded in **AVS Audio Editor** into a new format type (such as 44KHz/16-bit/stereo to 22KHz/8-bit/mono) use the **Convert Sample Type** command button on the **Edit Tab**.



Convert Sample Type directly processes samples within the file, or re-samples the data, so that the audio retains the same frequency and duration as the original file.

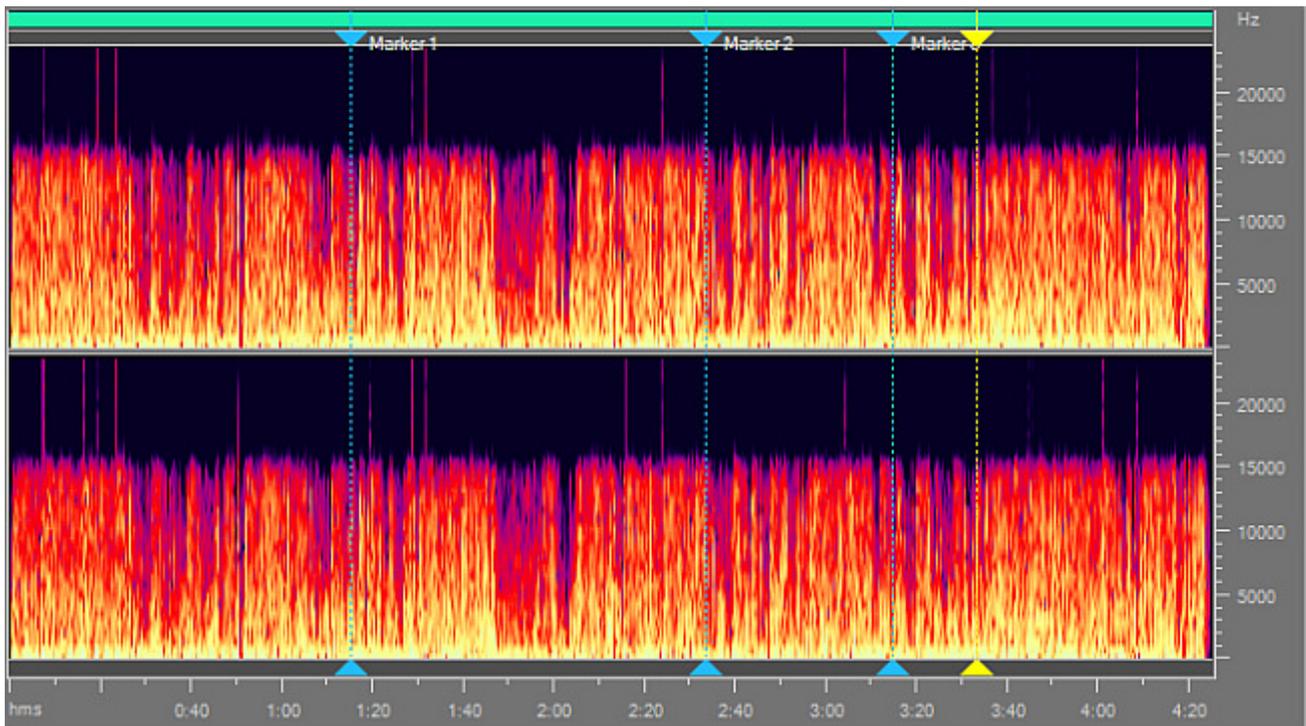
Every aspect of conversion can be adjusted to suit your needs. You can set the necessary **Sample Rate** (8'000 - 96'000 Herz), **Sample Size** (8-bit - 32-bit) and the **Number of Channels** (1/0 - mono -- 3/4+LFE 7.1 surround).

You can also use the advanced options of the resampling clicking the **Advanced >>** button (it will change into **<< Simple** once it is pressed). After you open the advanced settings panel of the **Convert Sample Type** window you will be able to set the following parameters:

- **Enable Dither** - use this option to apply an intentional form of noise, used to randomize the resampling errors thus preventing audio distortions. When you choose to enable dither, you can select the type of **Probability Density Function (p.d.f.)** used - **Triangular, Gaussian, Uniform** or **Colored** differing in type of randomizing in noise creation. It is also possible to change the **Noise Shaping** type - **A** or **B** - which is not actually dither, but rather a feedback process that has dither within it although it is used for the same purposes.
- **High Quality Resampling** - use this option to enable the high quality resampling, that will allow you to maximally avoid distortions. It will take more time and processor power than the sample type conversion without this option enabled.

Using Markers

The markers can be used to mark some moment in the audio to quickly return to it later or mark a selection to quickly copy or delete it.



To set a marker to some point in the audio you will need to click with the left mouse button within the audio timeline and after that press the **Add Marker**  button on the **File Tab**. Or double-click the left mouse button within the white bar between the **Display Range Bar** and **Waveform Display** (where the mouse standard cursor changes into a hand):

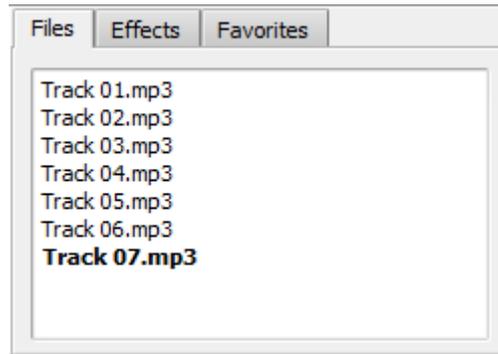


You can also set a marker to a selection. To do that either select a part of the audio with the mouse and click the **Add Marker**  button on the **File Tab**. Or make a selection and double-click the left mouse button within it on the white bar between the **Display Range Bar** and **Waveform Display** (where the mouse standard cursor changes into a hand):



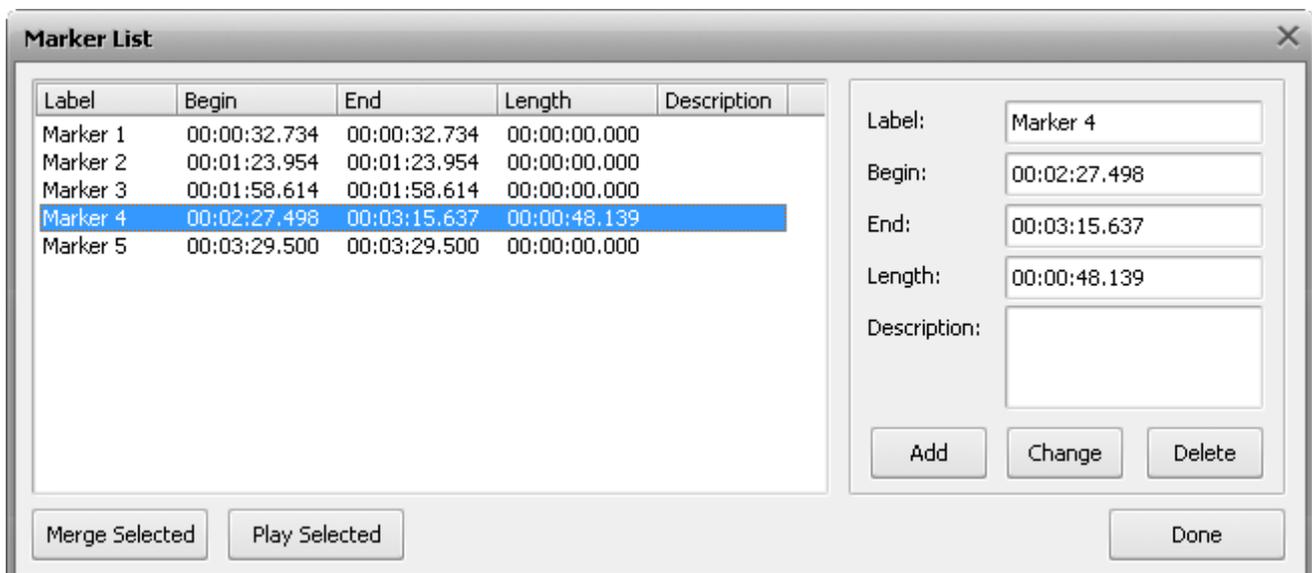
The set marker will have two boundaries - the first marking the beginning and the second the end of the selection.

Markers can be used to split the audio file into several separate ones. To do that you will need to set the markers to the necessary places and click the **Split by Markers** command button on the **File Tab**. The audio file will be divided into several ones that will begin and end at the points where the markers are placed. They will be named as **Unknown + the order number** and put to the **Files** tab of the **Effects and Filters Panel**:



After that you can edit each of the resulting files individually.

You can see the whole list of the set markers, change their names and place and delete the unnecessary ones using the **Marker List**. Press the **View Marker List** command button on the **File Tab** to open it:



Here you can also select several markers to merge them or playback only audio that is contained within the markers boundaries. Use the left mouse button and the **Shift** or **Ctrl** keys on the keyboard to select a group of markers or separate markers on the list.

Click the **Merge Selected** button to join together the selected marked fragments, or **Play Selected** to playback them (the **Play Selected** button will turn into **Stop** once you press it to let you stop playback).

In the right part of the **Marker List** window more marker properties can be entered manually: **Label** - to give a marker a more meaningful name, **Begin** and **End** to precisely alter the marker initial and final limit, **Length** to change the length of the markers selection and **Description** to give a marker a wider description so that you could understand later what this exact marker has been set for.

Click the **Add** button to create a new marker, **Change** to edit the current marker or **Delete** to remove the current marker from the **Marker List** and from the audio.

When you have finished with the marker editing, press the **Done** button to save the changes and close the **Marker List** window.

Editing Individual Channels

Normally all channels of the audio file are edited simultaneously. You can, however, edit channels independently by disabling the one or several of them. To do so, use the **Control** key of the keyboard together with the mouse pointer (the cursor will acquire **L** or **R** letter) and left-click to enable only the necessary channel for editing (the disabled channels will gray-out), another left-click within the disabled channel will re-enable it.



The possible cursor values include (for 8-channel audio, for instance):

- **L_f** - Left front channel
- **R_f** - Right front channel
- **C_f** - Central front channel
- **LFE** - Low-Frequency Effects (subwoofer) channel
- **L_r** - Left rear channel
- **R_r** - Right rear channel
- **L_s** - Left surround channel
- **R_s** - Right surround channel

Applying Effects to Individual Channels

You may apply a limited number of effects to individual channels. Channels that change the audio track duration or resample it will be applied to all audio channels regardless of whether they are disabled or not.

The following effects will be applied to all audio channels:

- Time Stretch
- Pitch Shift
- Rate Change
- Reverse

Pasting in Individual Channels

When pasting, the audio data is always overlapped with what is already there, since inserting only in one channel will put the channels completely out of phase.

Using Effects and Filters

The main audio editing options of the **AVS Audio Editor** include:

- **Amplitude effects**
- **Delay/Modulation effects**
- **Time Stretch/Pitch Shift effects**
- **Filters**
- **DX Filters**

All of them are available on the **Effects Tab** or on the **Effects and Filters Panel**.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. To apply an audio effect or filter to your audio file, follow these steps:

1. Add the audio file to the **Waveform Editing Space**.
2. Select the part of the file you would like to apply the effect to. Leave it unselected to apply the effect to the whole file.
3. Click the needed effect or filter button on the **Effects Tab** or double-click the effect on the **Effects and Filters Panel**.
4. Set the effect parameters in the opened window.



Note: if you change the effect settings, you can save them as a preset to be able to use them afterwards. Use the **Add** button to do that.

5. Click the **Preview** button to make sure that the changes are correct.
6. Click the **OK** button to apply the effect with the selected parameters and go back to the audio file editing.

Amplitude Effects

Using the **Effects Tab** or **Effects and Filters Panel** options you can increase or decrease the volume of a waveform or selection. The following **Amplitude Effects** are available:

- **Amplify**
- **Fade In/Fade Out**
- **Normalize**
- **Compressor**
- **Envelope**
- **Mute**
- **Invert**

Amplify

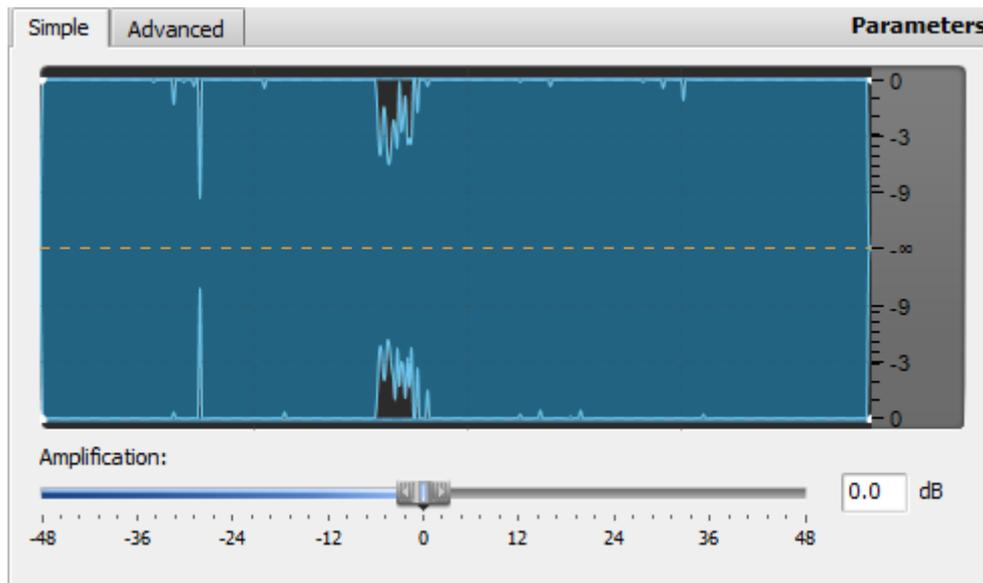
 The **Amplify** effect is used to increase or decrease the volume of the sound by means of increasing or decreasing its amplitude.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Amplify Settings - Simple Mode

The **Simple** mode allows to amplify/attenuate an audio signal and visually check the results on a waveform. The applied amplification will equally affect all sound channels of the edited file.

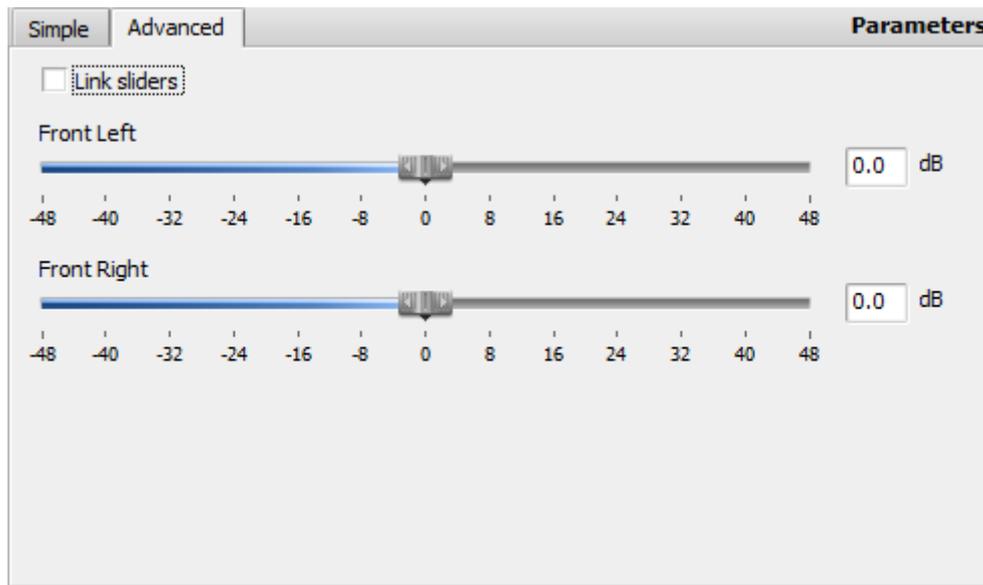


Use the slider of the **Amplification** scale to set the desired amplification of your audio file in dB. Alternatively, enter the desired amplification (gain) value in the text box next to the scale.

Setting a positive value will increase the sound volume of your audio file, setting a negative value will decrease sound volume. The more the set value (positive or negative) is, the more the resulting volume will be increased or decreased.

Amplify Settings - Advanced Mode

The **Advanced** mode allows to apply the **Amplify** effect individually to each sound channel. For this purpose there is a separate **Amplify** slider available for each channel.



To set the desired amplification for a particular channel, use the corresponding slider or text box. The setting is done in the same way as in the **Simple** mode (see above).

If you enable the **Link sliders** option, all sliders will change their position to that of the upper slider. With this option enabled, moving any slider will also move the rest of the sliders to the same position.

If a channel is not selected in the **Waveform Editing Space**, its respective slider will be disabled on the **Advanced** tab.



Note: The settings of the **Simple** and **Advanced** tabs can be changed independently. When you apply or preview the effect, the settings of the active tab are used.

Fade In and Fade Out

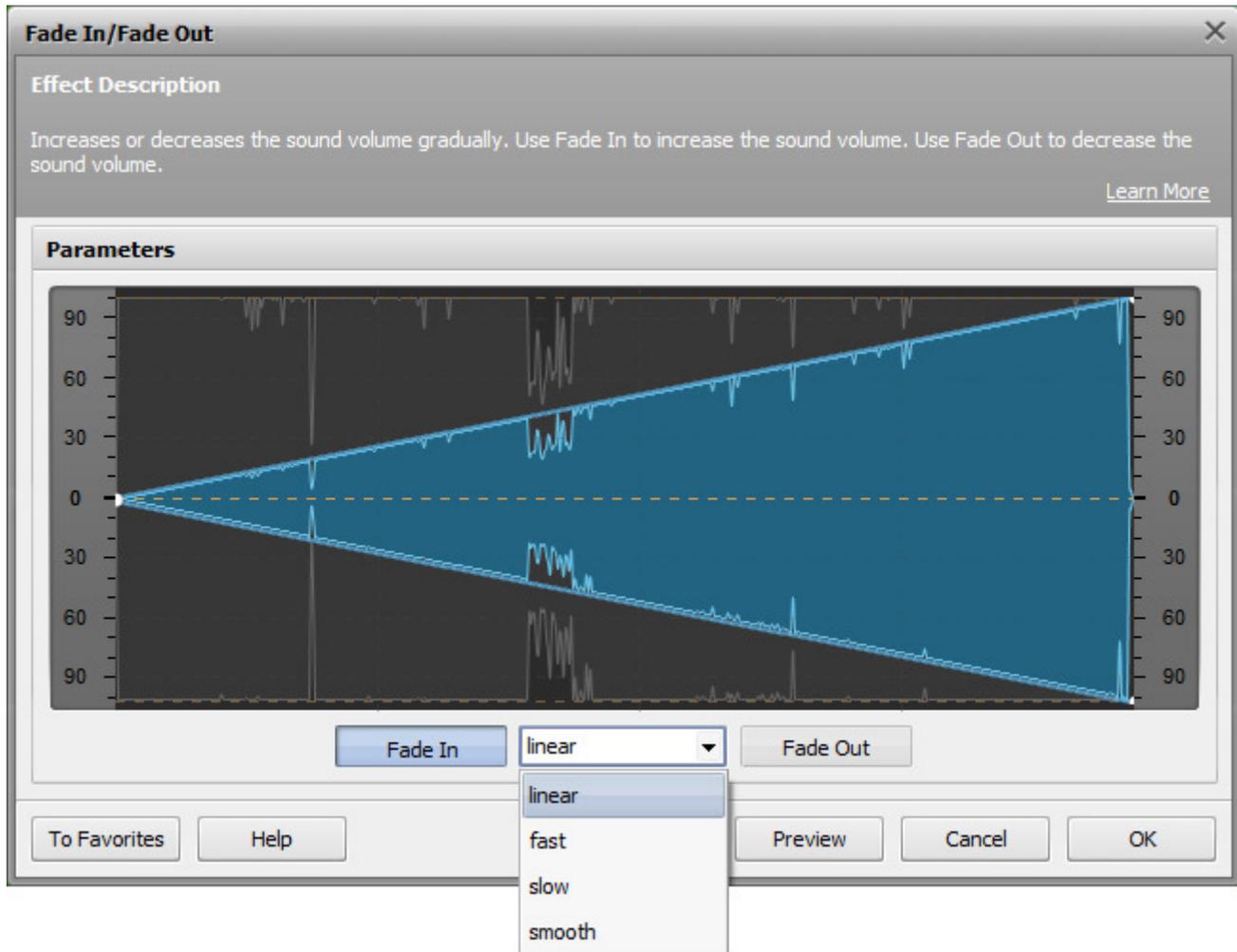
 The **Fade In** and **Fade Out** effects are usually used to smooth the boundaries of an audio track. Use the **Fade In** effect to gradually increase the sound volume. Use the **Fade Out** effect to gradually decrease the sound volume.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Fade In/Fade Out Settings

You may adjust the following parameters in the **Fade In/Fade Out** settings window:



First select whether you want to apply the **Fade In** or **Fade out** effect by pressing the corresponding button. Next select one of the four fade types from the **Spline** drop-down list:

- **linear** - use it to evenly increase or decrease the sound volume;
- **slow** - use it to slowly increase or decrease the sound volume;
- **fast** - use it to quickly increase or decrease the sound volume;
- **smooth**(selected by default) - use it to smoothly increase or decrease the sound volume.

The selected spline graph is displayed in the settings window so you may visually judge the way the sound of your audio file will be changed.

Normalize



The **Normalize** effect is used to even the volume of an audio track in accordance with the selected level.

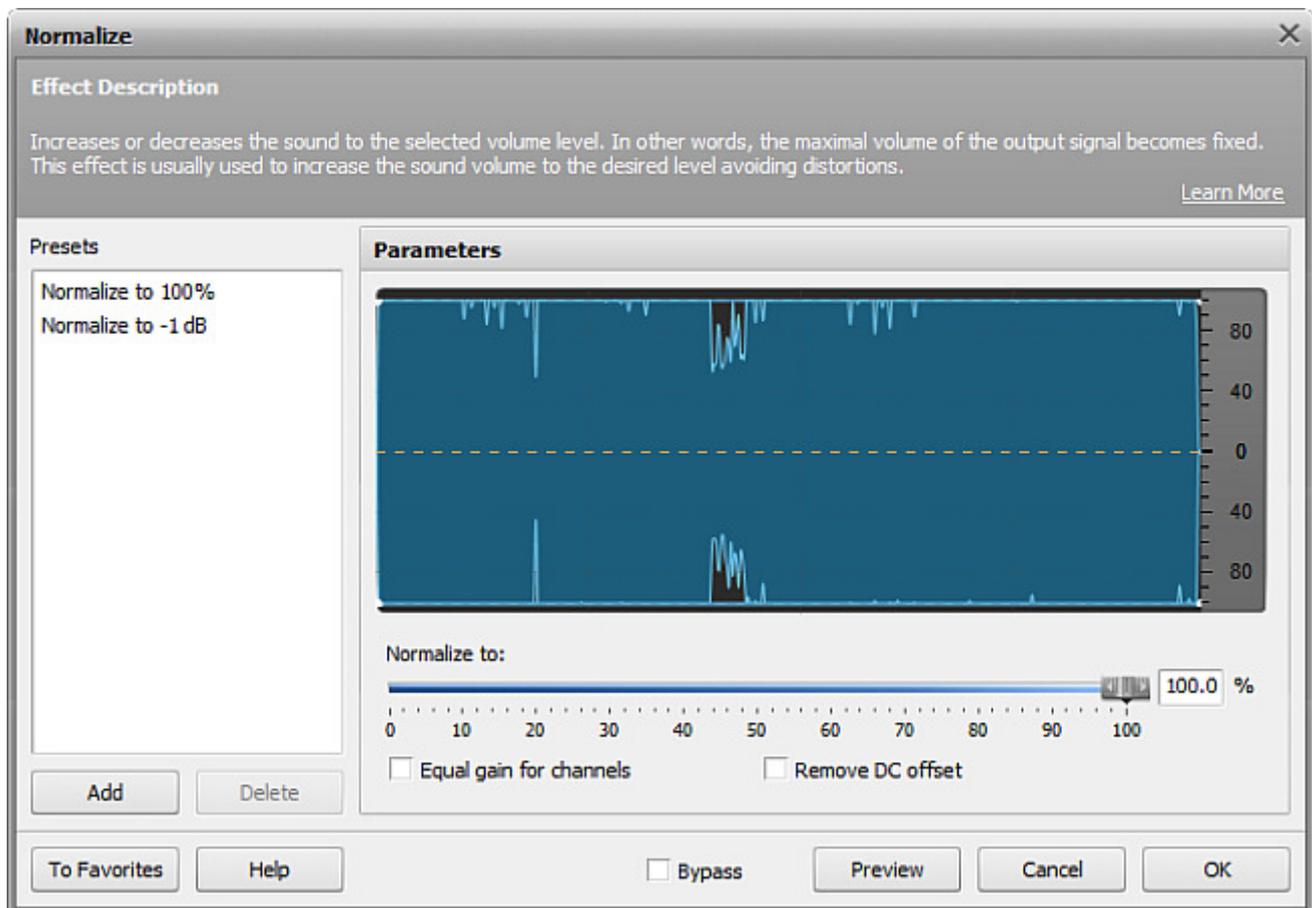
How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Normalize Effect Settings

The software will define the maximum level of the audio signal and calculate the ratio by which this maximum signal should be amplified/attenuated in order to fit the specified **Normalize to** level. If the maximum sound level of the audio file is above the specified **Normalize to** level, then the rest of audio will be attenuated by the same ratio. If the maximum sound level of the audio file is below the specified **Normalize to** level, then the rest of audio will be amplified by the same ratio.

The following **Normalize** effect parameters are available for editing:



- **Normalize to** - sets the maximum level of the output signal. If you set it to 100%, the maximum level of the input signal will be amplified so as to reach the maximum level of amplitude that **AVS Audio Editor** can produce without clipping. Setting it to 0% will mute the original audio completely.
- **Equal gain for channels** - tick this option to apply one and the same gain ratio to all channels. Enable this option to avoid changes in the stereo balance of the output audio.
- **Remove DC Offset** - tick this option to remove DC offset. Enable this option if the DC offset is present in the original file. The offset presence can be visually noticed on the waveform view: if an audio track is not centered along the horizontal line, it is advisable that the **Remove DC Offset** option is used to remove the offset.

Compressor



The **Compressor** effect is used to reduce the dynamic range of an audio signal - make quiet sounds louder, and loud sounds quieter. **AVS Audio Editor** will apply the **Compressor** effect along with the **Noise Gate** filter. This will help get rid of unnecessary noise by eliminating audio signal below the specified threshold.

How to Apply

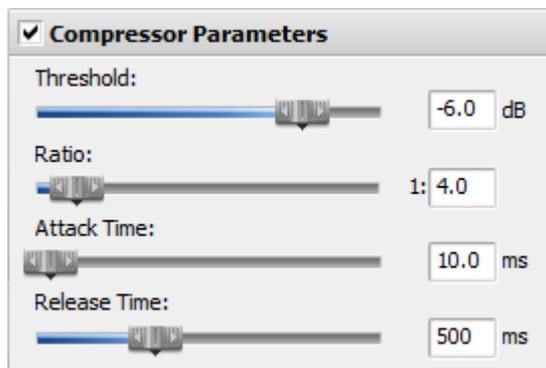
All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Compressor Settings

The Compressor effect allows to configure the following parameters:

Compressor Parameters

The following compressor parameters are available for editing:



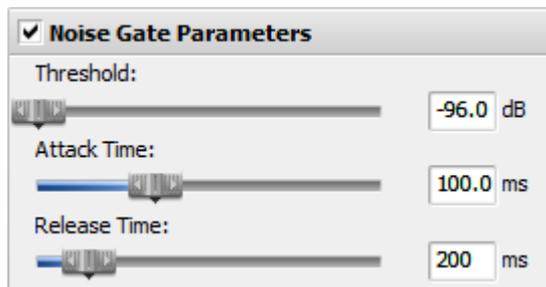
- **Threshold** - the level (in dB) at which the compressor becomes active. The compressor will affect only the sound above the specified threshold.
- **Ratio** - the amount of compression of the output signal in relation to the input audio. Setting the compression ratio to 1:1 will produce no compression effect. If you set ratio to 1:3 and threshold to -20, a -2 dB sound will be compressed down to -14 dB (the difference between -20 and -2 will be divided by 3).
- **Attack Time** - the period during which the compressor becomes active and decreases gain till it reaches the specified ratio.
- **Release Time** - the period during which the compressor remains active once the sound has fallen below the threshold.



Note: the attack and release times are used to smooth the compressor boundaries. The slower the attack and release are, the smoother the transition between affected and non-affected signal will be.

Noise Gate Parameters

The following Noise Gate parameters are available for editing:



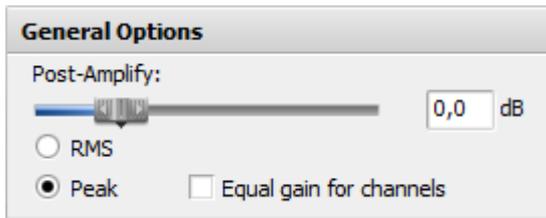
- **Threshold** - the level (in dB) at which the gate opens. The noise gate filter will affect only the sound below the specified threshold.
- **Attack Time** - the amount of time it takes the noise gate to change from closed to open once the signal exceeds the specified threshold.
- **Release Time** - the period during which the noise gate remains open once the signal has fallen below the specified threshold.



Note: the attack and release times are used to smooth the noise gate boundaries. The slower the attack and release are, the smoother the transition between affected and non-affected signal will be.

General Options

The following general parameters are available for editing:



- **Post-Amplify** - the amount of post-amplification used after the compressor has been applied. This will allow you to restore the original sound level after applying the compressor.
- **RMS/Peak** - a method of measuring the sound level. Tick the corresponding radiobutton to select the desired method.
- **Equal gain for channels** - allows to synchronize channels; the compression and noise suppression ratios, as well as the moments of noise gate opening will

be the same for all channels.

Envelope



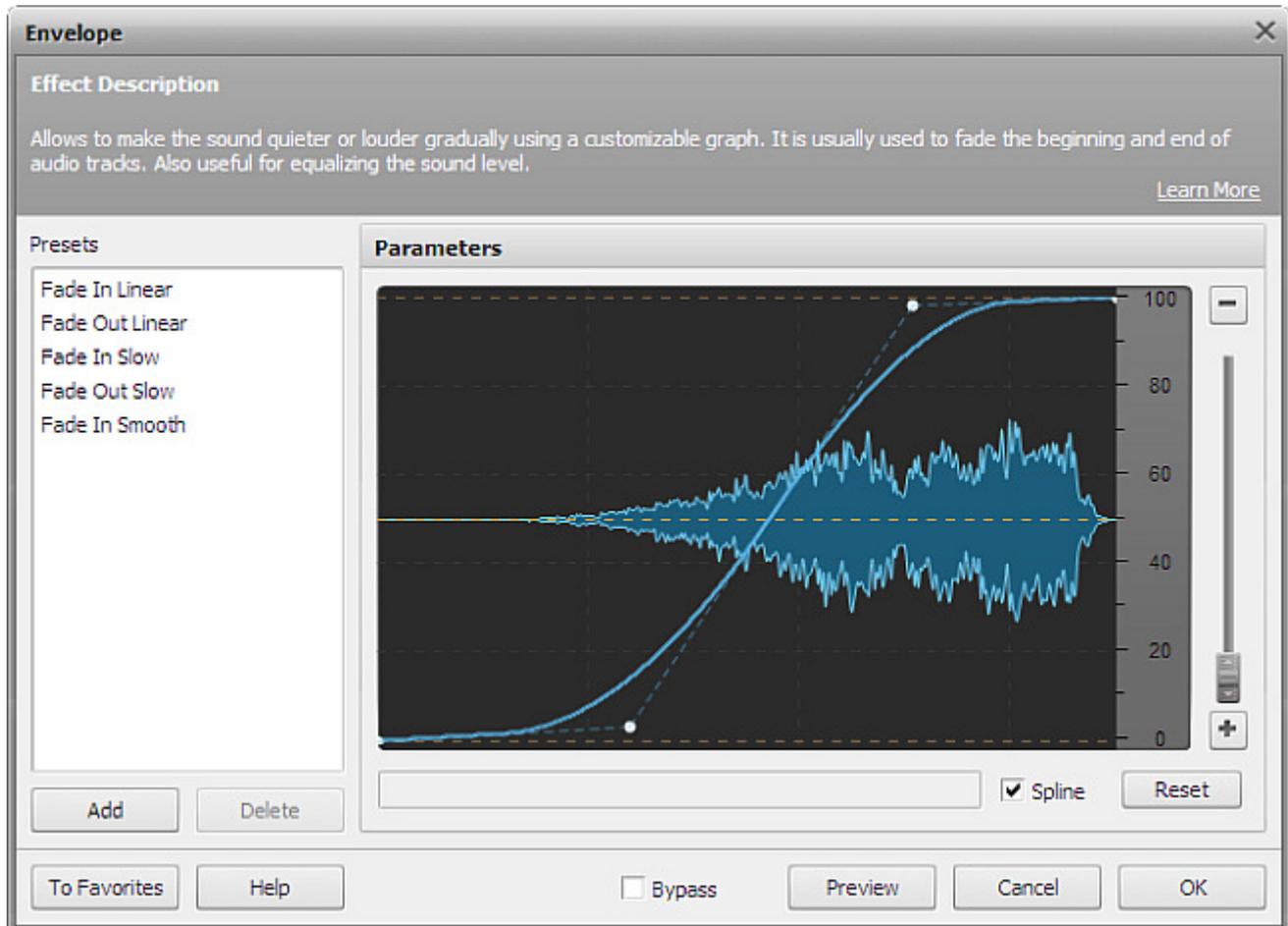
The **Envelope** effect allows you to increase the volume of some specific fragments of an audio track and decrease the sound volume of other fragments. It is usually used to smooth the beginning or end of audio tracks. It is also useful for creating audio loops and samples.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Envelope Settings

The **Envelope** effect is achieved by changing the audio file amplitude in accordance with the specified coordinates. To modify the amplitude, please use the following settings:



- use the **envelope scale** to set the maximum output level. If you set the scale to 400% with the envelope line going along the top, this will make the output sound 4 times as loud as the original. If you zoom in the scale up to 100%, the maximal output volume will correspond to the original, and you will be able only to decrease the output volume.
- use **envelope points** to modulate the volume of an audio track. To add a point, double-click on the envelope line. To delete a point, drag it anywhere beyond the limits of the waveform display. Drag the point upwards to amplify the original sound. Drag it downwards to decrease the sound volume.
- check the **Spline** option to smooth the envelope graph.

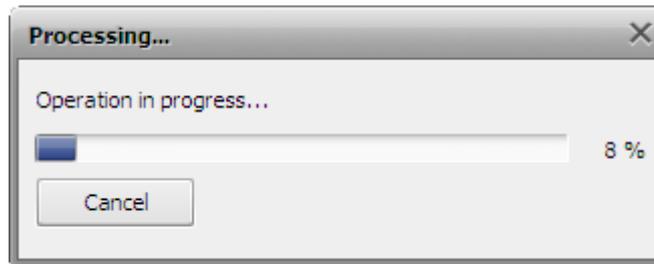
Mute



Mute effect is used to switch off the sound in the edited audio file.

How to Apply

1. Add the audio file to the **Waveform Editing Space**.
2. Select the part of the file you would like to apply the effect to. Leave it unselected to apply the effect to the whole file.
3. Click the **Mute** button on the **Effects Tab** or double-click the **Mute** effect on the **Effects and Filters Panel**.
4. Wait while the program applies the effect. Click **Cancel** at any time to abort the process.



Settings

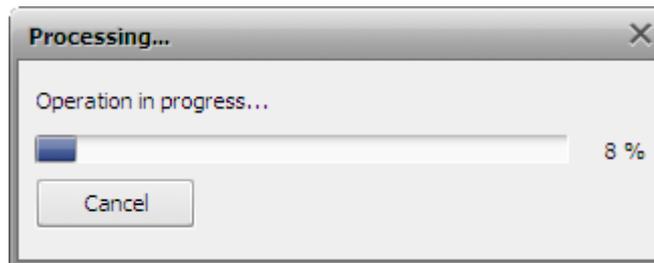
This effect does not have any settings and will be simply applied so that either the selected part of the audio file will be muted or the whole audio if nothing has been selected.

Invert

 The **Invert** effect inverts the sound around the horizontal axis. Usually this effect is used to change the polarity of one of the stereo channels in order to put them in phase. For more information, please refer to the Phase Analysis page.

How to Apply

1. Add the audio file to the **Waveform Editing Space**.
2. Select the channel you would like to apply the effect to by using the **Control** key of the keyboard together with the mouse pointer (the cursor will acquire **L** or **R** letter) and left-clicking.
3. Click the **Invert** button on the **Effects Tab** or double-click the **Invert** effect on the **Effects and Filters Panel**.
4. Wait while the program applies the effect. Click **Cancel** at any time to abort the process.



Settings

This effect does not have any settings and will be simply applied so that either the selected channel will be inverted or the whole audio if nothing has been selected.

Delay/Modulation Effects

Using the **Effects Tab** or **Effects and Filters Panel** options you can apply different effects to your audio. The following **Delay/Modulation Effects** are available:

- **Echo**
- **Chorus**
- **Flanger**
- **Phaser**
- **Reverb**
- **Vibrato**

Echo



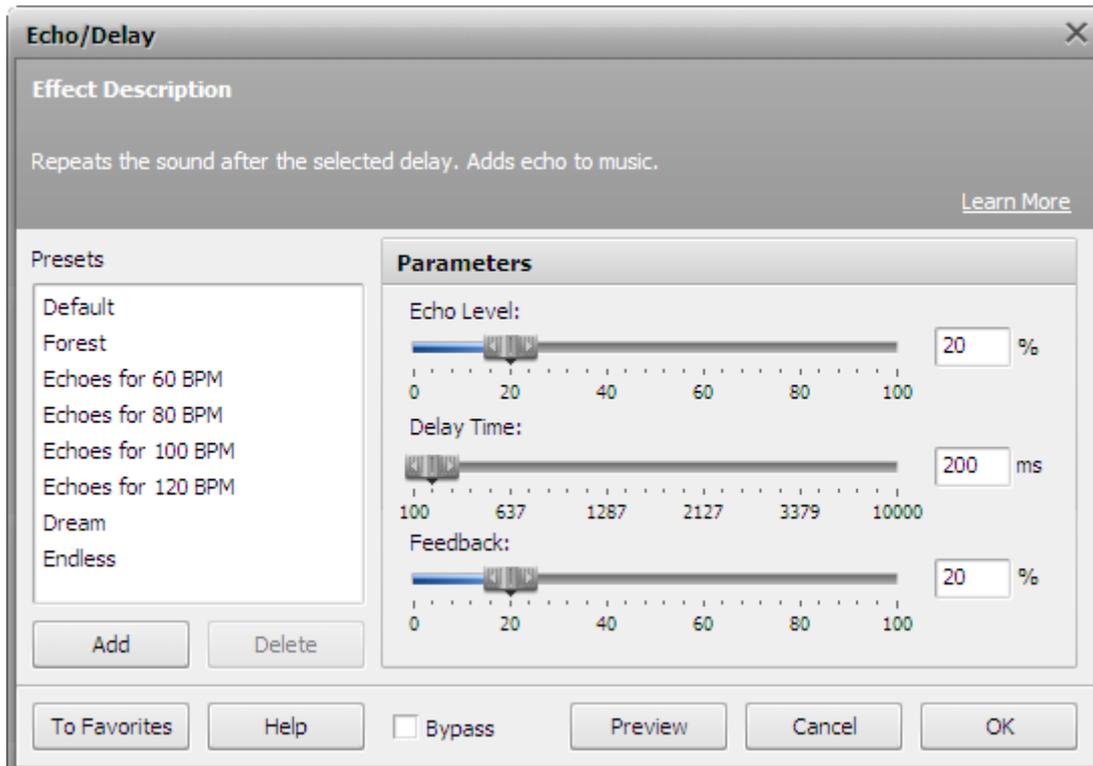
The **Echo** effect allows you to create an echo effect of your audio track by replaying the sounds of the selected audio portion after a certain period of time.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Echo Settings

To apply the effect, use one of the available presets or configure the effect parameters manually.



The following **Echo** settings are available for editing:

- **Echo Level** - defines the delayed sound volume level in relation to the original sound. If you set it to 100%, the original sound and its echo will have equal volume. If you set it to 0, no echo will be heard at all.
- **Delay Time** - sets the point in milliseconds at which echo starts behind the original signal. The minimal delay time is 100 ms. If you set a bigger value, the distance between the original and delayed sound will grow.
- **Feedback** - determines how much of the affected sound is sent back to the input to be processed again. Use it for multiple echoes.

Chorus

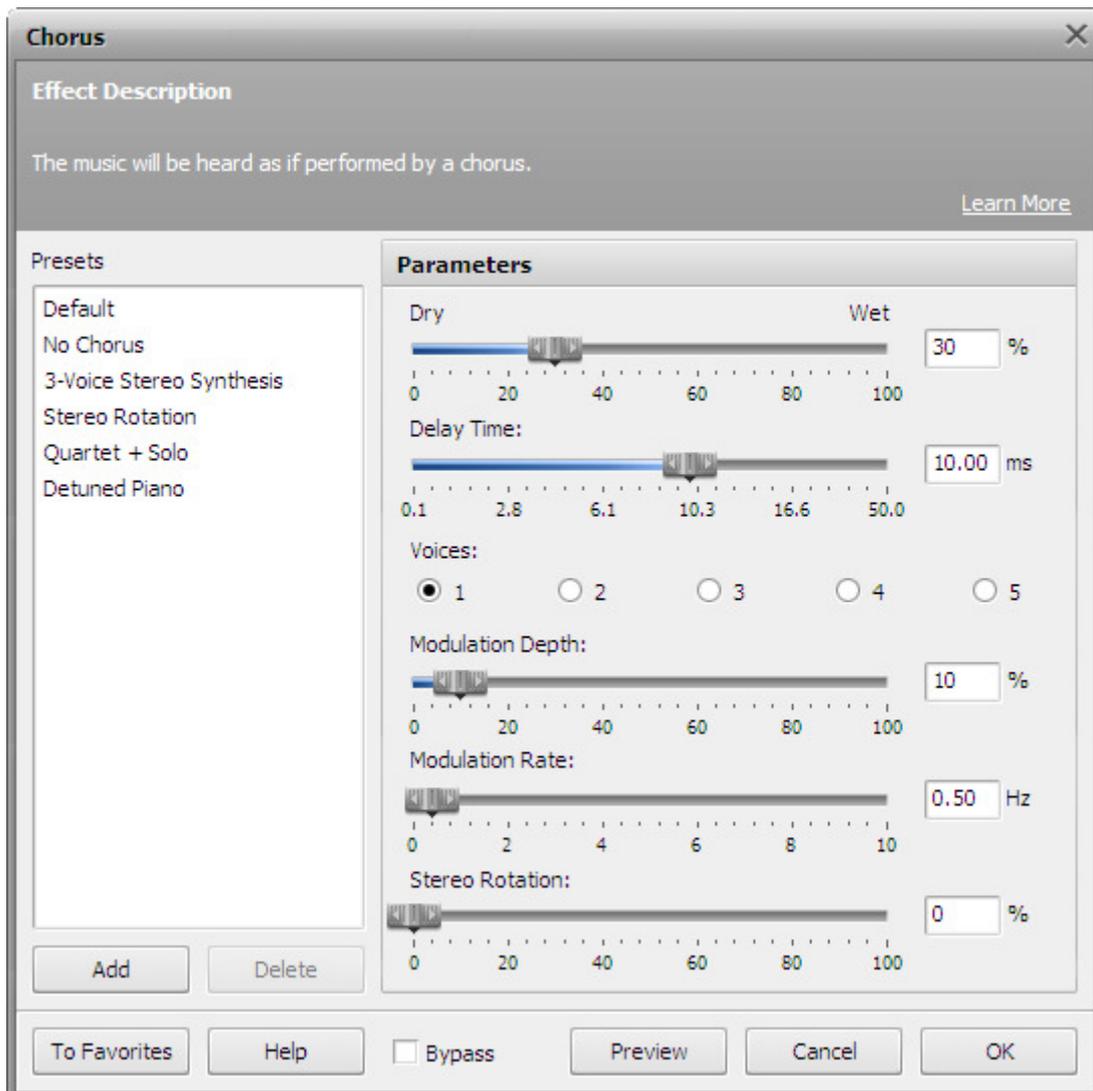
 The **Chorus** effect allows you to make your audio sound fuller. It can make a single instrument sound like there are actually several instruments being played or the recording of a vocal track sound like it was sung by two or more people singing in chorus.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Chorus Settings

The **Chorus** effect is achieved by adding one or more delayed signals to the original input. To apply the effect, use one of the available presets or configure the effect parameters manually.



The following **Chorus** settings are available for editing:

- **Dry/Wet Balance** - sets the amount of the affected sound as related to the original. The parameter values vary between 0% and 100%. If you set the balance to 50%, the original and affected sound will be mixed in equal proportions. Setting it to 100% will mute the original sound, setting it to 0% will mute the affected sound.
- **Delay Time** - sets the point in milliseconds at which echo starts behind the original signal. The default delay time is 10 ms. If you set a bigger value, the distance between the original and delayed sound will grow.
- **Voices** - defines the number of delayed signals that will be added to the original audio. A duplicate delay time is used for the second voice, a triple value is used for the third voice etc.
- **Modulation Depth** - controls the degree of delay time change. This parameter sets the limits within which the delay varies. The modulation depth values may vary from 0% to 100%. Setting the parameter to 0% will result in a constant delay time.

Setting it to 100% will make the delay time change from its original value (the one you set in the **Delay Time** field) to a duplicate **Delay Time** value.

- **Modulation Rate** - sets the number of delay time variations, in Hertz. This parameter controls how often the delay time oscillates. For example, if you set the parameter to 3 Hz, it means that the delay time will oscillate (make the full cycle from the smallest to the biggest delay value and back to the smallest) 3 times per second.
- **Stereo Rotation** - sets the left and right delays at separate values, creates a stereo rotation effect. The parameter values vary from 0% to 100%: 0% means no rotation; 100% corresponds to maximum stereo rotation effect.

Flanger



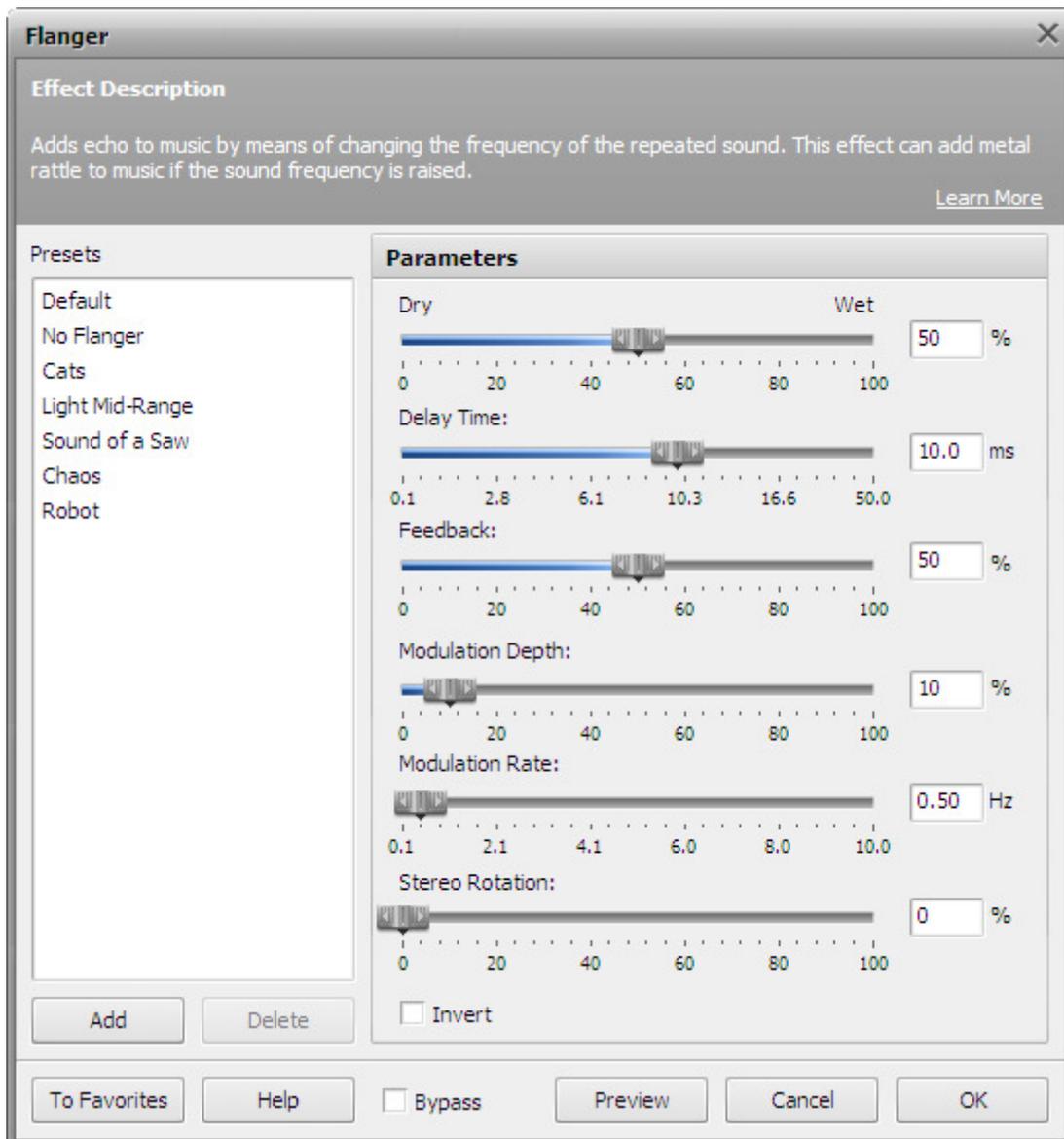
Flanger is a special case of the Chorus effect: it is created in the same way as Chorus. Flanging used to be created by putting a finger onto the tape reel's flange, thus slowing it down. Two identical recordings are played back simultaneously, and one is slowed down to give the flanging effect. Flanger gives a "whooshing" sound, like the sound is pulsating. It is essentially an exaggerated Chorus.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Flanger Settings

The **Flanger** effect is achieved by mixing a signal with a slightly delayed copy of itself, where the length of the delay is constantly changing. To apply the effect, use on the available presets or configure the effect parameters manually.



The following **Flanger** settings are available for editing:

- **Dry/Wet Balance** - sets the amount of the affected sound as related to the original. The parameter values vary between 0% and 100%. If you set the balance to 50%, the original and affected sound will be mixed in equal proportions. Setting it to 100% will mute the original sound, setting it to 0% will mute the affected sound.
- **Delay Time** - sets the point in milliseconds at which echo starts behind the original signal. The default delay time is 10 ms. If you set a bigger value, the distance between the original and delayed sound will grow.

- **Feedback** - determines how much of the affected sound is sent back to the input to be processed again.
- **Modulation Depth** - controls the degree of delay time change. This parameter sets the limits within which the delay varies. The modulation depth values may vary from 0% to 100%. Setting the parameter to 0% will result in a constant delay time. Setting it to 100% will make the delay time change from its original value (the one you set in the **Delay Time** field) to a duplicate **Delay Time** value.
- **Modulation Rate** - sets the number of delay time variations, in Hertz. This parameter controls how often the delay time oscillates. For example, if you set the parameter to 3 Hz, the delay time will oscillate (make the full cycle from the smallest to the biggest delay value and back to the smallest) 3 times per second.
- **Stereo Rotation** - sets the left and right delays as separate values, creates a stereo rotation effect. The parameter values vary from 0% to 100%: 0% means no rotation; 100% corresponds to maximum stereo rotation effect.
- **Invert** - inverts the affected sound when mixing it with the original.

Phaser



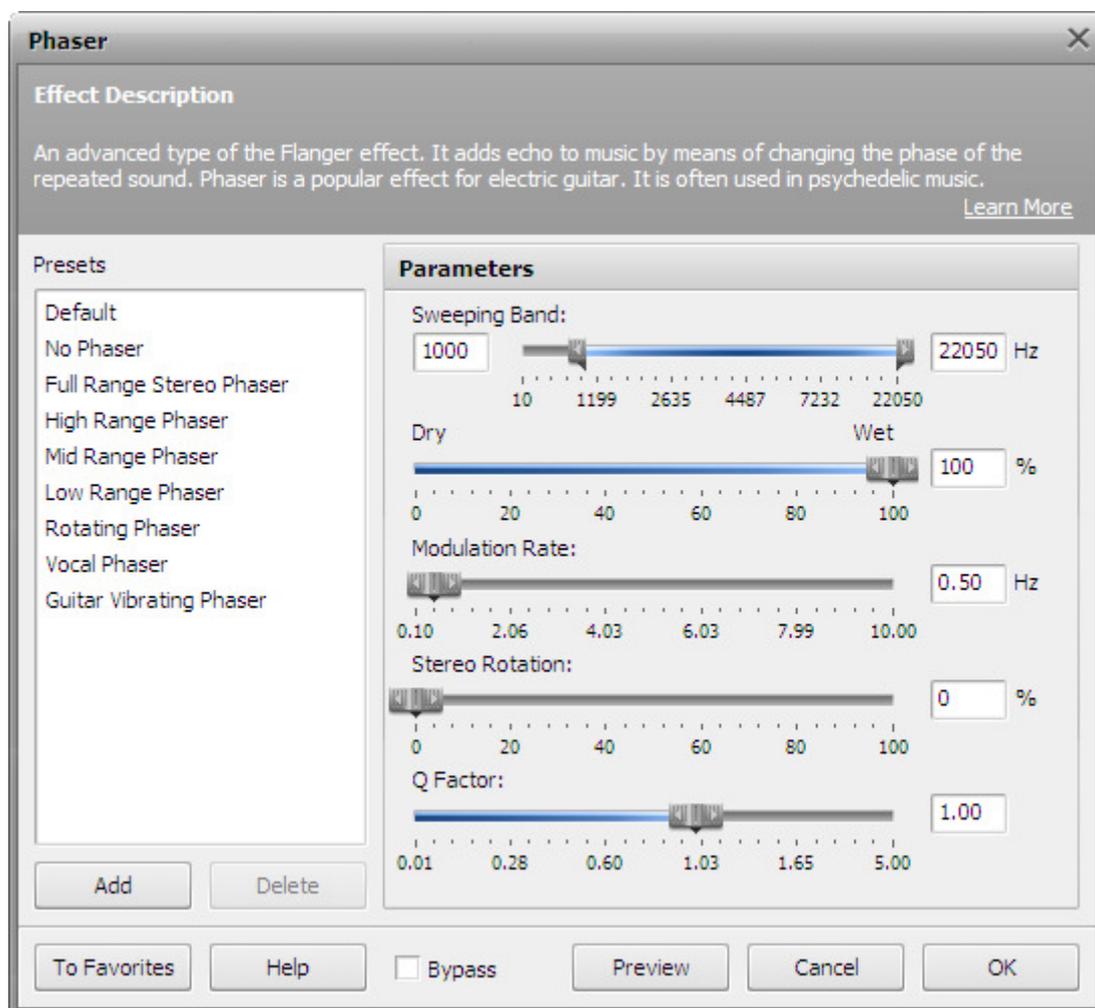
The **Phaser** effect is generally used to give electronic effect to natural sounds.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Phaser Settings

The phaser effect is created by means of passing a band of frequencies through a phase shifter and then mixing them with the original audio signal. The original and affected signals, when mixed, will cancel each other and create a notch if they are out of phase. If, however, they are partially out of phase, then partial cancellations occur. The phase shifter will change the phase of the signal by 180° at its center notch frequency (thus, putting the original and affected sound out of phase and cancelling them). The band of frequencies around the center notch frequency that will be affected by the phase shifter is determined by the Q factor.



The following **Phaser** settings are available for editing:

- **Sweeping Band** - determines the notch frequency oscillation range (the lowest and the highest notch frequency oscillation limits).
- **Dry/Wet Balance** - sets the amount of the affected sound as related to the original. The parameter values vary between 0% and 100%. If you set the balance to 50%, the original and affected sound will be mixed in equal proportions. Setting it to 100% will mute the original sound, setting it to 0% will mute the affected sound.
- **Modulation Rate** - sets the number of oscillations of the notch frequency per second. This parameter controls how often the notch frequency oscillates. For example, if you set the parameter to 3 Hz, the notch frequency will oscillate (make the full cycle from the lowest to the highest frequency value and back to the lowest) 3 times per second.
- **Stereo Rotation** - sets the left and right notch frequencies at separate values, creates a stereo rotation effect. The parameter

values vary from 0% to 100%: 0% means no rotation; 100% corresponds to maximum stereo rotation effect.

- **Q Factor** - determines the range of frequencies affected by the phase shifter. The bigger the value, the narrower the notch frequency band is.

Reverb

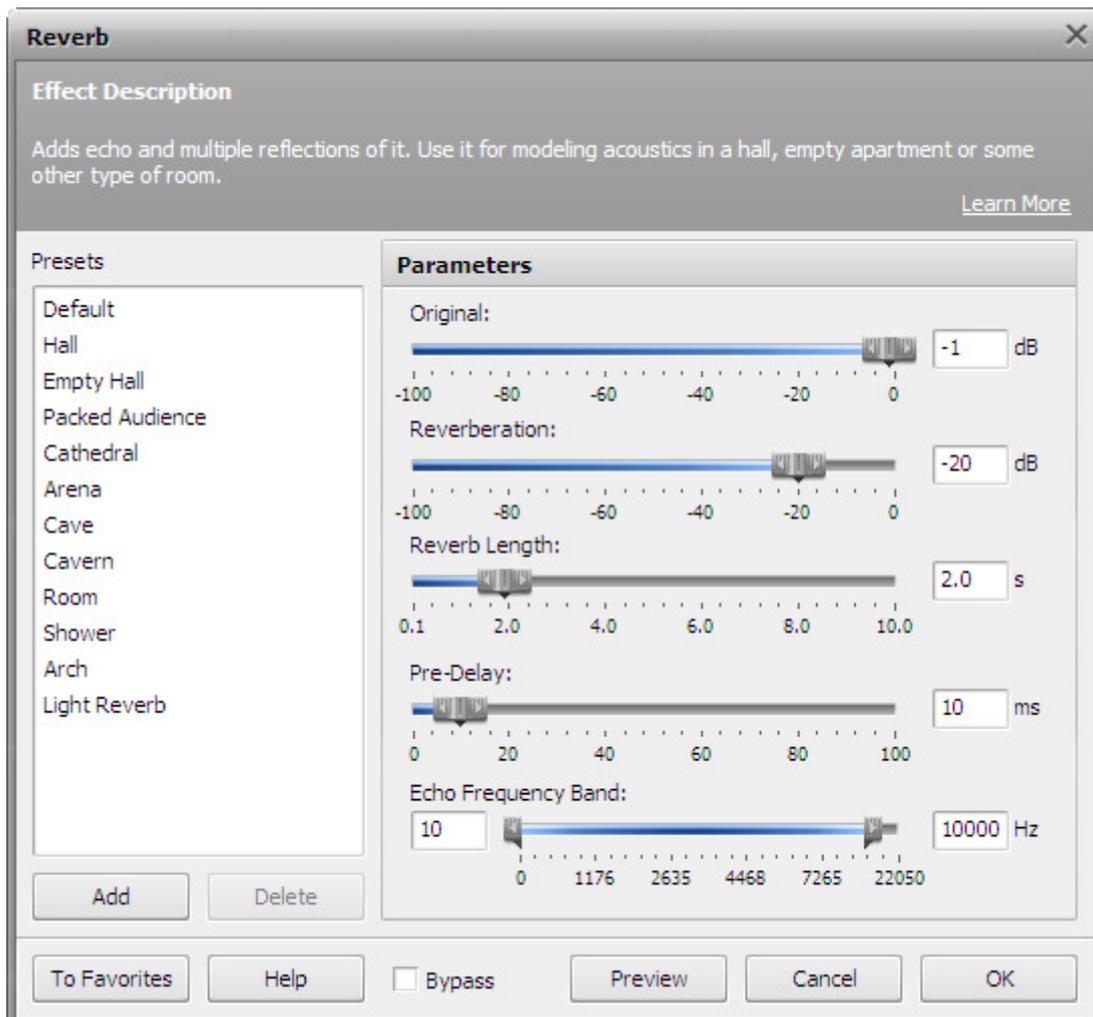
 The **Reverberation** filter helps you apply a particular effect when the sound stops but the reflections continue, decreasing in amplitude, until they can no longer be heard. The **Reverb** effect is used to simulate the acoustical effect of rooms and enclosed buildings.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Reverb Settings

To apply the effect, use on the available presets or configure the effect parameters manually.



The following **Reverb** effect parameters are available for editing:

- **Original** - sets the amplification value of the original audio signal that will be present in the output audio.
- **Wet** - sets the amplification value of the reverberated audio signal that will be present in the output audio.
- **Reverb Length** - defines the amount of time it takes the reverberations to fade out (decrease their amplitude gradually by 60 dB).
- **Pre-Delay** - sets the point in milliseconds at which reverberation starts behind the original signal. You may think of it as the time it takes the audio signal to reach the imaginary "wall" and start being reflected from it. Setting bigger values will simulate reverberations in a bigger room.
- **Echo Frequency Band** - sets the frequency range that will be processed and reverberated. Frequencies beyond this range will not be affected.

Vibrato



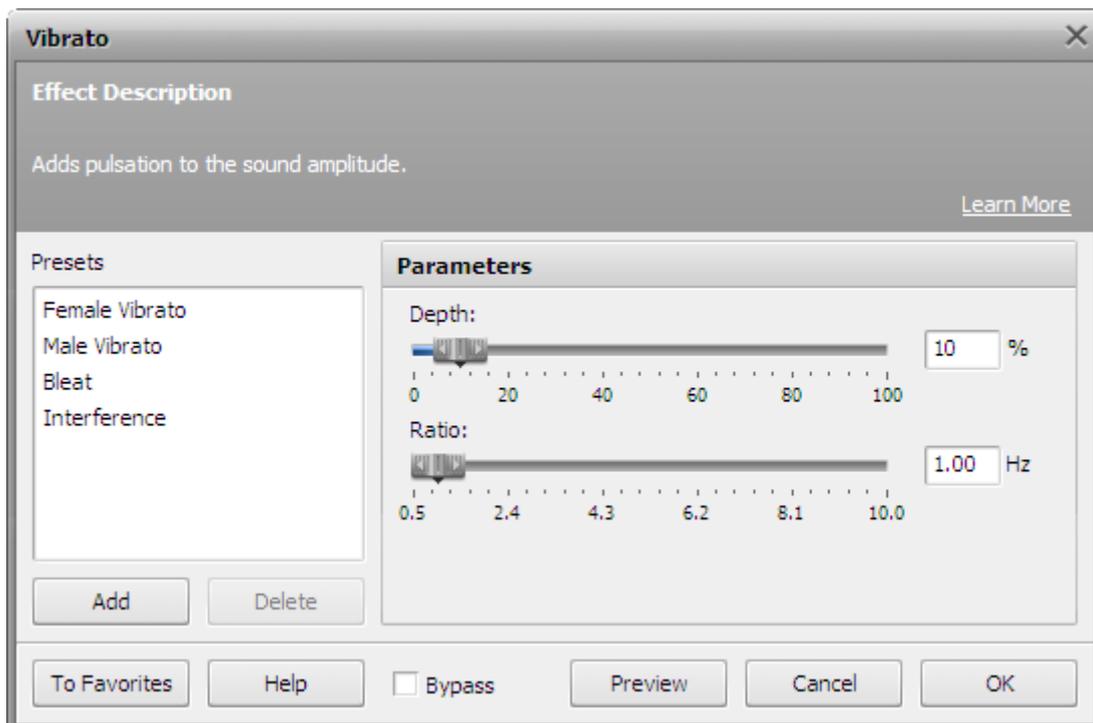
The **Vibrato** effect creates an effect of pulsating sound.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Vibrato Settings

The Vibrato effect is achieved by a periodical variation in the sound amplitude. To apply the effect, use on the available presets or configure the effect parameters manually.



The following **Vibrato** settings are available for editing:

- **Depth** - determines the amplitude variation limits. The smaller the value, the wider the amplitude variation range will be. Thus, setting depth to 100% will make the sound amplitude oscillate between silence and the maximum amplitude of the sound in your audio track. Setting it to 0% will result in no vibration at all.
- **Rate** - sets the number of amplitude oscillations per second. This parameter controls how often the sound amplitude oscillates. For example, if you set the parameter to 3 Hz, the amplitude will oscillate (make the full cycle from the minimum to the maximum amplitude and back to the minimum) 3 times per second.

Time Stretch/Pitch Shift Effects

Using the appropriate **Effects Tab** or **Effects and Filters Panel** options you can apply different effects to your audio. The following **Time/Pitch Effects** are available:

- **Tempo Change**
- **Pitch Shift**
- **Rate Change**
- **Reverse**

Tempo Change

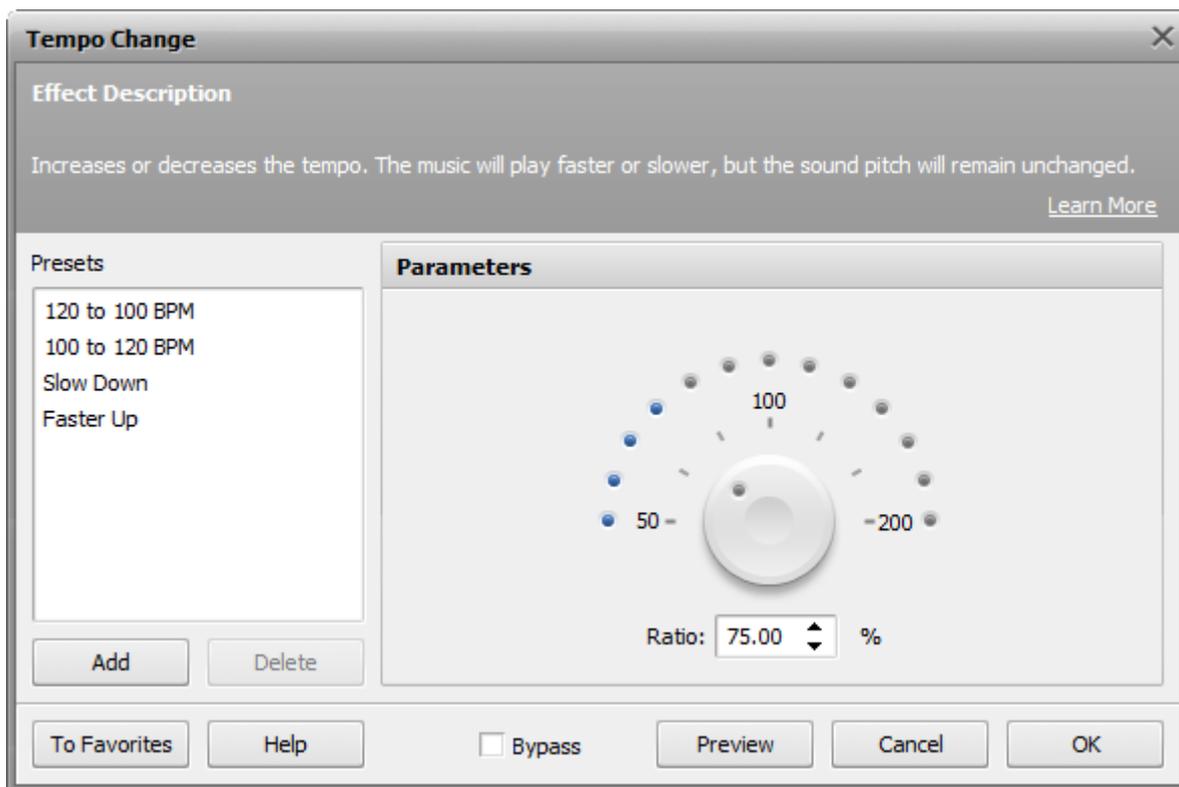


The **Tempo Change** (also called **Time Stretch**) effect allows to change the tempo, but keep the pitch the same throughout. The audio will play slower or faster, but its tone will remain unchanged.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Tempo Change Settings



Use the **Ratio** text field or the wheel cursor to set the desired tempo change level. Setting the parameter to 100% will correspond to the original sound. Values below 100% will cause the tempo to decrease. Values above 100% will cause the tempo to increase.

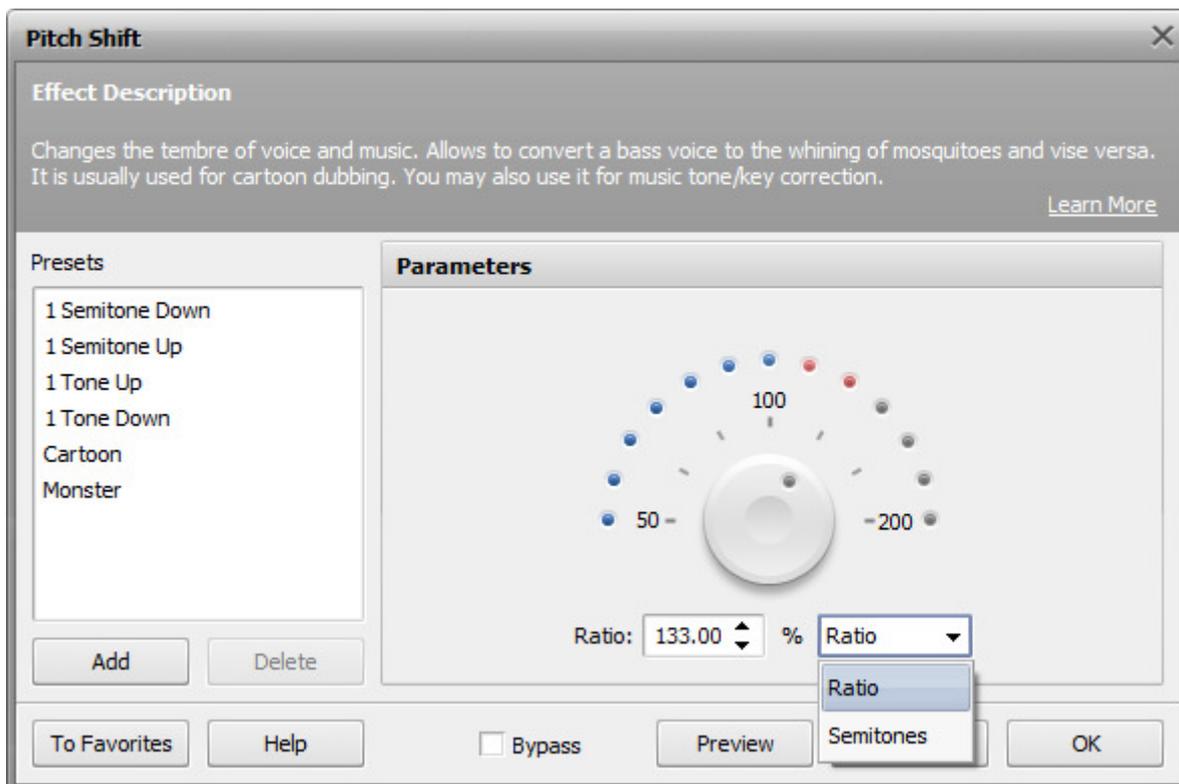
Pitch Shift

 The **Pitch Shift** effect shifts the frequency spectrum of the input signal. This will change the sound pitch (tone) without changing its tempo.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Pitch Shift Settings



Use the drop-down list to the right to select between the pitch shift types.

Select **Semitones** to shift the sound pitch by semitones. Next use the **Ratio** text field or the wheel cursor to set the desired pitch change level. Setting the parameter to 0.00 will correspond to the pitch of the original sound. Values below zero will cause the pitch to decrease. Values above zero will cause the pitch to increase. For example, if you set the ratio to 2.00, the resulting audio musical key will be raised by 2 semitones.

Select **Ratio** to to shift the sound pitch by percentage. Next use the **Ratio** text field or the wheel cursor to set the desired pitch change level. Setting the parameter to 100% will correspond to the pitch of the original sound. Values below 100% will cause the pitch to decrease. Values above 100% will cause the pitch to increase.

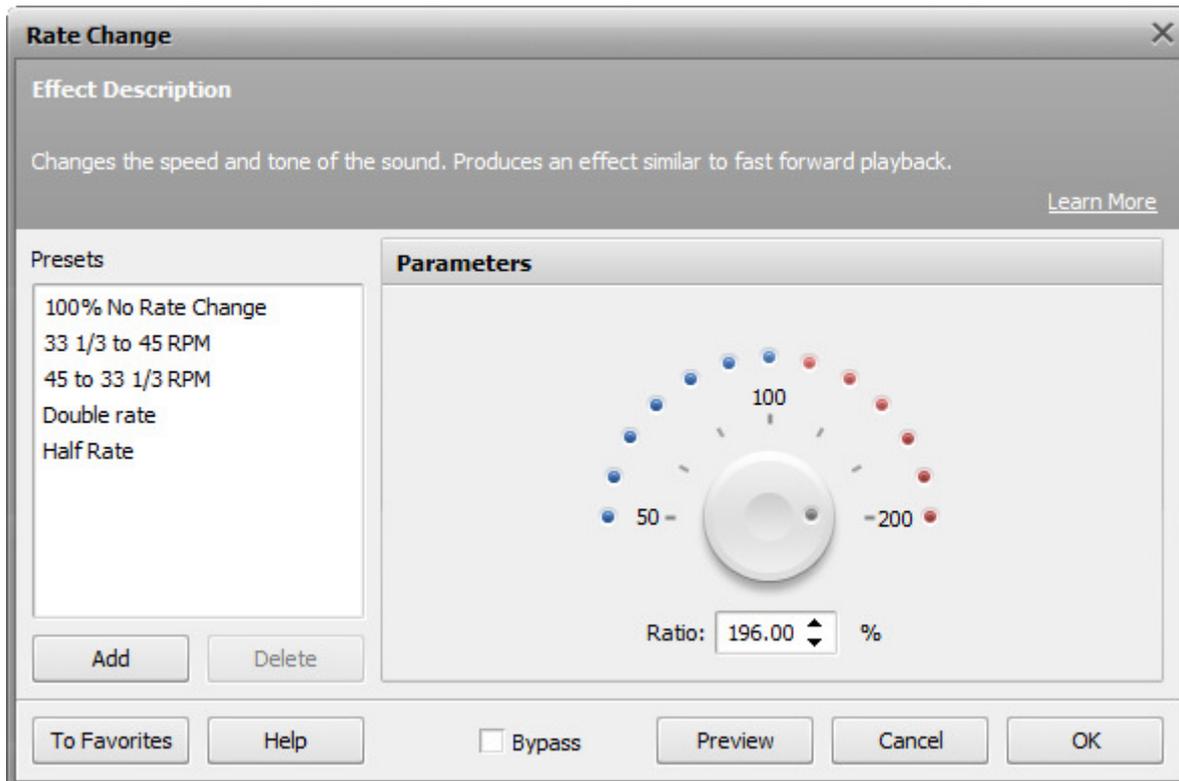
Rate Change

 The **Rate Change** effect allows you to simultaneously change the sound tempo and pitch. Decreasing the rate value will cause the tempo and pitch to decrease. Increasing it will cause the tempo and pitch to increase.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Rate Change Settings



Use the **Ratio** text field or the wheel cursor to set the desired pitch/tempo change level. Setting the parameter to 100% will correspond to the original sound. Values below 100% will cause the tempo and pitch to decrease. Values above 100% will cause the tempo and pitch to increase.

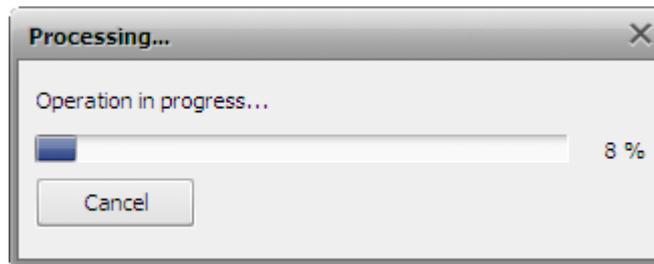
Reverse



The **Reverse** effect is used to make a selection play backwards by reversing the order of its samples. It creates an effect of reverse (backward) playback. It is also useful for creating special effects.

How to Apply

1. Add the audio file to the **Waveform Editing Space**.
2. Select the part of the file you would like to apply the effect to. Leave it unselected to apply the effect to the whole file.
3. Click the **Reverse** button on the **Effects Tab** or double-click the **Invert** effect on the **Effects and Filters Panel**.
4. Wait while the program applies the effect. Click **Cancel** at any time to abort the process.



Settings

This effect does not have any settings and will be simply applied so that either the selected part will be reverted or the whole audio if nothing has been selected.



Note: On multichannel (containing two and more channels) waveforms, all channels are reversed.

Filters

Using the appropriate **Effects Tab** or **Effects and Filters Panel** options you can apply different filters to your audio. The following **Filters** are available:

- **Equalizer**
- Parametric Filters
 - Pass filters:**High-Pass** and **Low-Pass**
 - Band filters:**Band-Pass** and **Band-Stop**
 - Notch/Resonance filters:**Resonance** and **Notch**
- **FFT**
- **Delete Silence**
- **Noise Gate**
- **Noise Removal**

Equalizer



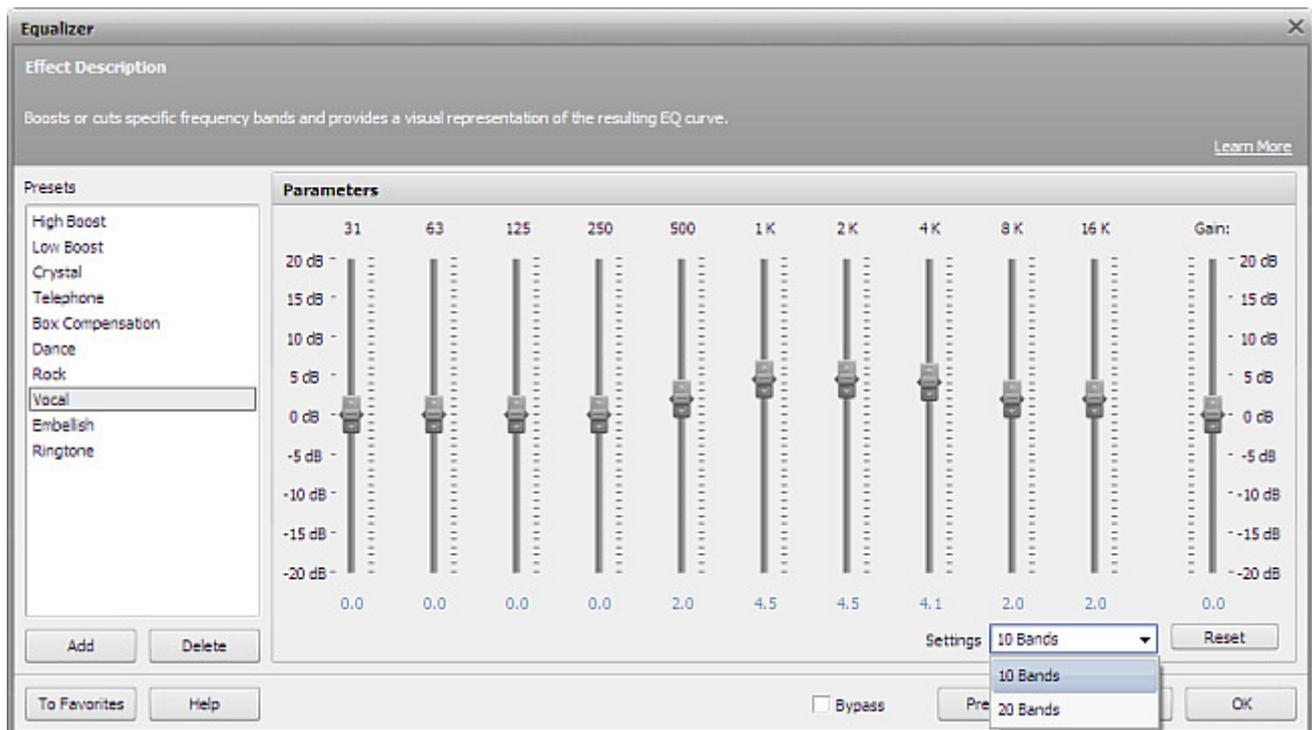
The **Equalizer** filter is used to change the frequency pattern of the audio file, attenuate some of the frequencies and amplify others.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Equalizer Settings

To apply the effect, select the number of frequency bands you would like to amplify/attenuate and then specify the desired amplification value for each frequency by moving the sliders of the vertical bars.



Use the **Settings** drop-down list to set the number of frequency bands you want to affect. You may select between 10 and 20 bands. The whole frequency range will be divided into 10 or 20 bands with the center frequency of each band indicated above its corresponding vertical bar. Selecting 20 bands will display more frequency bands which will help you build a more accurate equalization curve.

Use the sliders of the vertical bars with numbers to amplify/attenuate a particular frequency band. The software will allow you to amplify/attenuate each frequency band by a maximum of 20 dB. The current amplification/attenuation value will be displayed below the bar. You may also click the number and enter the needed value from the keyboard.

Use the **Gain** slider to the right to amplify/attenuate the whole audio track. The software will allow you to amplify/attenuate the audio signal by a maximum of 20 dB. Setting a positive value will amplify the sound, setting a negative value will attenuate it.

High-Pass Filter

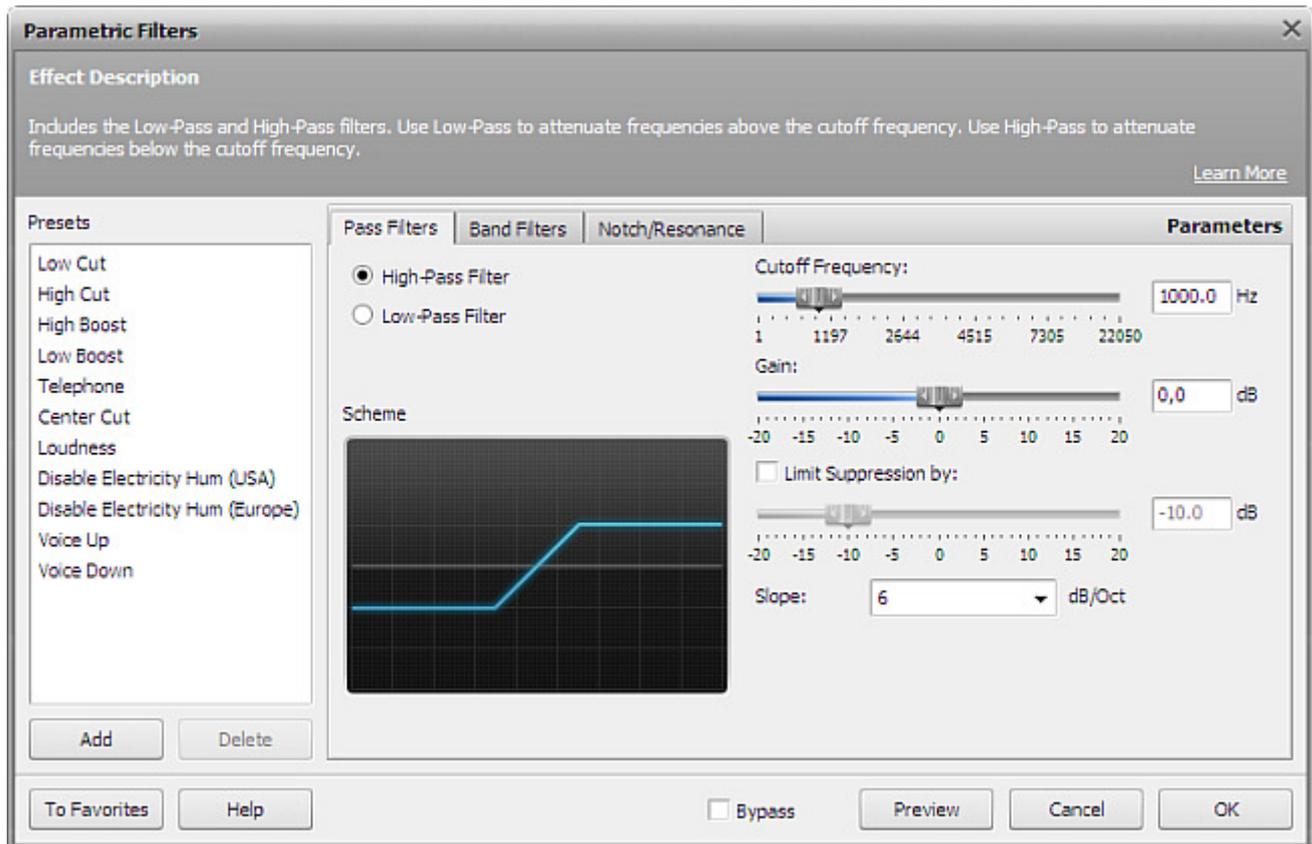
The **High-Pass** filter is used to to attenuate frequencies below the cutoff frequency. Like all parametric filters, this one is usually used to correct various imperfections in the original audio.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

High-Pass Filter Settings

To apply the effect, use one of the available presets or configure the effect parameters manually.



The following **High-Pass** filter settings are available for editing:

- **Cutoff Frequency** - sets the frequency point at which the whole frequency range will be divided into 2 bands – pass-band and stop-band. All frequencies below the **Cutoff Frequency** will be attenuated in accordance with the attenuation curve. All frequencies above the **Cutoff Frequency** will not be affected by the filter.
- **Gain** - sets the degree of amplification applied to the audio signal. Use it if you want to increase/decrease the resulting sound volume. The amplification will affect frequencies in both the pass- and stop-bands.
- **Limit Suppression by** - tick this option to limit the attenuation curve decay (lower shelf). The attenuation will stop at this point and the signal will not fall below this level.
- **Slope** - sets the steepness of the frequency attenuation curve. Use it to control the transition between the affected and non-affected frequencies. Higher values correspond to a higher attenuation growth in the stop-band.

Low-Pass Filter

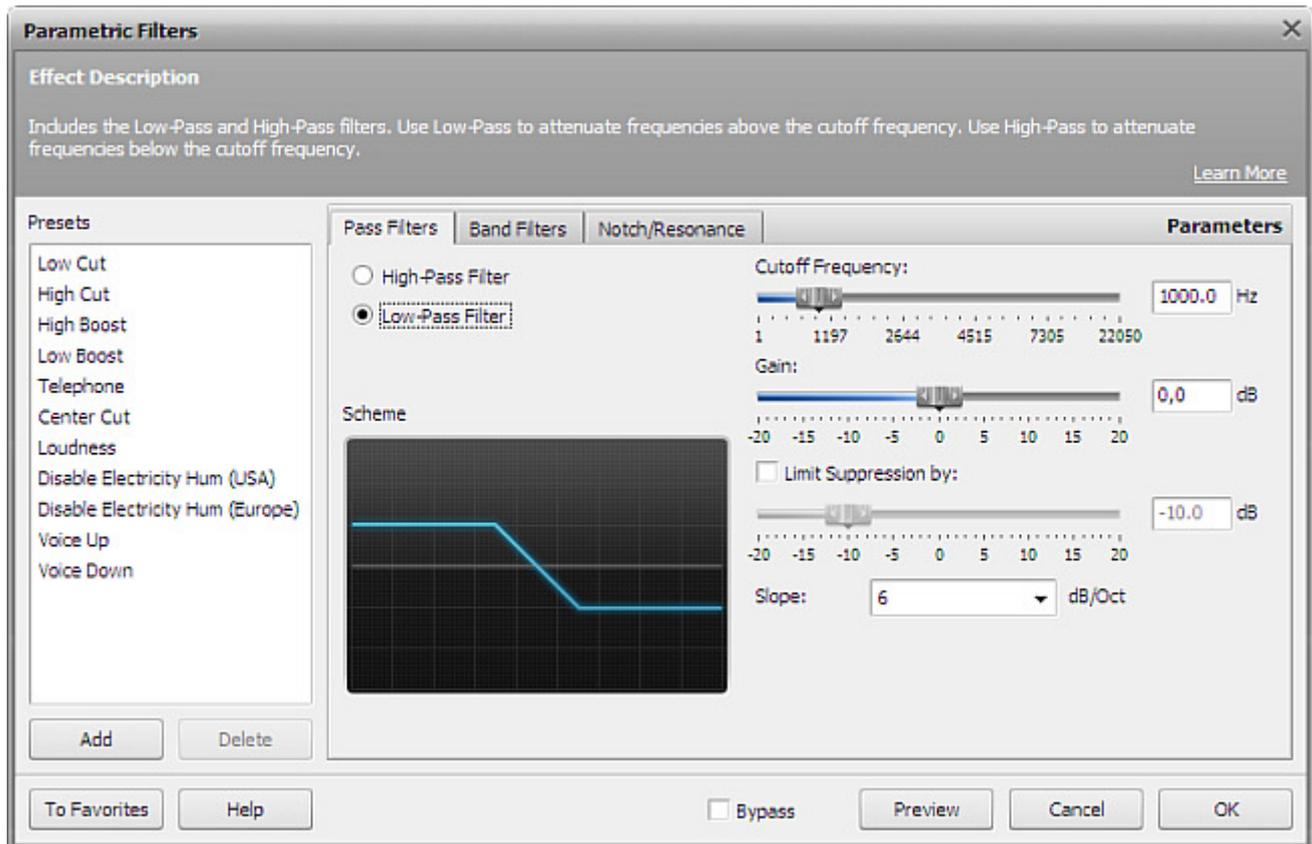
The **Low-Pass Filter** filter is used to to attenuate frequencies above the cutoff frequency. Like all parametric filters, this one is usually used to correct various imperfections in the original audio.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Low-Pass Filter Settings

To apply the effect, use one of the available presets or configure the effect parameters manually.



The following **Low-Pass** filter settings are available for editing:

- **Cutoff Frequency** - sets the frequency point at which the whole frequency range will be divided into 2 bands – pass-band and stop-band. All frequencies above the **Cutoff Frequency** will be attenuated in accordance with the attenuation curve. All frequencies below the **Cutoff Frequency** will not be affected by the filter.
- **Gain** - sets the degree of amplification applied to the audio signal. Use it if you want to increase/decrease the resulting sound volume. The amplification will affect frequencies in both the pass- and stop-bands.
- **Limit Suppression by** - tick this option to limit the attenuation curve decay (lower shelf). The attenuation will stop at this point and the signal will not fall below this level.
- **Slope** - sets the steepness of the frequency attenuation curve. Use it to control the transition between the affected and non-affected frequencies. Higher values correspond to a higher attenuation growth in the stop-band.

Band-Pass Filter

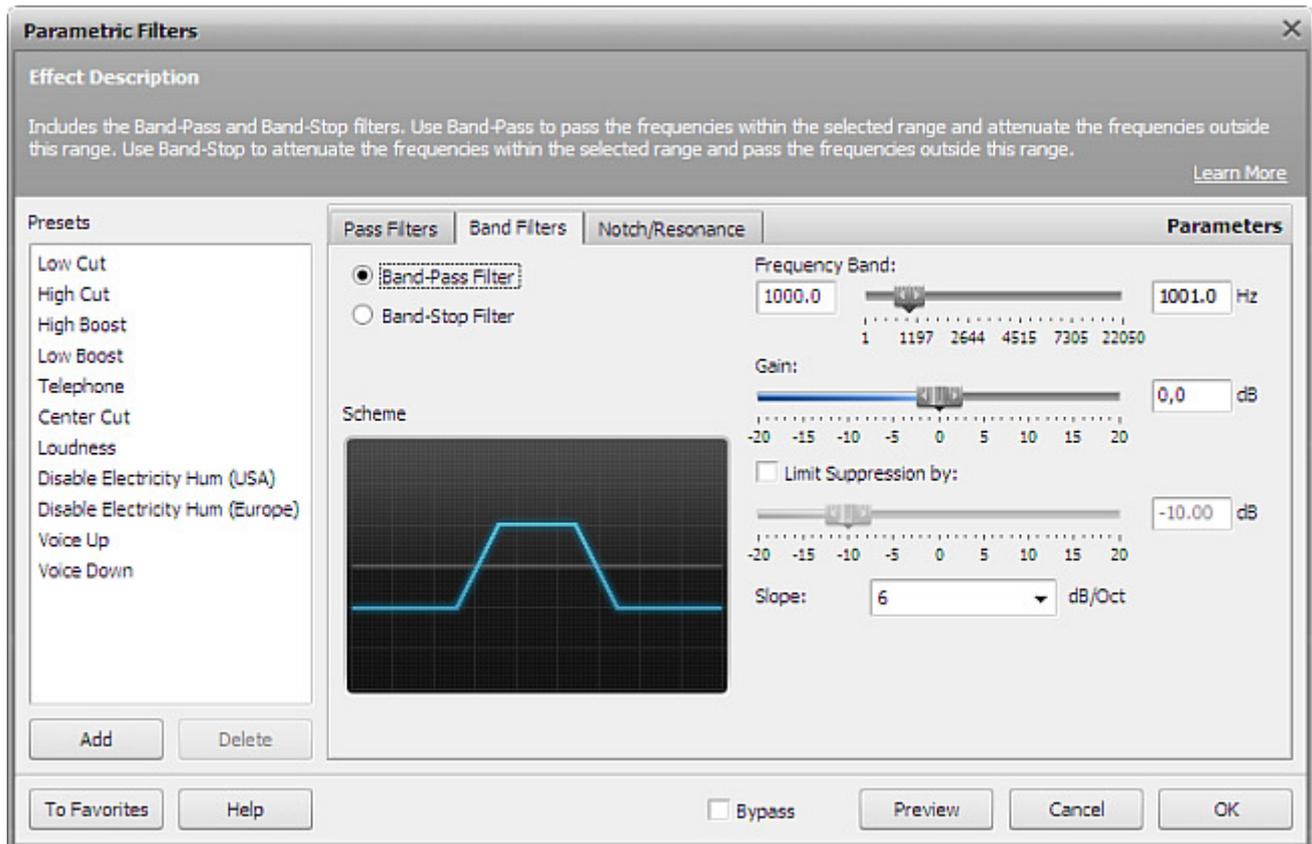
The **Band-Pass** filter is used to pass frequencies within the selected range and attenuate frequencies outside this range. Like all parametric filters, this one is usually used to correct various imperfections in the original audio.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Band-Pass Filter Settings

To apply the effect, use one of the available presets or configure the effect parameters manually.



The following **Band-Pass** filter settings are available for editing:

- **Frequency Band** - sets the frequency band which will not be affected by the filter. In other words, all frequencies outside the specified frequency band will be attenuated. Once a frequency goes beyond the specified frequency band limits, its amplitude will gradually decrease till it has no amplification (the signal will be very weak, or non-existent), unless you limit the suppression level in the **Limit Suppression by** field.
- **Gain** - sets the degree of amplification applied to the audio signal. Use it if you want to increase/decrease the resulting sound volume. The amplification will affect frequencies in both the pass and attenuation bands.
- **Limit Suppression by** - tick this option if you do not wish the frequency attenuation to reach its zero amplitude. Set the signal level in decibels at which the frequency attenuation will stop.
- **Slope** - sets the frequency attenuation rate. Use it to control the transition between the affected and non-affected frequencies. Higher values correspond to a sharper transition.

Band-Stop Filter

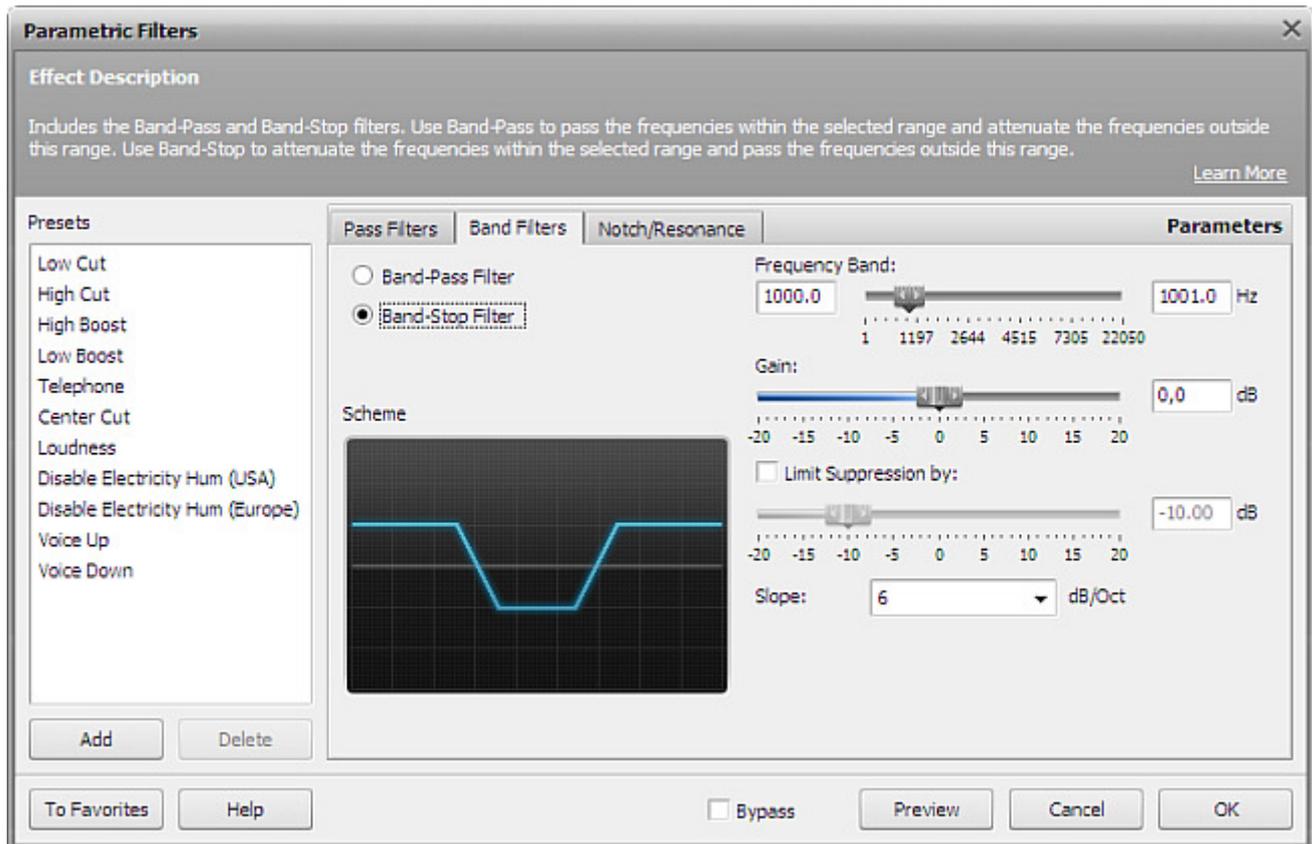
The **Band-Stop** filter is used to attenuate frequencies within the selected range and pass frequencies outside this range. Like all parametric filters, this one is usually used to correct various imperfections in the original audio.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Band-Stop Filter Settings

To apply the effect, use one of the available presets or configure the effect parameters manually.



The following **Band-Stop** filter settings are available for editing:

- **Frequency Band** - sets the frequency band within which the attenuation will take place. In other words, all frequencies outside the specified frequency band will not be affected by the filter. Once a frequency reaches the specified frequency band limits, its amplitude will gradually decrease till it has no amplification (the signal will be very weak, or non-existent), unless you limit the suppression level in the **Limit Suppression by** field.
- **Gain** - sets the degree of amplification applied to the audio signal. Use it if you want to increase/decrease the resulting sound volume. The amplification will affect frequencies in both the pass and attenuation bands.
- **Limit Suppression by** - tick this option if you do not wish the frequency attenuation to reach its zero amplitude. Set the signal level in decibels at which the frequency attenuation will stop.
- **Slope** - sets the frequency attenuation rate. Use it to control the transition between the affected and non-affected frequencies. Higher values correspond to a sharper transition.

Notch Filter

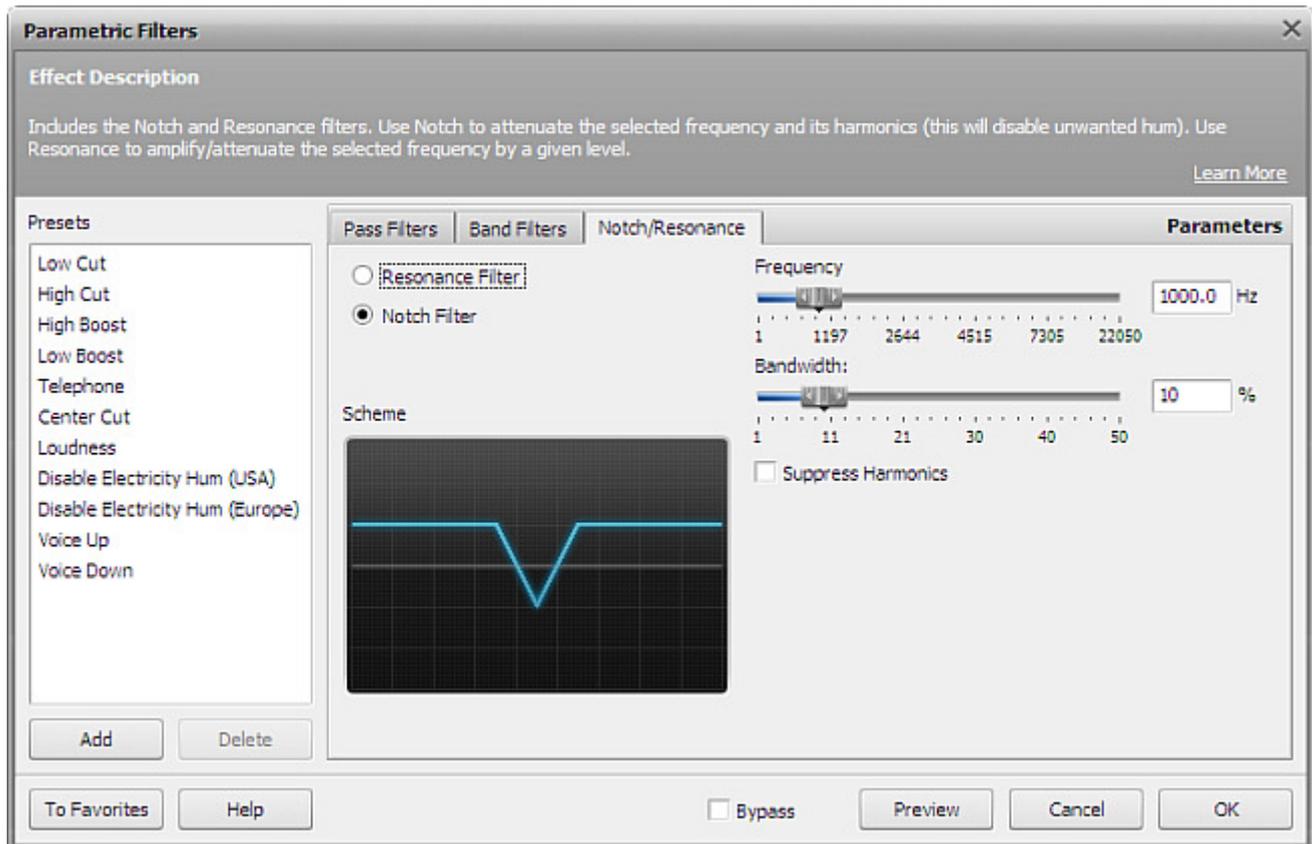
The **Notch** filter is used to attenuate the selected frequency and its harmonics. Like all parametric filters, this one is usually used to correct various imperfections in the original audio.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Notch Filter Settings

To apply the effect, use one of the available presets or configure the effect parameters manually.



The following **Notch** filter settings are available for editing:

- **Frequency** - sets the notch frequency (the one you want to attenuate).
- **Bandwidth** - specifies the band of frequencies around the notch frequency that will be affected by the filter in percent. The bigger the value, the wider the band.
- **Suppress Harmonics** - tick this option to suppress the notch frequency harmonics. The program will eliminate a maximum of 4 harmonics. Suppressing harmonics will help eliminate unwanted hum.

Resonance Filter

The **Resonance** filter is used to amplify/attenuate the selected frequency by a given level. Like all parametric filters, this one is usually used to correct various imperfections in the original audio.

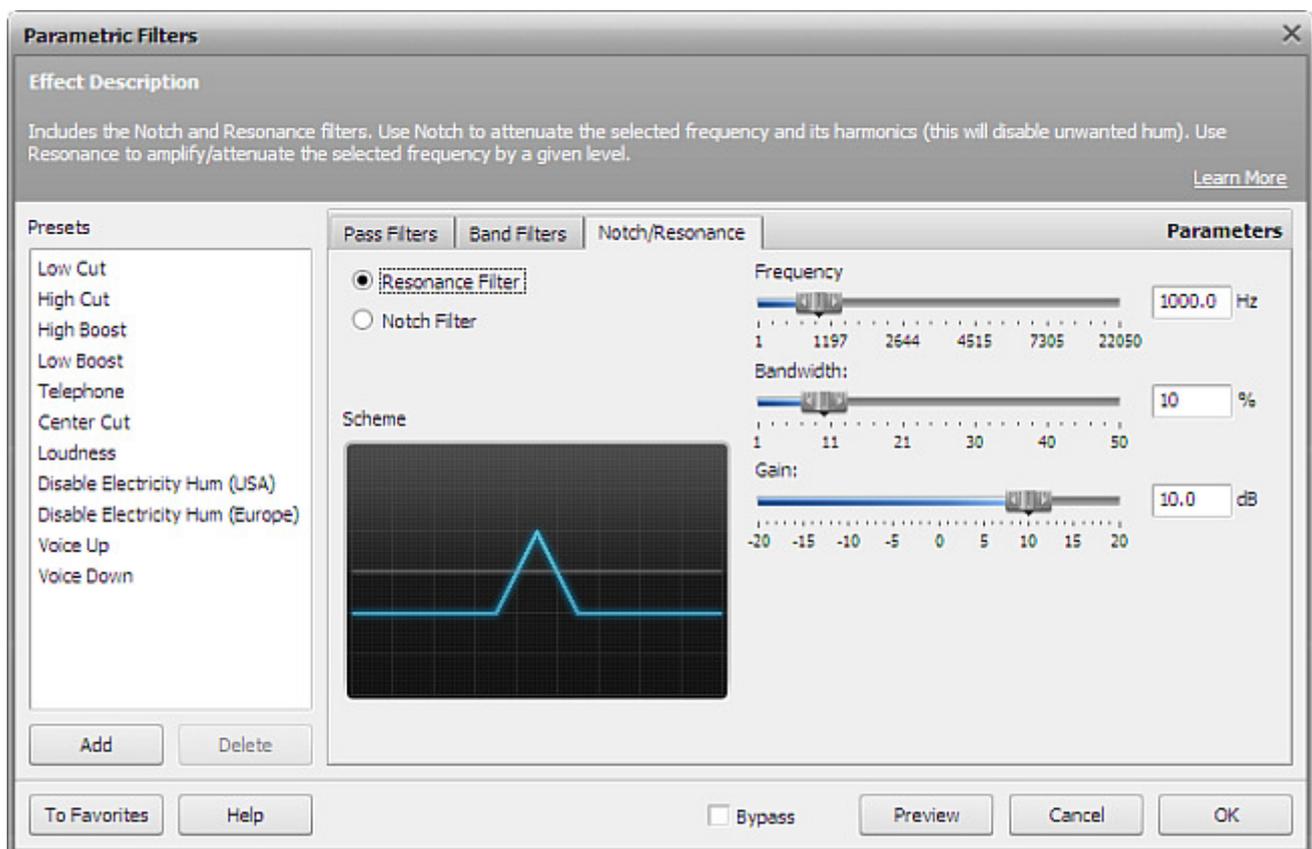
How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Resonance Filter Settings

Many of non-advanced musical instruments and acoustic systems tend to have unwanted resonances. If you discover such unwanted effect in your instrument, you'll be able to use Frequency Analysis to define the resonance frequency of you instrument and then attenuate it with the help of the **Resonance** filter.

To apply the effect, use one of the available presets or configure the effect parameters manually.



The following **Resonance** filter settings are available for editing:

- **Frequency** - sets the resonance frequency (the one you want to amplify/attenuate).
- **Bandwidth** - specifies the band of frequencies around the resonance frequency that will be affected by the filter in percent. The bigger the value, the wider the band.
- **Gain** - sets the level in decibels by which you want the resonant frequency to be amplified/attenuated.

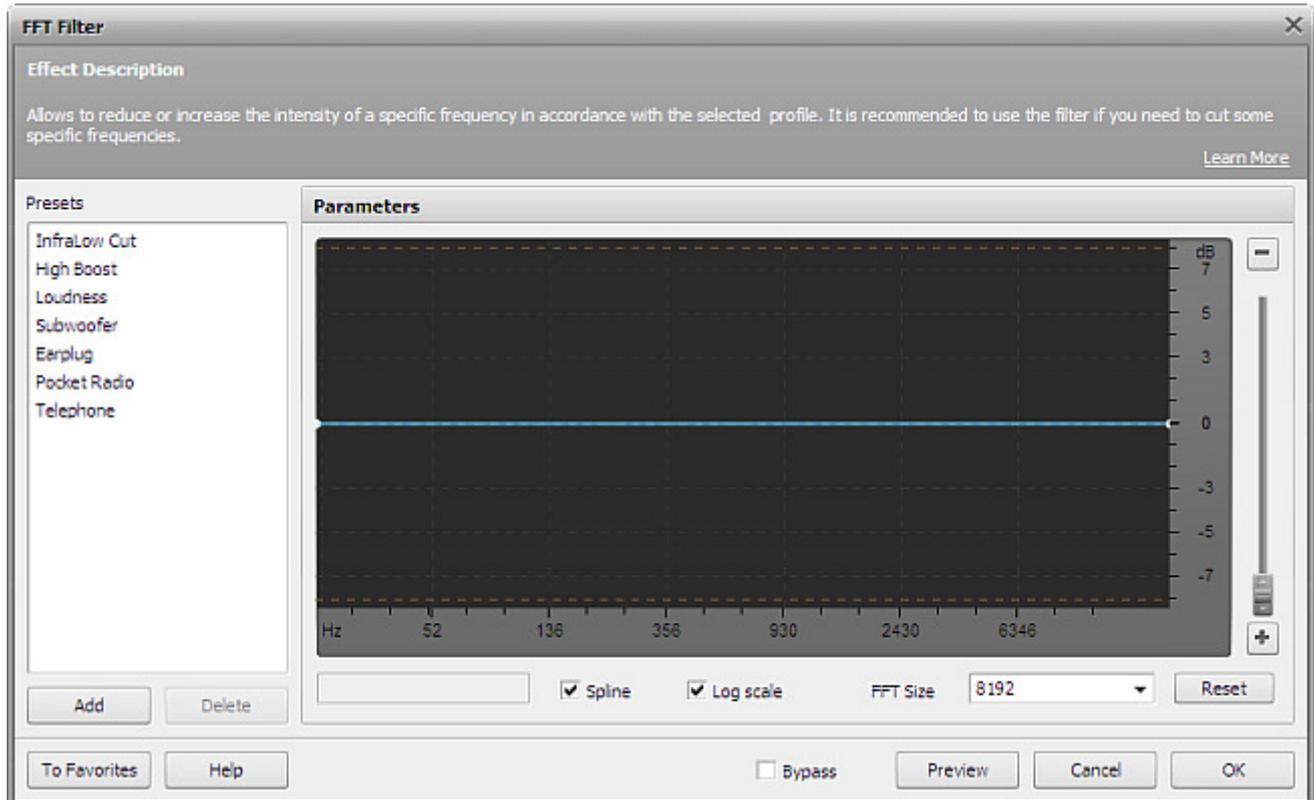
FFT Filter



The **FFT (Fast Fourier Transform) Filter** allows to reduce or increase the intensity of the frequency in accordance with the selected frequency profile. The filter smooths the signal during a time in limits of the FFT block. It is recommended to use the filter in the following cases: for the extreme filtering - when the processing result as a fact is more important than the possible smoothing effect; for the cutting of the specified frequencies. If you select a part of the file with the mouse, this effect will be applied to this exact part of the file. Otherwise the sound of the whole file will be affected.

To apply the effect you should do the following:

1. Add the audio file to the **Waveform Editing Space**.
2. Select a part of the file you would like to apply the effect to or leave it unselected to apply the effect to the whole file.
3. Click the **FFT Filter** button on the **Effects Tab** or double-click the **FFT Filter** on the **Effects and Filters Panel**.



4. Adjust the effect parameters or select one of the available presets from the list.



Note: if you change the effect settings, you can save them as a preset to be able to use them afterwards. Use the **Add** button to do that.

5. Click the **Preview** button to make sure that the changes are correct.
6. Click the **OK** button to return to the audio file editing.

Delete Silence

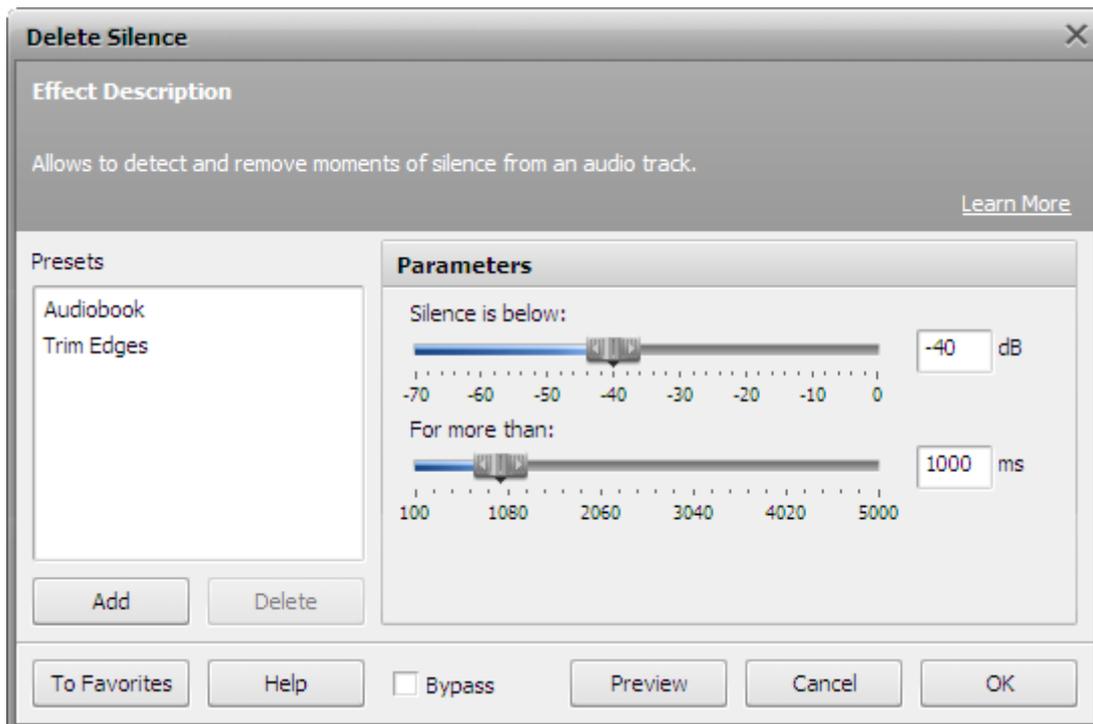
 The **Delete Silence** filter is used to detect and remove moments of silence from an audio track. You will need to tell the program what exactly to consider moments of silence by setting the corresponding parameters in the settings window (see below).

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Delete Silence Settings

To apply the effect, use one of the available presets or configure the effect parameters manually.



The following **Delete Silence** settings are available for editing:

- **Silence is below** - set the strength of the signal in decibels.
- **For more than** - sets the duration of audio signal in milliseconds.

All audio signal below the level set in the "Silence is below" field whose duration exceeds the value set in the "For more than" field will be treated as silence and removed from your audio track.

Noise Gate



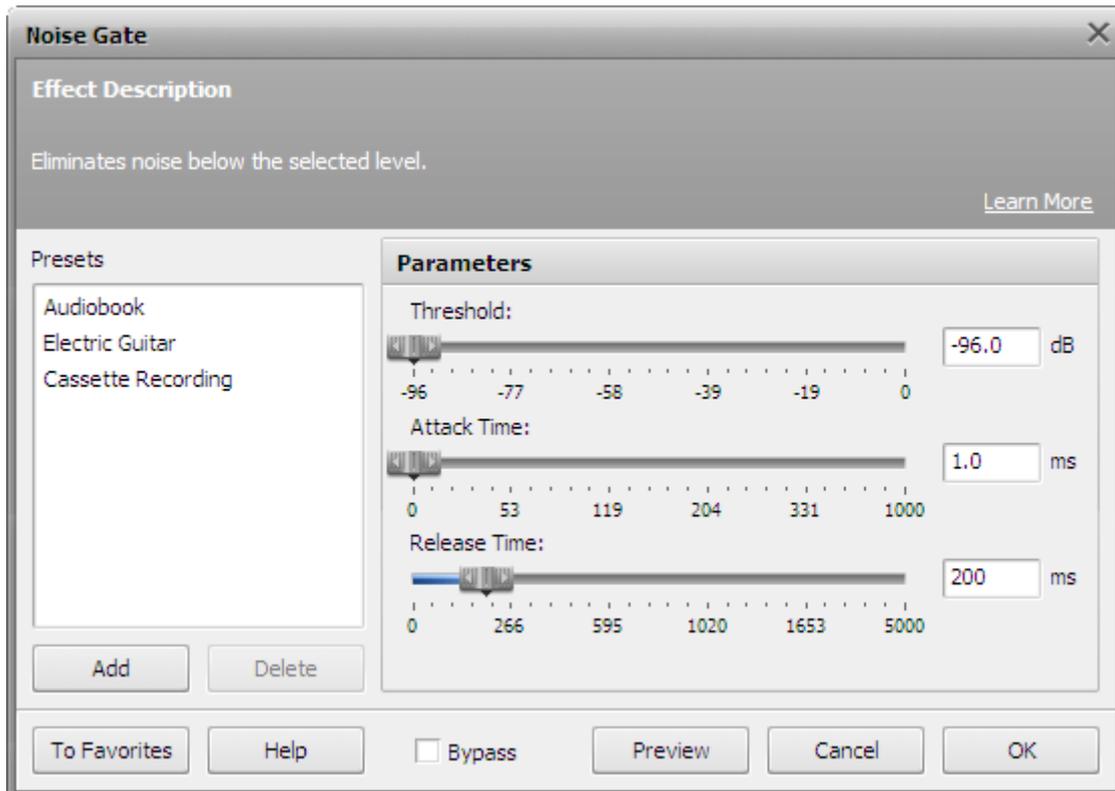
The **Noise Gate** filter is used to eliminate noise below the selected level.

How to Apply

All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Noise Gate Settings

To apply the effect, use one of the available presets or configure the effect parameters manually.



The following **Noise Gate** settings are available for editing:

- **Threshold** - the level (in dB) at which the gate opens. The noise gate filter will affect only the sound below the specified threshold.
- **Attack Time** - the amount of time it takes the noise gate to change from closed to open once the signal exceeds the specified threshold.
- **Release Time** - the period during which the noise gate remains open once the signal has fallen below the specified threshold.



Note: the attack and release times are used to smooth the noise gate boundaries. The slower the attack and release are, the smoother the transition between affected and non-affected signal will be.

Noise Removal



The **Noise Removal** effect is used to reduce noise and make the sound clearer. You'll need to specify an audio sample that will be treated as noise and then detect noise areas in the audio track in accordance with the specified noise profile.

How to Apply

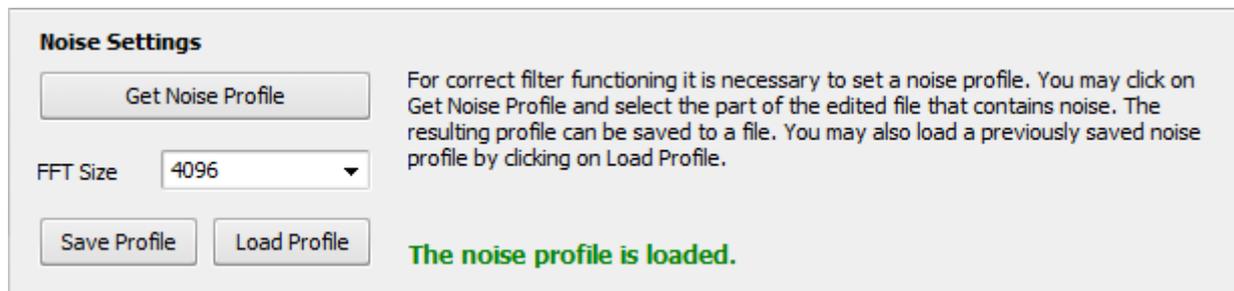
All effects and filters are applied in the same way in **AVS Audio Editor**. For detailed step-by-step instructions on how to apply effects to your audio files, please refer the **How to Apply** section of **Using Effects and Filters**.

Noise Removal Settings

Step 1. Configure Noise Settings

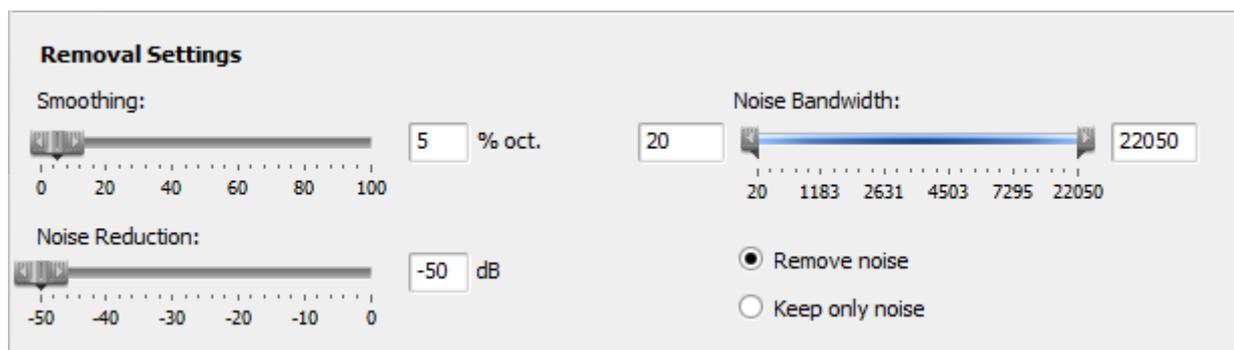
You'll need to tell the program what exactly it should treat as noise. You may do it in the **Noise Settings** section of the settings window. Click on **Get Noise Profile** and select the part of an audio track that contains all noise and no audio signal (usually a short audio portion at the beginning or at the end of an audio track is selected). The start and end times of the selected fragment will be shown in the opened **Get Noise Profile** window. When ready, click **OK** to load the selected fragment as noise profile. Alternatively, load a previously saved audio profile into the program.

After the profile has been successfully loaded, you will be notified with the appropriate message:



Step 2. Configure Removal Settings

Once the profile is loaded, you may proceed with eliminating noise from your audio track. Use the **Removal Settings** section of the settings window to configure noise removal parameters.



Move the **Bandwidth** sliders to set the band of frequencies that will be affected by the filter. The program will operate only within the band you set.

Enable the **Remove noise** option to remove the detected noise areas from your audio track. Enable the **Keep only noise** option to keep only noise areas and remove all the rest of audio signal.

Use the **Smoothing** parameter to smooth the frequency characteristic of the created noise profile. It defines the width of the smoothing corridor in percent of octave (% oct). For example, for a signal of 1000 Hz the corridor will be 500 (-1 octave) to 2000 (+ 1 octave) if you set the smoothing parameter to 100%. In other words, the bigger the percentage value, the wider the corridor.

Use the **Noise Reduction** parameter to set the degree of noise suppression in the detected noise areas in dB. Setting a bigger value will result in greater suppression. Setting the parameter to 0 will produce no effect at all.

Using DirectX Filters

AVS Audio Editor supports DirectX audio plug-ins and filters installed on your computer. DirectX is a development platform designed by Microsoft which provides an open standard for multimedia plug-ins. DSP modules, and other audio tools based on this standard, can be used by any application that supports the DirectX architecture.

With the help of DirectX plug-ins you can easily extend the range of powerful effects available in **AVS Audio Editor**.

DirectX plug-ins are very easy to use. To enable any of them just highlight an area to process and choose a filter from **DX Filters** drop-down menu on the **DX/VST Tab**.

Most DirectX plug-ins have their own settings dialog where you can configure the parameters of the effect. Some plug-ins may support real-time preview. But, of course, if you have some problems you will need to consult the documentation provided by the plug-in manufacturer.

Using VST Effects

AVS Audio Editor supports VST audio plug-ins and effects on your computer. VST (Virtual Studio Technology) effects and filters are native realtime plugins. Initially their standard was developed by Propellerhead/Steinberg. They usually provide a custom graphical user interface, displaying controls similar to the physical switches and knobs on audio hardware.

With the help of VST effects you can easily extend the range of powerful effects available in **AVS Audio Editor**.

In case you have VST effects on your computer you will need to point out their directory to **AVS Audio Editor** program. To do that press the **VST Effects Directories** button on the **DX/VST Tab**. The following dialog window will appear:



Use the **Add** button to select a directory with VST effects present and click **OK** to update the effects list. All the effects from the folder will be loaded automatically and will be available afterwards from the **VST Effects** drop-down menu on the **DX/VST Tab**.

You can also add the additional effects and remove the unnecessary ones in this window using the appropriate buttons.

VST effects are very easy to use. To enable any of them just highlight an area to process and choose a filter from the **VST Effects** drop-down menu on the **DX/VST Tab**.

Most VST effects have their own settings dialog where you can configure the parameters of the effect. Some plug-ins may support real-time preview. But, of course, if you have some problems you will need to consult the documentation provided by the plug-in manufacturer.

Generating Sounds

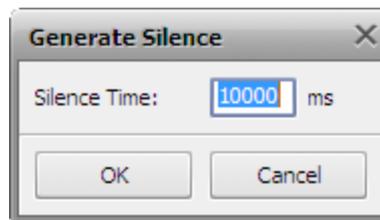
Sometimes you might need to create a sound containing some signals that exist in real world but you do not always have a possibility to record these sounds. **AVS Audio Editor** lets you create some of these signals using the internal methods.

You can add the following signals to your audio:

- **Silence**
- **Noise**
- **Signal**
- **Chirp**
- **DTMF Signals**

Generating Silence

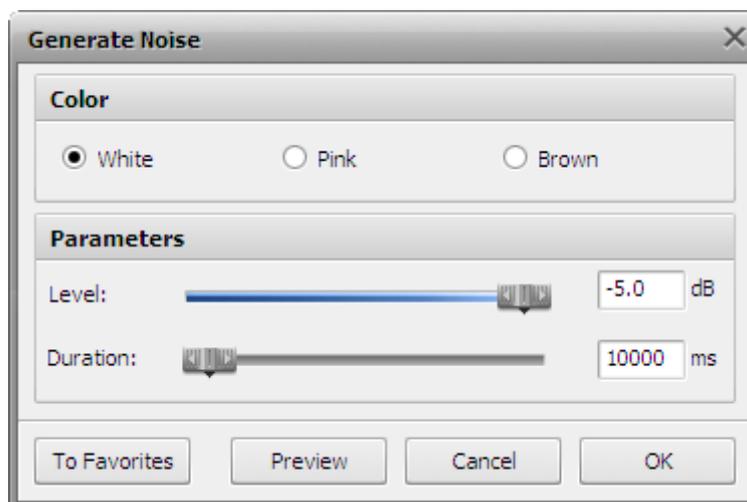
 To generate the absence of sound you need to open an audio file or create a new one, select the part of the file where you want to have a period of silence and click the **Silence** button of the **Generate Tab**. The following window will be opened:



You can set the silence duration in milliseconds. Click **OK** to apply the settings and close the effect window, or the **Cancel** button to discard the changes.

Generating Noise

 To generate the sound of noise you need to open an audio file or create a new one, select the part of the file where you want to have a period of noise and click the **Noise** button on the **Generate Tab**. The following window will be opened:



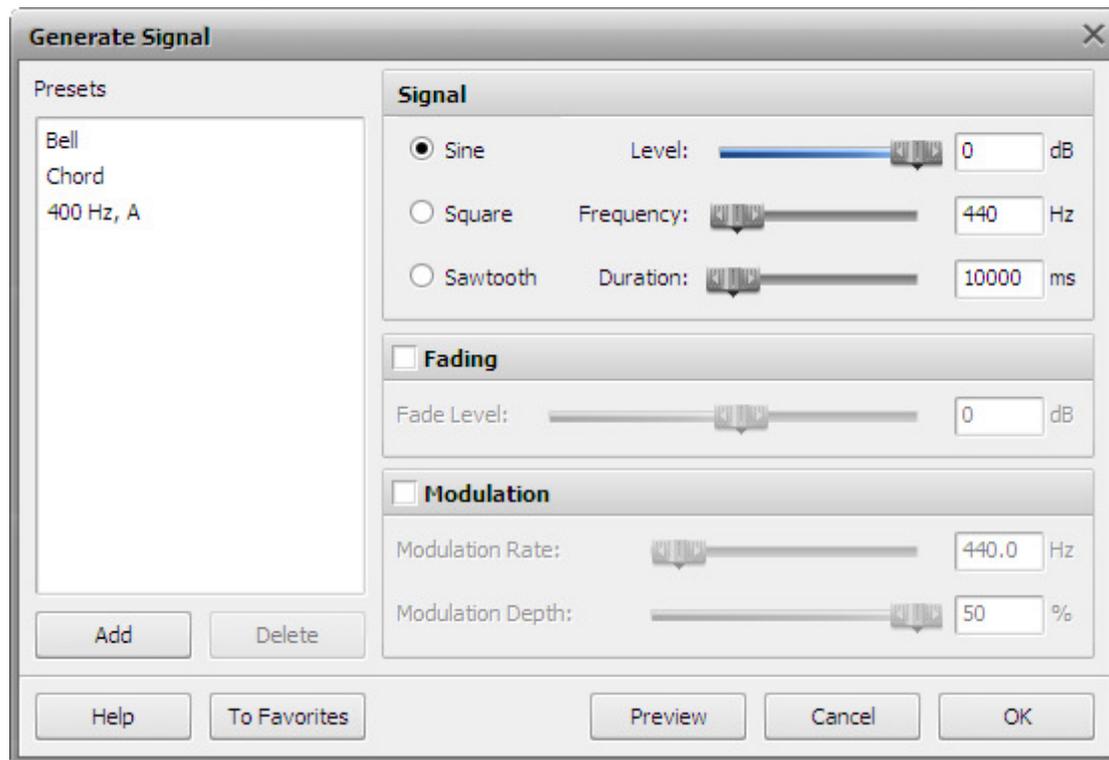
You can select the **"Color"** of the **Noise** signal (the available values include **White**, **Pink** and **Brown** types of noise different in sounding), set the noise loudness in decibels and duration in milliseconds.

Click **Preview** to make sure that the result is alright, **OK** to apply the settings and close the effect window, or the **Cancel** button to discard the changes.

Generating Signal



To generate the sound of some frequency you need to open an audio file or create a new one, select the part of the file where you want to have your signal and click the **Signal** button on the **Generate Tab**. The following window will be opened:



You can set the following signal properties:

- Select the type of the base signal used to create the sound:
 - select between the **Sine**, **Square** and **Sawtooth** type of the signal;
 - set the base signal **Level** in decibels;
 - select its main **Frequency** (in Herz);
 - set the signal **Duration** in milliseconds.
- Set the signal **Fading** properties:
 - **Fade Level**, the level of the signal change - increase or decrease - in time flow (in decibels);
 - **Fading Time**, the time in which the **Fade Level** is reached (in milliseconds).
- Select the **Modulation** parameters:
 - **Modulation Rate** used to change the signal applying a signal of different frequency over it (specified in Herz);
 - **Modulation Depth** - the intensity of the frequency modulation (in percent).

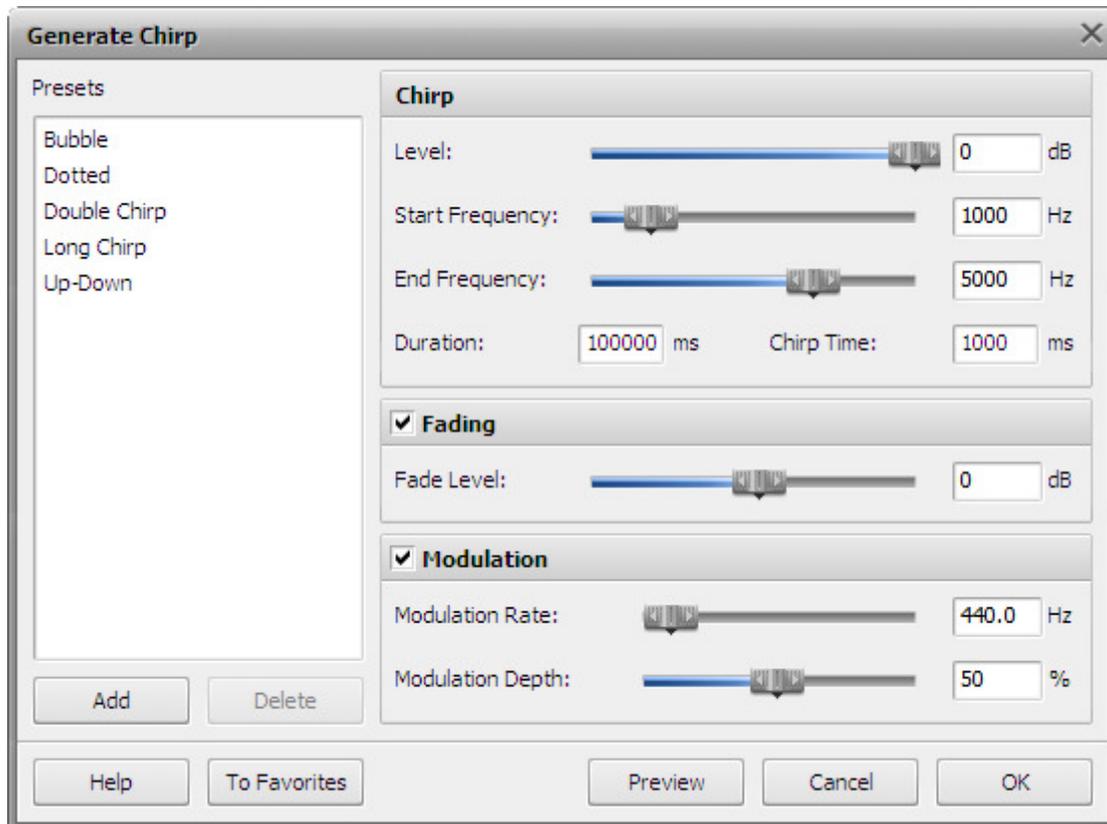
Note! **Note:** if you change the signal settings, you can save them as a preset to be able to use them afterwards. Use the **Add** button to do that.

Click **Preview** to make sure that the result is alright, **OK** to apply the settings and close the effect window, or the **Cancel** button to discard the changes.

Generating Chirp



To generate the chirp sound - a sound with the frequency increase or decrease - you need to open an audio file or create a new one, select the part of the file where you want to add the chirp signal and click the **Chirp** button on the **Generate Tab**. The following window will be opened:



You can set the following signal properties:

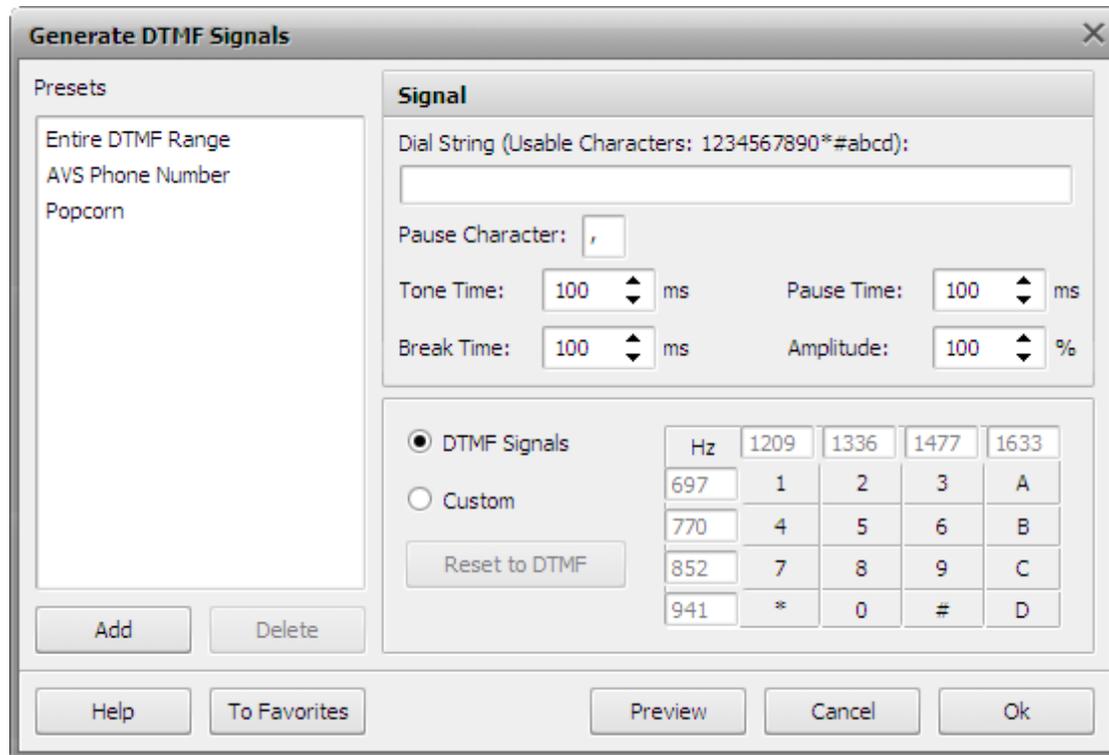
- Select the type of the base chirp signal used to create the sound:
 - set the base signal **Level** in decibels;
 - select its **Start Frequency** (in Herz);
 - select its **End Frequency** (in Herz);
 - set the signal **Duration** in milliseconds;
 - set the chirp effect duration (**Chirp Time**) in milliseconds.
- Set the signal **Fading** properties:
 - **Fade Level**, the level of the signal change - increase or decrease - in time flow (in decibels);
 - **Fading Time**, the time in which the **Fade Level** is reached (in milliseconds).
- Select the **Modulation** parameters:
 - **Modulation Rate** used to change the signal applying a signal of different frequency over it (specified in Herz);
 - **Modulation Depth** - the intensity of the frequency modulation (in percent).

Note! **Note:** if you change the signal settings, you can save them as a preset to be able to use them afterwards. Use the **Add** button to do that.

Click **Preview** to make sure that the result is alright, **OK** to apply the settings and close the effect window, or the **Cancel** button to discard the changes.

Generating DTMF Signals

**Dual-tone multi-frequency (DTMF) signal** is used for telephone signaling over the line in the voice-frequency band to the call switching center. To generate such signal you need to open an audio file or create a new one, select the part of the file where you want to have DTMF signaling and click the **DTMF Signals** button on the **Generate Tab**. The following window will be opened:



You can set the following signal properties:

Select the type of the base DTMF signals used to create the sound:

- enter the **Dial String** of the signal - the sequence of digits, letters and special symbols used for DTMF signals designation (the possible characters that can be used and determine each its own DTMF signal are ten digits - **1, 2, 3, 4, 5, 6, 7, 8, 9** and **0**, four letters - **a, b, c** and **d** and two special symbols - ***** and **#**);
- select the **Pause Character** used to mark pauses between the DTMF signals;
- select the signal **Tone Time** (in milliseconds) - the duration of a single signal;
- select the **Break Time** (in milliseconds) - the duration of a period of silence between single signals;
- select the **Pause Time** (in milliseconds) - the duration of a period of silence added to mark a pause between single signals;
- set the signal **Amplitude** in percent.

You can also select to use the classic DTMF signals matrix or determine one of your own. The DTMF keypad has a 4x4 matrix, with each row representing a low frequency, and each column representing a high frequency. Pressing a single key (such as '1') will send a sinusoidal tone of the two frequencies (697 and 1209 Hertz). The original keypads had levers inside, so each button activated two contacts. The multiple tones are the reason for calling the system multifrequency. In case you customize the DTMF signals matrix and want to return to initial values, press the **Reset to DTMF** button.

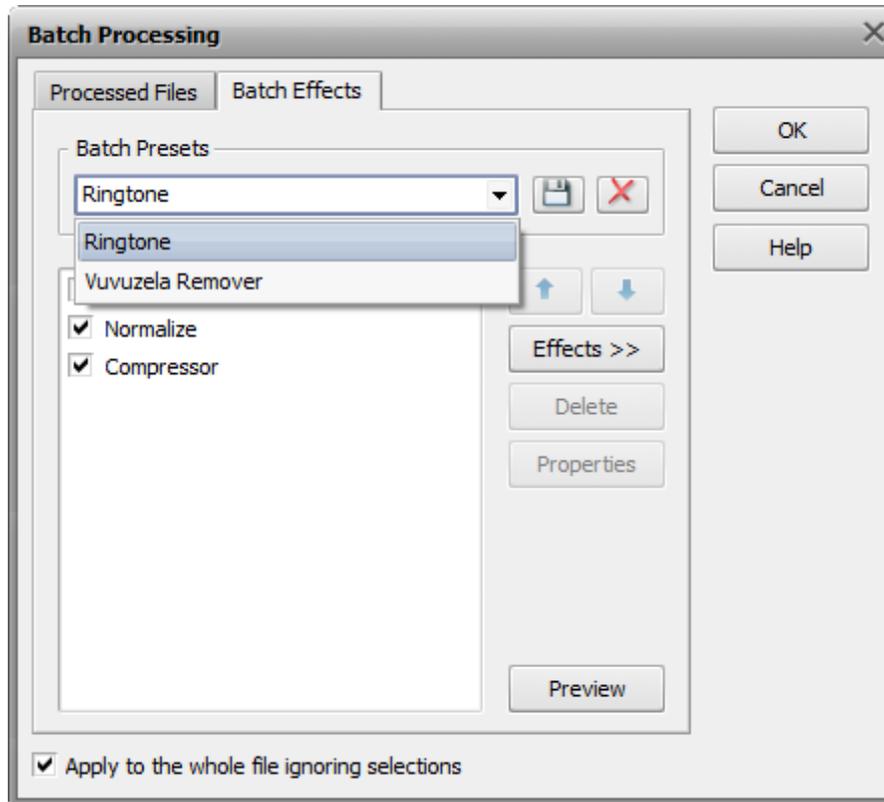
Note! **Note:** if you change the signal settings, you can save them as a preset to be able to use them afterwards. Use the **Add** button to do that.

Click **Preview** to make sure that the result is alright, **OK** to apply the settings and close the effect window, or the **Cancel** button to discard the changes.

Batch Processing

When you want to apply one and the same effect to several files, or several effects to one and the same file, or several identical effects to several files, it can be done quite easily with the help of the **Batch Processing** option of **AVS Audio Editor**.

Click the **Batch** command button on the **Effects Tab**. The following window will be opened:

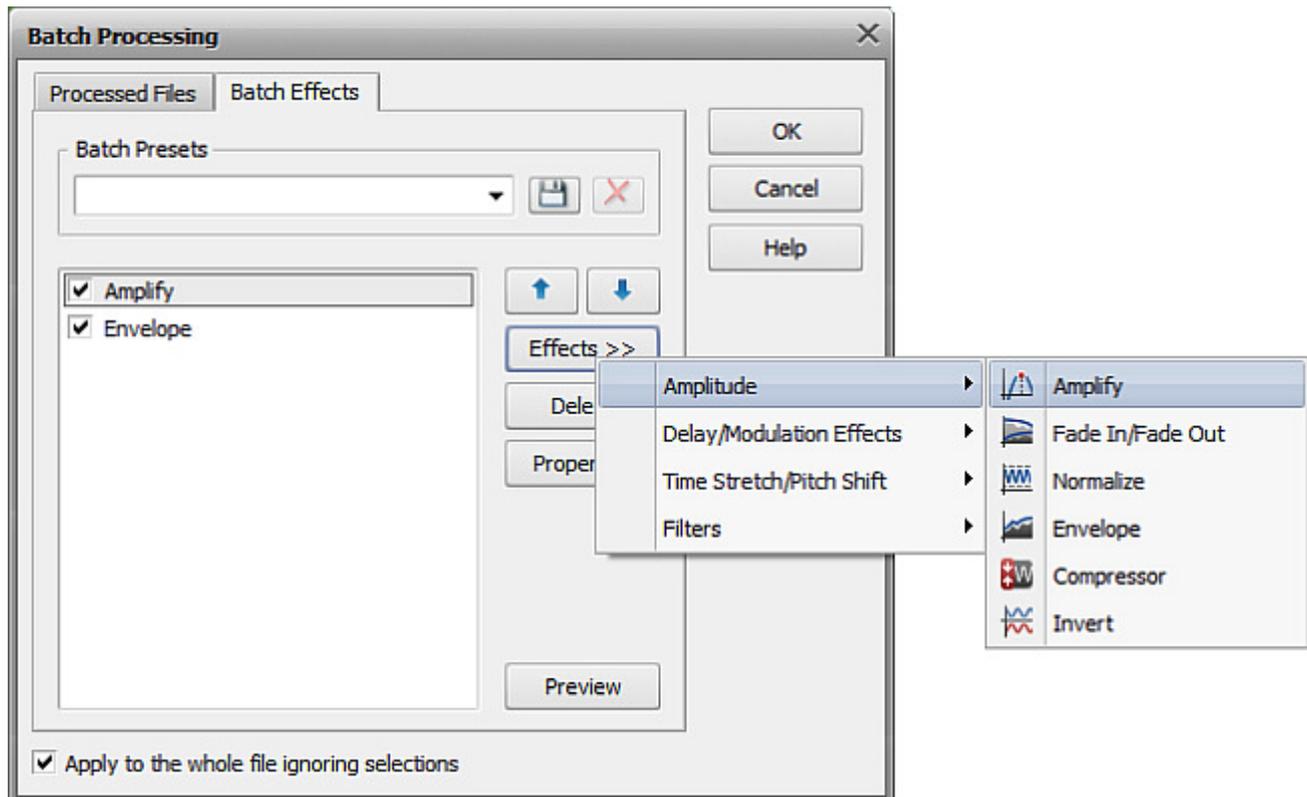


Here you can select the effects you would like to apply to the audio file (or files, if there are several of them loaded into **AVS Audio Editor**), the order in which they will be applied and their properties.

First of all you can select an effect preset from the list of available ones using the **Batch Presets** drop-down list. You can also delete the presets you no longer need using the **Delete Preset from List**  button. In case you change some of the preset settings (effects in the list, effect parameters, or something else) or create an absolutely new preset, you can save it using the **Save Preset** .

To change the presets order select one of the effects with the mouse and use the **Move Up**  and **Move Down**  buttons.

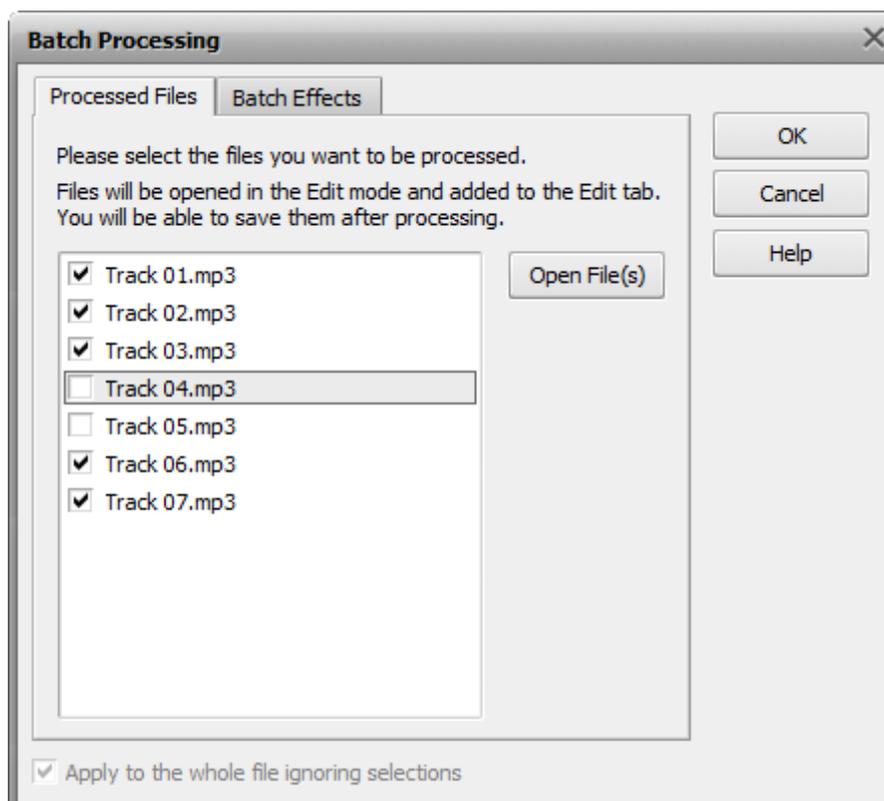
The new effects are added to the effects list using the **Effects >>** button. The list of all the available effects will be opened to let you select the necessary one:



After the effect is added its settings can be easily changed using the **Properties** button. Please, read the **effects description** section to find out more on each effect settings.

The effects are removed from the list with the help of **Delete** button (do not forget to select the effect to be deleted with the mouse before you press that button). The **Preview** button is used to make sure that the result of the effects application is the one you really need.

The list of files to be used for the effects application can be viewed and changed at the **Processed Files** tab:



Check the files with the mouse or uncheck the ones that should be excluded from the batch effects application. You can also load

additional files to the list using the **Open File** button. When ready press the **OK** button to accept changes and close the window, or **Cancel** to discard all the changes and continue work with **AVS Audio Editor**.

Note! **Note:** the **Batch Processing** is used to apply multiple effects to one or several audio files. To save multiple files please use the **Save All**  command button on the **Home** or **File** tabs.

Starting Mix Project

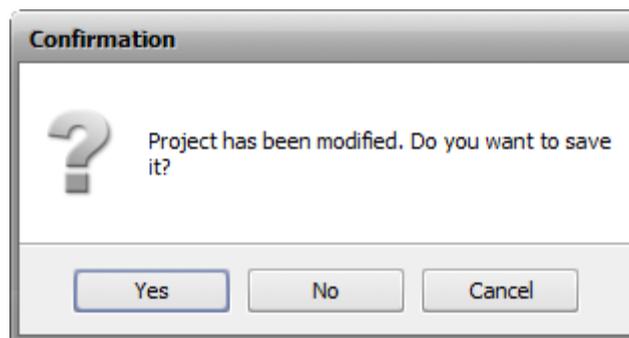
AVS Audio Editor lets you mix several audio tracks in one and apply audio effects to your mixes.

To start audio track mixing switch to the Mix tab of the Ribbon Command Bar.

The work in the Mix tab is done using the projects. Projects are very useful in case you do not have a possibility to create, edit and save your mix all at once. Using projects allows you to save the work done (even if it's not completed yet) to a file of special format and proceed with mixing next time by opening the previously saved project file.

Creating a New Project

Normally when you switch to this tab a new empty project is automatically created for you. But you can create a new project any time during mixing clicking the **New Project**  button on the Mix tab. You will be presented with the confirmation window asking you whether to save or not what's currently loaded in the mix area:



Pressing **Yes** will open the **Save Project** window and you'll be able to save the current mix as a project. Pressing **No** will just clear up all lines in the mix area without saving what's loaded there.

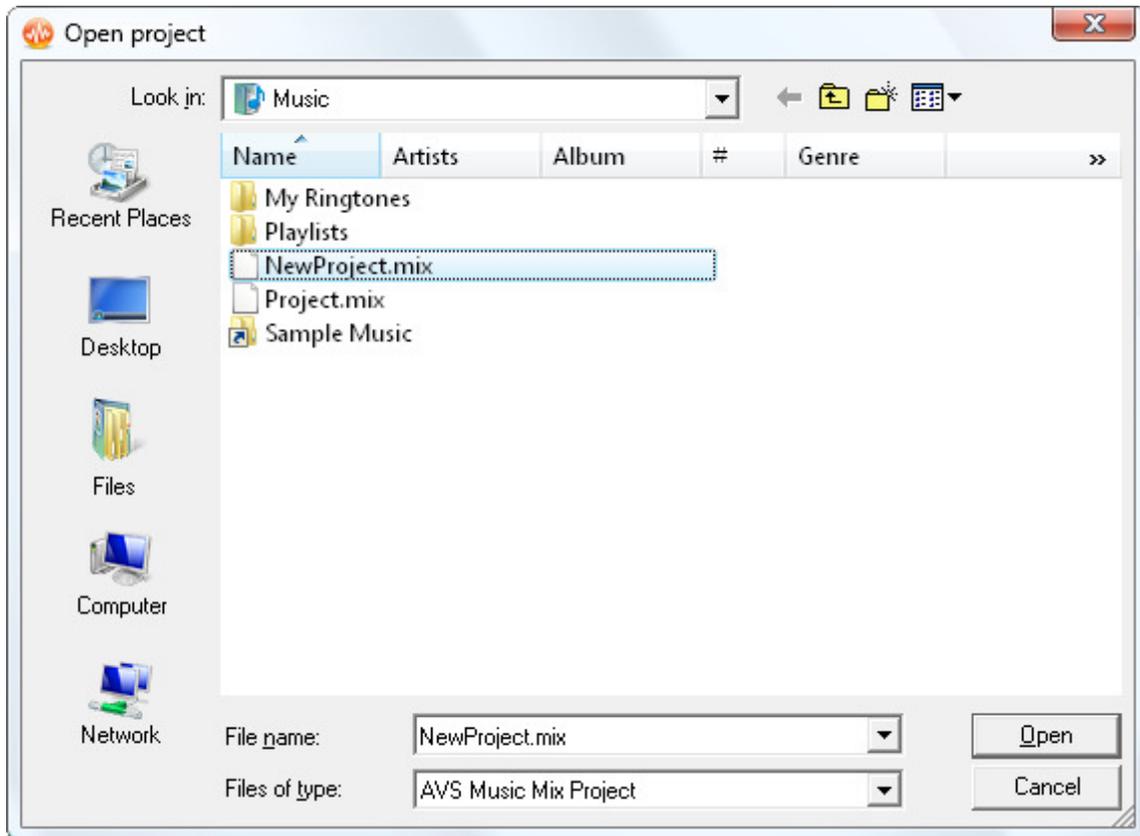
Once the project is created, add audio tracks that you plan to use in your mix in one of the following ways:

- To add a previously loaded file, switch to the **Files** tab on the **Effects and Filters** panel and double-click the needed file.
- To add a new file, click the **Open Audio File** button on the Mix tab, select the necessary file in the opened Windows Explorer window and click the **Open** button.
- Alternatively, right-click within the **Audio Mix** line of the **Mix Timeline**, choose the **Open and Add File to Mix** option, select the necessary file in the opened Windows Explorer window and click the **Open** button.

All the selected tracks will be added to the **Mix Timeline** and placed one under another.

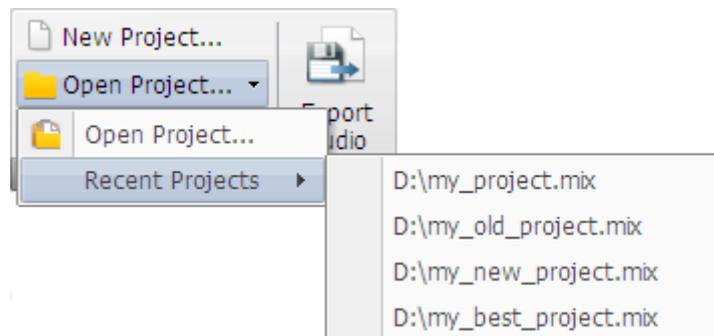
Opening an Existing Project

To open a previously saved mix project for further editing, click the **Open Project...**  button on the Mix tab.



In the opened window browse for the needed project and click **OK**. The selected project will be loaded to **AVS Audio Editor**.

Alternatively, click the arrow next to the **Open Project...** button and select the needed project from the **Recent Projects** list.



The selected project will be loaded into the program and displayed in the mix working space.

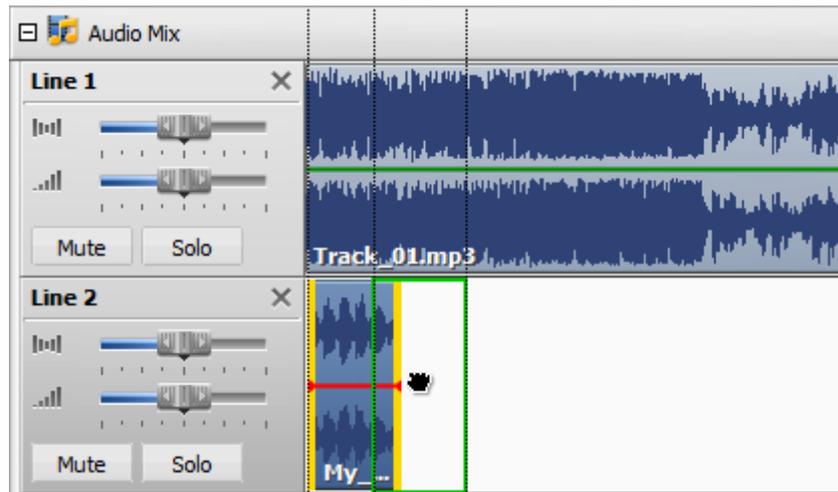
Editing Mix Project

All the audio tracks you add will be displayed within the **Mix Timeline** which helps to manage your files and edit them.

Changing Audio File Position

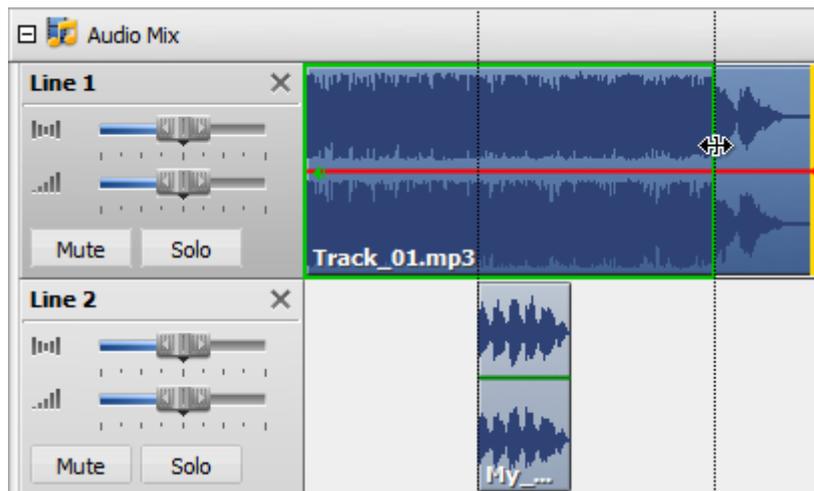
All the audio tracks added for mixing are placed one under another by default. This means they will be played back simultaneously. You may change the position of any audio track by dragging. Files may be dragged along their original line or placed to any other line below or above.

Click with the left mouse button within the audio track block and move it up or down to change the file order, or move it to the left/right to change the file position within its original line.



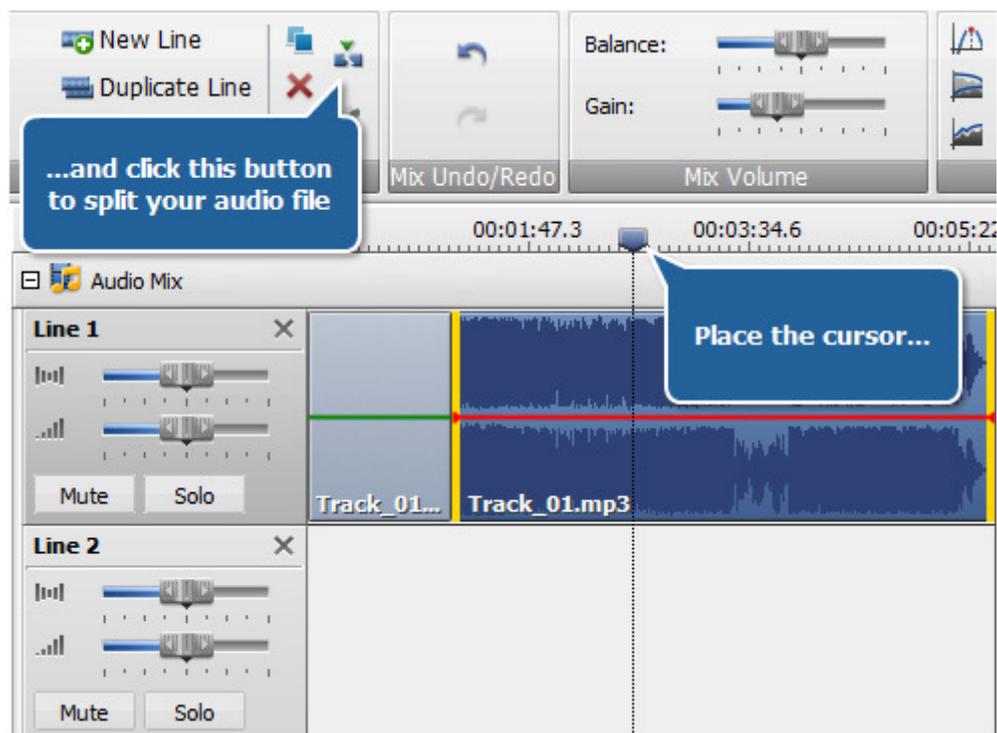
Trimming Audio File

To cut out the beginning or end of an audio track, drag its boundaries inwards.



Splitting Audio File and Deleting Parts

To split an audio track in parts, place the cursor where you want the file to be split and click the **Split Object** command button on the Mix tab or use the corresponding option of the right-click menu.



After that you can delete some unwanted parts selecting them with the mouse and clicking the **Delete Object** command button on the Mix tab or using the corresponding option of the right-click menu.

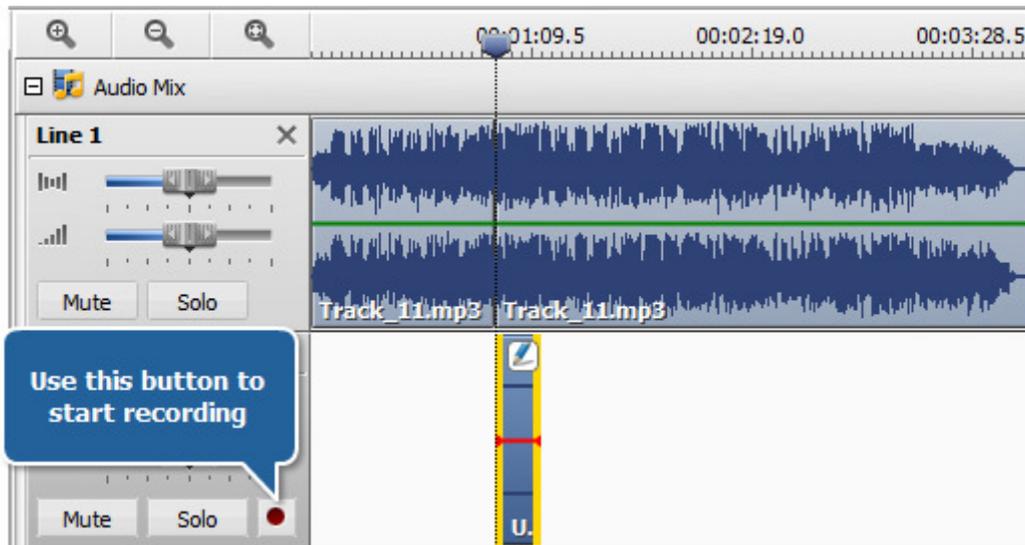
Adjusting Audio File Volume and Balance

To adjust the audio file volume and balance use the corresponding sliding bars on the left of the selected line. You can also mute the selected audio file or mute all the audio files except the selected one by clicking the **Mute** or **Solo** buttons respectively.

To adjust the volume and balance of **whole** audio mix use the sliding bars of the **Mix Volume** button group on the Mix tab.

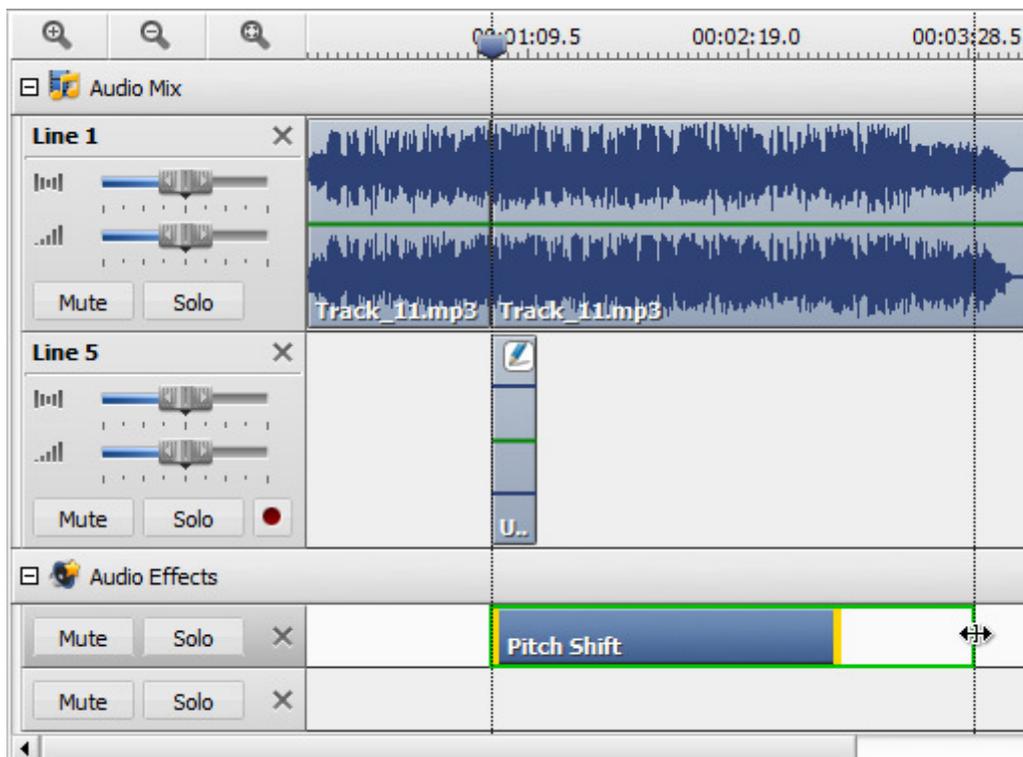
Inserting Audio Recording

To insert some voice recording place the mouse cursor within the **Audio Recording** line of the **Mix Timeline** where you want your recording to start and click the **Record** button on the left.



Applying Effects

To apply an effect to your audio mix, click the needed effect button on the Mix tab. The effect will be added to the **Audio Effect** line of the **Mix Timeline**. The applied effect covers only a part of your mix by default. You may change the effect application area by dragging its boundaries.



To change the applied effect properties, double-click it on the **Mix Timeline**. This will open the effect properties window. To learn more about configuring the parameters of some specific effect, please refer to the corresponding effect page in the **Using Effects and Filters** section.

To delete an effect, use the **Delete Object** command button on the Mix tab or right-click the effect and select the corresponding option of the right-click menu.

Note! **Note:** the effect will be applied to **ALL** audio tracks used in your audio mix.

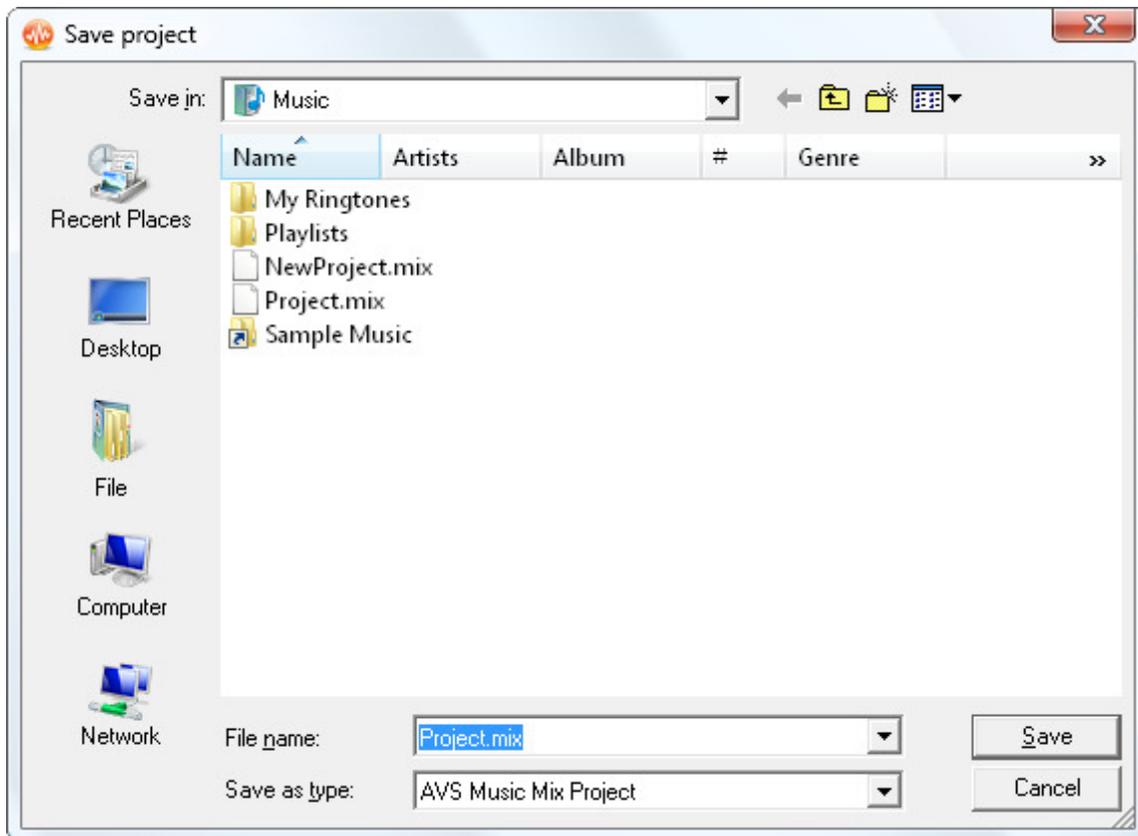
When the mix is ready use the **Playback** button on the **Bottom Toolbar** to check the results of your work.

Saving Mix Project

AVS Audio Editor allows you to export your created audio mix to an audio file or just save it to a **.mix** project file.

Saving to a Project File

Saving an audio mix to a project file gives you the possibility to go back to it later and make some further changes. To do that click the **Save Project** button of the **Mix Project** button group. The following window will be opened:



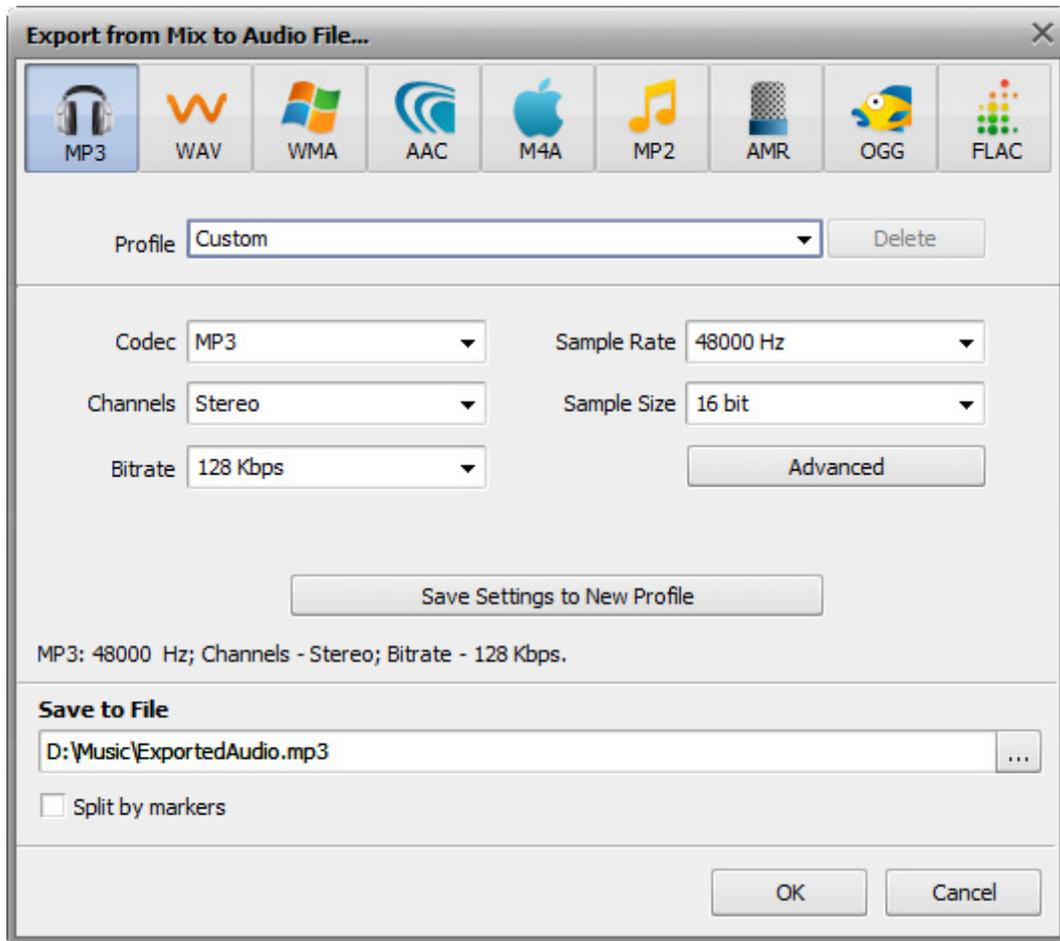
Type in the name for your project in the **File Name** field, select the project location from the **Folder** drop-down list and click the **Save** button.

A project file with the **.mix** extension will be saved to the specified location.

Saving to an Audio File

Saving the created audio mix to an audio file will produce an audio track of the selected format.

To do that click the **Export Audio** command button on the Mix tab. The **Export Mix to Audio File** window will be opened:



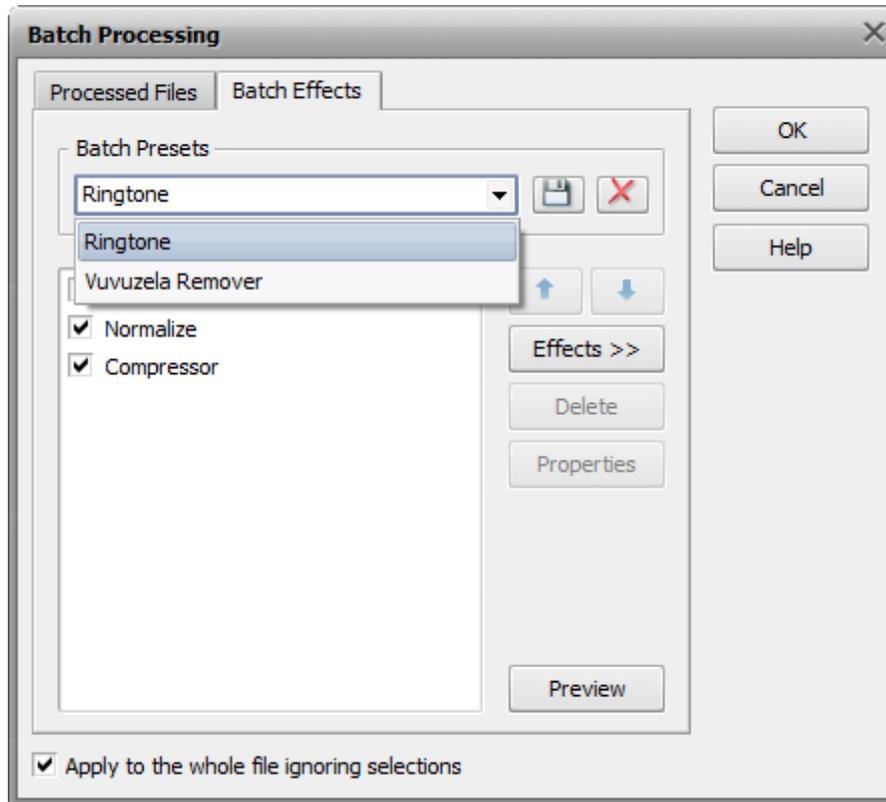
Select the output audio file format and specify the output audio parameters such as **Sample Rate (Frequency)**, **Bitrate** and the number of **Channels**.

Set a location for you audio track and click the **OK** button to start saving.

Batch Processing

When you want to apply one and the same effect to several files, or several effects to one and the same file, or several identical effects to several files, it can be done quite easily with the help of the **Batch Processing** option of **AVS Audio Editor**.

Click the **Batch** command button on the **Effects** tab. The following window will be opened:

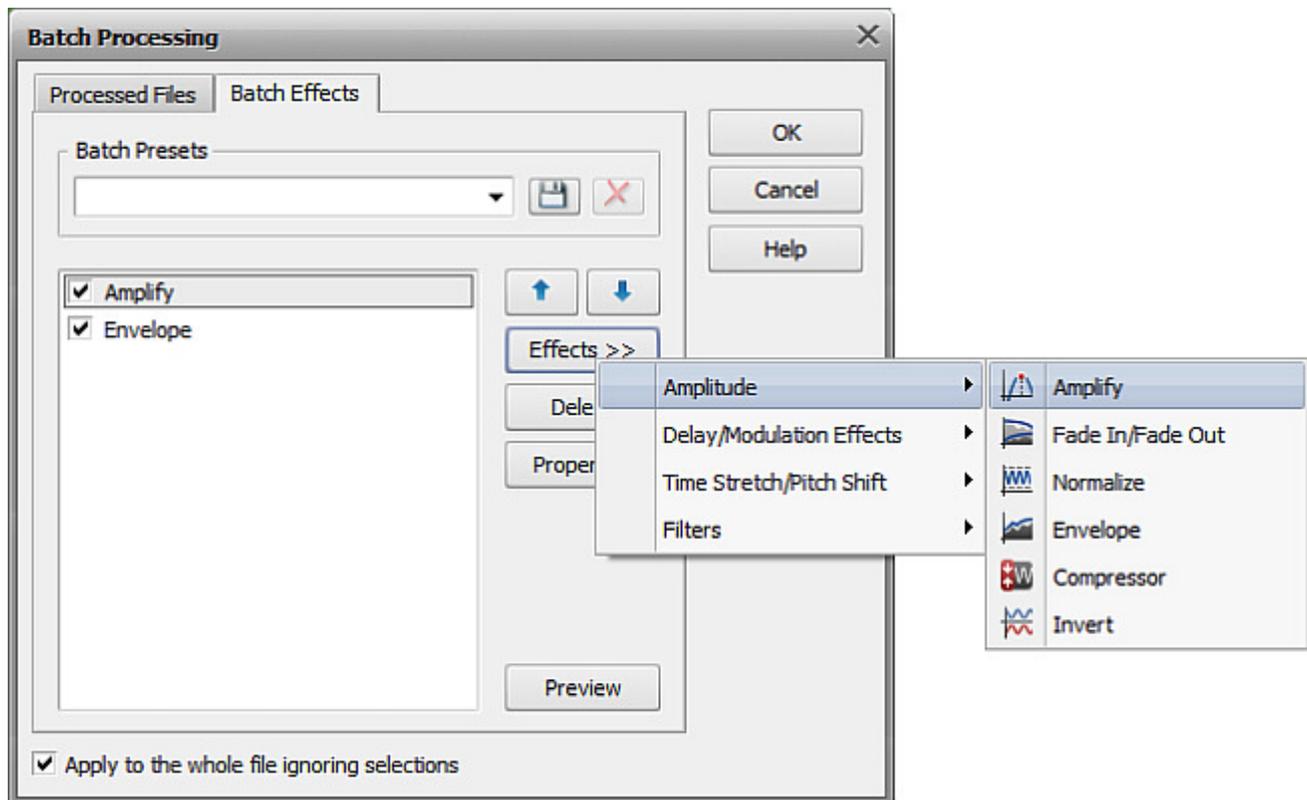


Here you can select the effects you would like to apply to the audio file (or files, if there are several of them loaded into **AVS Audio Editor**), the order in which they will be applied and their properties.

First of all you can select an effect preset from the list of available ones using the **Batch Presets** drop-down list. You can also delete the presets you no longer need using the **Delete Preset from List** Delete Preset from List button button. In case you change some of the preset settings (effects in the list, effect parameters, or something else) or create an absolutely new preset, you can save it using the **Save Preset** Save Preset button.

To change the presets order select one of the effects with the mouse and use the **Move Up** Move Up button and **Move Down** Move Down button buttons.

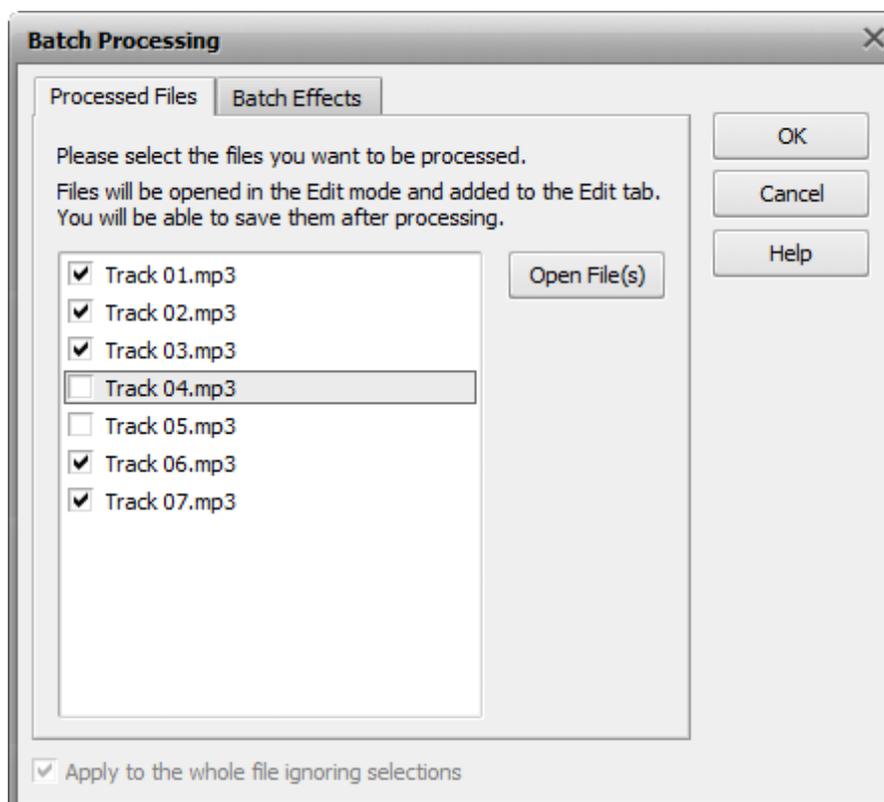
The new effects are added to the effects list using the **Effects >>** button. The list of all the available effects will be opened to let you select the necessary one:



After the effect is added its settings can be easily changed using the **Properties** button. Please, read the **effects description** section to find out more on each effect settings.

The effects are removed from the list with the help of **Delete** button (do not forget to select the effect to be deleted with the mouse before you press that button). The **Preview** button is used to make sure that the result of the effects application is the one you really need.

The list of files to be used for the effects application can be viewed and changed at the **Processed Files** tab:



Check the files with the mouse or uncheck the ones that should be excluded from the batch effects application. You can also load additional files to the list using the **Open File** button. When ready press the **OK** button to accept changes and close the window, or **Cancel** to discard all the changes and continue work with **AVS Audio Editor**.



Note: the **Batch Processing** is used to apply multiple effects to one or several audio files. To save multiple files please use the **Save All** Save All button command button on the **Home** or **File** tabs.

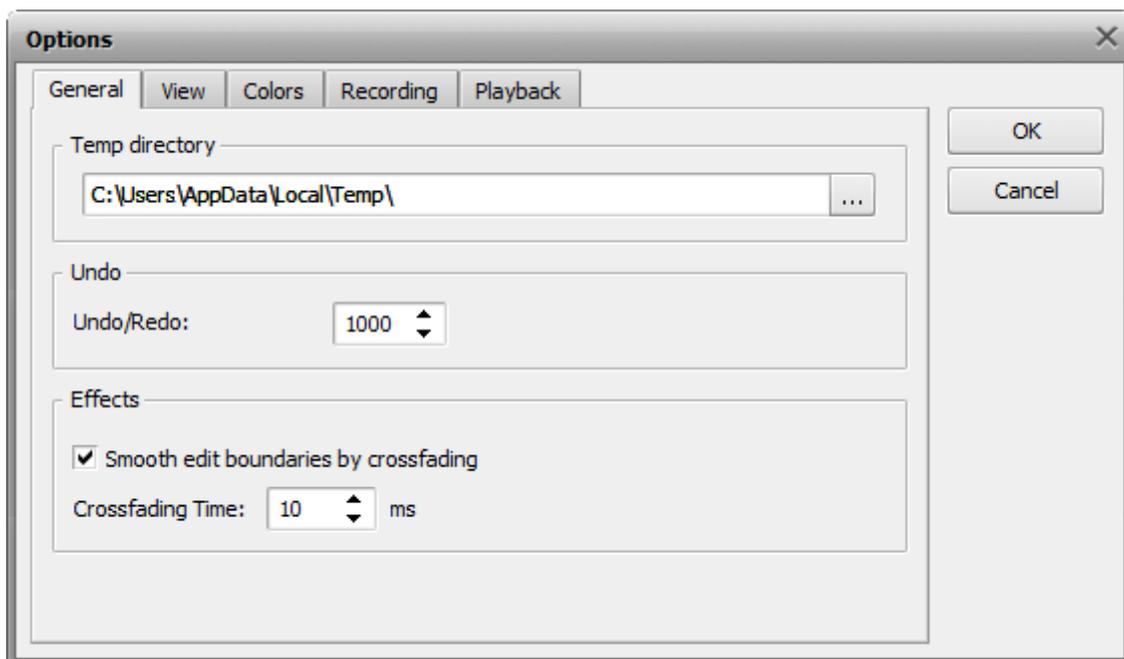
Setting AVS Audio Editor Parameters

You can customize **AVS Audio Editor** colors, devices and other settings so that it were convenient for you to work with the program. To open the **Options** window you need to click the **Options** Preferences button button on the **Tools** tab.

The **Options** window contains the following tabs:

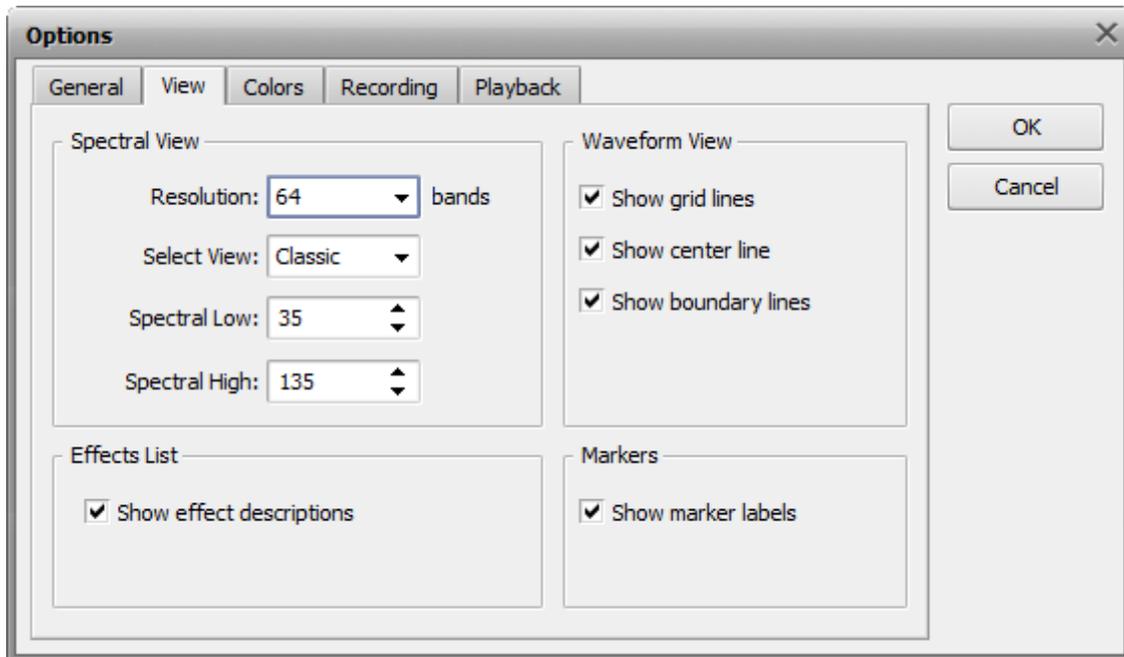
- **General settings**
- **View settings**
- **Colors settings**
- **Recording settings**
- **Playback settings**

The **General** settings tab lets you edit the following parameters:



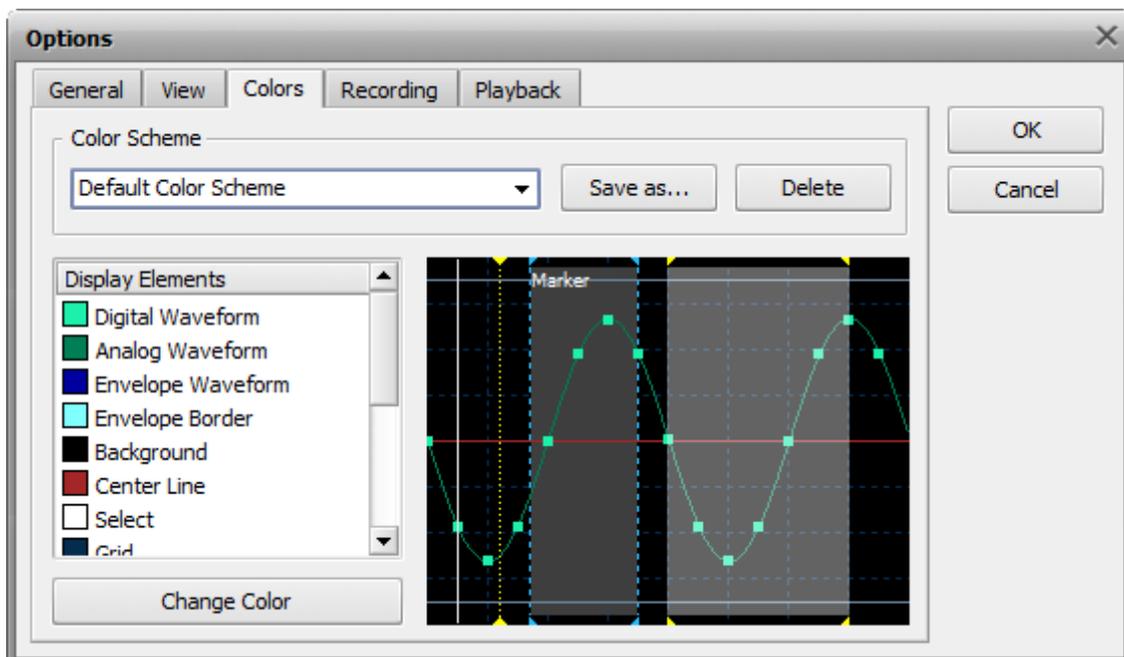
- **Temporary directory** - the temporary folder used to store the currently edited files. It should be large enough to store uncompressed versions of the edited files as **AVS Audio Editor** works with files in its own audio format.
- **Undo - Undo/Redo count** - the maximal number of undo/redo operations that can be performed.
- **Effects** - enable the **Smoothly edit boundaries by crossfading** option to determine the threshold of the effect. Normally when this option is enabled the effect will start gradually, there is no sharp transition from the audio without effect to the audio where the effect is applied. You can also set the **Crossfading time** (in milliseconds) during which this transition will be applied.

The **View** settings tab lets you edit the following parameters:



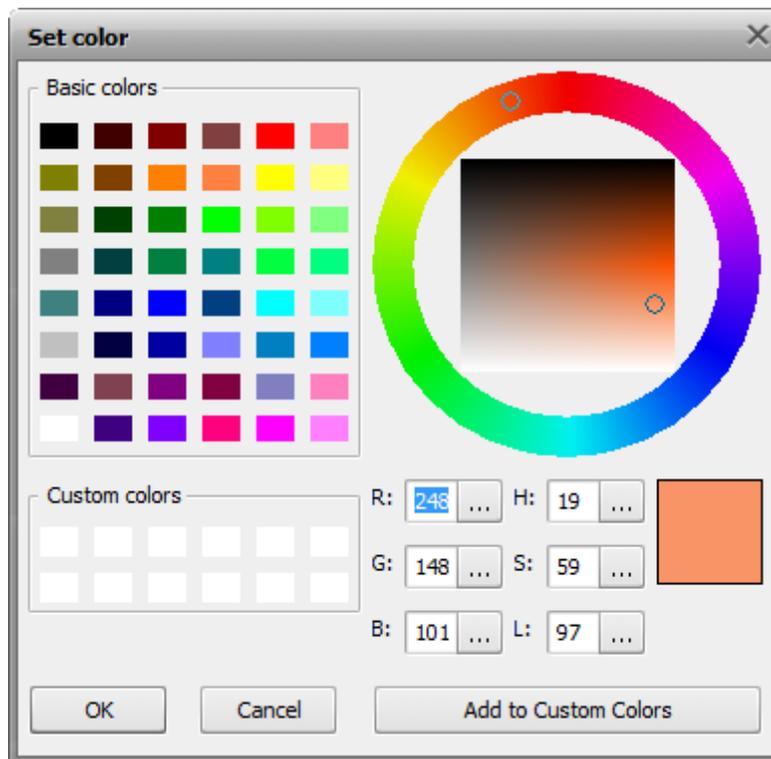
- **Spectral Display** - change the view of the audio timeline when the **Spectral View** is selected. You can change the **Resolution**, selecting the number of bands used to display a sample (**16 to 4096** bands), choose the **Spectral View** color scheme from the presets list or manually alter them using the **Spectral Low** and **Spectral High** fields (**0 to 360** values of colors selected from the color circle).
- **Effects list** - use this option to enable or disable the display of effect tooltips in the **Effects and Filters Panel**.
- **Waveform Display** - change the view of the audio timeline when the **Waveform View** or **Envelope View** is selected. You can enable/disable the grid lines, central line and the waveform boundary lines display.
- **Markers** - use this option to enable or disable the display of the marker names. When enabled this option will allow the markers to be shown with the names assigned to them automatically or by user in the **Markers List**.

The **Colors** settings tab lets you edit the following parameters:



The **Color Scheme** pull-down menu contains all the available color schemes for **Waveform View** and **Envelope View**. In case you would like to choose one, simply select it from the list. The currently selected color scheme will be displayed in the example window in the right part of the page.

The **Display elements** represent a list of all the elements within the waveform display with their current color. This list will change to reflect the changes made to any of the display elements. To change the color of any of the display elements you can double-click the necessary element or click the element and press the **Change Color** button. This will bring the color selection dialog:

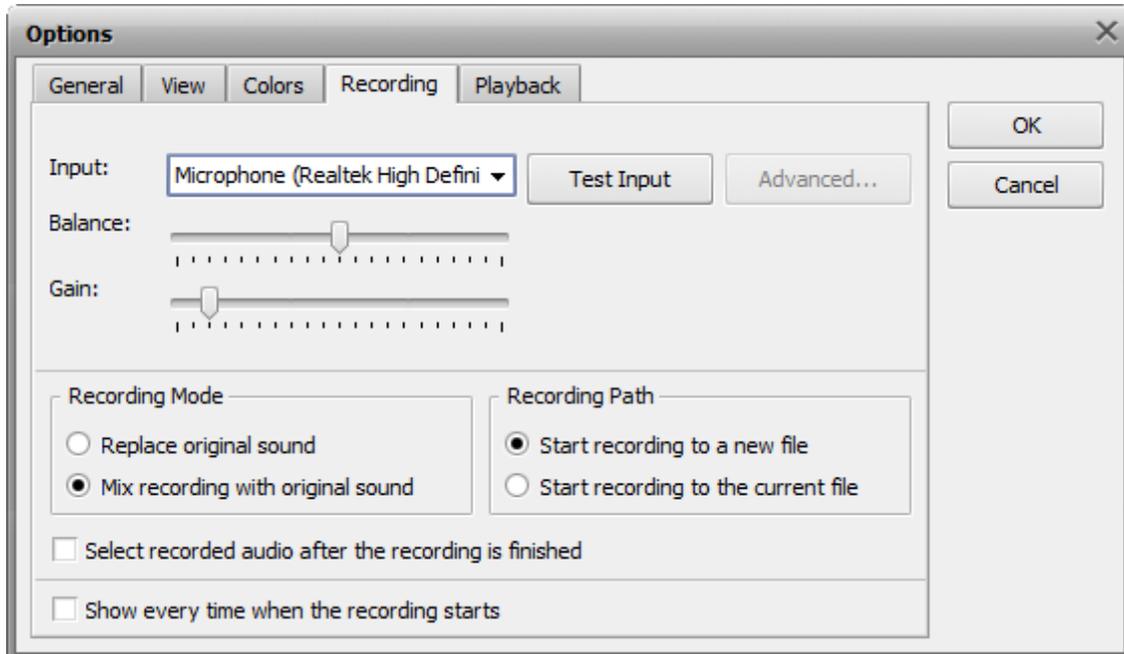


Use this dialog to choose a color for the currently highlighted item in the **Display Elements** list.

To save the current color scheme as a preset press the **Save As...** button.

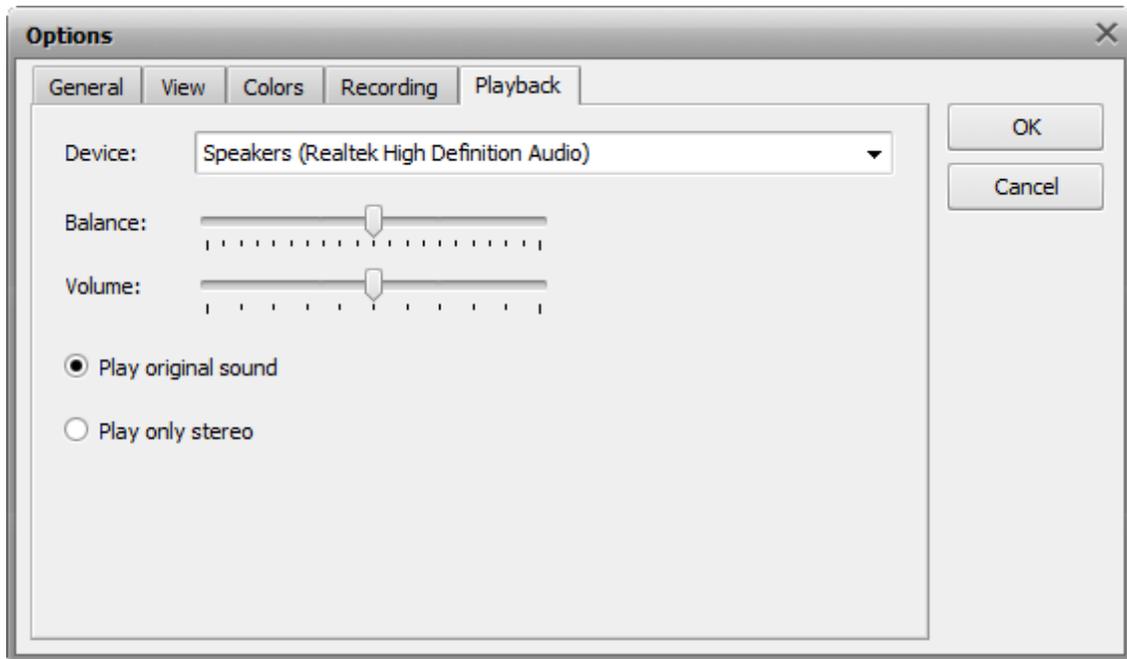
To delete the currently selected preset press the **Delete** button.

The **Recording** settings tab lets you edit the following parameters:



You can select the input device used for sound recording, the input jack of the selected device, set the balance, gain and some advanced device parameters, if any. You can also test the device to make sure it works. See the **Playing and Recording** section of help file for more detail on this.

The **Playback** settings tab lets you edit the following parameters:



You can select the input device used for playback, as well as set its volume and balance. See the **Playing and Recording** section of help file for more detail on this.

Bitrate

Bitrate is defined as the number of data points used to approximate the true wave form. Obviously, the higher the number, the more accurately the wave form is approximated.

For example, the current de facto standard is to encode MP3 at 192 kbps, or 192,000 bits per second. The CODEC takes the bitrate into consideration as it writes each frame to the bitstream. If the bitrate is low, the irrelevancy and redundancy criteria will be measured harshly, and more subtlety will be stripped out, resulting in a lower-quality product. If the bitrate is high, the codec will be applied with leniency, and the end result will sound better. Of course, the file size of the end product corresponds directly with the bitrate.

192 kbps is an example of a constant bitrate (**CBR**) mode. **Constant Bit Rate (CBR)** encoding maintains the same bitrate throughout an encoded file. All that means is no matter what, there will always be 192 kbps written into the bitstream. If your audio has moments of silence, it is captured and encoded at that rate, as are areas of very diverse frequencies (that might sound better if encoded to a higher bitrate).

Variable Bit Rate (VBR) is an **MP3** encoding method that's used when file size is not an issue. As it's name implies, the bitrate is varied throughout the file. The codec guesses which parts could benefit from more bits per second, and which can use less. The result is a much higher quality file.

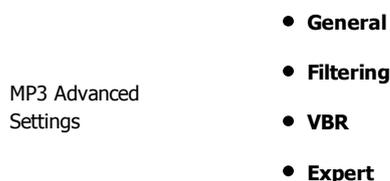
Frequency

Sampling **frequency** also impacts fidelity. The sampling frequency is essentially the number of times the sound event is quantized within a given time period. Sampling frequencies are specified in KiloHertz (KHz), a term meaning samples per second. The key in understanding how sampling frequency affects fidelity is the Nyquist sampling theorem. Basically, when applied to audio signals the Nyquist theorem states that the highest possible pitch in the sound is one-half that of the sampling frequency.

For example, "CD-quality" sound requires 16-bit words sampled at 44.1 KHz. Essentially this means 44,100 16-bit words (705,600 bits) are used to digitally describe each second of sound on a compact disc. The highest pitch possible is 22.05 KHz (approximately the top of human hearing range), which is half of 44.1 KHz.

MP3 Advanced Settings

MP3 Advanced window contains 4 tabs, which are:



General

At the **General** tab you can select the necessary **Channels**. The suggested choice includes the following variants: **Stereo**, **Joint Stereo**, **Forced Joint Stereo**, **Dual Channels** and **Mono**. See the description of each mode in the table below:

Channel	Description
Stereo	In this mode, the encoder makes no use of potentially existing correlations between the two input channels. It can, however, negotiate the bit demand between both channels, i.e. give one channel more bits if the other contains silence.
Joint stereo	In this mode, the encoder will make use of a correlation between both channels. The signal will be matrixed into a sum ("mid") and difference ("side") signal. For quasi-mono signals, this will give a significant gain in encoding quality. Unlike stereo, this mode does not destroy phase information that may be used by other encoders. This setting can be used to encode surround signals.
Forced Joint Stereo	This mode will force MS joint stereo on all frames. It's faster and it uses some special mid and side masking threshold.
Dual Channels	In this mode, the 2 channels will be totally independently encoded. Each channel will have exactly half of the bitrate. This mode is designed for applications like dual languages encoding (for example: English in one channel and French in the other). Using this encoding mode for regular stereo files will result in a lower quality encoding.
Mono	This option will generate a mono file, if the input file is a stereo file, the input stream will be downsampled to a mono file by averaging the left and right channel.

In the right part you will find the opportunity to set **Encoder Quality**. You can specify the output quality; thus you can trade off encoding time against sound

quality. The default (slider in the middle) is recommended for lower bitrates (<160 kbps), high quality (moving the slider in the direction of "better") for bitrates >160 kbps.

MP3 Advanced Settings

There are some built-in presets you can use. They have for the most part been subject to and tuned via rigorous double blind listening tests to verify and achieve this objective. These are continually updated to coincide with the latest developments that occur and as a result should provide you with nearly the best quality currently possible. You can find them in **Preset** and **Alt-Preset** lists.

Presets available:

Phone, SW, AM, FM, Voice, Radio, Tape, HiFi, CD, Studio, R3Mix.

Alt-Presets available:

Preset	Description
None	Do not use any alt-presets
Fast Standard	VBR mode preset. It should generally be understandable to most people with most music and is already quite high in quality. The resulting bitrate should be within the 170-210 kbps range, according to music complexity. Enables the new fast VBR method for a Standard preset. Its disadvantage is that often the bitrate will be slightly higher than with the normal mode and quality may be slightly lower also.
Standard	VBR mode preset. It should generally be understandable to most people with most music and is already quite high in quality. The resulting bitrate should be within the 170-210kbps range, according to music complexity.
Fast Extreme	VBR mode preset. If you have extremely good hearing and similar equipment, this preset will provide slightly higher quality than the "standard" mode. The resulting bitrate should be within the 200-240 kbps range, according to music complexity. The "fast"-option increases speed significantly but may give a tiny bit lower quality.
Extreme	VBR mode preset. If you have extremely good hearing and similar equipment, this preset will provide slightly higher quality than the "standard" mode. The resulting bitrate should be within the 200-240 kbps range, according to music complexity.
Insane	CBR mode preset. The "insane"-option gives you the current theoretical maximum quality possible. The output files are flat 320 kbps. Using this may be a little insane, since the difference in quality in between "extreme" and "insane" is minimal. However, if you simply don't care about file size, want maximum quality, or you have hardware that can't handle VBR files, then you could use this option.

Filtering

Highpass Filter cuts the lowest frequencies and passes the highest. The **Lowpass Filter** allows only the lower frequencies to be present into the output signal; it will cut the beautiful crystal sound of a violin (frequencies over 10 KHz), but if it could amplify rather than just pass the low frequencies, than it would enhance your favorite disco music with lots of percussions and bass.

MP3 Advanced Settings

Highpass filtering frequency (in kHz): Frequencies below the specified one will be cut off.

Width of Highpass filter (in kHz): The width of the highpass filter. The default is 15% of the highpass frequency.

Lowpass filtering frequency (in kHz): Frequencies above the specified one will be cut off.

Width of Lowpass filter (in kHz): The width of the lowpass filter. The default is 15% of the lowpass frequency.

Regarding to the Nyquist Sampling Theorem the sample rate have to be at least two times higher than the highest frequency of analog audio signal. For example, the human ear can detect sound across the frequency range of 20 Hz to 20 kHz. According to the sampling theorem, one should sample sound signals at least at 40 kHz in order for the reconstructed sound signal to be acceptable to the human ear.

So applying the low-pass analog filter to a file with sample rate of 8kHz you should take into consideration that the frequency parameter of this method have to be at least two times less (< 4 kHz) than the file's sample rate.

VBR

At **VBR** tab you should select Coding Method first. The description of each method can be found in the table below:

Coding Method	Description
None	Don't use VBR, instead it is encoding with a Constant Bit Rate (CBR)
Default	Use the default VBR method (currently set to VBR-MTRH)
Old	The functional approach, based on maskings, bisection in the bit domain
New	The approach, based on maskings and direct noise allocation
MTRH	A merger of old and new (VBR) routine
ABR	The Average Bit Rate (ABR) setting, the encoding principle is based on perceptual entropy, but more like CBR than VBR

MP3 Advanced Settings

Maximum VBR Bitrate: Allows to specify an maximum bitrate when using VBR (Variable Bit Rate), this selecting depends on what base bitrate you have chosen in the main encoder tab. It's recommended to leave this set to 320 kbit/s unless you want low quality VBR files.

VBR Quality: In VBR mode, you are able to specify a quality setting which will affect encoding bitrate allocation. If you use quality 0, the max bitrate will be reached easily, while using quality 9 the bitrate usually will be around the base bitrate. The lower the VBR quality value, the better the audio quality, but also the bigger the output file. Recommended setting for high quality VBR encoding is 1 or 0.

Write VBR Header: This tag is embedded in frame 0 of the MP3 file. It lets VBR aware players correctly seek and calculate playing times of VBR files.

Target bitrate for ABR: The allowed range of the ABR bitrate is 4 - 310 kbit/s, you can use any integer value within that range.

Expert

At **Expert** tab you can find the opportunity to set the following options:

- **include CRC-Checksum**
When enabled, the encoder will calculate the cyclic redundancy check (CRC) for the MP3 frames, and will add the CRC value to the MP3 stream.
- **comply as much as possible to ISO MPEG spec**
With this option, the encoder will enforce the 7680 bit limitation on total frame size. This results in many wasted bits for high bitrate encodings.
- **allow block types to differ between channels**
Allows the left and right channels to use different block types. Normally this is not allowed, only because the FhG encoder does not seem to allow it either.
- **short blocks**
Encode all frames using short blocks.
- **bit reservoir**
Enable bit reservoir.

MP3 Advanced Settings

You can also set **ATH Control** function, useful for low volume. ATH is used to approximate an equal loudness curve. Select one of the suggested modes:

ATH Control Mode	Description
Default	
Only	This option ignores the output of the psy-model and only use masking from the ATH. Might be useful at very high bitrates or for testing the ATH.
Disabled	Disable any use of the ATH (absolute threshold of hearing) for masking. Normally, humans are unable to hear any sound below this threshold.
Only for short blocks	Ignore psychoacoustic model for short blocks, use ATH only.

There are also three **De-emphasis** options: **default**, **0/15 microseconds** and **citt j.17**.

Available MP3 Parameters Combinations

You can find the available MP3 parameters combinations in the table below:

Frequency	Mono		Stereo	
	Minimum	Maximum	Minimum	Maximum
48000 Hz	64 Kbps	160 Kbps	128 Kbps	320 Kbps
44100 Hz	64 Kbps	160 Kbps	128 Kbps	320 Kbps
32000 Hz	32 Kbps	160 Kbps	64 Kbps	320 Kbps
24000 Hz	32 Kbps	80 Kbps	64 Kbps	160 Kbps
22050 Hz	32 Kbps	80 Kbps	64 Kbps	160 Kbps
16000 Hz	24 Kbps	80 Kbps	48 Kbps	160 Kbps
12000 Hz	24 Kbps	80 Kbps	48 Kbps	160 Kbps
11025 Hz	24 Kbps	80 Kbps	48 Kbps	160 Kbps
8000 Hz	16 Kbps	80 Kbps	32 Kbps	160 Kbps

Frequency	8000 Hz		11025 Hz		12000 Hz		16000 Hz		22050 Hz		24000 Hz		32000 Hz		44100 Hz		48000 Hz	
	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo
8																		
16	+																	
24	+		+		+		+											
32	+	+	+		+		+		+		+		+					
40	+	+	+		+		+		+		+		+					
48	+	+	+	+	+	+	+	+	+		+		+					
56	+	+	+	+	+	+	+	+	+		+		+					
64	+	+	+	+	+	+	+	+	+	+	+	+	+	+	+		+	
80	+	+	+	+	+	+	+	+	+	+	+	+	+	+	+		+	
96		+		+		+		+		+		+	+	+	+		+	
112		+		+		+		+		+		+	+	+	+		+	
128		+		+		+		+		+		+	+	+	+	+	+	+
144		+		+		+		+		+		+	+	+	+	+	+	+
160		+		+		+		+		+		+	+	+	+	+	+	+
192														+		+		+
224														+		+		+
256														+		+		+
320														+		+		+
384																		

MP3 supports all the combinations of the frequency and bitrate if the bitrate varies from Minimum to Maximum values for the current supported frequency.

Available MP2 Parameters Combinations

You can find the available MP2 parameters combinations in the table below:

Frequency	Mono		Stereo	
	Minimum	Maximum	Minimum	Maximum
48000 Hz	32 Kbps	160 Kbps	64 Kbps	320 Kbps
44100 Hz	32 Kbps	160 Kbps	64 Kbps	320 Kbps
32000 Hz	32 Kbps	160 Kbps	64 Kbps	320 Kbps
24000 Hz	32 Kbps	80 Kbps	64 Kbps	160 Kbps
22050 Hz	32 Kbps	80 Kbps	64 Kbps	160 Kbps

Frequency	22050 Hz		24000 Hz		32000 Hz		44100 Hz		48000 Hz	
	mono	stereo								
8										
16										
24										
32	+		+		+		+		+	
40	+		+		+		+		+	
48	+		+		+		+		+	
56	+		+		+		+		+	
64	+	+	+	+	+	+	+	+	+	+
80	+	+	+	+	+	+	+	+	+	+
96		+		+	+	+	+	+	+	+
112		+		+	+	+	+	+	+	+
128		+		+	+	+	+	+	+	+
144		+		+	+	+	+	+	+	+
160		+		+	+	+	+	+	+	+
192						+		+		+
224						+		+		+
256						+		+		+
320						+		+		+
384										

MP2 supports all the combinations of the frequency and bitrate if the frequency varies from Minimum to Maximum values for the current supported bitrate.

OGG Vorbis Options

In **OGG Vorbis Options** window you will be able to set the advanced options of your output file format. Two tabs are available that allow you to change the codec settings:

- **General** tab is used to change the encoding mode;
- **Advanced Encoder Options** tab is used to change some additional codec options useful for expert users.

The **General** tab lets the user choose either **Quality** or **Bitrate** based conversion.

In the **Quality mode** it is possible to encode the output file either at a **faster** speed or with a **better** quality:

OGG Vorbis Options

Use the slider to select the necessary quality/speed ratio and bear in mind that the better the quality is the more bitrate of the output audio file will be and the greater the file size will be. At the same time the files encoded at a higher speed will be less in size and worse in quality.

If you prefer the bitrate based encoding methods, three main modes are available:

1. **CBR Mode (Constant Bitrate)** - the output audio file will be encoded with the bitrate set using the **Current nominal bitrate** slider (in kilobits per second). The bitrate will always stay at the same value, although at lower values the quality might not be optimal for the output audio file.

OGG Vorbis Options

2. **VBR Mode (Variable Bitrate)** - the output audio file will be encoded with the variation of **Current minimal bitrate** and **Current maximal bitrate** (set in kilobits per second using the appropriate sliders). The bitrate will vary between the set values depending on the complexity of the audio being encoded. This method can maintain higher audio quality although the output file size is not as predictable as with **CBR** encoding mode.

OGG Vorbis Options

3. **ABR Mode (Average Bitrate)** - the output audio file will be encoded with the variation of **Current nominal bitrate** (set in kilobits per second using the appropriate slider). The encoder will try to reach a target average bitrate while allowing the bitrate to vary between different parts of the audio as well as in the **VBR** encoding mode. As it is a form of variable bitrate, this allows more complex portions of the material to use more bits and less complex areas to use fewer bits. However, bitrate will not vary as much as in **VBR** encoding mode.

OGG Vorbis Options

The **Advanced Encoder Options** are used to change some additional codec options and might be very useful for expert users that know what they are doing. You should be very careful though while changing these options as some of them may worsen the output file quality or make it incompatible with the device or player that will be used to playback the output file.

OGG Vorbis Options

The **Advanced Encoder Options** has the following options:

- **Apply Lowpass filter at** - is used to change the highest frequency that will be retained in the encoded file (specified in Herz).
- **Apply a Scale Factor of** - is used to change the amplification of the sound volume in the output file compared to that of the input file (measured in percent).
- **Use Impulse Noise Tune** - is used to define the amplitude above which the encoder will look for artifacts in transient peaks. Lower noise tune values improve transient response but increase bitrate (the values can be both positive and negative).

Use the **Reset to Default** button to return to default settings and values, or click **OK** to accept all the changes or **Cancel** to discard changes and close the window.

Available OGG Vorbis Parameters Combinations

You can find the available Ogg Vorbis parameters combinations in the table below:

Frequency	Mono		Stereo	
	Minimum	Maximum	Minimum	Maximum
48000 Hz	48 Kbps	160 Kbps	96 Kbps	320 Kbps
44100 Hz	48 Kbps	128 Kbps	96 Kbps	256 Kbps
32000 Hz	48 Kbps	112 Kbps	96 Kbps	224 Kbps
24000 Hz	40 Kbps	80 Kbps	80 Kbps	160 Kbps
22050 Hz	32 Kbps	80 Kbps	64 Kbps	160 Kbps
16000 Hz	32 Kbps	80 Kbps	64 Kbps	160 Kbps
12000 Hz	24 Kbps	40 Kbps	48 Kbps	80 Kbps
11025 Hz	24 Kbps	40 Kbps	48 Kbps	80 Kbps
8000 Hz	8 Kbps	16 Kbps	16 Kbps	32 Kbps

Frequency	8000 Hz		11025 Hz		12000 Hz		16000 Hz		22050 Hz		24000 Hz		32000 Hz		44100 Hz		48000 Hz	
	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo
8	+																	
16	+	+																
24		+	+		+													
32		+	+		+		+		+									
40			+		+		+		+		+							
48				+		+	+		+		+		+		+		+	
56				+		+	+		+		+	+		+		+		
64				+		+	+	+	+	+	+		+		+		+	
80				+		+	+	+	+	+	+	+	+		+		+	
96								+		+		+	+	+	+	+	+	+
112								+		+		+	+	+	+	+	+	+
128								+		+		+		+	+	+	+	+
144								+		+		+		+		+	+	+
160								+		+		+		+		+	+	+
192														+		+		+
224														+		+		+
256																+		+
320																		+
384																		

Ogg Vorbis supports all the combinations of the frequency and bitrate if the bitrate varies from Minimum to Maximum values for the current supported frequency.