AVS4YOU Programs Help

AVS Music Mix

www.avs4you.com

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.

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://www.avs4you.com/OnlineHelp/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://www.avs4you.com/OnlineHelp/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://www.avs4you.com/OnlineHelp/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.

Overview

AVS Music Mix is a compact and simple loop-based music composition and production tool for original music creation. The application allows you to save the created music or a part of the project with such popular audio formats as MP3, WAV, AMR, AAC, WMA and M4A. This easy program provides the most powerful, user-friendly environment for music and songs creation.

AVS Music Mix software has more than 120 loops included into installation, plus the additional loops package is available for registered users at **www.avs4you.com**.

Note: you can also import any supported audio to **AVS Music Mix** to improve or change it.

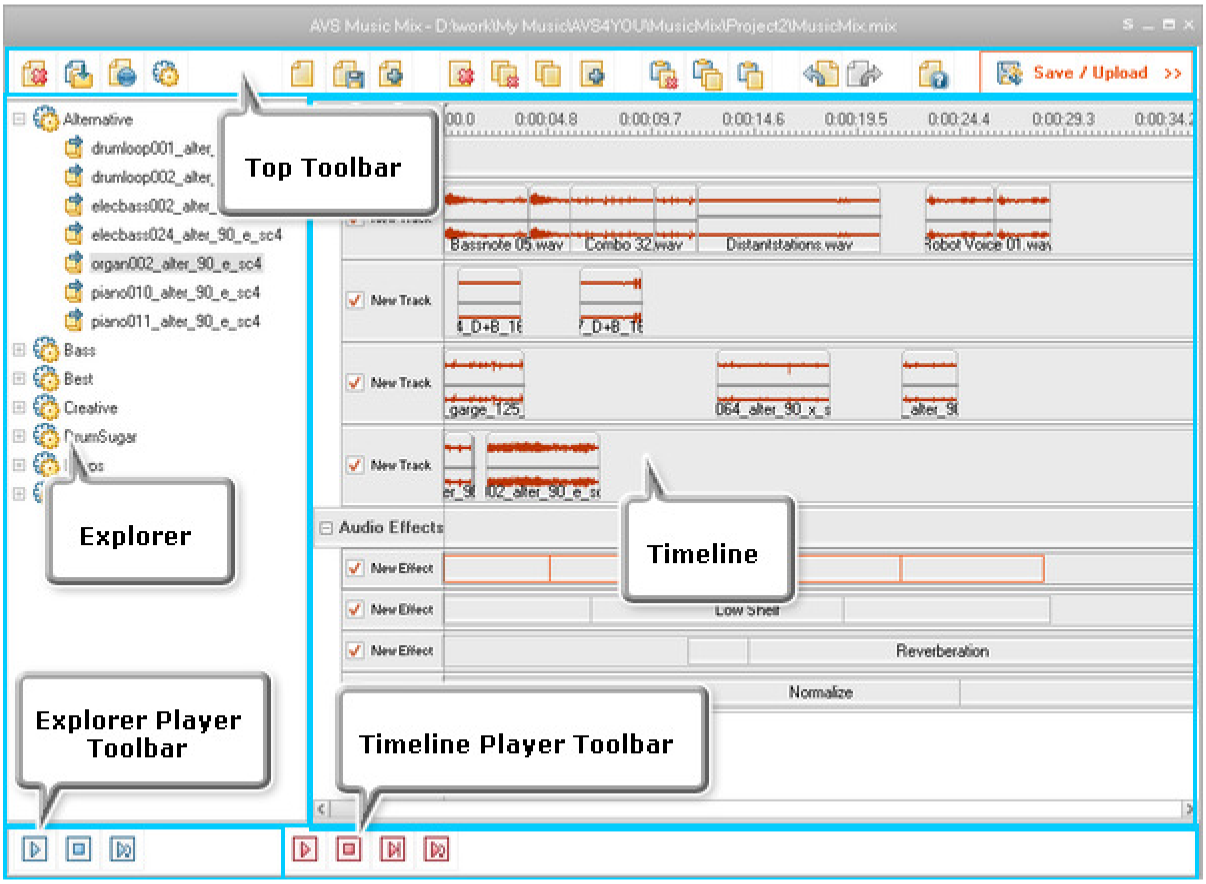
AVS Music Mix also gives you an ability to add various audio effects and filters, fade them in and out.

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

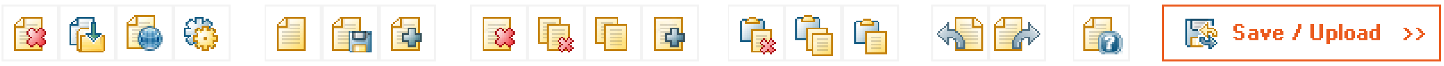
Interface Description

The interface of **AVS Music Mix** application could be metaphorically divided into five parts. They are:

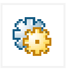
- **Top Toolbar**
- **Explorer**
- **Timeline**
- **Explorer Player Toolbar**
- **Timeline Player Toolbar**



Top Toolbar



Top Toolbar contains the following main buttons to let the user perform common operations quickly. You can find the description of each button in the table below:

Button	Name	Description
	**Delete samples from collection**	Use this button to delete the selected sample(s) from the collection.
	**Import your audio track**	Use this button to import any supported audio file to **AVS Music Mix**.
	**Sample sounds from Internet**	Use this button to load samples from the **AVS4YOU** web site.
	**Samples/Effects**	Use this button to switch between the **Effects and Samples** mode.
	**New project**	Use this button to create a new project.
	**Save mix-project**	Use this button to save the created project.
	**Load mix-project**	Use this button to load an existing project.
	**Delete all tracks**	Use this button to delete all of the tracks from the **Timeline**.
	**Delete the selected track**	Use this button to delete the highlighted track from the **Timeline**.
	**Duplicate the selected track**	Use this button to duplicated the highlighted track at the **Timeline**.
	**Add new line**	Use this button to add a new empty track to the **Timeline**.
	**Delete loop**	Use this button to delete the highlighted loop.
	**Duplicate loop**	Use this button to duplicate the highlighted loop.
	**Add loop from the selected collection**	Use this button to add loop from the collection to the **Timeline**.
	**Undo**	Use this button to undo **Timeline** operation.
	**Redo**	Use this button to redo **Timeline** operation.
	**Show Help**	Use this button to run **AVS4YOU** programs Help file.
 Save / Upload >>		Use this button to save the created project or the selected part of it into one of the supported digital audio formats.

Explorer

At **Explorer** part you can find the collections containing loops or imported audio tracks.

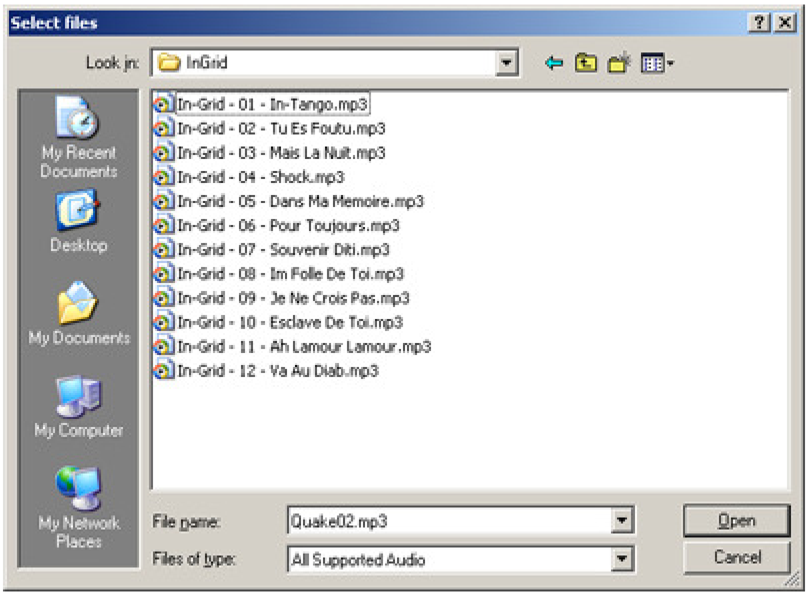
Note: you can organize the imported files into collection for their further comfortable use in your project.

To play a loop or imported audio just double click it in the **Explorer** part. You also can use **Explorer Player Toolbar**. You can find the description of each button in the table below:

Button	Name	Description
	**Play Loop**	Use this button to play the selected loop from the collection.
	**Stop**	Use this button to stop playing the loop.
	**Repeat Play**	Use this button to replay the selected sample from the collection.

To import audio to **AVS Music Mix:**

1. Click the **Import your audio track** button.
2. In standard explorer window please select file(s) to be imported.



3. Select the location for the imported files. It could be one of the existing collections or you can create a new one. Note that you can set the name of new created collection.

4. Click the **Import** button.

Timeline

AVS Music Mix has an easy to use working area which has a set of buttons. They let the user introduce various effects and modify the parameters of these effects.

AVS Music Mix allows you to apply one or several audio tracks/loops to your project. For convenience of use you can change the sequence of audio reflection or resize them at the Timeline. It's possible to create one track using fragments from several tracks. The simplest way to do that is use drag-and-drop method.

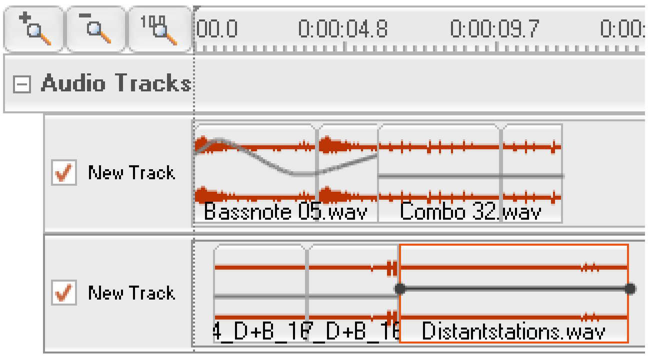
Note: this is how the way of displaying the whole audio track differs from the trimmed one:

Whole Audio Track

Trimmed Audio Track

Timeline Toolbar

Timeline toolbar allows to perform tasks such as introducing the limits of effect, boundaries of fade-in and fade-out areas, zooming in or out the details of your media.



Button	Description
	Zooms the selection in
	Zooms the selection out
	100% view
	Frames navigation
	Active Track
	Passive Track

Envelope

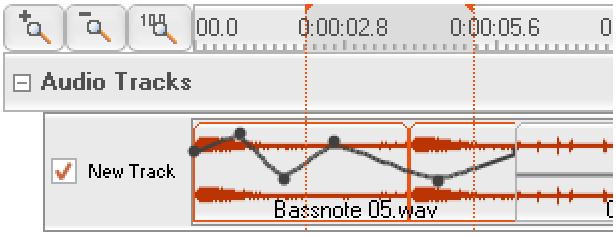
At each audio track reflected at the **Timeline** you can see an envelope that depicts the volume control. You can regulate it differently according to your project's details. To add a control point to the envelope, double-click within the envelope line (where the mouse cursor changes to a hand) and drag the control point to the location where you want it.

To move a point on the envelope, click and hold the point and drag it to a new location.

Note: in case you move the point more than 20 pixels over the track line up or down, the point will be deleted. You can use this feature to create smooth transition between two tracks. When the mouse cursor is located over a point, you will see it change from an arrow to a rhombus.

Selection

To select a part of your project, click the left mouse button and drag the slider along the navigation line. The selected area will be highlighted.



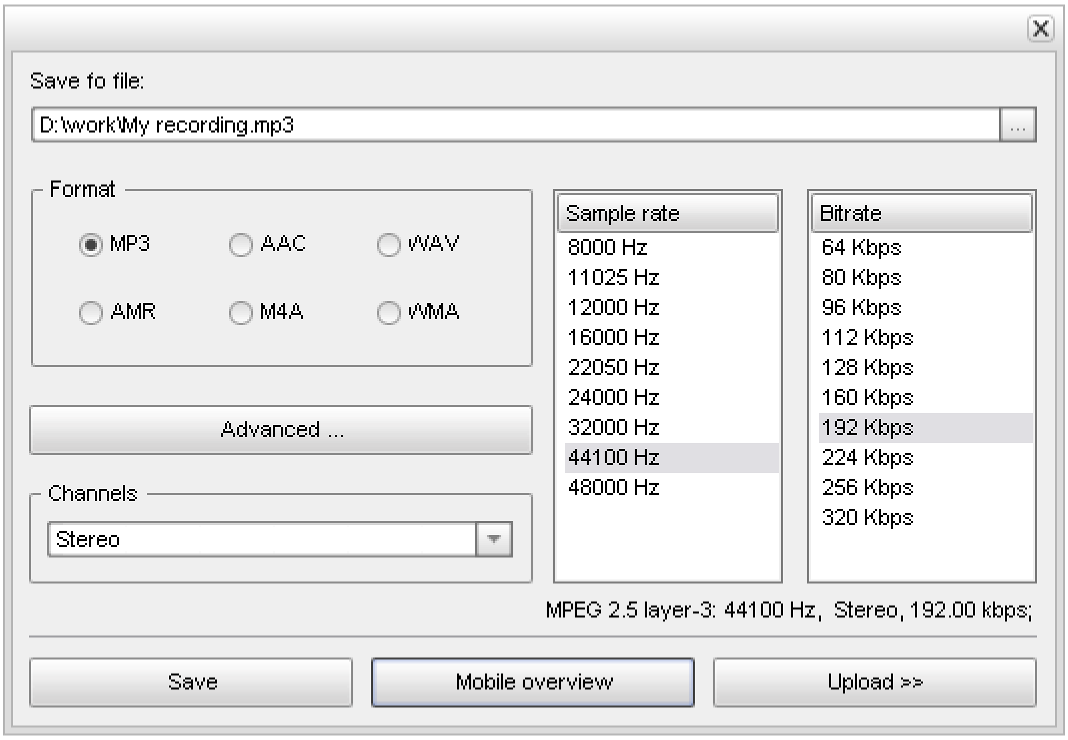
Timeline Player Toolbar

Timeline Player Toolbar consists of four buttons. See the detailed description of each button in the table below:

Button	Name	Description
	Play Mix	Use this button to play your project from the beginning to the end.
	Stop	Use this button to stop playing.
	Play from current position to the end	Use this button to play from current cursor position to the end of your project.
	Repeat Play	Use this button to replay.

Saving

AVS Music Mix offers the opportunity to save the created audio project in several common audio formats. To save the whole project click the **Save/Upload>>** button at the **Top Toolbar**. The program will suggest you to select the output file format and specify the output file format parameters such as **Sample rate (Frequency)**, **Bitrate** and **Channels**.



It is also possible to save a part of your project as a separate file. For that please select the necessary part at the **Timeline** and click **Save/Upload>>**

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

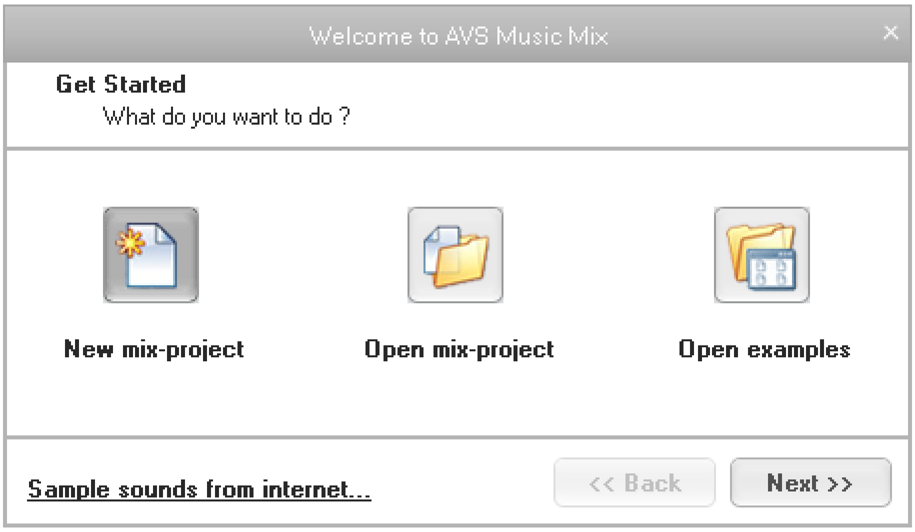
- **Available MP3 Parameters Combinations**

If you would like to upload the created audio to your mobile phone, make sure that it supports the format that you have selected, and click the **Upload>>** button. See the **AVS Mobile Uploader** program description for more detail on how to use it.

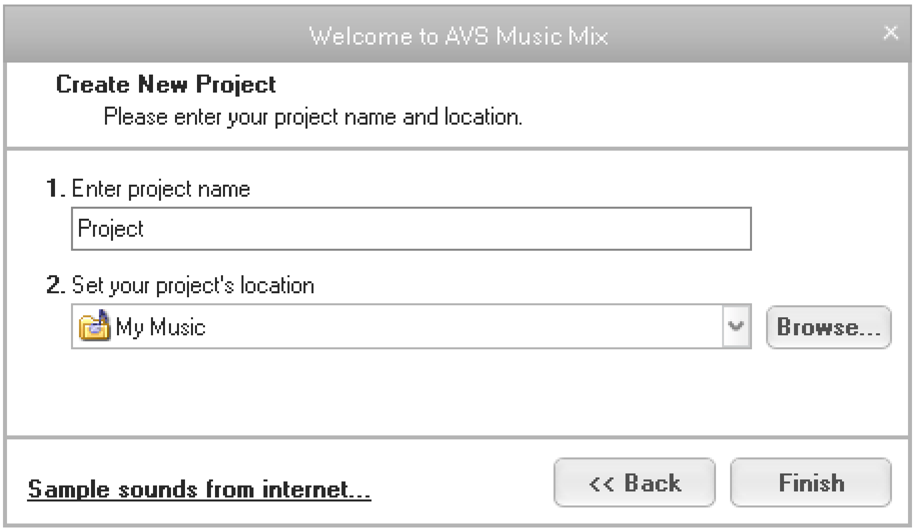
Working with Projects

Creating a project

At **AVS Music** start pane please select **Create new mix-project** item and click **Next>>**.



After that you should enter the file name for your project and set it's location. Click the **Browse...** button to open the standard Windows explorer menu and select the project location. After you are done, click the **Finish** button to finish creating a project. Please take into account that you will get an info message in case a project with the name you have typed already exists.



Saving a project

Saving your project lets you keep your current work, and then later open the file in **Music Mix** to make further changes. You can continue editing your project from where you left off when you last saved the project. When you save a project, the arrangement of loops added to the timeline and any other changes you have made are retained. To save the created project click the **Save mix-project** button .

Opening existing projects

You can use **Open existing mix-project** item in **Welcome to AVS Music Mix** window, that opens on the application start, and click **Open...** button. After that select the project to be opened in the standard explorer window and click **Open**. To open an existing project when the program is running, you should click the **Load mix-project** button . After that select the project to be opened in the standard explorer window and click **Open**.

Editing projects

To start a project and begin creating your music masterpiece, you need to add the imported audio to the **Timeline**. The loops on timeline become the contents of your project. The timeline displays your work in progress, as it reflects the timing of loops.

Effects and Filters

AVS Music Mix gives you an opportunity to apply various effects and filters to your mix projects. Using them you can achieve the desired sound. Below is the list of all the effects and filters and their brief description.

Effects and Filters:

- **Amplify**
- **Low Pass**
- **High Pass**
- **Band Pass**
- **Notch Filter**
- **Low Shelf**
- **High Shelf**
- **Vibrato**
- **Delay**
- **Pitch Shift**
- **Flanger**
- **Chorus**
- **Reverberation**
- **Compressor**
- **Expander**
- **Normalize**
- **Phaser**

Amplify

This function enables you to apply the same amount of gain change throughout the audio file.

Low Pass

The **Low Pass** filter lets you hear only low frequencies; it blocks higher frequencies.

High Pass

High Pass filter lets you hear only high frequencies. It attenuates frequencies below the certain cutoff frequency.

Band Pass

The **Band Pass** filter lets you hear a certain band of frequencies within an upper and lower range. Frequencies above and below this band are attenuated. The distance between the higher and lower cutoff frequencies in a band pass filter is called the bandwidth of the filter. The center frequency of a band pass filter is the maximum point of amplitude.

Notch Filter

The **Notch Filter** cuts specified frequency from audio data.

Low Shelf

The **Low Shelf** filter boosts or cuts frequencies below the cutoff, and passes frequencies above the shelf cutoff with no change made to their gain. Use this effect to enhance or diminish any amount of low frequency material in a sound.

High Shelf

The **High Shelf** filter boosts or cuts frequencies above the cutoff, and passes frequencies below the shelf cutoff with no change made to their gain. Use this effect to enhance or diminish any adjustable amount of high frequency material in a sound.

Vibrato

Vibrato equals to a cyclical changing of a certain frequency of the input signal.

Delay

You can use this function to create single echoes, as well as a number of other effects. Delays of 35 milliseconds (ms) or more will be perceived as discrete echoes, while those falling within the 35-15 ms range can be used to create a simple chorus or flanging effect. (These effects will not be as effective as the actual chorus or flanging effects, as the delay settings will be fixed and will not change over time).

Pitch Shift

The **Pitch Shift** effect shifts the frequency spectrum of the input signal. It can be used to disguise a person's voice, or make the voice sound like that of the "chipmunks", through to "Darth Vader". It is also used to create harmony in lead passages, although it is an "unintelligent" harmonizer.

Flanger

You can use this function to create a flanging effect by slightly delaying and phasing a signal at predetermined or random intervals. In the pull-down menu you will find a great variety of different flanger effects, to choose any of them just click the necessary one.

Chorus

You can use this function to set **Chorus Effect** that is used to mix the signal with a slightly delayed copy of itself, where the length of the delay is constantly changing.

Reverberation

You can use this function to set **Reverb Effect** that is used to simulate acoustic space, and consists of both early reflections and echoes that are so closely spaced that they are perceived as a single fading sound. Reverb is different from the basic echo function in that the delays are not repeated at regularly spaced intervals. Reverb function can create a wide range of high-quality reverb effects.

Compressor

This function enables you to reduce the dynamic range of an audio signal. For example, compressors can be used to eliminate the variations in the peaks of an electric bass signal by clamping them to a constant level (thus providing an even, solid bass line.) Compressors can also be useful in compensating the wide variations in the level of a signal produced by a singer who moves frequently or has an erratic dynamic range.

Expander

Expander effect is used to expand the dynamic range of an audio signal. Expander boosts the high-level signals and attenuates low-level signals.

Normalize

You can use the **Normalize** effect to achieve the greatest amount of amplification that will not result in clipping.

Phaser

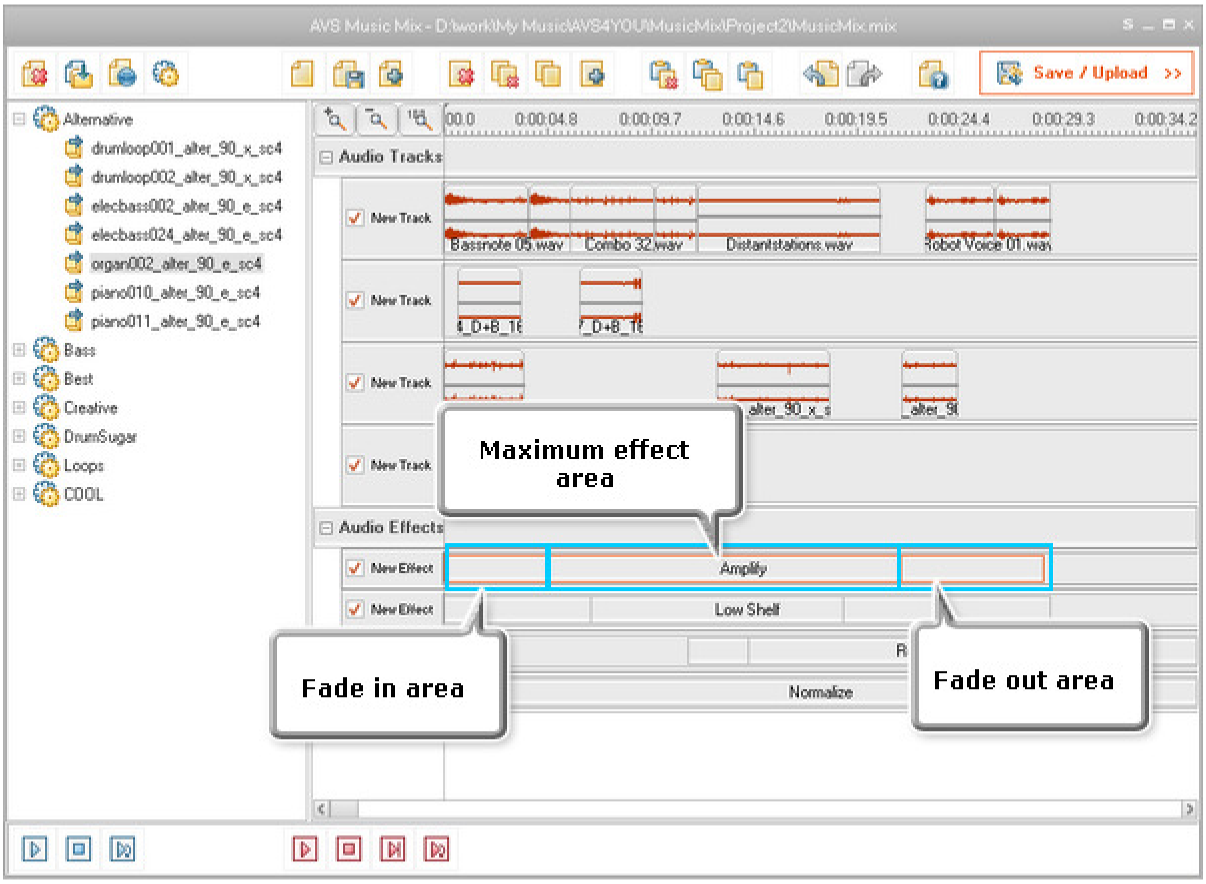
Here you can choose the desired effect from the options that you will find in the pull-down menu, in case you want to set your own **Delay Time**, **Mix Depth** and **Feedback Gain**, go to **View/Show/Hide Effect Panel**, on the effect panel open **Delay Effects/ Phaser**.

Working with Effects and Filters

AVS Music Mix gives you an opportunity to apply various effects and filters to your mix projects. Using them you can achieve the desired sound. All effects and filters are applied using the timeline tool.

Use this  button on the **Top Toolbar** to switch between the **Effects** mode and **Samples** mode.

Select an effect to apply from the **Effects and Filters** list. You can move it along the timeline, to affect a certain audio fragment, adjust the coverage, **Fade In** and **Fade Out** areas.



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.

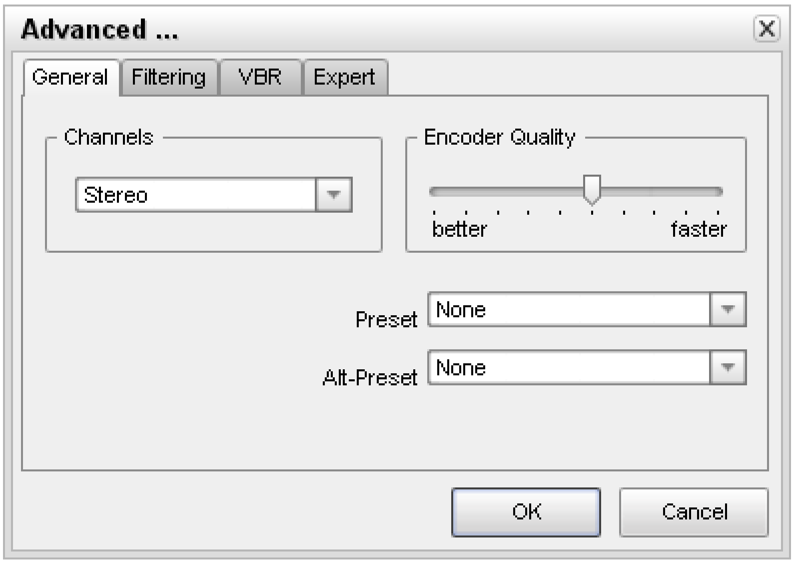
Frequency

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

MP3 Advanced window contains 4 tabs, which are:



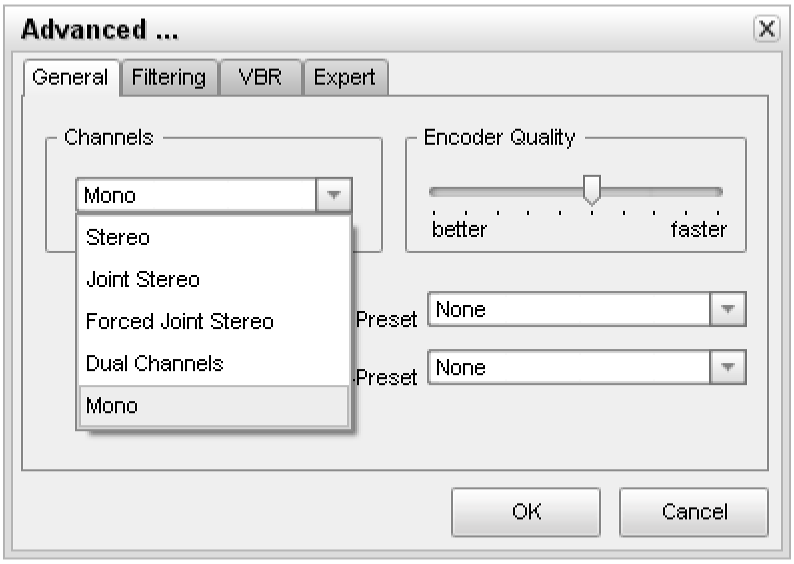
- **General**
- **Filtering**
- **VBR**
- **Expert**

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. This mode does not destroy phase information like IS stereo that may be used by other encoders. This setting can be used to encode DOLBY ProLogic 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 (normal) is recommended for the lower bitrates (<160 kbps), high quality for bitrates >160 kbps. The voice quality is more or less optimized to generate the best quality for voice



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.

You can also select **Mpeg Tag Version** out of **ID3 Ver.1** or **ID3 Ver.2**.

ID3 Ver.1

The audio format MPEG layer I, layer II and layer III (MP3) has no native way of saving information about the contents, except for some simple yes/no parameters like "private", "copyrighted" and "original home" (meaning this is the original file and not a copy). A solution to this problem was introduced with the program "Studio3" by Eric Kemp alias NamkraD in 1996. By adding a small chunk of extra data in the end of the file one could get the MP3 file to carry information about the audio and not just the audio itself.

The placement of the tag, as the data was called, was probably chosen as there were little chance that it should disturb decoders. In order to make it easy to detect a fixed size of 128 bytes was chosen. The tag has the following layout (as hinted by the scheme to the right):

Song title	30 characters
Artist	30 characters
Album	30 characters
Year	4 characters
Comment	30 characters
Genre	1 byte

If one sums the size of all these fields we see that 30+30+30+4+30+1 equals 125 bytes and not 128 bytes. The missing three bytes can be found at the very beginning of the tag, before the song title. These three bytes are always "TAG" and is the identification that this is indeed an ID3 tag. The easiest way to find a ID3v1/1.1 tag is to look for the word "TAG" 128 bytes from the end of a file.

As all artists doesn't have a 30 character name it is said that if there is some bytes left after the information is entered in the field, those bytes should be file with the binary value 0. You might also think that you cannot write that much in the genre field, being one byte big, but it is more clever than that. The byte value you enter in the genre field corresponds to a value in a predefined list. The list that Eric Kemp created had 80 entries, ranging from 0 to 79.

ID3 Ver.2

ID3v2 is a new tagging system that lets you put enriching and relevant information about your audio files within them. In more down to earth terms, ID3v2 is a chunk of data prepended to the binary audio data. Each ID3v2 tag holds one or more smaller chunks of information, called frames. These frames can contain any kind of information and data you could think of such as title, album, performer, website, lyrics, equalizer presets, pictures etc. The block scheme to the right is an example of how the layout of a typical ID3v2 tagged audio file may look like.

One of the design goals were that the ID3v2 should be very flexible and expandable. It is very easy to add new functions to the ID3v2 tag, because, just like in HTML, all parsers will ignore any information they don't recognize. Since each frame can be 16MB and the entire tag can be 256MB you'll probably never again be in the same situation as when you tried to write a useful comment in the old ID3 being limited to 30 characters.

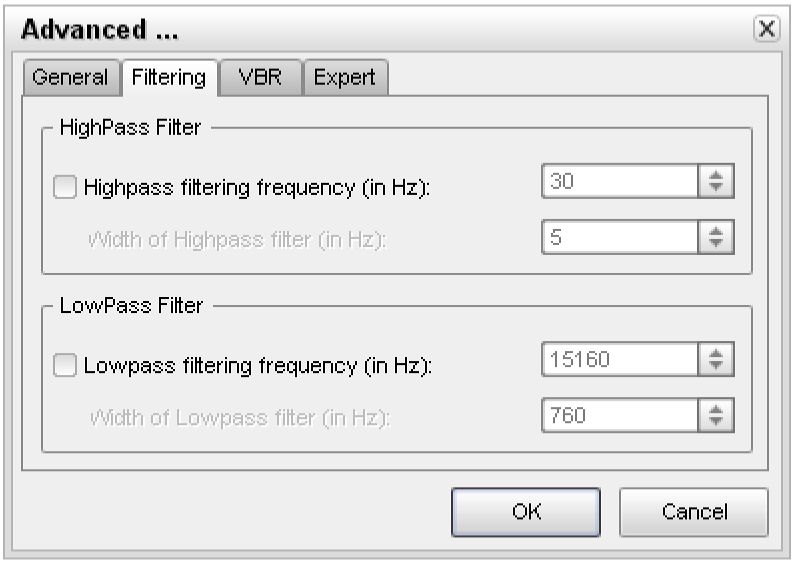
Speaking of characters, the ID3v2 supports Unicode so even if you use the Bopomofo character set you'll be able to write in your native language. You can also include in which language you're writing so that one file might contain e.g. the same lyrics but in different languages.

Even though the tag supports a lot of byte consuming capabilities like inline pictures and even the possibility to include any other file, ID3v2 still tries to use the bytes as efficient as possibly. If you convert an ID3v1 tag to an ID3v2 tag it is even likely that the new tag will be smaller. If you convert an ID3v1 tag where all fields are full (that is, all 30 characters are used in every field) to an ID3v2 tag it will be 56 bytes bigger. This is the worst case scenario for ID3v1 to ID3v2 conversion.

Since it's so easy to implement new functionality into ID3v2, one can hope that we'll see a lot of creative uses for ID3v2 in the future. E.g. there is a built-in system for rating the music and counting how often you listen to a file, just to mention some brainstorm results that are included. This feature can be used to build playlists that play your favorite songs more often than others.

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.



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.

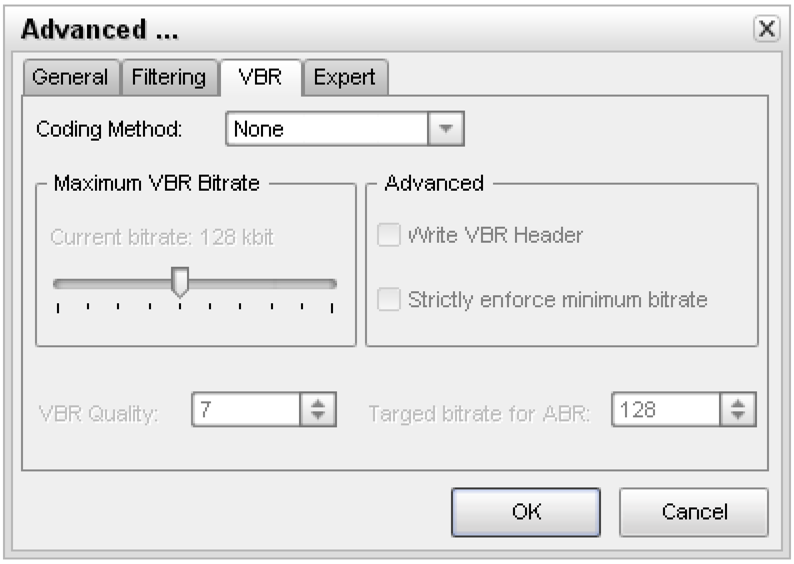
Regarding to the Nyquist Sampling Theorem the sample rate have to be at least two times higher than the highest frequency of analog audio signal. For example, the human ear can detect sound across the frequency range of 20 Hz to 20 kHz. According to the sampling theorem, one should sample sound signals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



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.

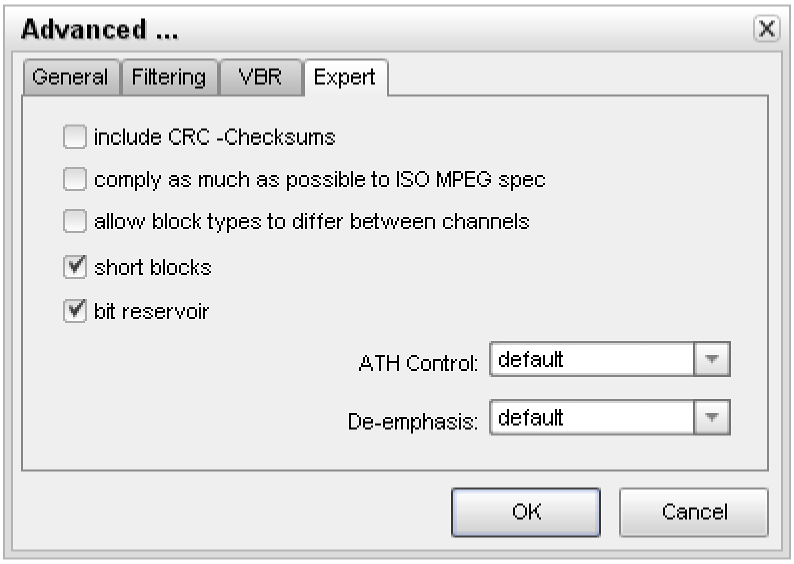
Target bitrate for ABR: The allowed range of the ABR bitrate is 4 - 310 kbit/s, you can use any integer value within that range.

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.



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**
48 KHz	64 Kbps	320 Kbps	128 Kbps	320 Kbps
44 KHz	56 Kbps	320 Kbps	112 Kbps	320 Kbps
32 KHz	40 Kbps	320 Kbps	80 Kbps	320 Kbps
24 KHz	32 Kbps	160 Kbps	64 Kbps	160 Kbps
22 KHz	32 Kbps	160 Kbps	56 Kbps	160 Kbps
16 KHz	32 Kbps	160 Kbps	40 Kbps	160 Kbps
12 KHz	24 Kbps	160 Kbps	32 Kbps	160 Kbps
11 KHz	16 Kbps	160 Kbps	32 Kbps	160 Kbps
8 KHz	8 Kbps	160 Kbps	8 Kbps	160 Kbps

Frequency	**8 KHz**		**11.025 KHz**		**16 KHz**		**22.05 KHz**		**24 KHz**		**32 KHz**		**44.1 KHz**		**48 KHz**	
Bitrate	**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	**16 KHz**		**22.05 KHz**		**24 KHz**		**32 KHz**		**44.1 KHz**		**48 KHz**	
Bitrate	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							+	+	+	+	+	+

Bitrate	**MONO and STEREO**	
	Minimum	**Maximum**
48 - 320 Kbps	22 KHz	48 KHz
40 Kbps	22 KHz	24 KHz
32 Kbps	22 KHz	48 KHz
8 - 24 Kbps	22 KHz	24 KHz

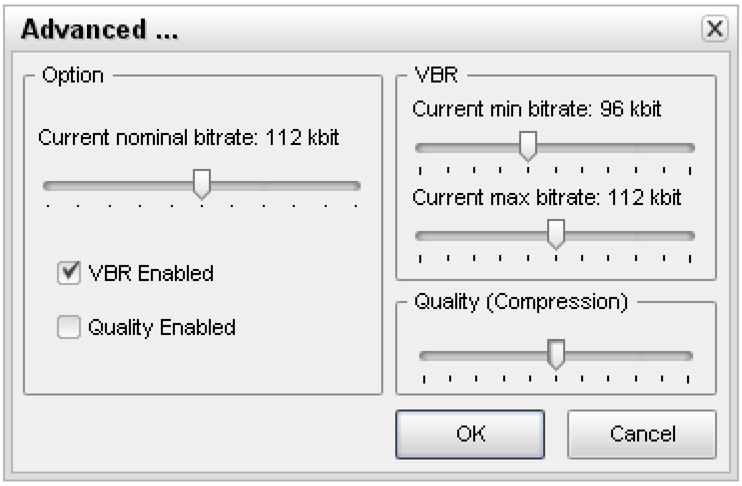
MP2 supports all the combinations of the frequency and bitrate if the frequency varies from Minimum to Maximum values for the current supported bitrate.

OGG Vorbis Options

In **OGG Vorbis Options** window you will be able to set the advanced options of your output file format.

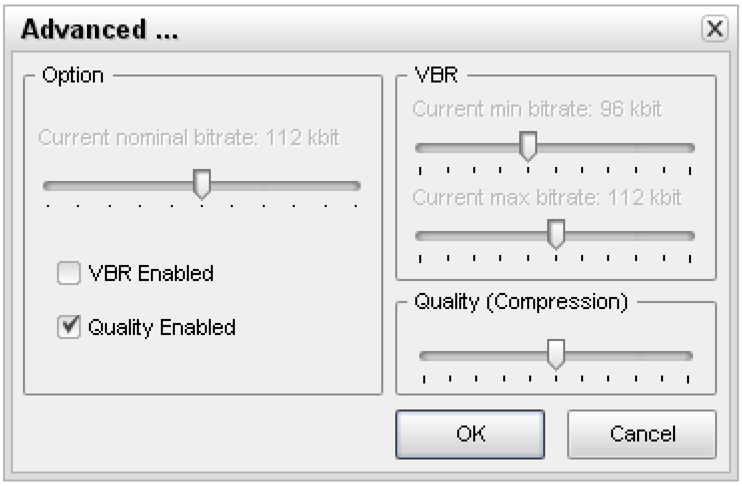
> **Note**: you can select either **VBR Enabled** or **Quality Enabled parameter**.

VBR Enabled



In case you enabled **VBR** by checking the **VBR Enabled** check-box, you will find the opportunity to set **Current nominal bitrate**, **Current minimum bitrate** and **Current maximum bitrate**.

Quality Enabled



In case the **Quality Enabled** check-box is checked, you can increase or decrease the quality of compression.

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**
48 KHz	48 Kbps	192 Kbps	48 Kbps	320 Kbps
44 KHz	48 Kbps	128 Kbps	48 Kbps	256 Kbps
32 KHz	48 Kbps	112 Kbps	48 Kbps	192 Kbps
24 KHz	32 Kbps	56 Kbps	48 Kbps	160 Kbps
22 KHz	32 Kbps	56 Kbps	48 Kbps	160 Kbps
16 KHz	48 Kbps	96 Kbps	48 Kbps	192 Kbps
12 KHz	32 Kbps	48 Kbps	48 Kbps	96 Kbps
11 KHz	48 Kbps	48 Kbps	48 Kbps	64 Kbps
8 KHz	8 Kbps	24 Kbps	16 Kbps	48 Kbps

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.