AVS4YOU Programs Help

AVS Audio Recorder

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

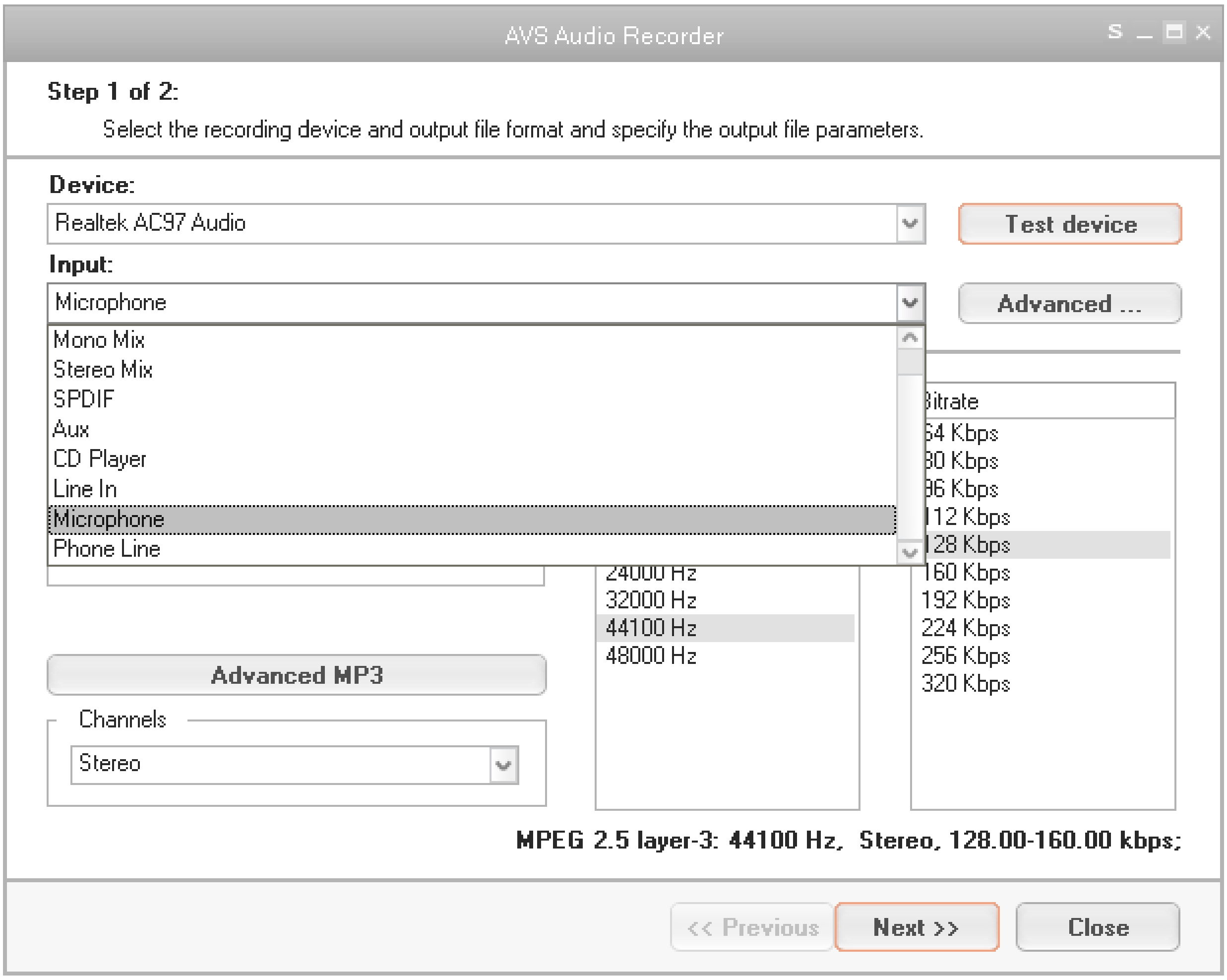
AVS Audio Recorder is a wizard-styled application, which allows you to record your own music, voice or other audio files of such popular formats as **MP3**, **MP2**, **MP+**, **AAC**, **M4A**, **AMR**, **WMA**, **WAV**, **ADPCM**. Just follow detailed step-by-step instructions of the program to create high-quality recordings.

Besides basic recording **AVS Audio Recorder** allows you to add the information to your output file such as **Title**, **Artist**, **Album** and **Comment**. For the most comfortable usage of the application you can play the recorded files directly from **AVS Audio Recorder** to make sure you've got the desired quality.

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

Step 1: Getting Started

AVS Audio Recorder wizard starts with the **Step 1 of 2** window. Please select the recording device you are going to use for recording. You can configure your input devices right from the **AVS Audio Recorder**, without need to go to the windows **Control Panel**. You can do that using the **Device** and **Input** drop-down lists in the **AVS Audio Recorder** main window:

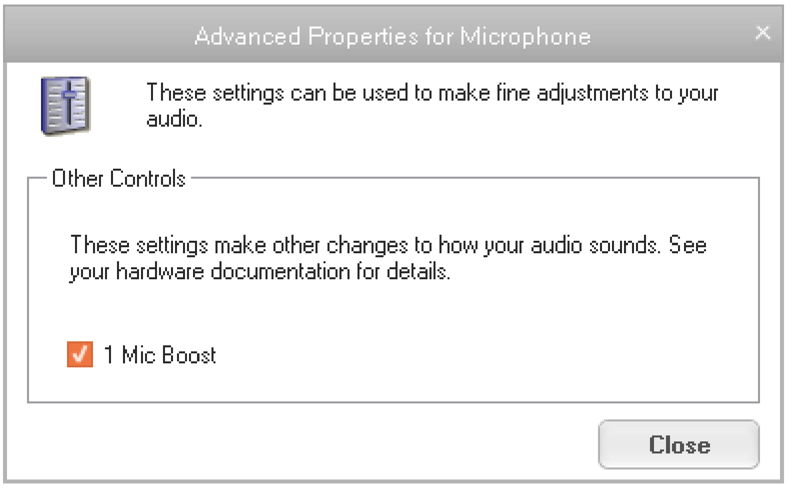


You can:

1. select the device that will be used for sound recording. 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;
2. select the input jack active on the device. The following input jacks might be available depending on your input device configuration:

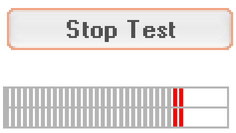
- **Mono Mix** - allows you to record the sound from a program player or a hardware tuner connected to your personal computer in mono mode;
- **Stereo Mix** - allows you to record the sound from a program player or a hardware tuner connected to your personal computer in stereo mode;
- **SPDIF** - allows you to record the sound from any external device connected to the digital input jack of your computer sound card;
- **Aux** - allows you to record the sound from any external device connected to the **Aux** (auxiliary) input of your computer sound card;
- **CD Player** - allows you to record the sound from a laser audio disc in you computer CD/DVD-ROM drive;
- **Line In** - allows you to record the sound from any external device connected to the **Line In** input of your computer sound card;
- **Microphone** - allows you to record the sound from a microphone connected to the **Microphone** input of your computer sound card;
- **Phone Line** - allows you to record the sound from an external device connected to the **Phone Line** input of your computer sound card;

3. for some devices it is also possible to change some advanced settings clicking the **Advanced...** button. The **Advanced Properties** window will pop up to let you configure the advanced device settings. You should consult your hardware documentation for more details on this settings:



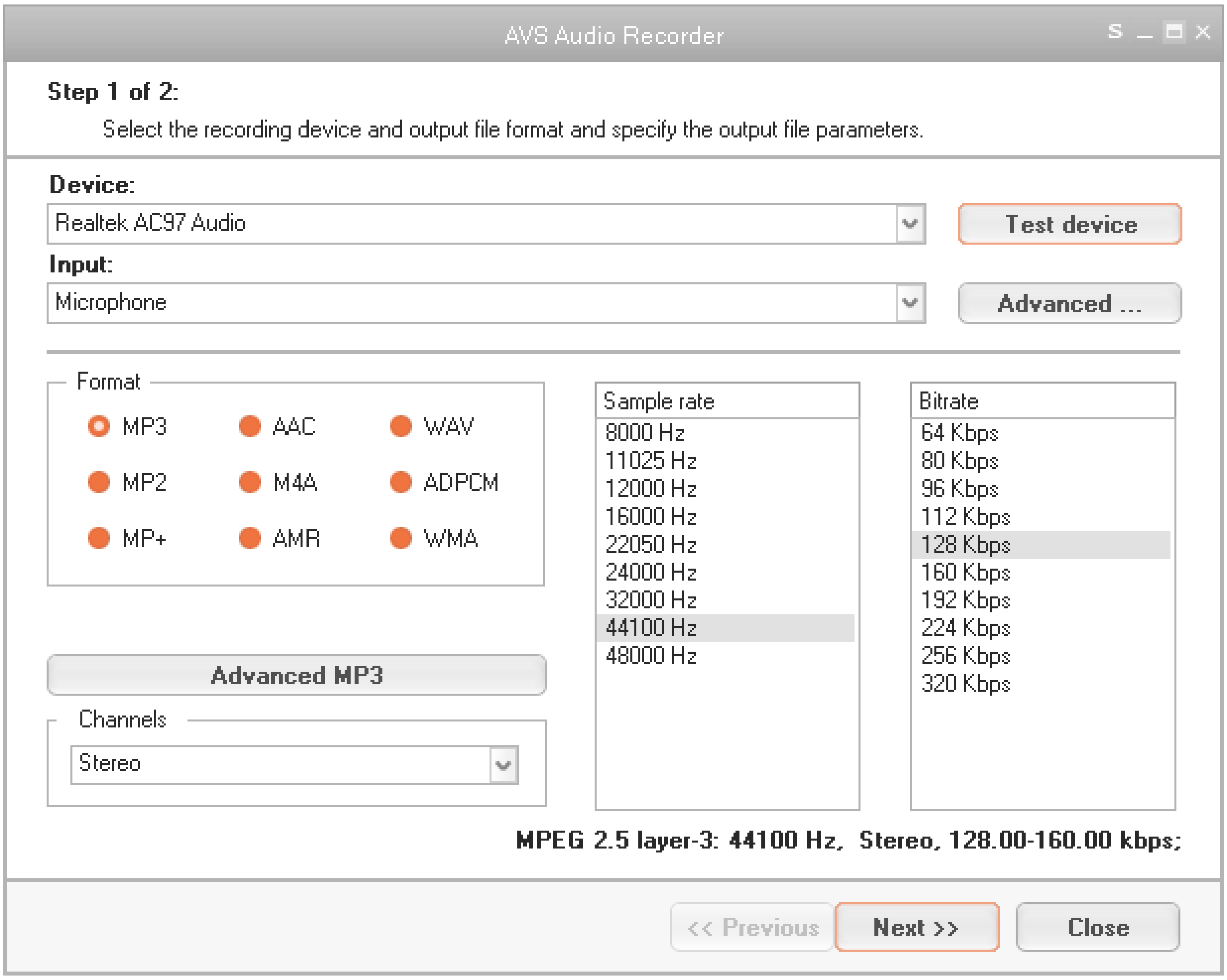
After you select all the settings for your input device you can click the **OK** button to accept the changes made and go on recording the sound from the selected and configured device.

4. test the device pressing the **Test device** button to make sure that the device is in working order:



You can visually judge the sound feedback during testing process. To stop testing click the **Stop Test** button.

After that 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: 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.

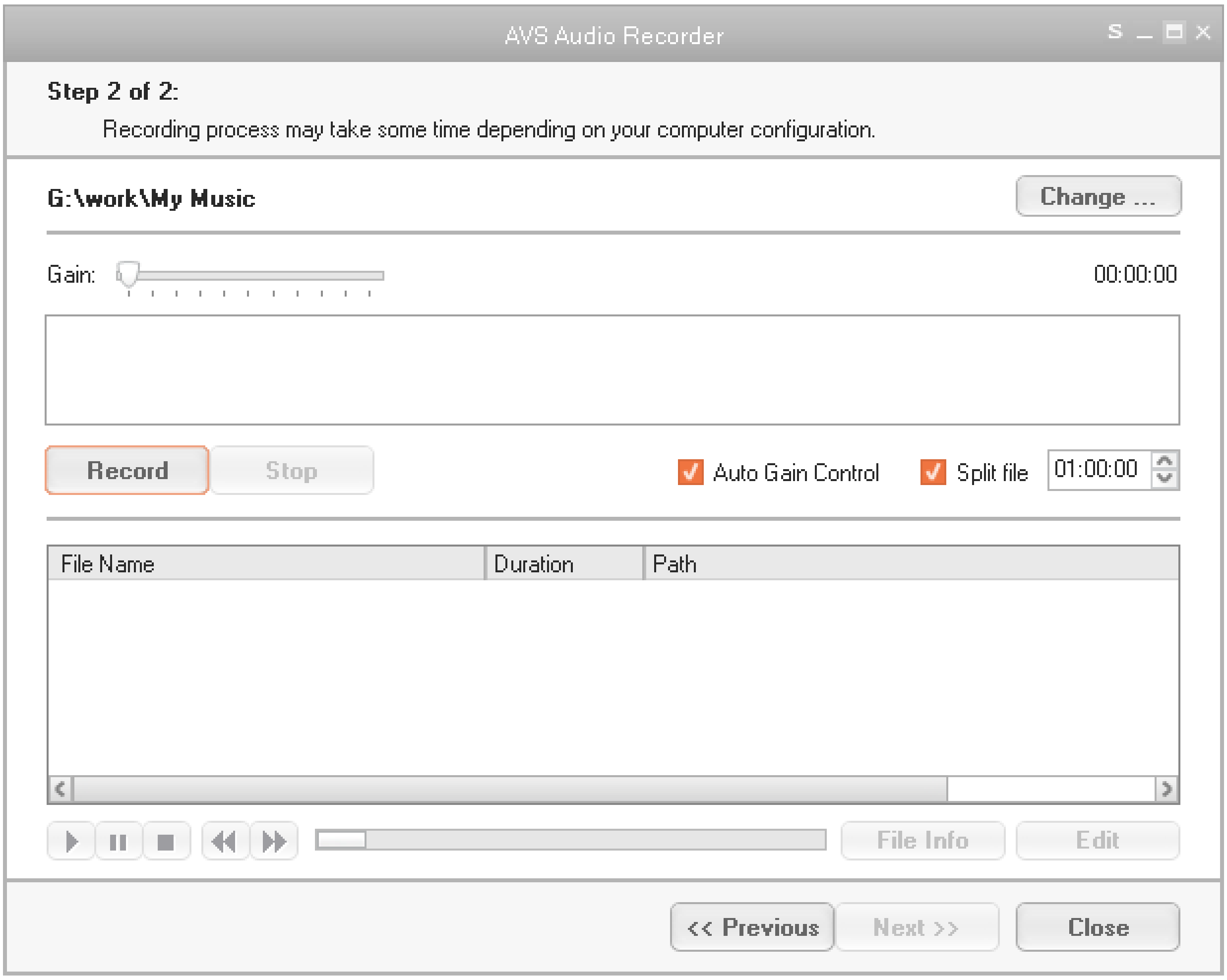
Note: it is possible to specify **Advanced MP3** parameters clicking the appropriate button. You can find the detailed information about these settings in the **Appendix** section. See also:

- **Available MP3 Parameters Combinations**
- **Available MP2 Parameters Combinations**

After the desired output format is selected please click the **Next>>** button.

Step 2: Recording and Verifying Files

AVS Audio Recorder will automatically locate all of the output files in **My Music** folder at your computer. You will be able to change output file location at **Step 2**. To do that click the **Change...** button and select desired output file location.

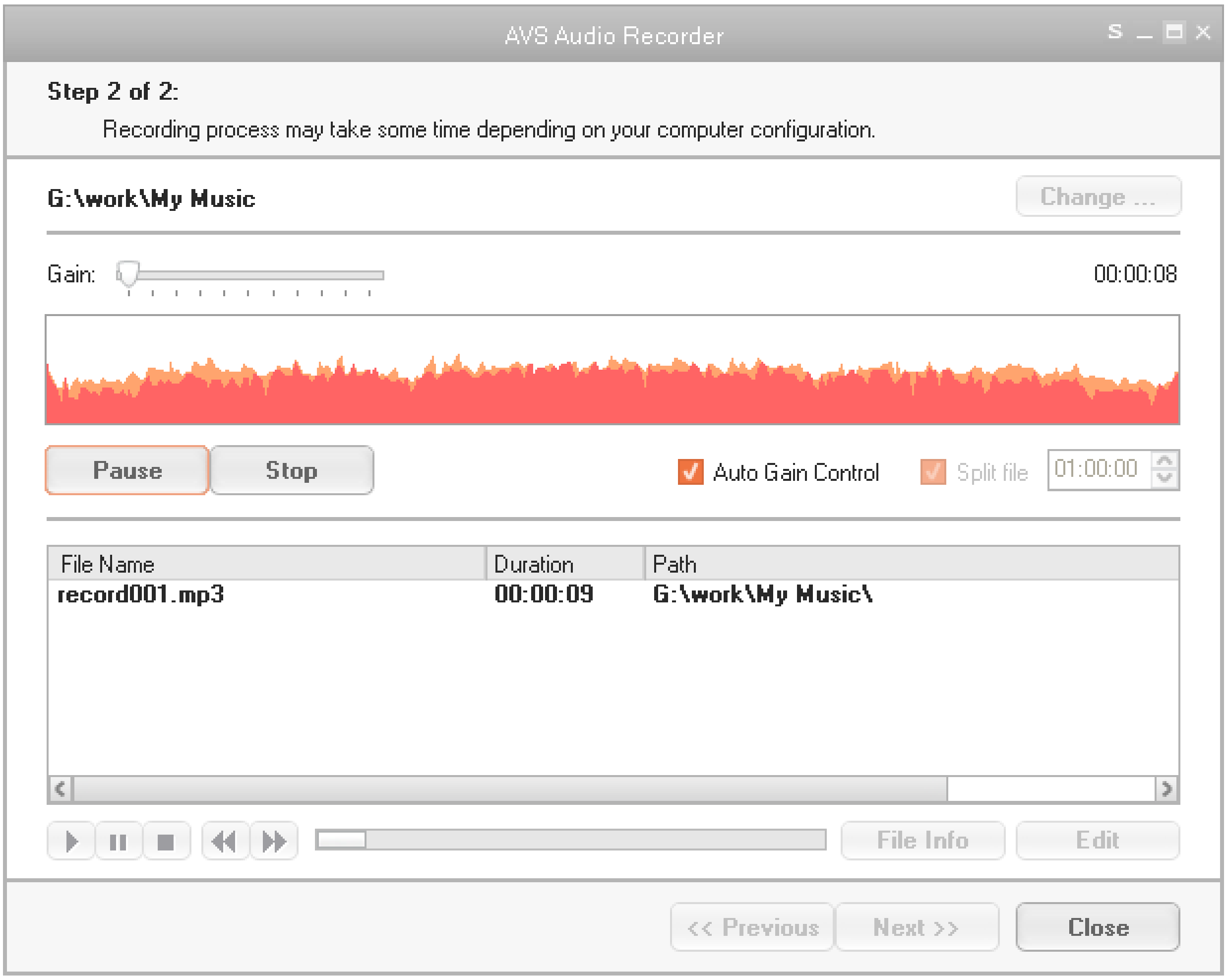


Note: you can split the recorded file into several parts. **AVS Audio Recorder** will suggest that you split your recordings by time. For example, if you want the recorded file to be split into several parts and you want each of these parts to be 60 minutes long, you should set the parameters as shown in the picture below:

Warning! If you don't want to split the output file, make sure that **Split file** check-box is unchecked!

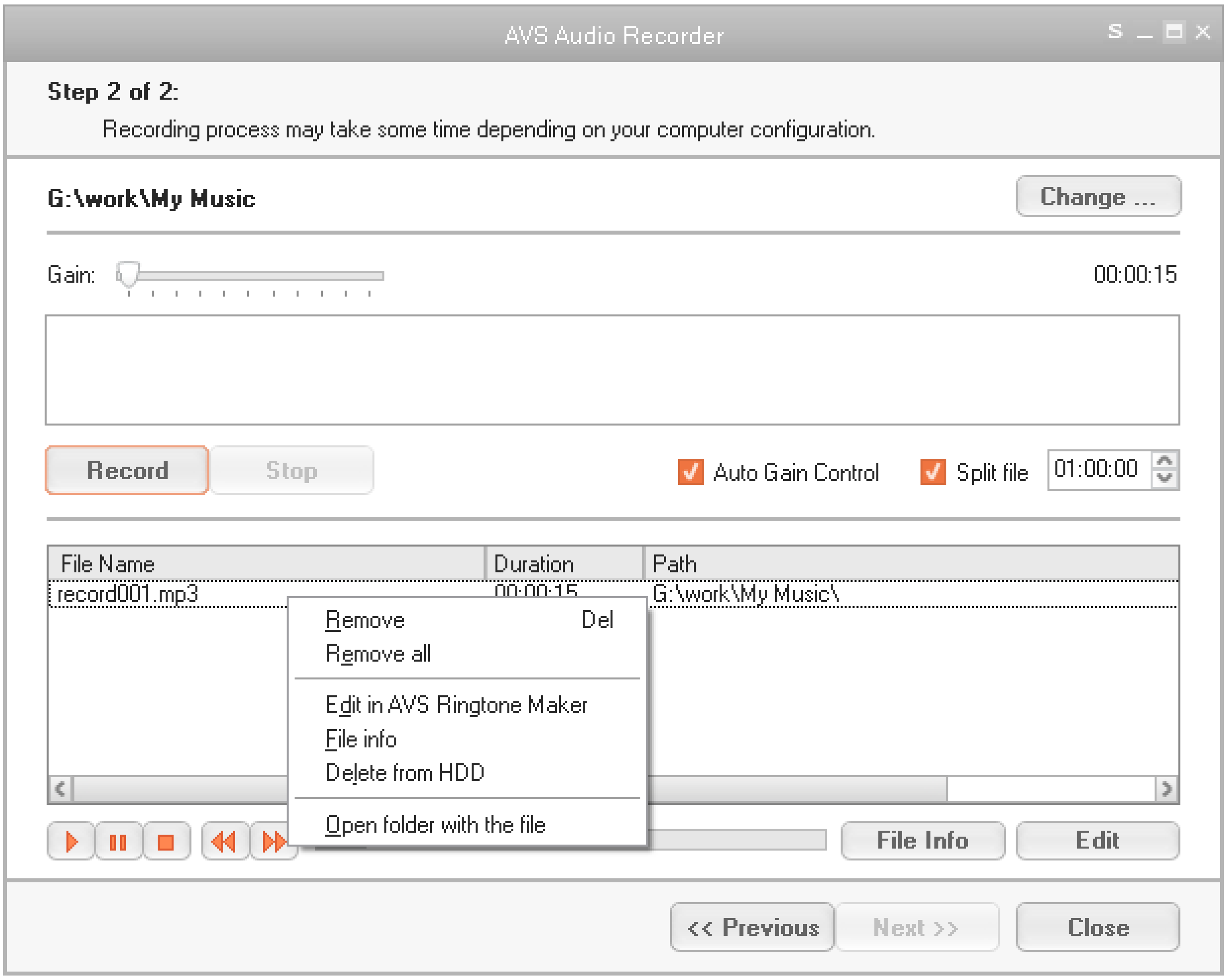
You can also select between auto and manual gain control. If you prefer to change the volume of the recording manually, please uncheck the **Auto Gain Control** check-box and use the **Gain** slider to select the gain level. Otherwise you can let the program automatically control the sound level during recording.

After that you can start recoding with the recording parameters you've set previously. To do that simply press the **Record** button and the recording will start:



To stop or pause recording use the **Stop** or **Pause** buttons accordingly. You can visually judge the sound feedback during testing process with the help of graphic representation area.

When the recording is done you can playback the recorded file or files to make sure you've got the desired quality.



Please use the following buttons to verify your file:

 To playback the selected file

 To pause the selected file

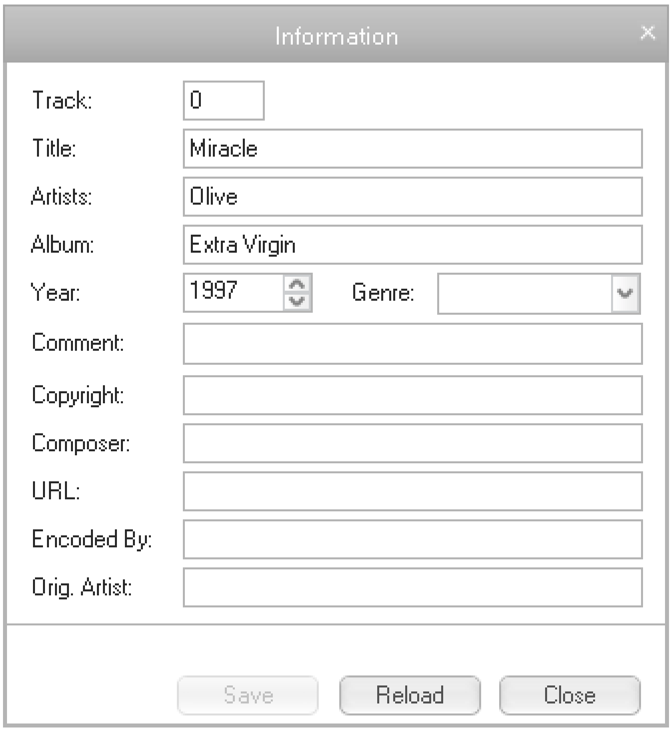
 To stop the selected file

 To go to the previous track

 To go to the next track

You can also edit the resulting files with the help of the **AVS Ringtone Maker** program. To do that click the **Edit** button to launch the application. See the appropriate section on how to work with **AVS Ringtone Maker**.

AVS Audio Recorder also allows you to modify extra text information in existing files. To do that click the **File Info** button and the **Information** window will appear.



Here you can change all of the available details such as **Track number**, **Title**, **Artist**, **Album**, you can add any **Comment**, specify **Year** and **Genre**.

After you finished, please click the **Step 1** button to return to the **first step** or **Close** to finish the work with the **AVS Audio Recorder** program.

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.

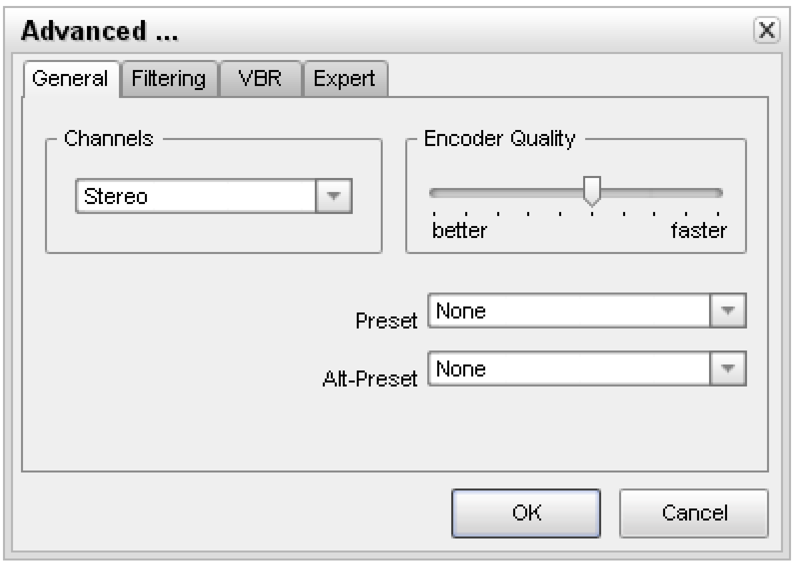
Frequency

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:



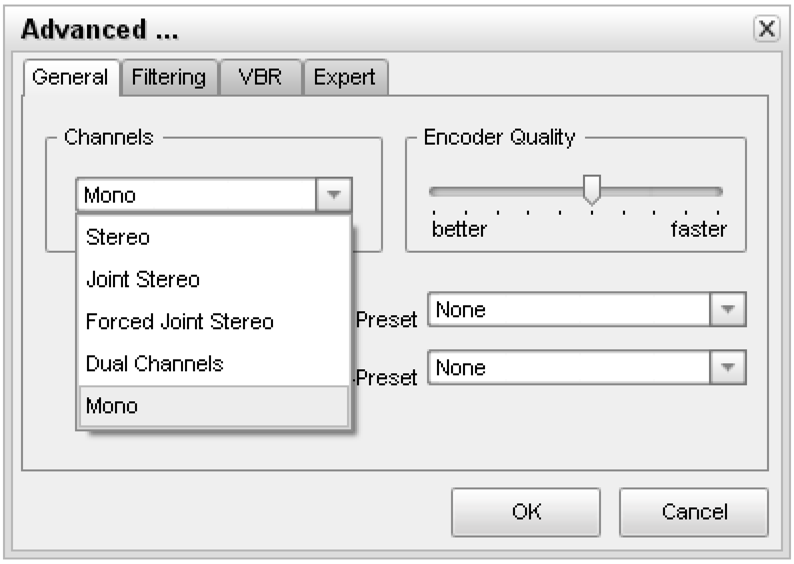
- **General**
- **Filtering**
- **VBR**
- **Expert**

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.

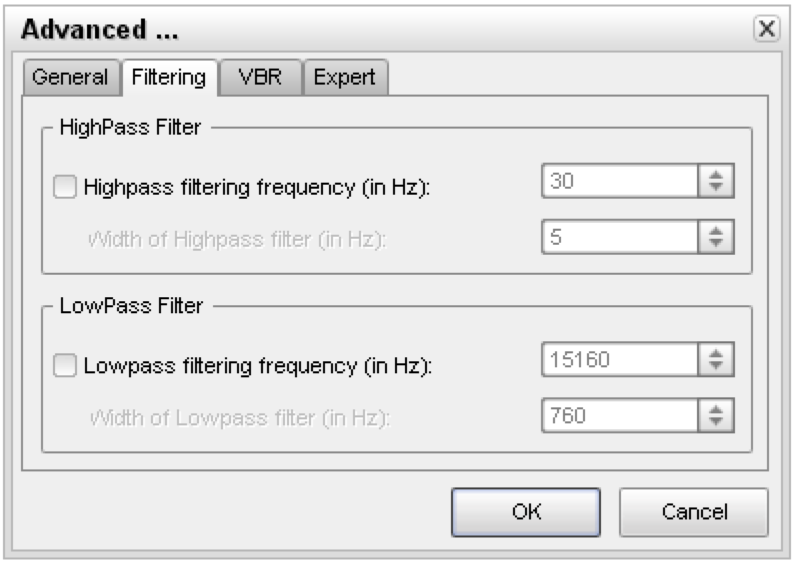
Speaking of characters, the ID3v2 supports Unicode so even if you use the Bopomofo character set you'll be able to write in your native language. You can also include in which language you're writing so that one file might contain e.g. the same lyrics but in different languages.

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.

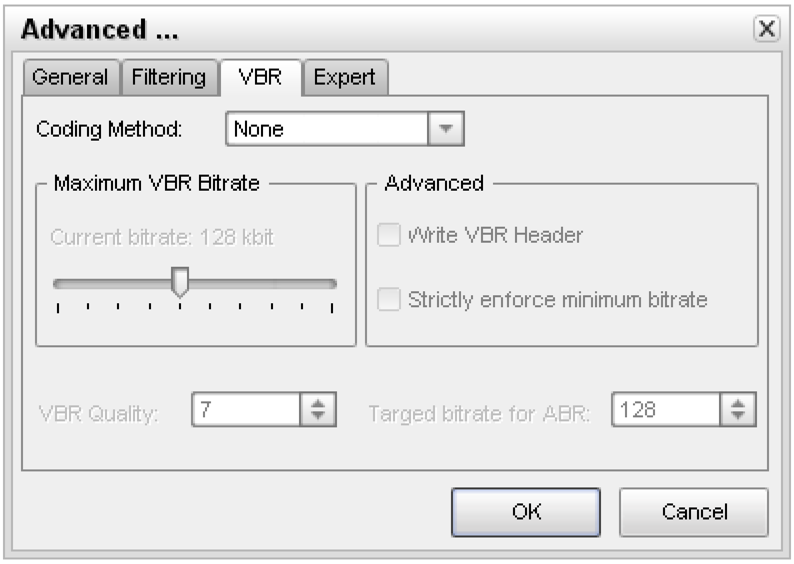
Regarding to the Nyquist Sampling Theorem the sample rate have to be at least two times higher than the highest frequency of analog audio signal. For example, the human ear can detect sound across the frequency range of 20 Hz to 20 kHz. According to the sampling theorem, one should sample sound signals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

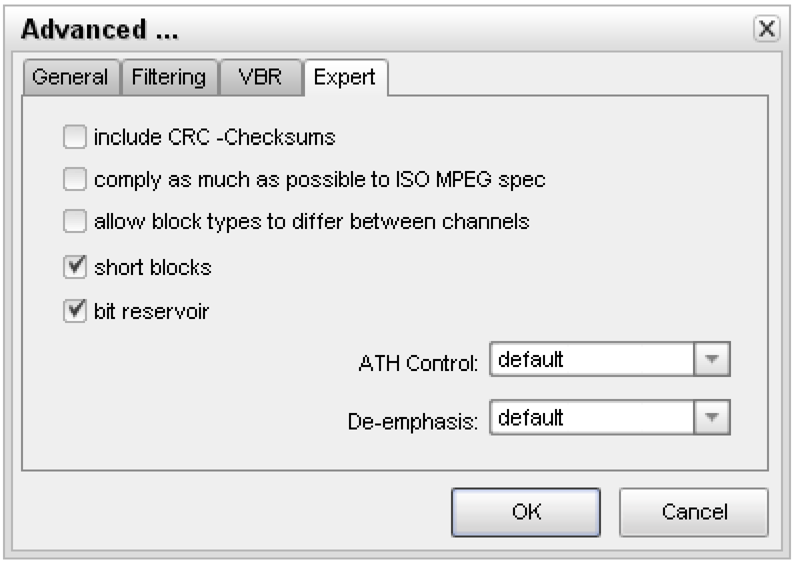
Target bitrate for ABR: The allowed range of the ABR bitrate is 4 - 310 kbit/s, you can use any integer value within that range.

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

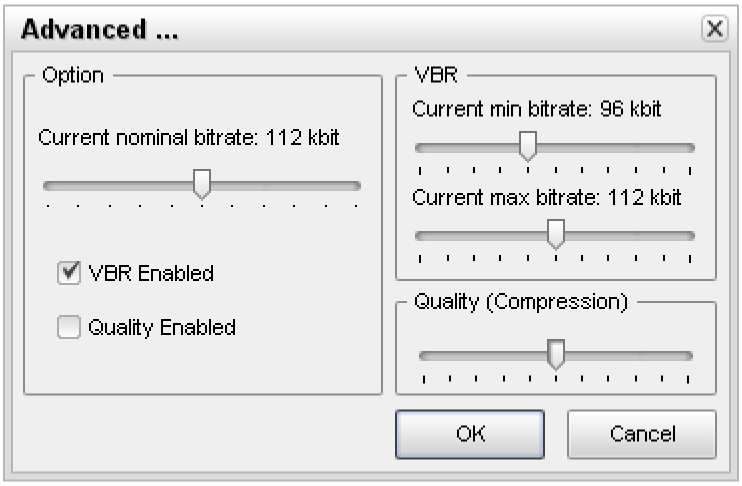
MP2 supports all the combinations of the frequency and bitrate if the frequency varies from Minimum to Maximum values for the current supported bitrate.

OGG Vorbis Options

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

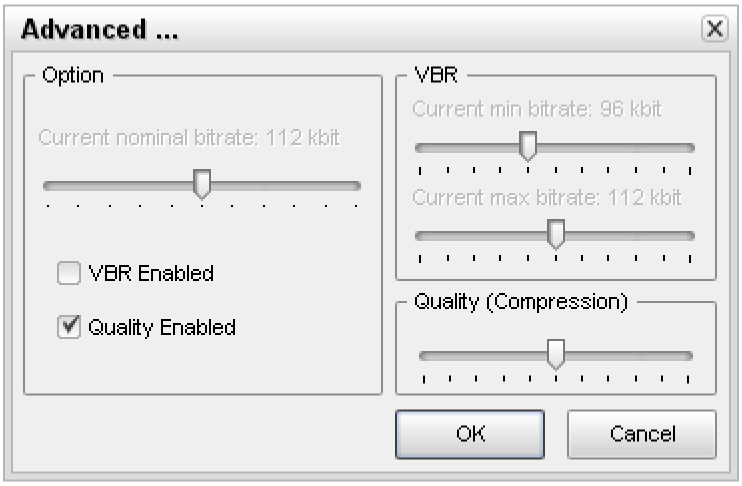
Note: you can select either **VBR Enabled** or **Quality Enabled parameter**.

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.

