

NCH Software

Switch Sound File Converter

This user guide has been created for use with
Switch Sound File Converter Version 11.xx

Technical Support

If you have difficulties using Switch Sound File Converter please read the applicable topic before requesting support. If your problem is not covered in this user guide please view the up-to-date Switch Sound File Converter Online Technical Support at

www.nch.com.au/switch/support.html.

If that does not solve your problem, you can contact us using the technical support contacts listed on that page.

Software Suggestions

If you have any suggestions for improvements to Switch Sound File Converter, or suggestions for other related software that you might need, please post it on our Suggestions page at

www.nch.com.au/suggestions/index.html.

Many of our software projects have been undertaken after suggestions from users like you. You get a free upgrade if we follow your suggestion.

Switch Sound File Converter

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Switch Overview

Switch is a program for converting audio files from one format to another. All major audio file formats can be loaded and converted, as well as some less well known formats.

Features

- Loads a variety of different audio file formats including Wave, MP3, Ogg, and many others.
- Converts to a variety of output formats including Wave, MP3, AIFF, Vox, Raw, and many others.
- Also supports converting to Ogg, and Flac formats (Plus version only).
- Ability to play tracks prior to conversion.
- Allows drag and drop of both files and folders.
- Perform batch conversions.
- Normalize while converting (evens out the volume level on convertings from different sources).
- Simple and intuitive interface.

System Requirements

- Mac OS X 10.3 or Later

Switch is just one component of the NCH Software suite of audio software for business. If you have not done so already, please visit <http://www.nch.com.au/software> to download many other audio related programs.

How to Use Switch

This is an overview of how to use Switch to convert your files from one format to another.

Adding and Removing Files

To convert your audio files into the desired format, begin by adding files into the file list window. Either click the "Add" button, or select File -> Add from the menu. In the browse window that opens, you can look for files of types mentioned on this page:

<http://www.nch.com.au/switch/kb/1405.html>.

Once your files have been added, if you wish to remove any of them, select the files and either click the "Remove" button, or select File -> Remove from the menu. To select multiple files at once, press the Command key and left-click on the files you want to select. To highlight all the files, either select File -> Select All from the menu, or press Command + A.

Playing Files

To play a file, double click it or left-click on the desired file and either click the "Play" button, or select File -> Play from the menu.

Selecting an Output Folder

Near the bottom of the main window, there is a field labeled "Save to Folder", where you can enter the file directory into which you wish to store the converted files. You can either type in the desired folder, or you can click the browse button on the right of the field. Doing so will bring up a window where you can browse for the destination directory on your hard drive. The last several directories you have chosen in the past are stored in a drop-down list, and can be accessed by clicking the arrow on the right side of the text field.

Selecting an Output Format and Configuring Encoder Options

The field labeled "Output Format" is where you can select the desired output file format for your files. Switch currently supports conversion to the file formats mentioned on this page:

<http://www.nch.com.au/switch/kb/1405.html>

Once you have selected a format to convert to, you can click the "Encoder Options" button to configure the settings for the format (see [Encoder Options](#) for more information on configuring options for any file format).

Converting Files

Once you have set the output folder and format, you can then select the files from the list you want to convert, and either click the "Convert" button, or select Convert -> Convert Files from the menu. If you hit the convert button without any files selected, all files in the list will be converted by default.

View Internal File Information

For those who are interested in the attributes or internal information of wave and mp3 files, Switch now has a feature that lets you access that information. For any wave, mp3 or dct file loaded into Switch, right-click on the file and select 'View Internal File Information' from the right-click menu.

View Extended Information

If you attempt to convert a file and something goes wrong with the conversion process, a short message will be displayed in the 'Extended Information' column in the Switch main window.

This message gives a brief summary of the specifics of the problem. If you wish to see a more detailed version of that message, right-click the file you attempted to convert and select 'View Extended Information'. You will see a pop-up window appearing that contains the more detailed message.

What Formats Does Switch Support?

Please refer to the following link for a list of formats currently supported by Switch:

[Click here to find out the formats supported by Switch](#)

There are a number of file formats that are only supported in the Switch Plus paid software version. These formats are also available, however, during the 14-day trial period of Switch Plus and during the first 14 days of Switch Free. These formats are as follows:

Convert From:

- aac
- aif
- aifc
- aiff
- amr
- ape
- au
- avi
- caf
- dct
- flac
- gsm
- m4a
- mid
- mov
- mp2
- mp3
- mp4
- mpc
- mpeg
- mpg
- mpga
- ogg
- opus
- raw
- voc
- vox
- wav
- wma

Convert To:

- aac
- adts
- aif
- aiff
- aifc
- amr
- au
- caf
- dct
- flac
- gsm

- m3u
- m4a
- m4r
- mov
- mp3
- mpc
- ogg
- opus
- pls
- raw
- rss
- spx
- vox
- wav
- wma
- wpl

If you have files to convert of the above format types, we strongly encourage you to make the most of the trial period to ensure Switch does what you want it to do. Switch does *not* have extra supported formats that are only accessible after purchasing a license, all supported formats are available during the trial. What you see during the trial is exactly what you get after a license purchase.

Editing Audio in WavePad

You may edit the selected file in the conversion list with an audio editor before conversion. To do so, click the Edit menu and select **Edit Audio in WavePad**. WavePad will open and your audio file will be loaded so you can perform basic editing, add sound effects, remove background noise, adjust frequencies, and more. Be sure to save any changes if you want the edited version to be converted in Switch.

To learn more about WavePad Audio Editing software, visit www.nch.com.au/wavepad

Editing Audio Tags

Switch supports editing tags to files of certain formats (e.g. ID3 tags for MP3 files, LIST INFO chunk of WAV files, etc). It also supports preserving tag information when converting to any supported formats.

Edit audio tags of a single file

To open the tag editor, select an audio file, click the Edit menu and select **Edit Audio Tags**.

Edit audio tags of multiple files

To open the tag editor, select two or more audio files, click the Edit menu and select **Edit Audio Tags**.

[Multiple Values] may be displayed if a tag has different values among files.

Edit audio tags of converted files

This function edits the tags of the output file. Function is enabled only after a successful conversion of a file.

To open the tag editor, select a file from the list, click the Edit menu and select **Edit Audio Tags of Converted Files**.

Saving to file

Select the course of action to take when saving tags to files. The options are:

- Apply changes to converted files only - Tags will be applied to succeeding conversions of the source file, but will not be saved to the source file.
- Apply changes to source files and converted files - Tags will be applied to the source file, and to succeeding conversions of the source file.

Tags will be applied to audio formats if the following conditions are met:

- File exists
- File is not Read-only
- File format supports metadata such as .aif, .aiff, .aifc, .flac, .m4a, .m4b, .m4r, .mp1, .mp2, .mp3, .ogg, .wav, and .wma.
- File is not part of an archive (Switch can open zip and rar files but only audio files will be listed or converted)

Restore Defaults

Click this button to restore previous settings by the user. Once the user clicks the OK button, the default values will be updated according to the latest values. Default values are specific per file.

Appearance

Tags which are not supported by the source file and the destination file format will have their values displayed as gray, or disabled for non-text controls (e.g. Artwork, Start/End date). For more information about tags, click [here](#).

Program Settings

Open Settings by clicking on the "Settings" button in the toolbar, or selecting "Switch" -> "Settings" from the menu. In Settings you can configure the following:

File

This dialog contains all options related to processing files. It is divided into a "Overwrite" section and an "Default Metadata" section.

Overwrite

Select the course of action to take when the destination file exists while converting files. The options are:

- Prompt for What to Do - always prompts whether you want to overwrite the file or not.
- Overwrite the file - overwrites the specific file being converted.
- Don't Convert the Original File - does not overwrite the specific file being converted
- Abort current and remaining conversions - cancels current file conversion and aborts all subsequent conversions as well.
- Append number to filename - appends a number to the filename of the converted file using a format that you can select. The format must include the strings "%filename%" and "%number%", but they may be in any order you like. You can add extra characters into the string and those characters will also appear in the converted file's filename.

Default Metadata

All converted files will include the default metadata entered in these fields:

- Comment - Description of the audio file or song.
- Copyright - Copyright information.
- Producer - Producer of the audio file.
- Encoder - Encoder of the audio file.
- Encoded By - Software used for encoding.

For more information about tags, click [here](#).

If title tag is empty, use file name

Not all source files have tags, which results in empty title tags after conversion. Check this box to use the file name as the title tag.

Conversions

This dialog contains all options related to converting files. It is divided into a "Formats" section and an "Errors" section.

Formats

Specify what to do if the file being converted is of the same format as the output format. The options are:

- Convert Anyway - converts the file as normal.
 - Check the sub-option if you don't want to recompress a file that has a source with a better compression rate
- Copy File Directly to Destination Folder - copies the file to the destination folder in its original non-converted form.
- Do Not Convert - do not convert the file at all.

Notes: The options in "Formats" will **always** be over-riden by the options specified in the "Overwrite" tab. For example, if you specify the "Convert Anyway" option in "Formats", but in "Overwrite" you specify "Don't Convert the Original File", then if the file exists the conversion will not go ahead.

Delete source file after conversion - if the conversion of the file was successful, then the original file will be deleted. If the conversion failed, the file will NOT be deleted. A successful conversion also includes copying of the file to the destination folder, as seen in the above option "Copy File Directly to Destination Folder".

Remember settings for reading files(only applicable to formats .raw and .vox)- If you are converting from .raw/.vox file. You can choose to use default Codec Settings for all other files of this type or set codec settings individually.

Errors

Specify what to do if there is an error produced with the file being converted. The options are:

- Prompt for What to Do - prompts whether you want to continue converting other files or abort converting altogether.
- Ignore Error and Continue Converting Other Files - continues converting other files even if any particular file produces errors.
- Abort File Conversions - cancels current file conversion and aborts all subsequent conversions as well.

Notes: If errors are produced on any file, they will be reported in the "Other Information" column for that file in the file list. To view extra information about the error, right click on the specific file and select "View Other Extended Information" from the menu.

Audio Processing

To 'normalize' is to adjust the volume so that the loudest peak is equal to (or a percentage of) the maximum signal that can be used in digital audio. Usually you normalize files to 100% as the last stage in production to make it the loudest possible without distortion. Another reason to normalize is to have multiple tracks sound equally loud, or to have equal average loudness.

The 'Peak' normalization method finds the sample of the greatest magnitude within the file. Normalization is then done with this value as the peak. With the Normalize Peak Level set to 100% (0dB), the whole file will be amplified so that the peak reaches 0dB.

The 'Average Loudness (RMS)' normalization method normalizes according to the file's average loudness, or volume. Multiple files normalized to the same peak level using this method will have equal average loudness. The 'Normalize Peak Level' for this method should be set much lower than for the Peak method, because the average loudness will always be lower than the peak sample.

The 'Peak Loudness (RMS)' normalization method attempts to normalize according to how loud the loudest part of the file will sound. This is the best method to use to make multiple tracks sound equally loud. As with Average Loudness, the 'Normalize Peak Level' for this method should be set lower than for the Peak method, because the peak loudness is lower than the peak sample. The actual algorithm used takes the RMS of each 50ms window in the file, ranks the windows from loudest to quietest, and then takes the 95th percentile of these as the 'peak'. Note that no adjustment is made for humans' differing perception of different frequencies.

When there are several audio tracks in a video file, select default audio track to work with

Some video files may contain several audio tracks. Check this box to select the default audio track to work with. Uncheck to select a audio track manually.

Naming (Switch Plus feature only)

These options define how you wish to structure the output format of converted files. For .cda input files, the option 'Access Internet database to retrieve CD Album Details' will use your Internet connection to access the Album and Track name details from the MusicBrainz online database.

1. Artist - TrackName.ext
2. Album - Artist - TrackName.ext
3. Custom Setup (see below)

The Custom Setup is an advanced setting that allows you to decide how the output files will be stored on your computer. This involves the writing of a string of characters that denote information related to the file. An example of how to write a string is as follows:

[%artist% %album% %genre%]\{%tracknumber% %trackname% %year%}

This will convert your files to a folder named "Artist - Album - Genre", with filename "Tracknumber. Trackname - Year.fileextension".

All the arguments that can be used to form the custom string are as follows:

%artist% - the artist name

%album% - the album name

%extratrackinfo% - extra track information (may not exist in all cases)

%genre% - the genre of the album (e.g. rock, jazz)

%tracknumber% - the track number

%trackname% - the name of the extracted track

%year% - the year the album was released

[] - anything inside these brackets will be treated as a folder

{} - anything inside these brackets will be treated as a filename

These details are viewable in Edit Audio Tags functionality (CTRL+T) of Switch.

You can have multiple levels of folders set up, but you are only allowed to have one filename specified in the string.

Other

This dialog contains general options that can't be grouped into a specific category.

Sound Playback - Here you can select which audio device to use. Useful for computers with multiple sound cards installed.

Confirmation - Select these options to be notified by Switch when you remove files from the list, or when your hard drive space drops below the specified amount.

Duplicate Files - To prevent slowing of Switch when working with long lists, select this option to prevent duplicate files from being added to the list.

Finishing Notification - When converting a long list of files, you may find it useful to use the finishing notification options to alert you to completion.

Using the Switch Command-line Tool

The Switch command-line tool allows you to use Switch to convert files from the Windows command line. This can be used to let other software convert files with Switch or to automate batch conversion.

The Switch executable is usually located at /Applications/Switch.app/Contents/MacOS/Switch

Usage: /Applications/Switch.app/Contents/MacOS/Switch *[options] [files to add]*

"Options" is where you can issue a series of commands to Switch. The options are:

-convert - convert all files loaded into Switch.

-clear - remove all files loaded into Switch.

-addfolder [FOLDER_NAME] - add a folder of files into Switch.

-format [FORMAT_NAME] - specify the output file format for which to convert the loaded files.

The format names are the file extensions for each format (e.g. .wav, .mp3).

-outfolder [FOLDER_NAME] - specify the output folder to store the converted files. Do NOT end the folder path with a backslash '\' character.

-settempfolder [FOLDER_NAME] - specify the temporary folder that Switch will store files in when playing or converting files. If the specified folder is not valid then no action will take place.

-overwrite [OVERWRITE_FLAG] - specifies what to do when the converted file already exists. The flags are "PROMPT" (Prompt for What to Do), "ALWAYS" (Overwrite the File), "NEVER" (Don't Convert the Original File), "CANCEL" (Abort All Conversions), "APPENDNUMBER [format]" (Append number to filename - [format] is optional, if blank it will use the format already specified in Switch).

-settings [FORMAT_NAME] [SETTINGS_OPTIONS] - specifies the settings configuration to use for the specified format. For more detail on what parameters to use, see further down the page.

-insettings [FORMAT_NAME] [SETTINGS_OPTIONS] - specifies the settings configuration to use for formats requiring decoding prior to conversion (e.g. .vox, .raw) For more detail on what parameters to use, see further down the page.

-hide - hide the Switch window.

-show - show the Switch window.

-exit - exit Switch.

"Files to Add" is where you type in the files you want to load into Switch. All typed filenames must use the full filepath of the name, and circumfix with inverted commas.

Examples:

Switch -format MP3

./Switch -outfolder ~/Music;

./Switch -hide

./Switch "~/Music/sound.mp3"

./Switch -convert "~/Music/music.mp3" -format .wav -overwrite ALWAYS

./Switch -convert "~/Music/music.ogg" -format .mp3 -overwrite APPENDNUMBER

"%filename%-%number%"

Unless you are in the Switch executable's folder you will need to replace ./Switch from above with the full path /Applications/Switch.app/Contents/MacOS/Switch like this:

/Applications/Switch.app/Contents/MacOS/Switch -convert "~/Music/music.mp3" -format WAV

Settings Options:

The following details what parameters to enter when using the "-settings" and "-insettings" options. Note the order in which the parameters are written is the order in which they must be typed. Additionally, all parameters must be entered otherwise the command will not take effect.

WAV [CODEC] [SAMPLE_RATE] [CHANNELS]

- CODEC - either "PCM8", "PCM16", "ALAW" or "MULAW" (note ADPCM is not supported at this time)
- SAMPLE_RATE - entered in Hz e.g. 8000
- CHANNELS - either 1 or 2

Note the possible combination of values that can be entered depend largely on the codec, so you would have to open Switch and determine beforehand what values you can use for the codec in question.

Example: -settings .wav PCM16 44100 2

MP3 ENCODER_TYPE CBR[BITRATE] [HIGH_QUALITY] VBR[MIN_BITRATE] [MAX_BITRATE] [QUALITY] [CHANNEL_MODE] [ERROR_PROTECT]

ENCODER_TYPE - either "CBR" or "VBR"

CBR OPTIONS:

- BITRATE - entered in kbps e.g. 128
- HIGH_QUALITY - either "TRUE" or "FALSE"

VBR OPTIONS:

- MIN_BITRATE - entered in kbps e.g. 128
- MAX_BITRATE - entered in kbps e.g. 192
- QUALITY - a value from 0 to 9
- CHANNEL_MODE - either "STEREO", "JOINT", "FORCE" or "MONO"
- ERROR_PROTECT - either "TRUE" or "FALSE"

Examples:

- -settings .mp3 CBR 128 TRUE STEREO TRUE
- -settings .mp3 VBR 128 192 4 MONO FALSE

OGG ENCODER_TYPE Quality[QUALITY] VBR[MIN_BITRATE] [MAX_BITRATE] [CHANNELS] [STRIP_COMMENTS]

ENCODER_TYPE - either "Quality" or "VBR"

Quality OPTIONS:

- QUALITY - a value between 0 and 10

VBR OPTIONS:

- MIN_BITRATE - entered in kbps e.g. 128
- MAX_BITRATE - entered in kbps e.g. 192
- CHANNELS - either 1 or 2
- STRIP_COMMENTS - either "TRUE" or "FALSE"

Examples:

- -settings .ogg Quality 5 2 TRUE
- -settings .ogg VBR 96 192 1 FALSE

MPC [QUALITY_PRESET]

- QUALITY_PRESET - one of the following: Thumb, Radio, Standard, Extreme, Insane

Example: -settings .mpc Standard

AIFF [SAMPLE_RATE] [CHANNELS]

- SAMPLE_RATE - one of the following: 6000, 8000, 11025, 16000, 22050, 32000, 44100, 48000, 64000, 88200, 96000, 196000
- CHANNELS - either 1(Mono) or 2(Stereo)

Example: -settings .aiff 44100 1

AAC/M4A ENCODER_TYPE ABR[BITRATE] VBR[QUALITY] [FORCE_MPEG2] [TNS]

ENCODER_TYPE - either "ABR" or "VBR"

ABR OPTIONS:

- BITRATE - entered in kbps e.g. 128

VBR OPTIONS:

- QUALITY - a value between 0 and 10

- **FORCE_MPEG2** - either "TRUE" or "FALSE"
- **TNS** - either "TRUE" or "FALSE"

Examples:

- `-settings .aac ABR 128 FALSE FALSE`
- `-settings .m4a VBR 100 TRUE TRUE`

FLAC [COMPRESSION_LEVEL] [SAMPLE_RATE] [CHANNELS]

- **COMPRESSION_LEVEL** - a value between 0 and 8
- **SAMPLE_RATE** - entered in Hz e.g. 8000
- **CHANNELS** - either 1 or 2

Example: `-settings .flac 6 44100 2`

VOX/RAW [CODEC] [SAMPLE_RATE] [CHANNELS]

(Note: applies to both "-settings" and "-insettings" commands)

- **CODEC** - a number between 0 and 8, representing the codec type. The codecs on offer are (in order):
 - Dialogic ADPCM
 - G.711 uLaw
 - G.711 aLaw
 - 16 bit PCM (Intel Endian)
 - 8 bit unsigned
 - 8 bit signed
 - 16 bit PCM (Big Endian)
 - 32 bit float (Intel Endian)
 - 32 bit float (Big Endian)
- **SAMPLE_RATE** - entered in Hz e.g. 8000
- **CHANNELS** - either 1 or 2

Examples:

- `-settings .vox 0 44100 2`
- `-settings .raw 3 32000 1`
- `-insettings .vox 2 22050 2`
- `-insettings .raw 4 8000 1`

AMR [BITRATE]

- **BITRATE** - the bitrate at which you want to encode the AMR file (in kbps). The choices are 4.75, 5.15, 5.9, 6.7, 7.4, 7.95, 10.2 and 12.2.

Examples:

- `-settings .amr 4.75`
- `-settings .amr 10.2`

DCT [SAMPLE_RATE] [CHANNELS]

- **SAMPLE_RATE** - one of the following: 8000, 11025, 16000, 22050
- **CHANNELS** - one of the following: 0(Same as source channels), 1(Mono), 2(Stereo), 6(5.1 Channels), 8(7.1 Channels)

Example: `-settings .dct 11025 1`

M3U/PLS/WPL/RSS [URL]

- **URL** - the URL of the site where you will host the playlist file. URLs should start with the `http://` or `ftp://` protocol string as appropriate.

To configure the settings for the generated MP3 file, use the standard MP3 settings syntax to configure it (i.e. `-settings .mp3 etc. etc.`)

Examples:

- `-settings .rss "http://www.music.com/"`
- `-settings .m3u "http://livebands.com/rock/"`
- `-settings .pls "ftp://latinmusic.net/samba/"`

Encoder Options

This page provides links to our knowledge base pages for information on the settings available for each encoder supplied with Switch.

- [aac / m4a / m4r encoder options](#)
- [caf encoder options](#)
- [aiff encoder options](#)
- [au encoder options](#)
- [flac encoder options](#)
- [rss / m3u / pls / wpl encoder options](#)
- [mp3 encoder options](#)
- [ogg encoder options](#)
- [opus encoder options](#)
- [raw / vox encoder options](#)
- [wav encoder options](#)

Same as source

For some formats, this is an encoder option when you want the sample rate and number of channel(s) of output file to be the same as input source. Please be careful when using this option because the specific sample rate and number of channel(s) on the input may not be available for the output format. In this case, the conversion may fail or just use a default valid encoding option.

Converting M4P files

M4P files are songs purchased via Apple iTunes, and they are protected by Apple's "Fairplay" DRM (Digital Rights Management). M4A files may be DRM protected as well and are treated the same as M4P files.

M4P files can only be opened with iTunes on a computer or devices such as iPod, iPhone or iPad. The computer or device should be authorised using an iTunes account prior to playing, transferring or burning the files.

Switch cannot play nor convert an M4P file but you can obtain an unprotected (M4A) version of the file using iTunes.

There are two ways to unprotect an M4P file:

1. Using iTunes Plus

Please follow the steps in the Apple web page

- [Apple information about iTunes Plus](#)

2. Burn Audio CD in iTunes and convert the audio CD using Switch

Please follow the steps below to burn the M4P files to an Audio CD using iTunes.

- 1. Open iTunes and login using Apple ID and password.
- 2. Make a new Playlist and add MP4 files you wish to convert.
- 3. Insert a blank CD or a virtual CD.(see Note)
- 4. Select the Playlist.
- 5. Select menu File->Burn Playlist to Disk.
- 6. In the Burn Settings dialog select Audio CD and Gap between songs set to None.
- 7. click Burn to start writing to CD and wait until iTunes finishes writing the CD.
- 8. Open Switch.
- 9. Select menu File->Add CD to import the CD into Switch.
- 10. Select desired output format and click Convert. Switch will convert all tracks in the CD to the new format.

Note:

It is possible to use a Virtual CD by installing a virtual CD software. In this case no real CD is required. In Burn Settings (Step 6) select a virtual CD drive instead of a real CD drive.

Shortcut Keys Reference

Add Files	Command+O
Add Folder	Shift+Command+O
Remove	Command+Backspace
Remove All Files	Shift+Command+R
Play/Pause File	Space
Close Play Window	Esc/Enter
Select All	Command+A
Settings	Command+,
Convert Files	Return (Enter)
Edit Audio Tag	Command+T
Edit File (in WavePad)	Command+E
Help	Command+?
Exit Program	Command+Q
Go to Start (in Play Window)	Home
Go to End (in Play Window)	End
Rewind (in Play Window)	Arrow Left
Forward (in Play Window)	Arrow Right

Output Options

Sets options used for conversion such as encoder settings, and metadata written to the output file.

To set output options, click the **Options...** button near the selection of Output Format

Encoder Options

For information about encoder options, go to [Encoder Options](#) topic.

Default Metadata

All converted files will include the default metadata entered in these fields:

- Comment - Description of the audio file or song.
- Copyright - Copyright information.
- Producer - Producer of the audio file.
- Encoder - Encoder of the audio file.
- Encoded By - Software used for encoding.

For more information about tags, click [here](#).

If title tag is empty, use file name

Not all source files have tags, which results in empty title tags after conversion. Check this box to use the file name as the title tag.

Restore Defaults

Click this button to restore previous settings by the user. Once the user clicks the OK button, the default values will be updated according to the latest values

Send an Audio File

Send an Audio File

Switch allows you to send the file by either email or direct internet connection. You can select how to send a file by either of the following:

- Go to Tools menu, and click "Send Files..."
- Go to Convert menu, and click "Convert and Send Files..." - this will let you convert a file first, and send the output file

Email

If you select this option, Switch will send the file as an attachment to an email.

Enter the email address of the recipient in the Email Address box.

Email - Settings

By default Switch uses the MAPI system to send email using your existing email software (Eudora, Outlook, Lotus Notes etc.). To use this you must have email software installed and set as the "default MAPI server". The advantage of this system is that your email software manages the email. However, there can be some problems with some email software. If you have MAPI problems, try the alternative internal SMTP.

If you select the "Internal Simple SMTP" option, email is sent directly by Switch to your server. You must enter the SMTP mail host used by your ISP for sending of mail (call them if you do not know it).

If, when using Internal Simple SMTP, it does not work, it might be that your server requires an authenticated login (a username and password) to send email. If so, append :username:password to the SMTP server (i.e. smtp.yourserver.com:username:password).

Direct Internet Connection

Switch can send files directly to an internet server. This is faster than the email option but requires that your recipient have access to a web hosting (FTP) service to store the files.

Select the Direct Internet Connection option and enter the Server, User Name, Password and Directory as setup for your FTP server for your recipient.

Converting to CDA files

Why CDs are needed to convert .cda (Compact Disk Audio) file?

The .cda files that we see in our Windows explorer are generated by the Windows CD drivers. They act as a pointer or shortcut to the CD location of a specific track only and do not contain any musical information, or the actual audio file or music.

Thus, if then we attempt to convert a .cda file, it will require the CD that contains the actual audio file. If CD is not available during the .cda conversion, you will get an error as it cannot find the audio file that it points to.

NCH Software Suite

This is a useful way to browse all the software available from NCH Software

You can see a set of products by type like Audio, Video and so on and view the product. From there you can try out the product and it will download and install it for you to trial. If you already have the product installed then you can click "Run It Now" and the program will be launched for you.

There is also a list of features for products in the category. Click on a feature, such as "Edit a Video File", to install a product with that ability.

Search

Search our website for products matching any keywords you type.

See more of our software

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See the latest discounts we are offering for purchasing our products.

Opus Encoder Options

Encode audio into the Opus format.

Bitrate

Target bitrate in kbit/sec (6-256 per channel) In VBR mode this specifies the average rate for a large and diverse collection of audio. In CVBR and Hard-CBR mode it specifies the specific output bitrate. Default for ≥ 44.1 kHz input is 64kbps per mono stream, 96kbps per coupled pair.

Use default bitrate encoding

In default mode, encoder will choose bitrate automatically. For ≥ 44.1 kHz input is 64kbps per mono stream, 96kbps per coupled pair.

Use variable bitrate encoding

In VBR mode the bitrate may go up and down freely depending on the content to achieve more consistent quality.

Use constrained variable bitrate encoding

Outputs to a specific bitrate. This mode is analogous to CBR in AAC/MP3 encoders and managed mode in vorbis coders. This delivers less consistent quality than VBR mode but consistent bitrate.

Use hard constant bitrate encoding

With hard-cbr every frame will be exactly the same size, similar to how speech codecs work. This delivers lower overall quality but is useful where bitrate changes might leak data in encrypted channels or on synchronous transports.

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A party who intends to seek arbitration must first send to the other, by certified mail, a written Notice of Dispute ("Notice"). The Notice to NCH should be addressed to:

Legal Department
NCH Software, Inc.
6120 Greenwood Plaza Blvd, Ste 120
Greenwood Village CO, 80111
USA

("Notice Address"). The Notice must (a) describe the nature and basis of the claim or dispute; and (b) set forth the specific relief sought ("Demand"). If NCH and you do not reach an agreement to resolve the claim within 30 days after the Notice is received, you or NCH may commence an arbitration proceeding. The amount of any settlement offer made by NCH or you shall not be disclosed to the arbitrator.

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B. The arbitrator may award declaratory or injunctive relief only in favor of the individual party seeking relief and only to the extent necessary to provide relief warranted by that party's individual claim. **YOU AND NCH AGREE THAT EACH MAY BRING CLAIMS AGAINST THE OTHER ONLY IN YOUR OR ITS INDIVIDUAL CAPACITY, AND NOT AS A PLAINTIFF OR CLASS MEMBER IN ANY PURPORTED CLASS OR REPRESENTATIVE PROCEEDING.** Further, unless both you and NCH agree otherwise, the arbitrator may not consolidate more than one person's claims, and may not otherwise preside over any form of a representative or class proceeding. If this specific provision is found to be unenforceable, then the entirety of this arbitration provision shall be null and void.

C. Notwithstanding any provision in this Agreement to the contrary, we agree that if NCH makes any future change to this arbitration provision (other than a change to the Notice Address) you may reject any such change by sending us written notice within 30 days of the change to the Arbitration Notice Address provided above. By rejecting any future change, you are agreeing that you will arbitrate any dispute between us in accordance with the language of this provision.

D. To opt out of this Arbitration Agreement and class action waiver send an Opt Out notice to the Notice Address stating "I am electing to opt out of the Arbitration Agreement and class action waiver contained in the Legal Terms applicable to my purchase of an NCH product." Your Opt Out Notice must include the date and proof of purchase. The Opt Out Notice must be postmarked no later than thirty (30) days after the date of purchase. A separate Opt Out Notice must be sent for each product purchased.

Effects - Amplify

Amplify

To 'amplify' is to increase the loudness or volume of the selected region. To make a part of the recording softer or louder, select it and then use the menu Effects -> Amplify. The volume is entered in percent (100 being no change, 50 being -6dB softer or 200 being +6dB louder).

Effects - Normalize

Normalize

To 'normalize' is to adjust the volume so that the loudest peak is equal to (or a percentage of) the maximum signal that can be used in digital audio. Usually you normalize files to 100% as the last stage in production to make it the loudest possible without distortion. Another reason to normalize is to have multiple tracks sound equally loud, or to have equal average loudness.

The 'Peak' normalization method finds the sample of the greatest magnitude within the file. Normalization is then done with this value as the peak. With the Normalize Peak Level set to 100% (0dB), the whole file will be amplified so that the peak reaches 0dB.

The 'Average Loudness (RMS)' normalization method normalizes according to the file's average loudness, or volume. Multiple files normalized to the same peak level using this method will have equal average loudness. The 'Normalize Peak Level' for this method should be set much lower than for the Peak method, because the average loudness will always be lower than the peak sample.

The 'Peak Loudness (RMS)' normalization method attempts to normalize according to how loud the loudest part of the file will sound. This is the best method to use to make multiple tracks sound equally loud. As with Average Loudness, the 'Normalize Peak Level' for this method should be set lower than for the Peak method, because the peak loudness is lower than the peak sample. The actual algorithm used takes the RMS of each 50ms window in the file, ranks the windows from loudest to quietest, and then takes the 95th percentile of these as the 'peak'. Note that no adjustment is made for humans' differing perception of different frequencies.

The Perceived Loudness (dBA) normalization method uses A-weighted decibels which is an expression of the relative loudness of sounds in air as perceived by the human ear. Normalization is made because the human ear is less sensitive at low audio frequencies, especially below 1000 Hz, than at high audio frequencies.

The Integrated Loudness (EBU) normalization method uses the R-128, an international standard for loudness normalisation and maximum level of audio signals. Several platforms use this such as Spotify (-14dB), YouTube (-14dB), Apple Music (-16dB), Facebook (-18dB), and TV / Radio (-23dB). See [EBU R-128](#) for more details.

Effects - Compressor

Dynamic Range Compressor

A dynamic range compressor limits the volume levels of a sound recording so that it stays within a certain loudness range.

An example of where it is used is in TV broadcasting, where it ensures that the volume levels of ads are perceived as being louder than the television program itself (without any change in the actual broadcast volume).

It also has a use for recording audio from one medium to another, where the two mediums are not capable of handling the same range of volume levels (e.g. A CD can handle a much greater range than a cassette tape).

The Dynamic Range Compressor dialog has two tabs: "Simple" and "Graphic". Changing settings on the Simple tab will also change the graph on the Graphic tab, but not vice versa as the graph allows more control. There is also an "Advanced Compressor Settings" dialog for adjusting more advanced features.

The Simple Tab

The "Simple" tab of the Dynamic Range Compressor dialog contains settings called "Limiter", "Compressor", and "Noise Gate". While these sound like three different things, they are more accurately viewed as three different ways of using the dynamic range compressor.

The "Limiter" defines the maximum decibel level that the sound recording will be allowed to rise up to. So if, for example, the Limiter Threshold was set to -2dB, then you would never hear the volume level of the recording get louder than -2dB. Any signal over the limiter threshold would be clipped, which would probably cause distortion. Note that setting the Limiter Threshold to 0dB effectively turns the limiter off, because 0dB represents the loudest signal possible in a digital recording.

The "Compressor" reduces the volume of any sound which exceeds its "Threshold" setting. When a signal exceeds the threshold, the compressor gradually attenuates the sound to bring it down below the dB level, and does it in such a way that the listener will not be aware the attenuation is occurring. The compressor differs from the limiter in that the compressor does allow sounds to go above its threshold (for a short time), whereas the limiter does not.

The "Ratio" setting defines the ratio of the reduction in volume of sounds which exceed the compressor threshold. For example, if the ratio is 4:1 and the volume exceeds the threshold by 4dB, then the volume will be reduced to only exceed the threshold by 1dB. Note that a ratio of 1:1 means that there will be no change in volume; it effectively turns the compressor off.

The "Noise Gate" works similarly to the Compressor, except that it reduces the volume of sound below its Threshold. This can be useful for reducing or removing softer background noise from a recording.

You will find that the maximum Compressor Threshold you can set is the same as the current Limiter Threshold value. This basically means that, in any situation, the sound will start to attenuate at the Compressor Threshold, but will never be heard louder than the Limiter Threshold. Similarly, the maximum Noise Gate Threshold you can set is the same as the current Compressor Threshold.

The Graphic Tab

The "Graphic" tab of the Dynamic Range Compressor dialog shows a graph which represents the relationship between input and output volumes. The horizontal axis shows input volumes in dB from -60dB to 0dB. The vertical axis shows output volumes on the same scale. The graph will be changed by changes to settings on the Simple tab, but changes to the graph will not be reflected on the Simple tab, because it is possible to represent a wider variety of settings on the graph than is possible in the controls on the Simple tab. When the dynamic range compressor is applied it will use the settings from the Graphic tab.

To change the graph, click and drag the black vertex markers, or click anywhere else to create a new vertex. To remove a vertex, right-click on it.

Advanced Compressor Settings

Clicking on the "Advanced" button in the Dynamic Range Compressor dialog will open the Advanced Compressor Settings dialog. In it are controls for the following properties of the compressor:

-Input Level Sensing - Peak or RMS:

-This controls how the compressor determines the audio level. "Peak" sensing looks at the highest point in the window of audio which it examines. It will almost always give a higher reading than "RMS" sensing, which uses an average, or Root Mean Square of the window to determine the audio level. RMS sensing more closely corresponds to the audio level which a human listener would perceive.

-Compressor Response:

-

-Attack:

-The time (between 0 and 1000 milliseconds) that it will take to apply the gain adjustment. The total gain adjustment required will be gradually introduced over this period.

-Release:

-The time (between 0 and 5000 milliseconds) that it will take to remove the gain adjustment once gain adjustment is no longer needed. This is the opposite of attack.

-WindowLength:

-The length (between 10 and 50 milliseconds) of the window to use when calculating the current audio level. A shorter window responds to level changes more rapidly, but anything less than 50ms will start to respond inconsistently to bass, since 50ms (20Hz) is the wavelength of the lowest human-audible sound.

-LookAhead:

-How far ahead (between 0 and 100 milliseconds) to look at the input level when determining the output gain adjustment. This can cause the compressor to start responding to a change in volume before it happens. If this value is the same as the attack time, then the full gain adjustment could be made by the time the louder signal is reached.

-Side-Chain Equalizer:

-This determines how strongly the compressor should weight different audio frequencies when determining the input level. For example, to compress only when there is a loud bass sound, turn the Bass level up and/or reduce the MidRange and High levels.

-Auto Makeup Gain:

-When this option is selected compressor automatically makes up the gain lost in the compression process. Select this option if you want to amplify the compressor output to the original audio level.

Dynamic Range Compressor Presets

The following presets have been defined for your convenience. A preset will change the settings of the dynamic range compressor, after which you can make further adjustments if necessary. The presets are: -Default:

-Pressing the "Default" button will cause the compressor to have no effect. It sets the output levels to be exactly the same as the input levels, and also resets the advanced settings to their defaults.

-Fast Compressor:

-This compression preset will cause any spikes over -20dB to be rapidly reduced, but will not cause distortion. It uses peak input level sensing and a fast attack, which will reduce the volume of transient sounds (such as a snare drum hit), but may also change their characteristic sound. Compare this with the Smooth Compressor preset below.

-Smooth Compressor:

-This preset reduces the volume more gradually when the signal climbs above -20dB. The slow attack time will mean that transients (such as snare drum hits) will not be changed, or if they are then they will be uniformly reduced, thus their characteristic sound will not be significantly altered.

-Heavy Compressor:

-This preset uses a lot of compression whenever the average volume climbs over -30dB, resulting in a very uniform dynamic range. This can be useful for making the quieter parts of music with a large dynamic range (such as classical music) easier to hear in noisier environments, such as in a car or a restaurant.

-Hard Limit:

-This preset does not allow any sounds to exceed -12dB. This may cause distortion due to clipping in some tracks.

-Soft Limit:

-This limit allows short spikes over -6dB, but will prevent longer durations of audio over this threshold.

-Noise Gate:

-This will remove soft sounds from a track. This can be useful for removing the crackle of a record player during silences, or background noises in a dictation.

Effects - Equalizer

Equalizer

An equalizer changes the frequency response of a signal so it has different tonal qualities. After you select Effects menu -> Equalizer you will see a dialog containing three different Equalizer representations. Use the tabs at the top to select between the Visual Equalizer, Graphic and Parametric Equalizer views.

Visual Equalizer

Left click on any point to create a new band point. To remove a band point right click on it. To assist you with shaping the Equalizer graph in the way you want, there is a preset list that displays the most common sorts of filters used in the Equalizer graph. You can choose any preset filter from the list and then manipulate the filter to achieve the effect you desire. The list of filters to choose from and how you can shape them are explained below. Note that all fields where a frequency value is entered can have a maximum value of 20000 (Hertz).

Graphic Equalizer

The Graphic Equalizer uses discrete sliders to set the gain or attenuation of a signal at a particular frequency. You can select how many sliders you would like to manipulate by entering a value between 3 and 20 in the box at the top of the display. When you change the number of sliders you would like to utilize, the frequencies are automatically allocated to best span the audible frequency range from 20Hz to 20kHz. Selecting presets allows you to easily configure common filters such as low pass or high pass. Note that when you change the Graphic Equalizer, the Visual and Parametric Equalizer views are not changed, as the changes in the three views are not compatible.

Parametric Equalizer

The Parametric Equalizer is similar to the Graphic Equalizer, but with more control. Here you can adjust the frequency and bandwidth of the individual sliders by left clicking on the frequency or Q values below each slider. Frequency must be set between 20Hz and 20,000 Hz. The Q parameter must be set between 0.05 and 20. A higher Q causes the gain or attenuation peak at the frequency to be much sharper, and therefore less likely to impact adjacent frequency content, while a lower Q applies the modification more smoothly across the frequency spectrum.

-Band Pass Filter

-Keeps only those frequencies in the audio between a certain range. -Start Frequency

-The lower cutoff frequency value, in Hertz.

-End Frequency

-The upper cutoff frequency value, in Hertz.

-Slope Length

-The width of the slope extending from the lower and upper cutoff points, in Hertz.

-Amplitude

-The degree that the frequencies outside the cutoff range are suppressed. 6dB means the volume is reduced to one-half, 12dB means the volume is reduced to one-quarter. Maximum value is 60dB.

-Band Stop/Cut Filter

-Keeps all frequencies in the audio except those between a certain range.

-Start Frequency

-The lower stop frequency, in Hertz.

-End Frequency

-The upper stop frequency, in Hertz.

-Slope Length

- The width of the slope extending from the lower and upper stop points, in Hertz.
- Rejection
 - The degree that the frequencies inside the stop range are suppressed. 6dB means the volume is reduced to one-half, 12dB means the volume is reduced to one-quarter. Maximum value is 60dB.
- High Pass Filter
 - Keeps only those frequencies in the audio above a certain value.
- Pass Frequency
 - The point at which all frequencies above are to be kept, in Hertz.
- Slope Length
 - The width of the slope extending from the pass frequency, in Hertz.
- Low Pass Filter
 - Keeps only those frequencies in the audio below a certain value.
- Pass Frequency
 - The point at which all frequencies below are to be kept, in Hertz.
- Slope Length
 - The width of the slope extending from the pass frequency, in Hertz.
- Notch Filter
 - Attenuates the frequencies in the specified range to very low levels and passes all other frequencies unaltered. There is no slope - frequencies are either attenuated or not.
- Start Frequency
 - The lower cutoff frequency value, in Hertz.
- End Frequency
 - The upper cutoff frequency value, in Hertz.
- Boost Filter
 - Either attenuates or boosts frequencies in the specified range and passes all others unaltered.
- Start Frequency
 - The lower boost/cut frequency value, in Hertz.
- End Frequency
 - The upper boost/cut frequency value, in Hertz.
- Slope Length
 - The width of the slope extending from the lower and upper boost/cut points, in Hertz.
- Amplitude
 - The degree that the frequencies inside the boost/cut range are either boosted or cut. 6dB means the volume is boosted to twice the original amount, and 12dB means the volume is boosted to four times the original amount. 20dB.
- High Pass Shelf Filter
 - Attenuates signals of frequencies below the cut frequency and passes all others unaltered.
- Start Frequency
 - The lower cut frequency value, in Hertz.
- Slope
 - The width of the slope extending from the lower and upper cut points, in Hertz.
- Rejection
 - The degree that the frequencies inside the cut range are cut. 6dB means the volume is attenuated to about half the original level, and 12dB means the volume is attenuated to about a quarter of the original level.
- Low Pass Shelf Filter
 - Attenuates signals of frequencies above the cut frequency and passes all others unaltered.
- Start Frequency

- The lower cut frequency value, in Hertz.
 - Slope
 - The width of the slope extending from the lower and upper cut points, in Hertz.
 - Rejection
 - The degree that the frequencies inside the cut range are cut. 6dB means the volume is attenuated to about half the original level, and 12dB means the volume is attenuated to about a quarter of the original level.
- If you are using the equalizer simply to drop lower frequencies, you should always try the High Pass filter first (Effects menu -> High Pass Filter), because it is better and faster for very low frequencies.

Effects - Envelope

Envelope

The 'envelope' is the change in volume of the selected region over time. This can be used to make fine adjustments to the volume over time or even more crude changes like fade in or fade out.

Select the region you want to change the volume over and use the Levels tab -> Envelope or Menu -> Effects -> Envelope. Click and drag on any point to adjust its volume (right click removes the point). Click the Set Flat button to reset the volume and remove extra volume points.

Effects - Stereo Pan

Stereo Pan

The stereo pan effect allows you to change how loud the sound is that comes out the left or right speaker. For example if you had a stereo recording with all the sound coming out of only one speaker, you could use the pan effect to "center" the sound yourself. You can also make a centered sound change move one from speaker to the other as the sound file plays.

Select the region you want to change the pan for and choose Effects -> Stereo Pan. Click on a point and move it upwards for an increase in volume on the left speaker, or move it downwards for an increase in volume on the right speaker.

Please note the stereo pan effect only works on stereo files. If your file is not stereo you must first convert it to stereo by choosing Edit -> Convert Channels -> Stereo.

Effects - Echo

Echo

An echo is a repeat of the sound after a short time (usually 400 - 1000ms). It sounds a bit like the person is in a large stadium or is shouting between two mountains.

To add echo select the region and use the menu Effects -> Echo then specify the duration and amplitude of the echo. The duration is the length of time after which the sound repeats - usually this is between 400 and 1000ms. The amplitude can be between 1 - 99% (99 being a very loud echo).

Effects - Pitch Shifter

Pitch Shifter

Pitch Shifter is a sound effect that raises or lowers the pitch of audio signals. You can adjust pitch shifter speed by dragging the slider in the settings.

Effects - Reverb

Reverb

Reverb is many small reflections of the sound that come after a set time. It usually occurs when someone is speaking in a room, hall, etc. More reverb is called wet, no reverb is called dry. When you select the reverb effect, you will see a dialog with two tabs.

Simple

The first tab of the reverb effect allows you to adjust the reverb level and time. The reverb level is the amplitude - 99 is very wet, 0 is dry. The time can be between 100 and 800ms - 200ms sounds like a small room or 800ms a large hall. If you add too much reverb it can sound like the person is in a pipe or in the bathroom.

The Simple tab also includes preset options to choose from, depending on how large the space being simulated is. Click the play button at the bottom of the tab to preview the reverb effect on your audio.

Room Design

The second tab of the reverb effect allows you to specify the dimensions of a room, the position of the source and listener, and the room absorption with preset options for the materials that make up the walls, floor and ceiling of the room. Click the play button at the bottom of the tab to preview the reverb settings on your audio.

Effects - Phaser

Phaser

The phaser sound effect is created by mixing a slightly delayed signal with the original. You can set the delay in ms (default 5ms) and the wet dry gain in percent. 100% is wet. 0% is off/dry.

Effects - Flanger

Flanger

A Flanger sound effect is similar to the phaser except that the delay is slowly modulated over time. You specify the starting delay time (default 5ms), the frequency of modulation in times per second (default 0.5Hz which is 2 seconds) the depth of modulation (default 70%) and the wet dry gain (100% for wet, 0% for dry, default 60%).

Effects - Vibrato

Vibrato

The vibrato sound effect is a pulsating of the pitch at a depth and frequency specified by the user. The higher the Frequency (Hz) set, the more often the pulses will be heard, and the higher the Depth (semitones), the wider the fluctuation in pitch will be.

Effects - Tremolo

Tremolo

The tremolo sound effect is similar to the vibrato effect, except that the amplitude pulsates rather than the pitch. The higher the Frequency (Hz) set, the more often the pulsation will be heard, and the higher the Depth (%), the deeper the fluctuation in volume.

Effects - Doppler

Doppler

The doppler effect simulates the sound of a passing vehicle, which has a high pitch while approaching, shifting to a low pitch when traveling away from the listener. Specify the Velocity (in km/h) of the passing source; a higher velocity will result in a higher starting pitch and lower ending pitch. Adjust the Listener Horizontal and Vertical Positions to indicate the listener's horizontal and/or vertical position to the passing source; play around with the values to achieve different combinations of pitch.

Effects - Wah-Wah

Wah-Wah

As the name suggests, the effect modulates a specified frequency band within the sample, which results in the characteristic "Wah wah" sound. The effect is a bandpass filter with its center frequency (not to be confused with the center frequency parameter, below) alternating between a min frequency and max frequency (specified by the center frequency and depth parameters) and from max frequency to min frequency. The frequency of alternating direction is represented as a triangular wave with a frequency specified by the wah frequency parameter.

Resonance: also known as Q or emphasis, this parameter controls the resonant peak of the bandpass filter. This value determines the sharpness of the wah-wah effect. Higher values produce more resonant/peaky tones.

Depth: this parameter determines the frequency range swept by the bandpass filter. Its range is specified as a percentage of the range (0 to center frequency). If the value of the percentage of the range (0, center frequency) is specified as X, the min and max frequencies are (center frequency - X) and (center frequency + X).

Center Frequency: This parameter is the center frequency of the bandpass filter sweep, and is used to determine the min and max frequencies as mentioned above.

Wah Frequency: This is the frequency of alternating the direction of the sweep, or the frequency of the wah-wah sound. It is the frequency of the triangular wave described above.

Effects - Chorus

Chorus

The chorus sound effect is used to make one voice or one instrument sound like 3 voices or instruments by playing the original with variably delayed and slightly pitch changed copies of the original.

Note: Chorus is a very useful way to make a mono source sound more stereo. You should convert your file to stereo first before using Chorus.

Effects - Distortion

Distortion

While normally we do everything to reduce distortion, sometimes you want to add it. It is popular for use with guitars. The distortion is measured between 0.0 (off) and 1.0 (clipping). You also specify the level where it kicks in in dB.

For a more consistent sound, you should apply Dynamic Range Compression first before you add distortion.

Effects - AM Radio Effect

AM Radio

This simulates an AM Radio. We have made it accurately simulate a 'good' AM radio. To make it worse, apply the effect twice. For a really bad sound, paste mix some soft white noise (use the Tone Generator tool) to simulate bad reception.

Effects - Telephone Effect

Telephone

This simulates the audio down a telephone line. It simulates a 'good' telephone line. To make it worse apply the effect twice and paste mix soft white noise.

Effects - Reverse

Reverse

This effect reverses the selection in the same way playing a record or tape backwards would.

Effects - Fading

Fade In

To fade in use the menu Effects -> Fade In.

Fade Out

To fade out use the menu Effects -> Fade Out.

Fade Out and Trim

The fade out and trim option is a combined function which fades out over the selection then marks the end of the selection as the end of the file. This is frequently used at the end of music tracks.

CrossFade

The CrossFade tool allows you to mix together voice and music in a variety of different ways. You can, for example:

- Fade out a music track while fading in another track,
- fade out a music track and cue in a voice track at full volume (or vice versa), or
- overlay the end of one voice track with the start of another track.

To use the tool, first select the region of audio you want to perform the crossfade on. If you want to crossfade between two files, you must combine the two files together first into one file. Next, go to Effects menu -> CrossFade. A window will appear, showing a graph and a number of data fields.

The graph is divided into two sections, the top section shows the fading in part of the audio, the bottom shows the fading out. The area that the crossfade is to be performed on is highlighted in blue, and surrounded by markers showing the start and end of the crossfade region. There is a one second portion of the waveform on either side of the highlighted section, which is there to provide a better view of the crossfade.

If you hover your mouse over any part of the graph, you can see what parts of the graph correspond to what time in the audio waveform.

The data fields work as follows:

- Start and End Selected Positions
 - Tells you the start and end times of the audio you selected in the waveform. **Note:** These times do NOT correspond to the start and end times you see in the graph window! Read on!
- Gap Time
 - This says how long the crossfade region will be, in milliseconds. This time may be modified when the crossfade is performed, if the fade in and fade out times are larger than this value. **Note:** If this time is shorter than the audio you selected then the middle of the selected audio will be lost as a result of the crossfade.
- Fade In Time
 - The length of time to fade in the end of the selected audio. For example, if you select 5000ms of audio and a Fade In Time of 1000ms, then the last 1000ms of your selection will fade in over the last 1000ms of the crossfade.
- Fade Out Time
 - The length of time to fade out the beginning of the selected audio.
- Fade Type
 - The shape of the transition over time of the fade in or out. You can select from Linear, Logarithmic, Exponential or Sinusoidal fade types.
- Linear - Volume change will occur evenly over time.
- Exponential - Volume change will start slowly to begin with and then rapidly become faster towards the end.

- Sinusoidal - Volume change will start slowly, then increase rapidly, then slow down again towards the end.
- Logarithmic - Volume change will start rapidly and slow down towards the end.

So with the above information in mind, the crossfade will work as follows:

1. A Fade-Out buffer will be created with a length of the Gap Time. At the start of the buffer will be the start of your audio selection, fading out over the Fade Out Time.
2. A Fade-In buffer will be created with a length of the Gap Time. At the end of the buffer will be the end of your audio selection, fading in over the Fade In Time.
3. The Fade-In and Fade-Out buffers will be mixed together, and replace your audio selection.

Effects - Speed and Pitch Changing

Simple Speed and Pitch Change

This plays the recording faster or slower which in turn increases or decreases the pitch too. This function is useful to correct slow or fast tapes.

Speed Change

Normal speed changes (i.e. "Simple Speed and Pitch Change" above) changes the pitch in proportion to the speed. If you want to change the speed but keep the pitch the same use this function. Speed can change the duration of the audio. The time duration (in seconds) can also be adjusted using this effect.

Pitch Change

This changes the pitch of the recording without changing the speed (i.e. the converse of the above). Change of semitones can also be adjusted using this effect

Pitch Speed Profile

This allows you to specify how much to change pitch, speed, or pitch and speed at any point in the file, using a graph.

Effects - Reduce Vocals

Reduce Vocals

If you want to reduce the vocals from a music track you can use this effect. This effect will attempt to identify the voice in the left-to-right spectrum of a stereo recording and remove it. The recording must be stereo (from an original stereo source like a CD - simply converting a file to stereo will not work). It will also remove any instruments near the voice in the stereo spectrum.

Perform Only Simple Channel Subtraction - This option works best if the vocals are recorded in the center of the stereo recording with instruments spatially separated. If the stereo is simply a mono copied to the left and right channels, the lack of spatial separation will result in silence rather than a relative reduction in vocals.

Note: it is impossible to remove the vocals perfectly without the original mix track. You will notice some instruments might be removed too and some vocal remain. The effect will also not work on some files which have previously encoded in a highly compressed form like mp3 (because this remove some stereo depth).

Effects - Voice Change

Voice Change

The Voice Changer allows vocal distortion by changing pitch, semitones, cents, and timbre, by modulating tone, and adding whisper/noise to the voice.

Audio Cleanup - Noise Reduction

There are two ways of reducing noise. The slow but accurate "Spectral Subtraction" method - usually used where noise is really a problem - and the fast "Multiband Noise Gates" method - usually just automatically on batch voice recording jobs.

Sometimes using both (spectral always must be first) then multiband gates works very well.

- Spectral Subtraction

- -Automatic Method

-This approach will automatically estimate what is noise and what is not. It usually works well on voice and is nice and easy to use; just select the region and apply the effect.

- Manual Method

- To use this you must:

1. Select a short part of 'noise only'. Usually this is from a gap in the audio.
2. Select Effects -> CleanUp -> Noise Reduction -> "Grab noise sample from selected area for spectral subtraction".
3. Select the entire file.
4. Select Effects -> CleanUp -> Noise Reduction -> "Spectral subtraction based on noise sample".

- Multi-Band Noise Gate

- Multi-Band Noise Gate removes all the audio data below a given threshold in the audio file.

A good noise threshold for most audio files is usually between -30dB and -20dB.

Noise Gate

A noise gate is a filter which controls the volume of an audio signal. Any part of your audio which is below the Threshold will be attenuated by the amount you specify.

- Threshold

- Audio falling below this threshold will be attenuated.

- Hold

- The period of time (in milliseconds) to wait before applying the attenuation.

- Release

- The period of time (in milliseconds) taken to fully apply the attenuation.

- Attack

- The period of time (in milliseconds) taken to fully remove the attenuation.

- Attenuation

- The amount to attenuate the audio signal when it falls below the threshold.

Noise Removal Wizard

An easy-to-use noise removal wizard can be found on the Tools tab. The wizard guides you through choosing the best parameters based on noise type description, then applies the noise reduction. To use the wizard, click the Noise Removal button on the Tools tab.

Audio Cleanup - Click/Pop Removal

Auto Click/Pop Removal

This tool allows you to apply a repair of a single click/pop artifact. To use it properly, you must zoom right in to the artifact and select a small region around it. Then select Tools menu -> Auto Click/Pop Removal. The repair will be performed straight away.

Parametric Click/Pop Removal

This tool is designed to remove click and pop sounds from recordings. It is ideal for those who have recorded music onto their computer from LP records and want to repair any defects caused by dust and scratches on the vinyl.

To use the tool, click Tools menu -> Parametric Click/Pop Removal. In the window that appears, you can configure settings for the following fields:

-Click Sensitivity

-This is the degree of aggressiveness (as a percentage) that will be applied by the tool when searching for click and pop artifacts. If you don't know what to enter, you can start by leaving it at 50%. The more a piece of audio is damaged, the higher you may have to set it. Moderately damaged audio can require settings of 60% - 80%. Be careful though - if you set it too high, the tool will start thinking parts of the audio are actually clicks/pops. If you set it too low of course, the tool will think some clicks/pops are part of the audio. Try experimenting to find the right value, and note that the level you apply to one file may be different to the level you apply in another file.

-Maximum Click Length

-This is the maximum length that a click lasts in your audio, in milliseconds. As a general guide, use 450ms if you don't know what to enter. 350ms is appropriate for audio with only small amounts of defects, whereas 550ms or 650ms is appropriate for audio with lots of defects.

Audio Cleanup - High-Pass Filter

High-Pass Filter

A high-pass filter (sometimes called a low cut filter) removes all low frequencies below a specified Hz. This is useful if you want to make your recording sound 'clearer' or less 'muddy'. It is very usual to use a high-pass filter of about 250Hz on all voice recordings to improve intelligibility.

Audio Cleanup - Low-Pass Filter

Low-Pass Filter

A low-pass filter removes all high frequencies above a specified Hz. This is useful if you want to make your recording sound 'clearer'. It is very usual to use a low-pass filter of about 1600Hz on all voice recordings to improve intelligibility.

Audio Cleanup - Automatic Gain Control

Automatic Gain Control

Automatically adjusts the amplitude to compensate for variations at the input, in order maintain an suitable amplitude at the output. Changes the level of the file up and down over periods of time (time frame: 5 seconds)

De-esser is applied to avoid the AGC increasing "S" sounds and making them too loud (default for voice: 2200Hz).

High-Pass filter is applied to avoid low frequency bass making the AGC reduce volume (default for voice 450Hz).

Other features which can be used with AGC:

- Normalization – changes the level of the whole file equally (time frame: whole file duration)
- Dynamic Range Compressor – instantaneous regulation of volume so softer sounds are increased and louder sounds are decreased (time frame: 100ms)

Audio Cleanup - DC Offset Correction

DC Offset Correction

Often when you record audio using bad electronics the recording has a constant 'DC' level throughout the file. Because the ear cannot hear this you will not notice it until you attempt to edit in other audio when you can hear horrible clicks. If you think this is the problem you can run DC Offset Correction over the entire recording before you begin to edit. Another (and possibly better) way to deal with this problem is to run a high pass filter (say at 50Hz) over the recording.

General - Find and Play Recordings

Find and Play Recordings

Open the Find and Play Recordings dialog box from the Find & Play menu item, or pressing the Find & Play Toolbar button.

All recordings in the specified recording folder will be listed.

Playing a Recording

To play a recording select it and press the spacebar . When the Play Control Box opens you can move back and forward within the recording using the left and right arrow keys. To pause playback press the spacebar. To stop and go to the beginning press Esc. To move to the beginning press Home. To move to the end press End. To go back to the list of recordings press Enter.

Edit Audio

This will open the selected recording in WavePad: A full featured professional sound editor for Windows. When editing audio files you can cut, copy and paste parts of recordings and, if required, add effects like echo, amplification and noise reduction. WavePad works as a wav editor or mp3 editor but it also supports a number of other file formats including vox, gsm, real audio, au, aif, flac, ogg and more.

WavePad is designed to be a very easy and intuitive to use for audio editing. Within minutes you will be able to open or record a file and edit it. But if you take time to explore the other features you will find many powerful tools for editing audio designed with the professional sound engineer in mind. WavePad comes in two versions a free basic version and a masters edition, both with a full-featured trial period.

Save As

This will save the selected recording to a different position in a local or networked folder.

To save the recorded file on your hard drive, select the recording and click the "Save" button. Enter the file name and click OK.

Send to Email Now

The selected file will be attached to a default email and sent to the address you specify.

Burn to CD-ROM

The selected files will be imported into Express Burn. Express Burn is also the fastest CD/DVD writing program in the world using proprietary optimizing systems. It will also create an index file which it will begin burning immediately to a data CD.

Compress and Backup

Are your file sizes getting too large, or do you wish to ensure you have a backup of your audio? Compress and Backup will provide you the choice of burning straight to a CD after compression or to a Backup Folder. It can compress in 3 different formats. The choice of compression depends on the particular audio settings that determined the quality of the sound file that was recorded. Please test that the compression type is sufficient in size and quality of audio for a sample file. Compressing and backing up does not effect your original audio recording.

Convert Sound File

Converts the audio file(s) to a different format using Switch. Switch audio file converter is very easy to use. With the selected files added into Switch, select the format you want to use, and then click the convert button. Switch supports converting to wav (PCM, ADPCM+, aLaw+, uLaw+, and others), mp3, au, aif/aiff, gsm, vox, raw, ogg, flac, .rss, .amr+ and more.

Transcribe

Allow simple and easy dictation of files using Express Scribe. This computer transcriber application features variable speed wave playback, foot pedal operation, file management and more.

Deleting a Recording

To delete recordings, select them and click on the Delete button. You can select more than one recording by holding down the Shift key while selecting.

Folder Tab

By default, Find and Play lists only recordings in the local record folder.

You can change the default folder for holding your audio files or you can change the alternative folder to suit your needs.

If you want to play files over your local area network, install this program on the other computer and point it to the folder that the Find and Play Window is referencing.

View Filter Tab

The View Filter Tab allows searching and filtering on what is displayed in the list view.

You can search or filter what is shown by the text in each column, by the duration of the audio or by the date of the recording.

Play Options Tab

Set the sound device you use to play your audio file, also set 'Variable Speed Slow' and 'Variable Speed Fast' to vary the speed of playback.

Clicking 'Update Now (Rescan for new recordings)' will add or remove any recordings that have been made or deleted since the last update.

Screen References - Select Stream

Select an audio stream **Audio Stream:**

Select an audio stream from the file

Don't ask me again

Suppress the popup dialog to select audio stream

Screen References - Custom Resolution

This dialog allows you to set up the resolution for your video during saving. To set a custom resolution, click the Save button on the toolbar and select the Disc: Make Data Disc, Computer/Data, Portable Device, Image Sequence, or Stereoscopic 3D options. From the Resolution drop menu, select Custom.

In the Custom Resolution dialog that opens, input the width and height in pixels and click OK. If **Keep aspect ratio** checkbox is available, you can check it to retain the same aspect ratio when changing the width or height.

Screen References - Custom Resolution (360-Degree Video)

This dialog allows you to set up the resolution for your 360-degree video during saving. To set a custom resolution, click the Save button on the toolbar and select the 360 Video File option. From the Resolution drop menu, select Custom.

In the Custom Resolution dialog that opens, input the width and height in pixels and click OK. The width must be exactly double the height for 360-degree videos.

Screen References - Custom Frame Rate

This dialog allows you to set up the frame rate (number of frames per second) for your video. To set a custom frame rate, click the Save button on the toolbar and select either of the Disc: Make Data Disc, Computer/Data, Portable Device, or Stereoscopic 3D options. From the Frame Rate drop menu, select Custom.

In the Custom Frame Rate dialog that opens, input the frame rate and click OK.

The default frame rate is **Variable Frame Rate (VFR)** but the dialog allows the user to select **Constant Frame Rate (CFR)** by selecting the **Constant frame rate** checkbox.

Screen References - Detected Format

This dialog shows you the detected format of the video sequence during the saving process. To detect the format of your video, click the Export Video button on the Home tab bar and select either of the following options: Video File, 3D Video File, Portable Video, Lossless Video, DVD Movie Disc, DVD Data Disc, Image Sequence, YouTube, Flickr, Dropbox, Google Drive, OneDrive, or Vimeo.

Click the Detect button next to the Preset field. Detect will adjust the video output resolution format based on your monitor screen size.

If you want to use a different format you can select the format from the Preset drop-down list. Click OK to use the selected format as the output format.

Screen References - DVD Bitrate Settings

This dialog allows you to specify bitrate for your DVD video. To open this dialog, click the Save icon in the toolbar and select the Disc save option and click the Make DVD Movie option. Click the Advanced button to open the DVD Bitrate Settings dialog.

The default option is to calculate the bitrate automatically. To set the bitrate manually, select the manual option and adjust the video and audio bitrates.

Screen References - Choose Base Folder to Mirror

This feature allows you to copy the folder structure (folders and their subfolders) of your "base" folder into another location. This is ideal for if you want to convert your entire music library, including it's folder structure, to a new location.

Choose base folder to mirror

Click this checkbox if you want the folders that contain the files you have added to Switch to be copied with the newly converted files.

When you hit the convert button a window will open asking you to choose your base folder to copy. This folder needs to contain (at some point) all the files you have added to switch for conversion. For example, if you have added all the files in "C:\MyMusic\artists\albums" your choices for base folder would be "MyMusic", "Artists" and "Albums."

The new base folder and all its subfolders will be copied to the output location you have specified in the output field in Switch. So if your output folder has been set to "C:\Documents\Converted Music" and you specify your base folder as "C:\MyMusic" the converted files will be copied to "C:\Documents\Converted Music\MyMusic.." with the folders that contain your music being recreated there.