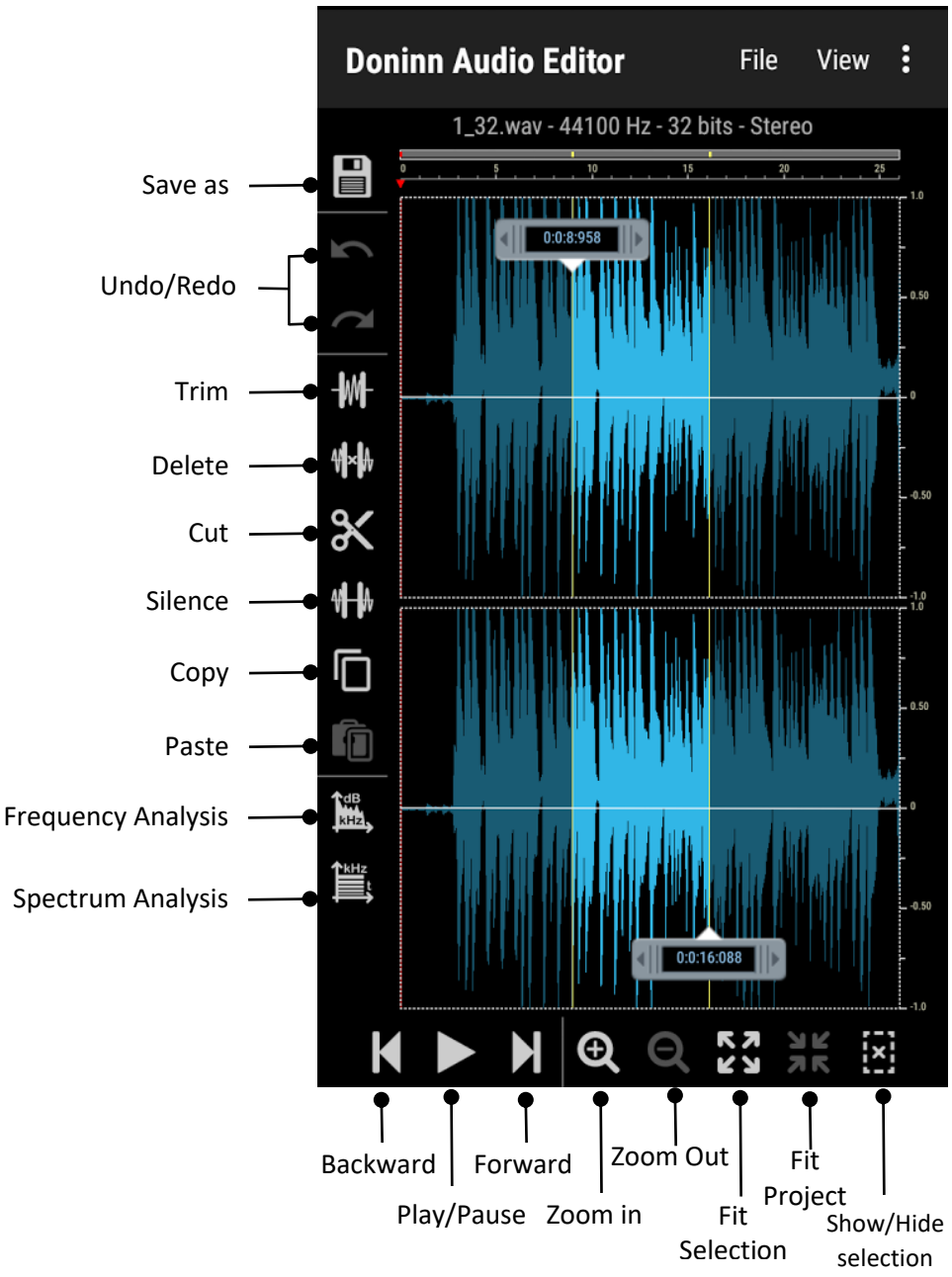
Doninn Audio Editor

User Guide

CONTENTS

EDITOR

Main window



Menu File

Save

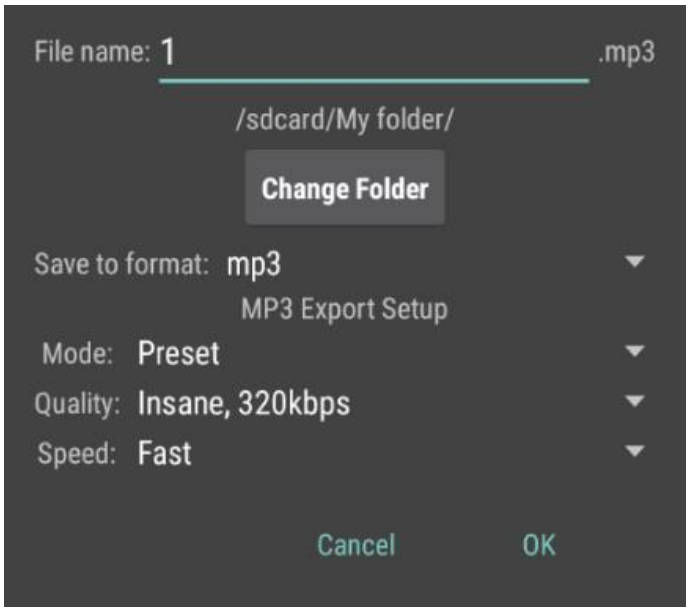
Saves changes to the current file.

Save as

Saves the current file in any available format under a different filename and in the specified folder.

Available formats: mp3, flac, wav, ogg.

Mp3 Export Setup



Mode

There are four different ways to the trade-off between the size and quality of the exported files: *Preset, Variable, Average, Constant.*

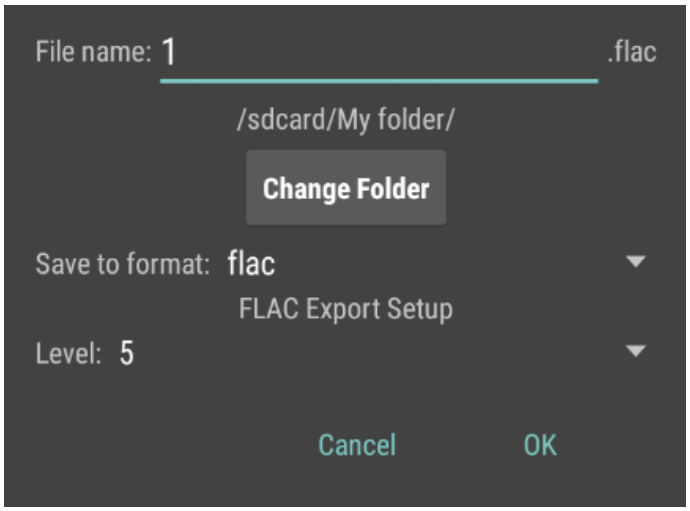
Quality

This list lets you select the bit rate in kbps (kilobits per second) for encoding your file. A higher bit rate always gives better quality but at the expense of a larger file size and vice versa. The bit rates are absolute for average and constant bit rate modes, but expressed as a range for the variable and preset modes (except for the Insane preset).

Variable Speed

Choice of speed is available when using variable bit rate encoding.

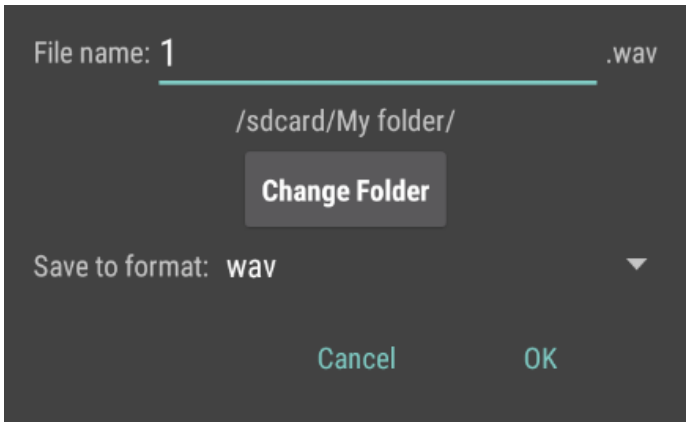
FLAC Export Setup



Level

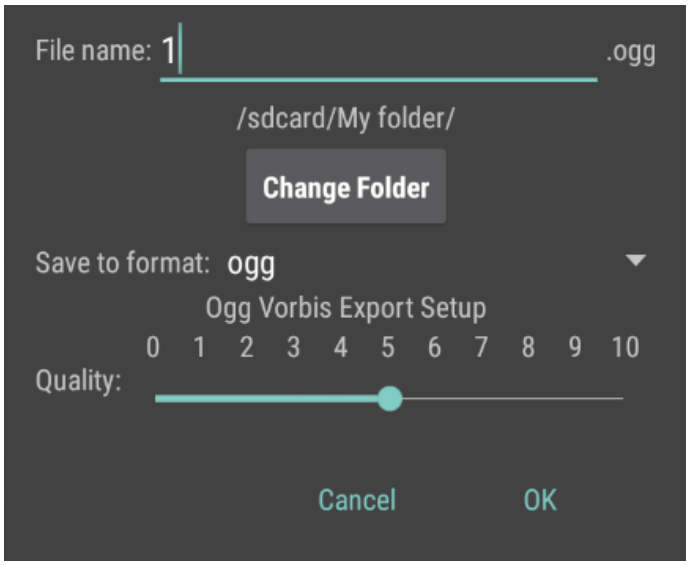
The amount of file size compression used to losslessly pack the audio data. The level settings range from 0 to 8. Level 0 is optimized to encode as fast as possible. Level 8 is optimized to pack as efficiently as possible, so produces slightly smaller files than level 0 at the expense of taking longer to encode.

WAV Format



WAV format has no options, as this container file format for storing uncompressed audio, without any loss of quality.

Ogg Vorbis Export Setup



Quality

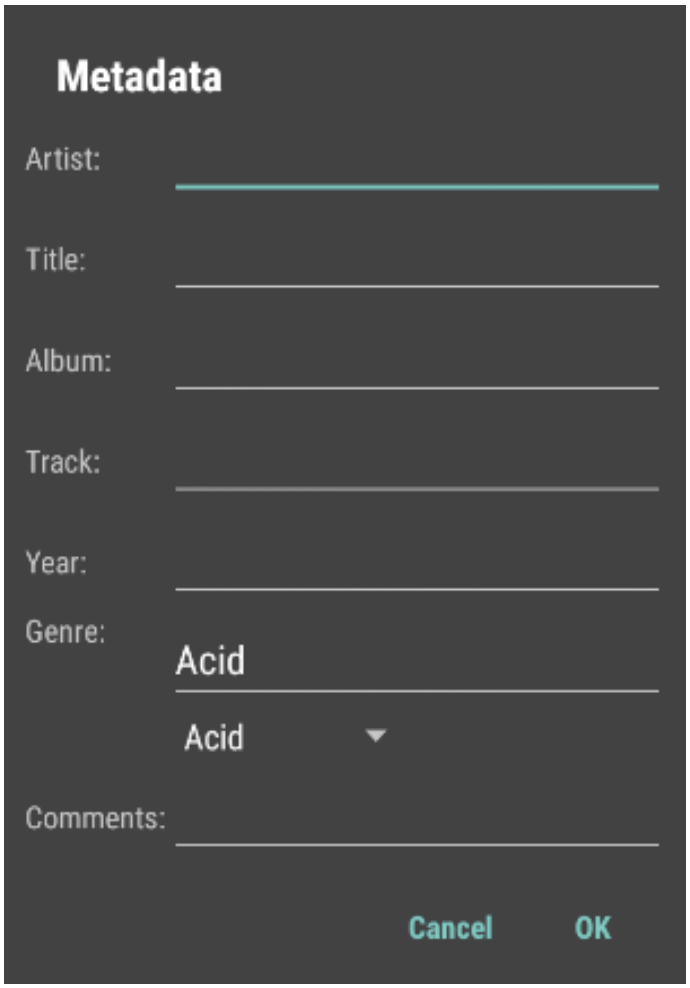
Choose a quality setting from 0 (lowest) to 10 (best). The default level of 5 typically produces a file of about the same size as default 128 kbps MP3 encoding, but of higher

quality. Higher OGG quality settings mean a larger file size, because a higher overall bit rate will be used.

Save selection as

Saves the selected area as a new file in any available format under a different filename and in the specified folder.

Metadata



Edit metadata tags.

As ringtone

Sets the current file as a ringtone.

Exit

Exiting the program

Menu View

Hide Selection/Show Selection

Controls the display of selection sliders on the main editor window.

Select all

Selects the entire audio track.

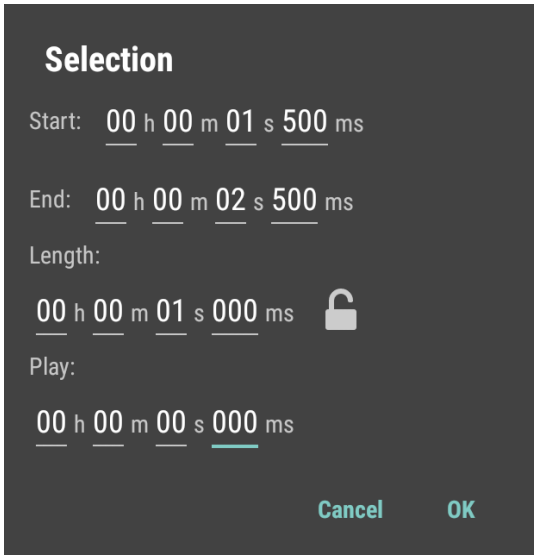
Select to start

Shifts the left selection slider to the beginning of the file.

Select to end

Shifts the right selection slider to the end of the file.

Set range



In the dialog window, you can specify the beginning and end of the selection, or the total duration; and also specify the position of the play marker. If you need to select a part of the fixed-length audio track, you can click on the lock and edit the beginning, then the end will be calculated automatically, or vice versa.

Fit Selection

Scales the selected area of the signal horizontally to the entire screen of the sound waveform.

Fit Project

Inscribes a sound wave in the entire screen.

Zoom In

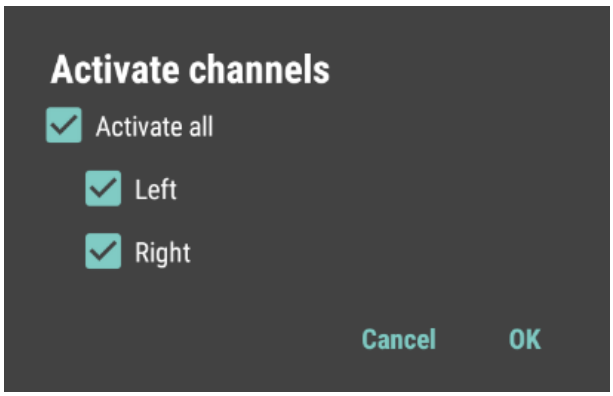
Increases the waveform display horizontally.

Zoom Out

Zooms out to a lower magnification level.

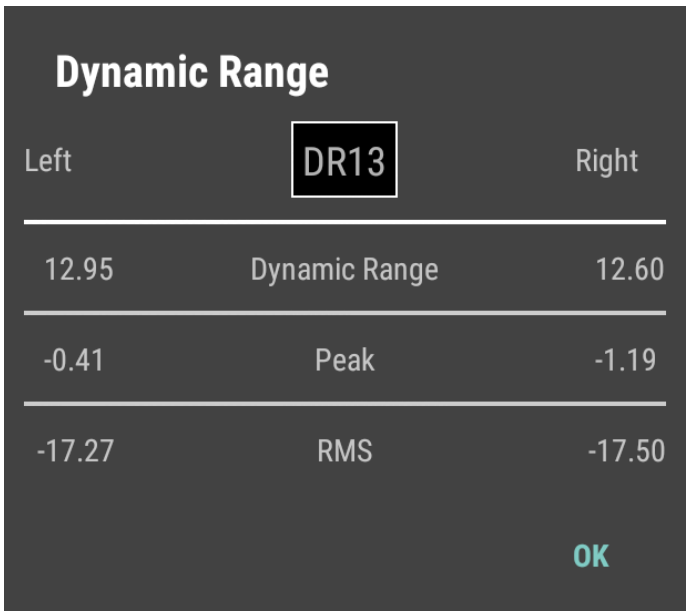
Channels

Controls the activity of channels of audio tracks.



Dynamic Range

The dynamic range expresses the degree of deviation of the signal from the average within the audio track.



Dynamic Range (DR)

The DR Value is the difference between the peak and the top 20 average RMS measurements.

Peak

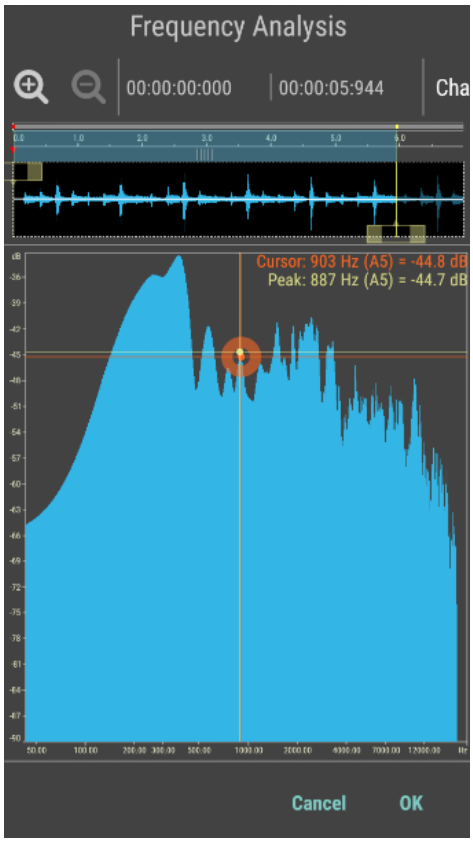
The peak value is the highest measured peak value in dBFS.

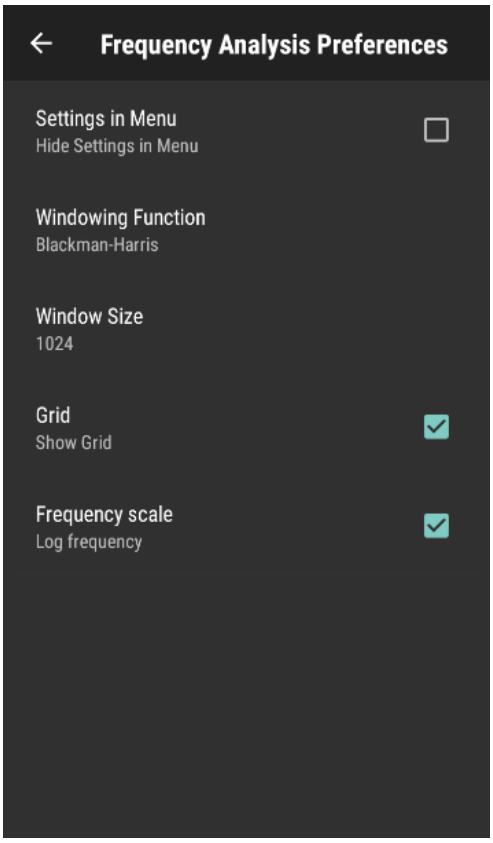
RMS

The RMS – an established loudness measurement standard – is determined by gathering approximately 10,000 pieces of loudness information within a time span of 3 seconds (dB/RMS).

Frequency Analysis

This takes the selected audio (which is a set of sound pressure values at points in time) and converts it to a graph of frequencies (the horizontal scale in Hz) against amplitudes (the vertical scale in dB).





Frequency Analysis Preferences

Windowing function

Determines the Fast Fourier transform shape. The following window functions are available: Barlett, Hamming, Hanning, Blackman, Blackman-Harris, Welch, Gaussian (a = 2.5), Gaussian (a = 3.5), Gaussian (a = 4.5).

Window Size

Specifies the Fast Fourier Transform size. Higher sizes report frequency data more accurately but they require longer processing times.

Grid

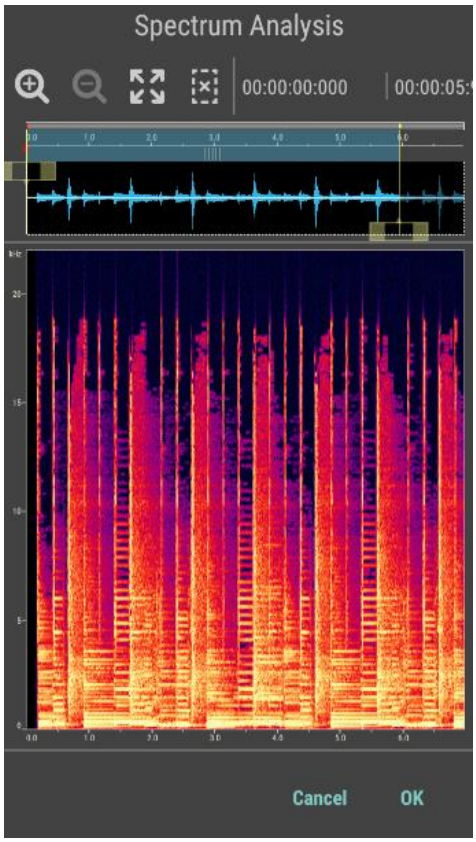
Check this to show gridlines in this frequency display.

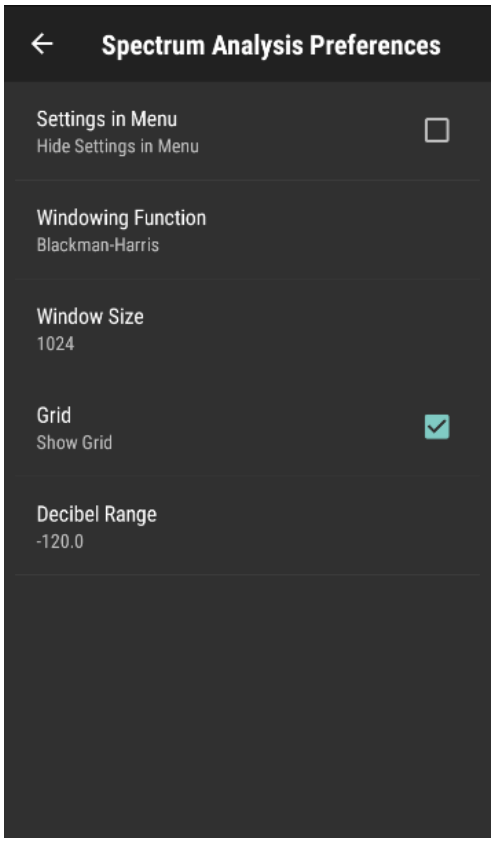
Frequency scale

Displays the frequency scale either logarithmically (reflecting human hearing) or linearly (providing more detail for upper frequencies).

Spectrum Analysis

The spectral display shows a waveform by its frequency components, where the x-axis (horizontal ruler) measures time and the y-axis (vertical ruler) measures frequency. This view lets you analyze audio data to see which frequencies are most prevalent. Brighter colors represent greater amplitude components. Colors range from dark blue (low-amplitude frequencies) to bright yellow (high-amplitude frequencies).





Spectrum Analysis Preferences

Windowing Function

Determines the Fast Fourier transform shape. The following window functions are available: Barlett, Hamming, Hanning, Blackman, Blackman-Harris, Welch, Gaussian (a = 2.5), Gaussian (a = 3.5), Gaussian (a = 4.5).

Window Size

Specifies the number of vertical bands used to draw frequencies. As you increase resolution, frequency accuracy increases, but time accuracy decreases.

Grid

Check this to show gridlines in this spectral display.

Decibel Range

Changes the amplitude range over which frequencies are displayed. Increasing the range intensifies colors, helping you see more detail in quieter audio. This value simply adjusts the spectral display; it does not change audio amplitude.

Menu Edit

Undo

Reverts the last editing operation.

Redo

Restores the previous editing operation that was just undone.

Cut

Removes the selected audio data and places these on the clipboard.

Copy

Copies the selected audio data to the clipboard.

Paste

Either inserts the clipboard contents at the position of the selection cursor, or replaces a selected area with the clipboard contents.

Delete

Deletes the selected audio data.

Trim

Deletes all audio but the selection.

Trim start

Deletes all audio data, excluding the area from the beginning to the selection cursor or to the right marker.

Trim end

Deletes all audio data, excluding the area from the selection cursor or the right marker to the end.

Silence Audio

Replaces the currently selected audio with absolute silence.

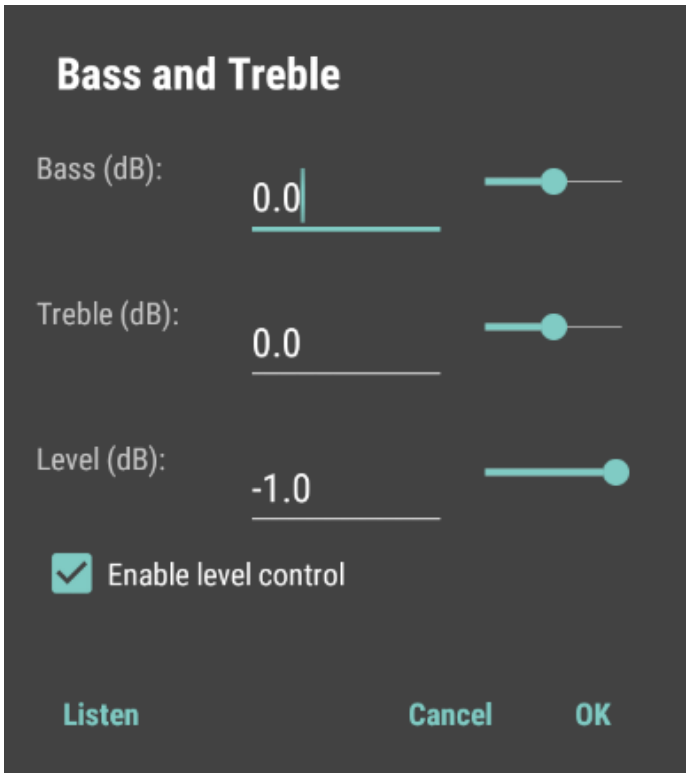
Swap Stereo Channels

Swaps the right and left channels.

Menu Effects

1. Bass and Treble

Bass and Treble effect increases (boosts) or decreases (cuts) the low frequencies (bass) or the high frequencies (treble) of the audio.



Bass (dB)

The amount of gain (amplification above 0 dB or attenuation below 0 dB) to bass (low) frequencies.

A positive value increases the bass, and a negative value reduces it.

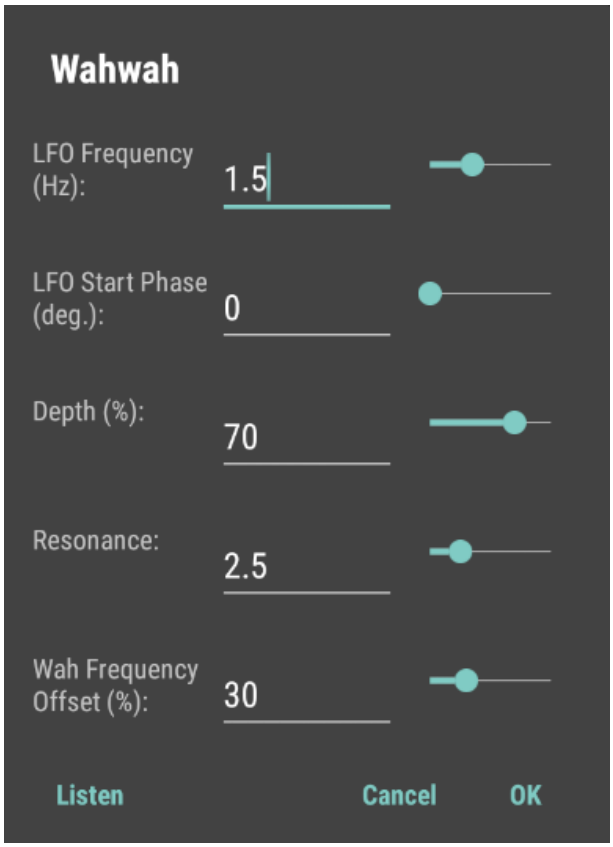
Treble (dB)

The amount of gain (amplification above 0 dB or attenuation below 0 dB) to treble (high) frequencies. A positive value increases the high frequencies; a negative value decreases the high frequencies.

Volume (dB)

Changes the overall level from -30 dB to 0 dB.

2. Wahwah



LFO Frequency

The speed at which the bandpass filter is swept back and forth.

LFO Start Phase

The start position of the LFO cycle. This determines whether the bandpass filter starts at its lowest, mid or highest frequencies.

Depth

Determines the range of frequencies that are swept through by the bandpass filter. Higher values will sweep the filter to higher frequencies and so give more variation to the sound quality over a complete LFO cycle. Lower values will give a more constant sound effect.

Resonance

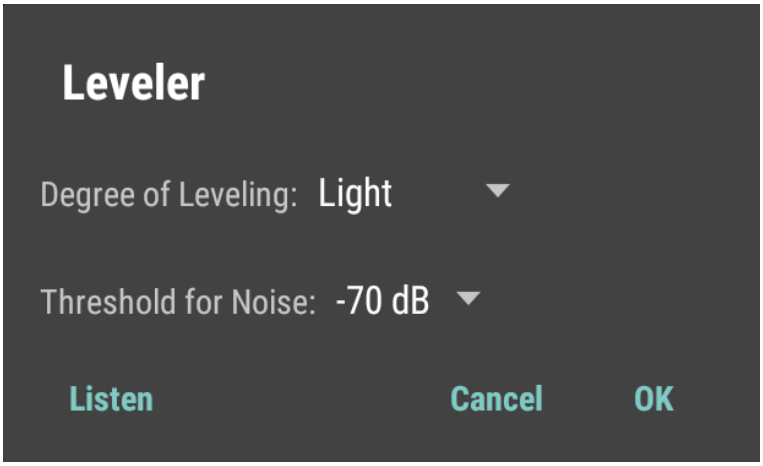
Determines the degree of resonance in the bandpass filter. Higher values give a more "peaky" effect.

Wah Frequency Offset

Determines the "base" frequency of the bandpass filter. Higher values will shift the filter's frequency range upwards. To achieve a Wah effect that's in the low frequencies the frequency offset needs to be set to a low value.

3. Leveller

Leveler is designed to reduce the difference between loudness and softness, which makes the audio easier to hear in noisy environments.



Degree of Leveling

There are 5 levels of alignment applied to the signal.

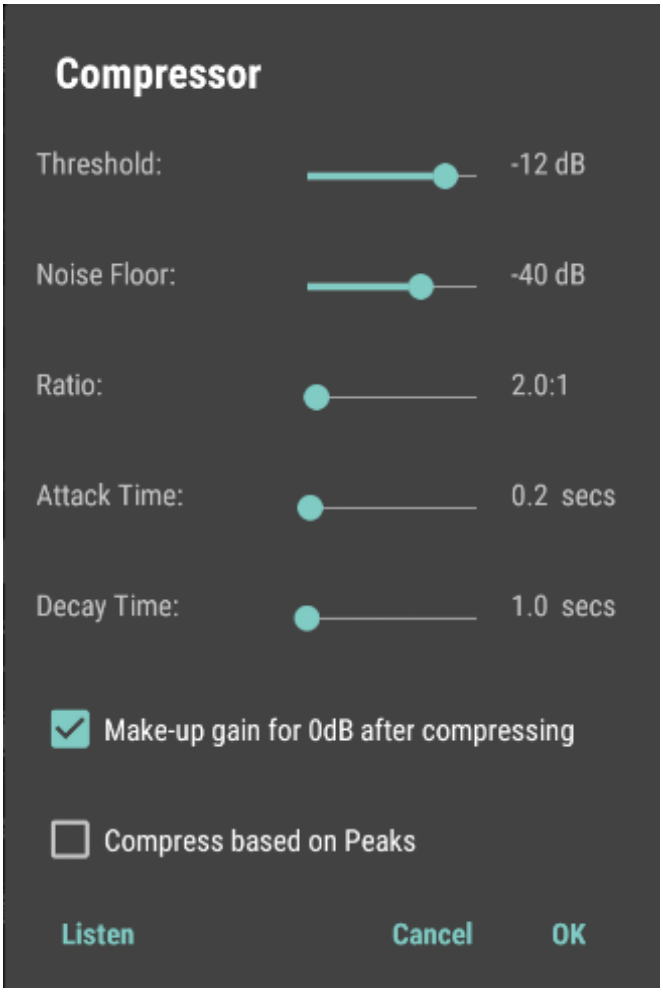
Threshold for Noise

The noise threshold parameters range from -20 dB to -80 dB at -5 dB intervals.

4. Invert

Invert flips the audio samples upside-down, reversing their polarity.

5. Compressor



Threshold

The input level at which compression begins.

Noise Floor

Adjusts amplitude after compression.

Ratio

The compression ratio applied to the sound when it passes the threshold level.

Attack Time

Determines how quickly compression starts after audio exceeds the Threshold setting.

Release Time

Determines how quickly compression stops when audio drops below the Threshold setting.

Make-up gain for 0 dB after compressing

Amplifies the resultant audio after compression to a peak level of 0 dB.

Compress based on Peaks

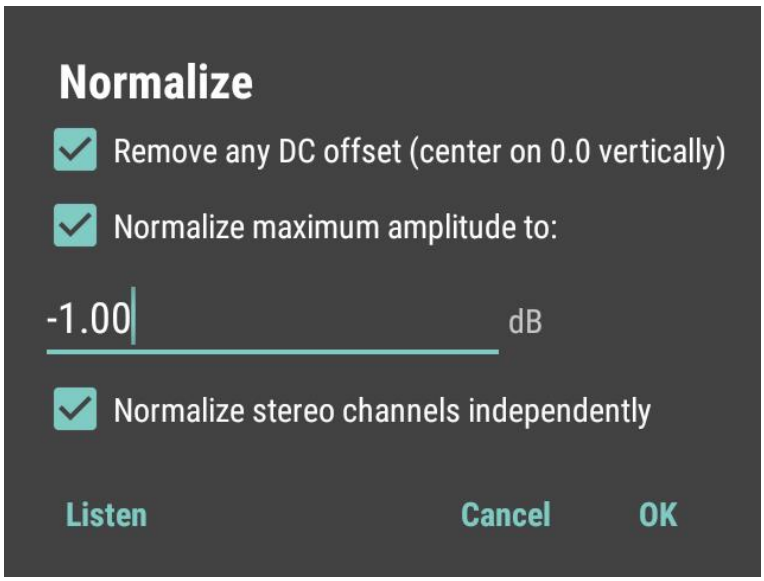
The audio with a peak level within the threshold range will be increased.

6. Mono to stereo

Converts a mono file to stereo, by duplicating the track.

7. Normalize

The effect is used to equalize the signal level. It provides gain in a given range, that is, the maximum amplitude is a fixed number.



Remove any DC offset (center on 0.0 vertically)

When this box is checked Normalize attempts to remove any DC offset by centering the waveform on the 0.0 amplitude level.

Normalize maximum amplitude

The initial default value for the maximum amplitude is -1 dB. Enter a more negative value to normalize to lower amplitudes.

Normalize stereo channels independently

When this box is unchecked, Normalize will work on the channels of a stereo track as a pair and change the level of both channels by the same amount.

When this box is checked, Normalize will adjust the amplitude separately for the left and right channels of a stereo track.

8. Fade Out

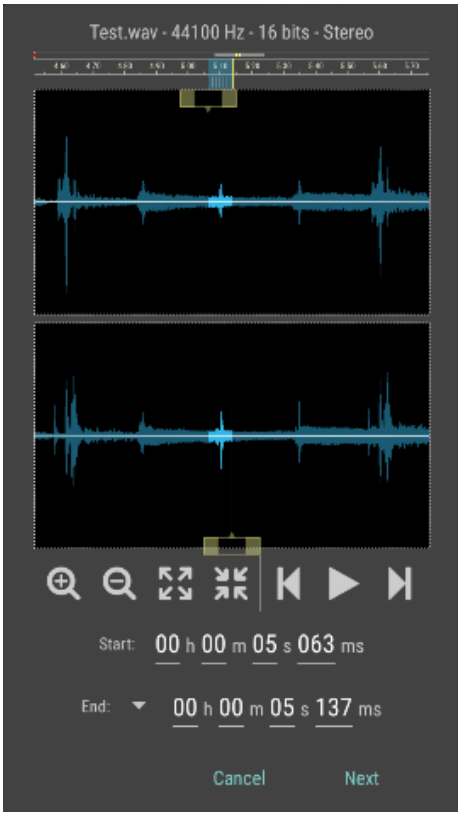
The effect smoothly reduces the signal amplitude to zero.

9. Fade In

The effect smoothly increases the amplitude of the signal.

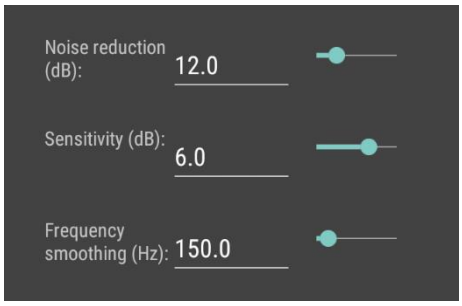
10. Noise reduction

Noise Reduction effect reduces background noise with a minimal reduction in signal quality. This effect can remove a combination of noise, including tape hiss, microphone background noise or any noise that is constant throughout a waveform.



Step 1.

Select a few seconds of just noise, in order for the application to know what to filter out.



Step 2.

Noise Reduction (dB)

Determines the level of noise reduction. Use the lowest value that reduces noise to an acceptable level. Higher values than necessary may make the noise even quieter, but will result in damage to the audio that remains.

Sensitivity (dB)

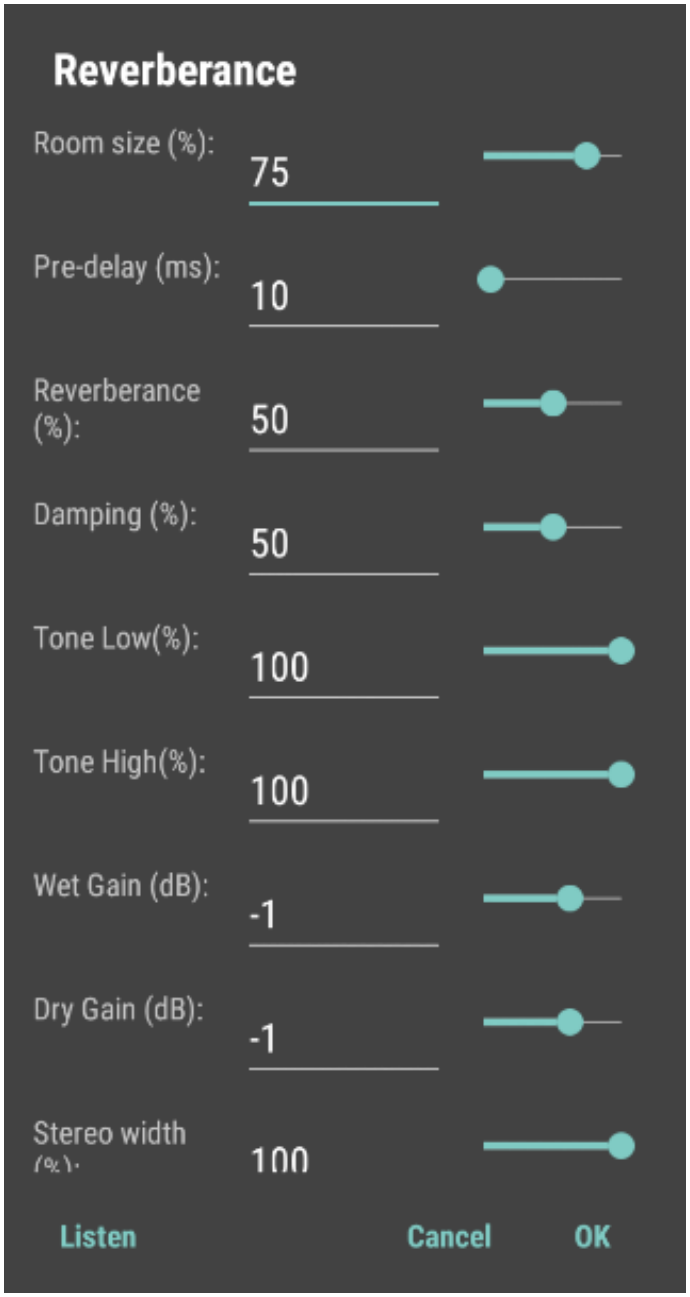
Controls how much of the audio will be considered as noise. Greater sensitivity means that more noise will be removed, possibly at the expense of removing some of the desired signal as well. Lower values may result in the appearance of artifacts in the noise-reduced audio. Set this control to the lowest value that achieves effective noise removal without the introduction of artifacts.

Frequency Smoothing (Hz)

Retains desirable audio in specified frequency bands between found artifacts. Lower settings remove more noise but may introduce audible processing.

11. Reverberance

Reverberation is created when any sound is heard in a closed space, resulting in reflections from the surfaces of the walls causing a large number of echoes, then the sound slowly fades due to the absorption of sound waves by walls and air. With the reverb effect, you can simulate different rooms.



Room Size (%)

Sets the size of the simulated room. A high value will simulate the reverberation effect of a large room and a low value will simulate the effect of a small room.

Pre-delay (ms):

Determines how many milliseconds the reverb takes to build to maximum amplitude.

Reverberance (%)

This determines how long reverberation lasts after the original sound is reflected.

Damping (%):

Characterizes the ability of the room material to reflect sound.

Tone Low (%)

Setting this control below 100% reduces the low frequency components of the reverberation, creating a less "boomy" effect.

Tone High (%)

Setting this control below 100% reduces the high frequency components of the reverberation, creating a less "bright" effect.

Wet Gain (dB)

Applies volume adjustment to the reverberation component in the mix. Increasing this value relative to the "Dry Gain" increases the strength of the reverb.

Dry Gain (dB)

Applies volume adjustment to the original audio in the mix. Increasing this value relative to the "Wet Gain" reduces the strength of the reverb.

Stereo Width (%)

Sets the apparent "width" of the Reverb effect for stereo tracks only. Increasing this value applies more variation between left and right channels, creating a more "spacious" effect. When set at zero, the effect is applied independently to left and right channels.

Wet Only

When this control is checked, only the wet signal (added reverberation) will be in the resulting output, and the original audio will be removed.

Presets:

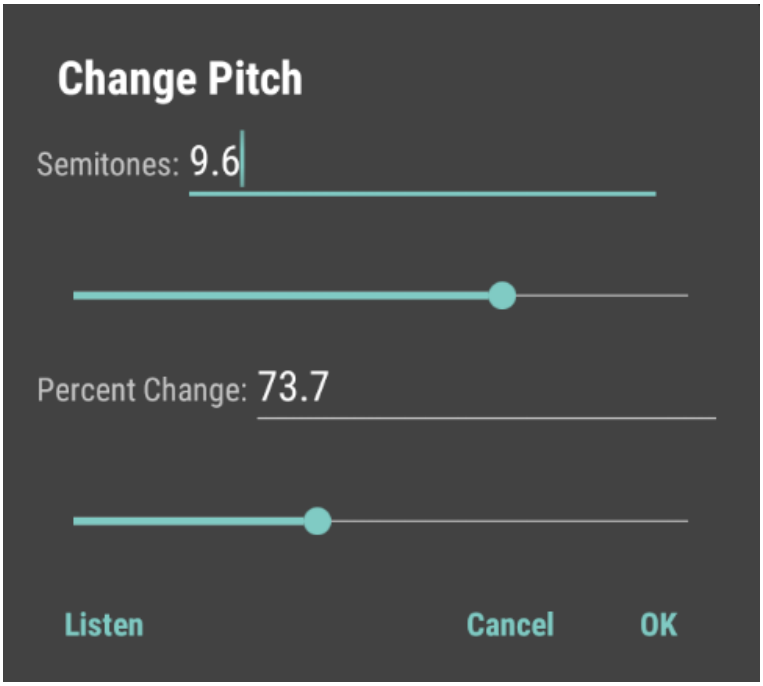
There are 9 presets of reverb are available for use.

12. Reverse

Reverse reverses the selected audio, so that the end of the audio will be heard first and the beginning last.

13. Change Pitch

Used to change the pitch without changing the tempo (speed).



Semitones:

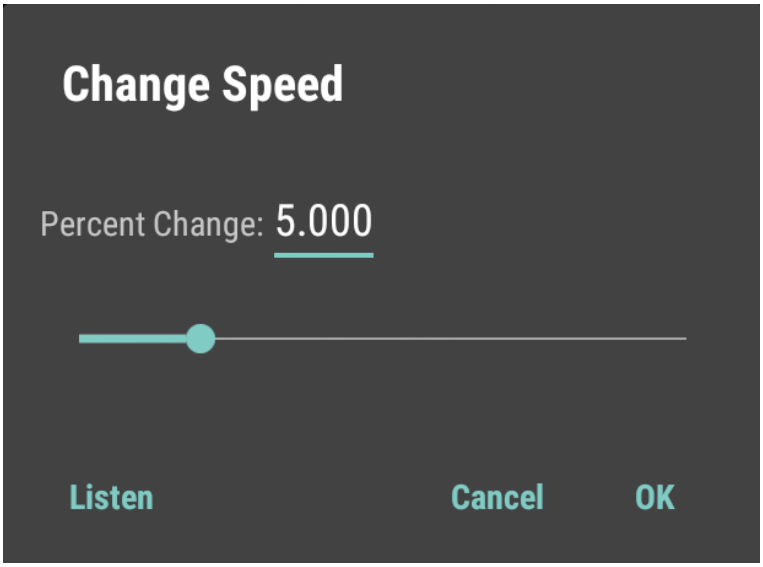
Transposes pitch in semitone increments, which equal musical half notes.

Percent Change

Adjusts pitch in fractions of semitones.

14. Change Speed

Changes the speed of the audio, affecting its tempo, pitch and frequency. When the speed decreases, all frequencies become lower. With increasing speed, all frequencies become higher.

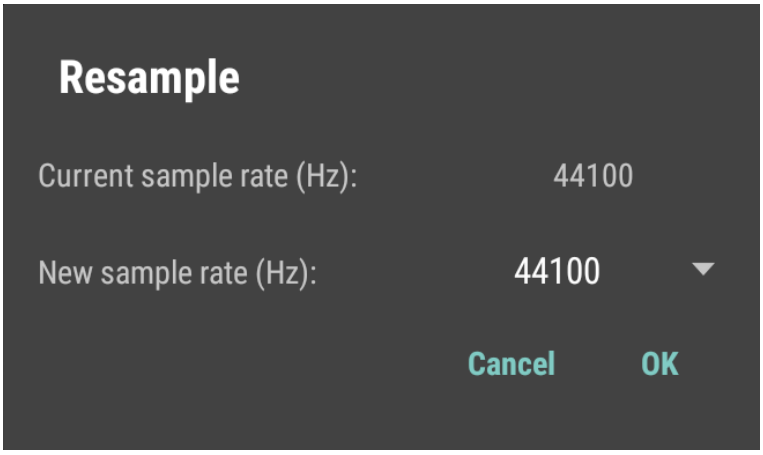


Percent Change

Change the speed of the audio in percent.

Valid values are from -99% to 400%.

15. Resample



Current sample rate (Hz)

Displays the current sample rate.

New sample rate (Hz)

Enter the desired sample rate

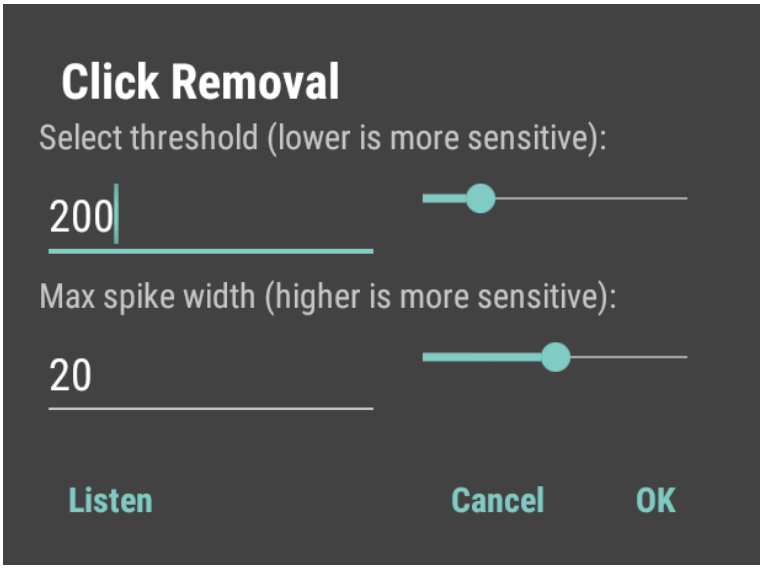
Available values range from 8000 Hz to 2 822 400 Hz.

16. Stereo To Mono

Converts stereo track into the mono track, combining left and right channels equally by averaging the volume of both channels.

17. Click Removal

The effect is designed to remove individual clicks on audio tracks.



Select Threshold

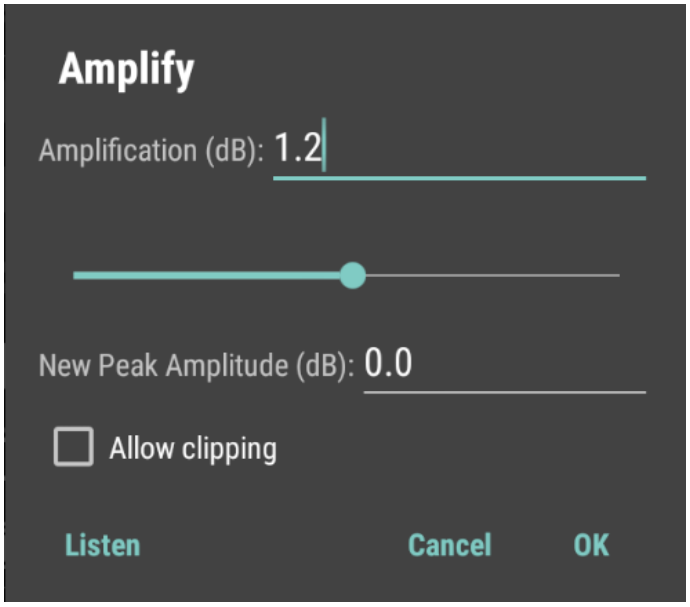
Entering a lower value or moving the slider left will detect softer clicks. Setting this too low may cause false click detection and damage the audio. Setting it too high may leave audible clicks that you'd rather it removed.

Max Spike Width

Enter a value, or move the slider, to set the length of the spike that is considered to be a click. Setting this too high may cause false click detection and damage the audio. Setting it too low may leave audible clicks that you'd rather it removed.

18. Amplify

The effect is used to change the volume of the audio track.



Amplification (dB)

Positive values make the sound louder, negative values make it quieter.

New Peak Amplitude (dB)

Type in the value you would like for the new peak amplitude of your track. As you type, the Amplification input box will be updated.

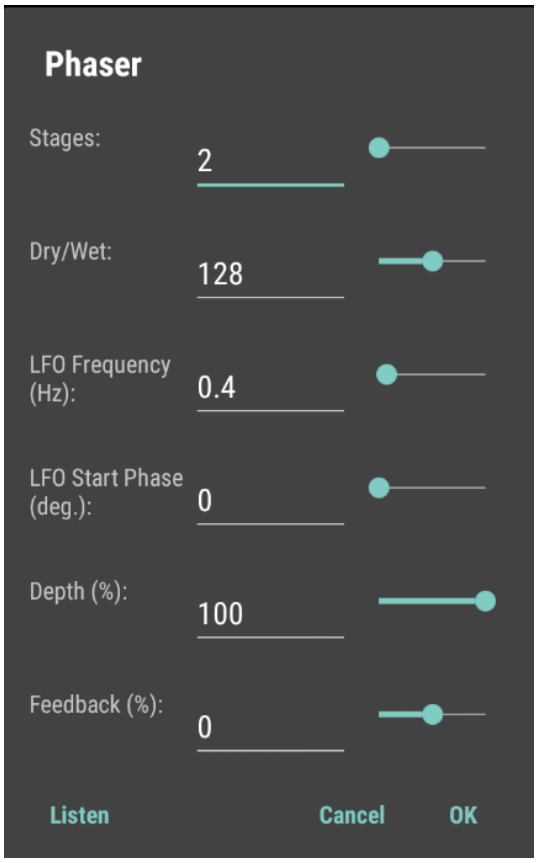
Allow Clipping

If this box is not checked, and you attempt to enter an Amplification value that will result in a New Peak Amplitude of greater than 0 dB, the "OK" or "Listen" button will become inactive. This will prevent you from applying too much amplification. If this

box is checked you can apply as much amplification as you want, possibly creating a distorted sound.

19. Phaser

The effect works by combining phase shifts with the original signal. The movement of the phase shifts is performed using a Low Frequency Oscillator (LFO).



Stages

Specifies the number of phase-shifting filters. A higher setting produces denser phasing effects.

Dry/Wet

When set to 0, only the "Dry" (unprocessed) signal is produced. When set to 255 (maximum), only the delayed signals are produced. Because the effect's sound results from phase interaction, the effect sounds strongest when the Dry/Wet mix is set at the default halfway position (128).

LFO Frequency (Hz)

This is a low frequency oscillator (LFO) control that adjusts the rate at which the effect sweeps up and down across the frequency range.

LFO Start Phase (deg.)

The start position of the low frequency oscillator is adjustable between 0 and 360 degrees.

Depth

The depth control governs how high the filter frequencies sweep.

Feedback (%)

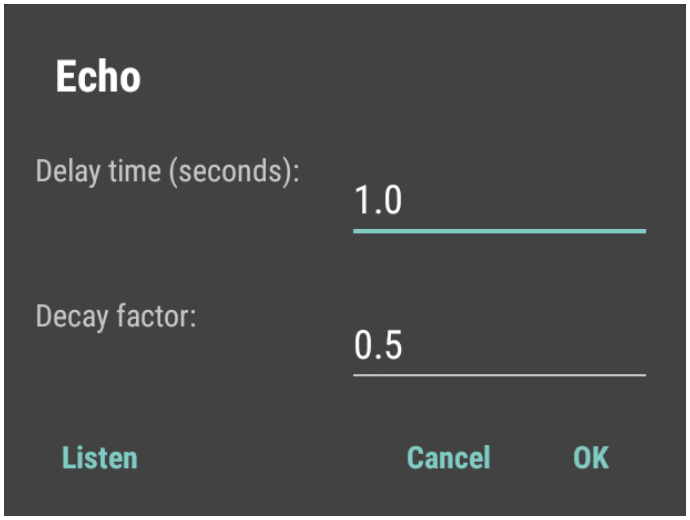
By adjusting the feedback control, the processed signal can be passed back through the effect, creating a more pronounced effect. At high settings, the effect will become highly resonant, similar to a Wahwah effect. At negative settings an inverted signal is fed back into the effect, creating another variant of the effect.

20. Equalization

The effect allows you to increase the volume of some frequencies and reduce other.

21. Echo

This effect repeats the audio you have selected again and again. The delay time between each repeat is fixed, with no pause in between each repeat.



Delay time (seconds)

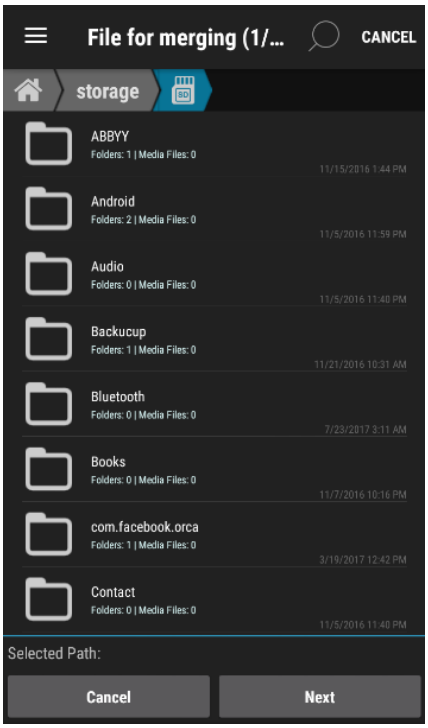
Specifies the number of seconds between each echo.

Decay factor

Usually a number between 0 and 1. A value of 0 means no echo, and a value of 1 means that each echo is exactly as loud as the original. A value of 0.5 reduces the amplitude or loudness of each echo by half each time, so the audio dies out quite slowly.

Merging

This function allows you to insert the selected area from another file into the current file in the specified position.

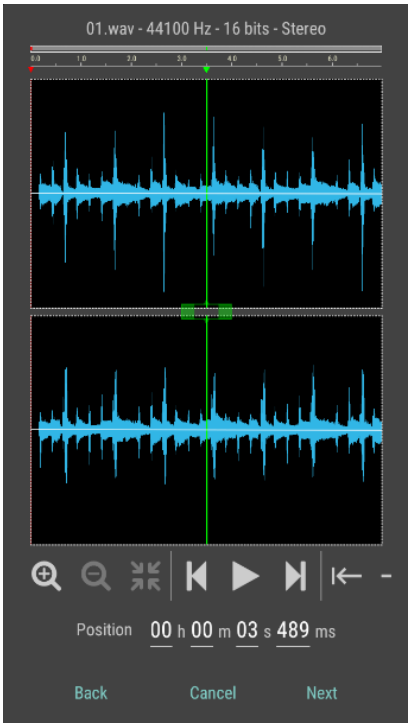


Step 1

Select the file for merging in the file manager.

Step 2

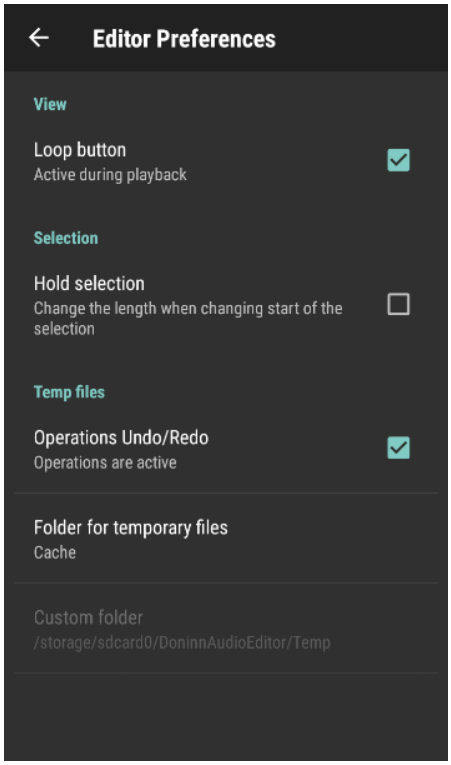
Select the desired range using the right and left markers.



Step 3

Select the position of the selection cursor to insert the selected range in the current edited file.

Editor Settings



View

Loop button

Controls the visibility of the button during playback in the editor window.

Selection

Hold selection

Controls the fixation of the left selection marker.

Temp files

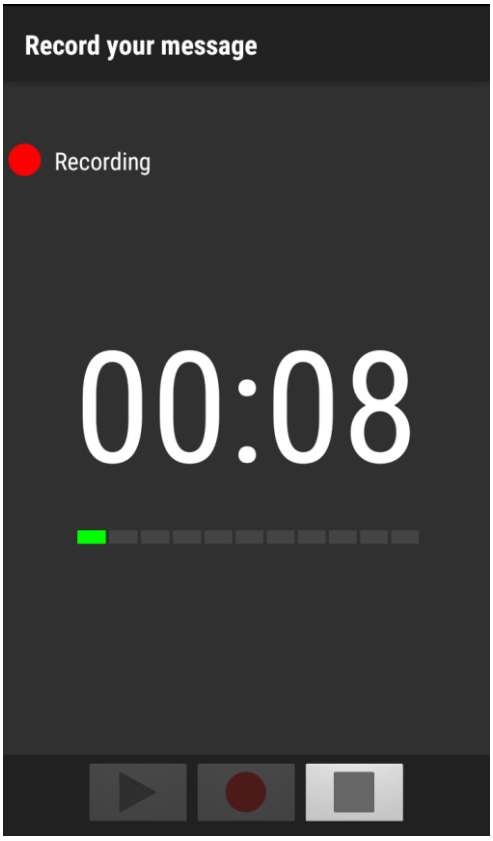
Operations Undo/Redo

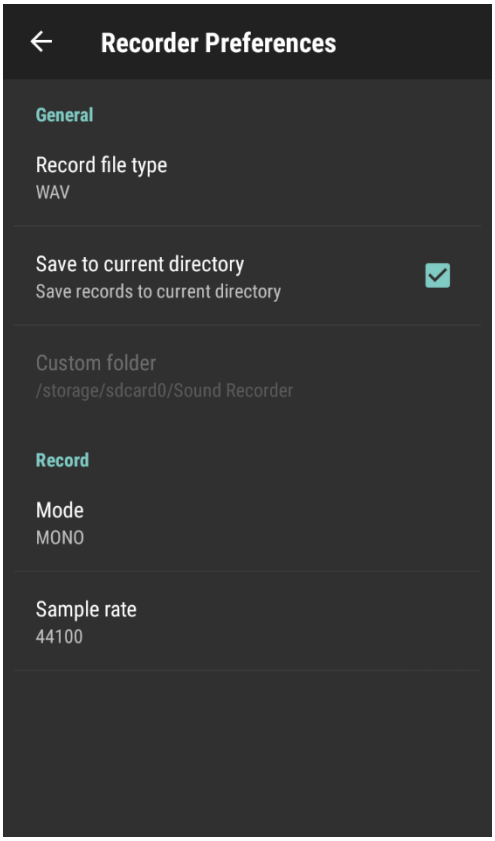
Controls the availability of the Undo and Redo operations. Disabling these commands saves space on the device.

Folder for temporary files

This setting allows you to specify the location for temporary files.

RECORDER





Recorder Preferences

General

Record file type

Allows you to select the recording file format. There are two formats available: wav and mp3.

Save to current directory

Saves the recorded files to the current folder. When the state is inactive, the entry is in the user folder.

Custom folder

Allows you to specify the location for saving the recorded files.

Record

Mode

Allows you to specify the type of recorded files: Mono / Stereo

Sample rate

Allows you to specify the sampling rate of files.

Quality (for mp3 format)

Allows you to select the bitrate in kbps for the file.

Speed (for mp3 format)

Allows you to select the speed. There are two modes available: Fast, Standard.

FILE MANAGER

The main features

- Cut, copy, paste, rename, delete media files and folders in the internal memory and on the external SD Card.
- Asynchronous operation of the file manager.
- Multiple selection of files and folders.
- Search and sort audio files and folders by name, size and date.
- Show hidden files and folders, as well as the protected folders.
- Caching metadata for fast loading and displaying the contents of folders.
- Ability to install the folder when the application starts (Home folder).
- Convenient and informative audio player with the ability to move to the currently playing file in other folders.

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