1. What this document covers.

If a church is relying on music files (eg, WAV or MP3 files) then there will be occasions when it is helpfully to know how to manipulate those files to obtain a more appropriate version of the music. Typical requirements may be:

- Convert MP3 files to WAV file
- Changing the length of a song (eg drop out a verse, reduce the introduction)
- Change the tempo and pitch
- Add 'fade in' and 'fade out'
- Recording your own music or audio (eg sermons)

All these aspects can be easily handle by a freeware program called 'Audacity', and this document describes how it is done.

2. What are 'wav' and 'mp3' sound files?

(a) What are sound files?

There are many file formats for storing sound on a computer. These sound files are normally music, but can be any audible sound. The two most common formats for storing sound are 'wav' (wave) files and 'mp3'

These two file types serve different purposes as they either optimize the 'sound quality' or 'minimize file size'. The trade-off between quality and file size is necessary as it takes a lot of storage space to store a high quality file, and for the internet is necessary to have as small as file size as possible, because large files take time to load. (A one minute piece of music in a 'wav' file may take up to 10 minutes to download).

So the common problem is how to make the most of the strong points of each file type, and how convert from one to another? Most programs that do this conversion also have other features that enable you to 'fix' up the sound file and make it better, eg:

- Delete quiet times at the beginning or the end (or even in through the file)
- 'Normalize' the file (turn the volume up to a 'standard' so all tracks are about the same volume)
- Do 'fade ins' and 'fade outs'
- Record sounds
- Play sounds
- Change the compression density
- Converting to different file types

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(b) The program we shall use.

There are many excellent commercial programs that will perform these functions, some even including video processing.

This course is based upon the excellent freeware software called 'Audacity'. This program runs on Windows, Linux and Mac operating systems.

This software is downloaded from: https://www.audacityteam.org/download/

Installation is straightforward.

(These notes are based on V2, release March 2012)



(c) What is a 'wav' file?

The best way to visualize a 'wav' (wave) file is that it is a numerical representation of the actual sound wave.

Normally there are 44100 samples per second, and for each sample a number between -32768 to +32768 (it takes 2 bytes to store each number, hence one second requires at least 88200 bytes).

So a wave file gives the best recording quality for signals up to about 15,000cps, sufficient for most of us with normal hearing.

Sound 'wav' files do come in a number of varieties; all aimed at reducing file size and hence sound quality somewhat. But file sizes are still too large for the internet. Some of the options are:

- mono or stereo (obviously stereo requires twice as much space as mono)
- the number range can be stores in one byte (-128 to +128) instead of two bytes. One byte is often OK for speech. Four byte numbers can also be used
- The number of samples per second can be altered. The normal is 44100.

It is best to regard the 'wav' files as your source files, and store your sounds at the best quality, ie stereo at 44100 samples a second and 2 or 4 byte. If you keep this as your

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source, it is always easy to produce lower quality files from it by using programs like 'Audacity'. However, you cannot create a high quality file from a low quality file.

(d) What is a 'mp3' file?

In the 1990's there was a developing need to compress sound files for the internet and portable digital storage devices.

Around about 1995 the MP3 compression technique began to become popular.

Most listeners accept a MP3 bitrate of 128 kilobits per second as near enough to CD quality, and this gives a compression ratio of about 1/11th of the original wave file. What the 128kb/sec actually means does not really matter, except to remember that it indicates very good quality. Bit rates of less than 128 indicate a lowering in quality (and also in file size).

The calculation routine to do this compression is not public domain, and therefore is not usually free. There are some programs available that do this conversion, and these are called 'encoders'. Audacity uses one of the few free encoders that are available, namely the LAME MP3 encoder which allows Audacity to export MP3 files. Not all encoders produce the same quality of output file. My experience is that the LAME MP3 encoder does a good job even at low compression rates.

On the internet many sites offer their music files at different compression rates. A 128kb/sec compression rate is regarded as 'High fidelity', whereas a compression rate 64kb/sec is regarded as medium fidelity. Rates lower than that are regarded as 'low fidelity'.

When preparing files for MP3 compression, it is always adviseable to check the result to make sure it is acceptable. Even at the same compression rate, some music sounds good, while others sounds distorted. In general, I have found the Audacity/LAME combination produces a good quality result allowing for the amount of compression (eg 48kb/sec is a compression of 1/30 of original wave file size).

3. Processing the input wave file.

(a) The source file.

Always start with the best quality sound that is available. Like the saying, "you can't make a silk purse out of a sow's ear" likewise no program is able to dramatically improve sound quality, so start with the best you have. This would normally be one of the following:

- A 'wave' file
- Direct recording into your sound card (more on this later)
- A high quality MP3 file.

Source files should be treated with the greatest respect like a historic document. Once altered and over written, source files cannot be easily restored. So any changes made to the source file should be limited to those changes that you want to permanently make. This would include such things that do not affect sound quality, eg:

- Removal of silent periods at the start and end
- Normalise the file (this just adjusts the overall volume)

Even when removing such things as 'background rumble' or 'hiss', it safer to make either a new version and retain the original file.

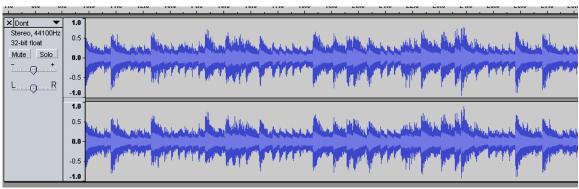
(b) Opening a file

The usual Windows style 'file open' is used to open the input file, eg

File Edit View Project Generate Effect Analyze Help New Ctrl+N Ctrl+N Ctrl+N Image: Ctrl+D	(🚗) Don	t						
Open Ctrl+O	<u>F</u> ile <u>E</u>	dit	⊻iew	<u>P</u> roject	<u>G</u> enerate	Effe <u>c</u> t	Analyze	<u>H</u> elp
	<u>N</u> ew		Ctrl+N		10			
Close Ctrl+W	<u>O</u> pen			Ctrl+O		0.01		
	<u>C</u> lose		Ctrl+W		1-	10		
Save Project Ctrl+S	Save Project			Ctrl+S				

Audacity will open a number of file formats including 'wav' and 'mp3'.

When the file is open in Audacity, the full wave form is squeezed into the window display, eg:



If necessary, this display of the wave can be expanded through the use of the buttons:

The first 2 provide zoom out and zoom in. The 3 rd button fills any highlighted areas of the wave in the current window width. The 4th button brings it back to the initial state of the whole wave file being fitted into the screen area.

This zooming can be to the point where individual samples can be seen:

which allows for editing of the indivual values (for removing clicks etc).

The two tracks show the stereo channels.

(c) Selecting part of the wave.

By clicking on the wave, a vertical line is positioned at that point on the score.

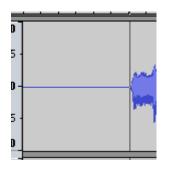
The exact position is shown in the bottom left of the screen:

		•			
Click and drag to move left selection boundary.					
Project rate:	44100	Cursor: 0:00.3	19407 min:sec	[Snap-To Off]	



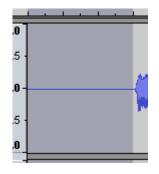
This vertical line become the point of focus of the zoom commands.

So by zooming out we are able to see more clearly that point of the wave, eg if I click near the beginning of the curve I am able to see clearly the quiet start area, eg:



Hold down the left button and drag across the area or wave, and the selected area will become highlighted:

(Note: Ctrl/A will highlight the whole wave).



To remove highlighted area, click elsewhere on the wave.

(d) Simple editing of the selected area.

The area that is highlighted can be cut and pasted, or worked on by the more specialised functions in the 'Effect' menu (more this later).

The cut and paste functions are in the tool bar:

-//0//- 00	1 <u>0</u> 12	¢,	-1001-	01H0

the 1st one being 'cut' which removes the highlighted area and moves the wave left to replace the area that is removed.

We shall highlight the 'quiet' area at the start of the wave file.

Simply by pressing the cut button the quiet piece is removed, and the wave file moved to the left.

(Note: if cutting and pasting whole verses, it is best to start and end on just the start of the rise of the verse. In this way, you don't hear click caused by the pasting process, which can happen if you cut and paste during a quiet time).

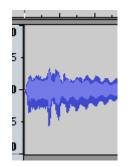
The tooltips on the other buttons are self explanatry and are variations on the cut, copy

and paste commands.

(e) 'Effects' editing of the selected area.

The 'effect' menu provides a large number of options.





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Its is recommended that the input file be normalised. Normalisation is a process that looks at the loudest part of the wave, and adjusts that to 97% of the maximum, and the rest of the file proportionally.

To put it more simply, it turns everything up to the maximum volume. This is very useful for preparing tracks for your CD or iPod, as all tracks are about the same volume, and you don't have to keep adjusting the volume control.

(i) Normalise

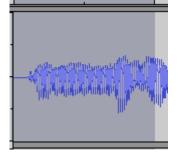
To normalise the file, highlight the whole wave (ctrl/A) and then select 'normalise' in the effects menu. A dialogue box will appear:

Normalize	×			
Normalize by Dominic Mazzoni				
Remove any DC offset (center on 0 vertically)				
☑ Normalize maximum amplitude to -3 dB				
Preview Cancel OK				

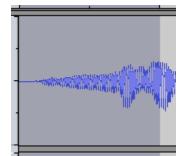
Press 'OK' and you will proably see the whole wave expand vertically.

(ii) 'fade in' & 'fade out'

Select the area for the 'fade in' (or 'fade out')



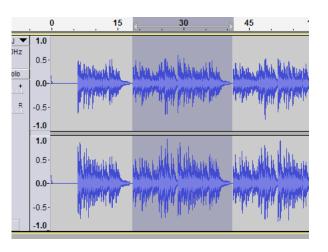
Select 'fade in' from the 'Effects' menu. You can see the fade in in the wave shape



(iii) cut/Insert out a verse

Look for start and end of each verse (ie, where the music goes quieter) and then select from the start of one verse to the start of the next verse).

The do a cut (Ctrl/C) to remove the verse, a copy (Ctrl/X) the verse, and then paste it into some other area of the song.



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4. Saving the output file.

The 'file' menu provides options to save the current wave as a certain file type. Within each type is the option to just save part of the current file or the whole file.

(a) Saving as a 'wav' file.

Caution was previously expressed about over writing the original source files. That caution is restated here.

Under the 'file' menu there is a an 'Export' file option. This will bring up a dialogue box very similar to the normal 'save as' function in most programs.

•			4
File name:	IComeWithJoy-StBotolph-SPiano-CAM	•	Save
Save as type:	MP3 Files	•	Cancel
	Other uncompressed files AIFF (Apple) signed 16 bit PCM WAV (Microsoft) signed 16 bit PCM		Options
	GSM 6 10 WAV (mebile)		

There is 'Save as type' field on this dialogue box which gives the option of a variety of files types.

(b) Saving as a MP3 file.

If you are going to save as an MP3 file, it is necessary to set the compression bit rate. This is done in the ''Options' button on the 'Export' screen.

(i) install 'encoder' – 1st time only

In the 'Edit/Preference' menu there is an option to setup a Library.

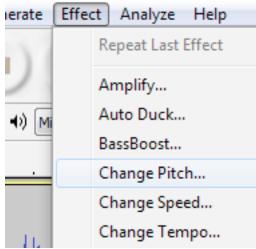
Preferences: Libraries				
Devices Playback Recording Quality Interface	MP3 Export Library MP3 Library Version: LAME 3.99.3 MP3 Library: Locate LAME MP3 Library: Download			
···· Tracks ···· Import / Export ···· Extended Import ···· Projects ···· Libraries	FFmpeg Import/Export Library FFmpeg Library Version: FFmpeg library not found FFmpeg Library: Locate			
Spectrograms Directories	FFmpeg Library: Download			

Click on the 'Download' button to go to the Audacity website to obtain the encoder (called LAME)

For Mac users, a similar process is required, but the encoder file has a different name. For some reason the Mac version of the file does not generate files of the same quality, especially at high compression rates. You are probably better to save your files as WAV files.

5. Changing the musical qualities of a file.

Audacity V2 (released March, 2012) now does have quality functions to allow you to change pitch, tempo. These are under the 'Effects Menu:



Each of these three functions are fairly simple to use.

When using these functions, do no over-write for source recording, because once they are save to you original file they cannot be reversed. Always save to a new file, and keep your source unchanged.