Making a Segment for the Broadcast

Introduction

The following is a description on how to make the best audio file for your club's news segment for optimum quality and hence best listening pleasure.
Although there are numerous audio production/editing programs available today, we strongly recommend Audacity as the program-of-choice because of its ease of use, and that it is free to use in the public domain. It also is undergoing continuous development and new features are being added continually.

Equipment needed

1) A PC running either MS Windows, or Linux.
2) The Audacity program, and the Lame_enc.dll file.
3) A soundcard. Most PCs have an onboard soundcard and these are most suitable.
4) A microphone or inexpensive headset, compatible with the soundcard.

Setup

Install the Audacity program and ensure that the Lame_enc.dll file, which is used to generate the MP3 file, be installed into the same folder which holds the Audacity program file. Please note that the Lame_enc.dll file is not part of Audacity and must be downloaded and installed separately.
The following parameters should be set, (using Audacity Ver 1.2.6):
Edit, Preferences, Audio I/O, Channels = 1(Mono)
Edit, Preferences, Quality, Default Sample Rate = 44100 Hz,
Edit, Preferences, Quality, Default Sample Format = 32 bit float
Edit, Preferences, File Formats, Uncompressed Export Format = WAV (Microsoft 32 bit float)
Edit, Preferences, File Formats, MP3 Export Setup, Bit Rate = 64, and
Project Rate (bottom LH corner of main panel) = 44100 Hz. This is most important.

Check operation

Press the red ***Record*** button and note a new ***Recording Track*** is displayed. Speak into the microphone and note that the recording meter indicates a suitable level, and the recorded waveform is displayed in the ***Recording Track***.
If no level is generated check that the ***Mic*** is selected from the ***Volume Control***, ***Options***, ***Properties***, ***Recordings***, within the operating system.
Press the (square) ***Stop*** button to stop the test recording.
Click on the ***X*** in the ***Recording Track*** to delete the test recording when completed.

Presentation

It is preferred that a script be produced prior to the recording process. This script will contain all the information to be transmitted with a minimum number of words rather than an ad-hoc live presentation which will usually incur a number of “umms and errs” as the presenter tries to remember what should be said.
The script can easily be written in MS Wordpad or MS Notepad, and edited as required.
The script should be read aloud prior to the recording to familiarize the presenter with the

flow of the script. Inflexions can be included to add emphasis where required. Remember to add your name and callsign (if relevant) and your club's name, either at the beginning, or the end, of your presentation.

Adjusting levels/voice quality

Particular attention should be paid to getting the audio level, quality, and signal-to-noise-ratio "just right".

The audio level should ***occasionally*** peak at 0, with the average level approximately –3 to –6 as read on the level meter. The level can be adjusted by the ***Microphone Gain*** slider control, and/or speaking closer to the microphone itself. Be aware that speaking too close to the microphone creates a ***"sp'tt"*** which can be most annoying to the listener and should be avoided. A simple ***"sp'tt"*** filter can be made by wrapping the microphone in a piece of thick material, eg an old sock.

The use of a desk microphone is ok, providing the presenter is not too far away from it. It is easy for reflections from nearby cupboards and walls to encroach, making the presentation sound ***"echo-ie".*** This should be avoided. Perhaps the best method to use is to speak close to, and across, the microphone face, thus reducing the ***"sp'tt"*** effect.

A practice run should be made of a few sentences to adjust the recording levels. View the result and note the waveform which should be ***"rounded".*** If the recording level is too high the waveform will be ***"flat-topped"*** and therefore compressed, and if to low, will appear ***"thin"*** and ***"spikey".***

The signal-to-noise-ratio (S/N ratio) is a measure of how much background noise is presented, and is measured reasonably accurately by noting the audio level when not speaking. This should fall to less than –35 during silence periods. If this level cannot be achieved then either the microphone level is too high introducing background noise, or there is "earth loop hum" introduced. A common cure for this is to ground the PC case.

Recording your segment

When the recording levels and quality are best adjusted, commence the recording. It is best practice to leave a short silent piece at the start which can easily be edited out later. If a mistake is made it is not necessary to re-do the whole recording, either continue on or revert to the start of the last sentence. The flexible editing commands are very easy to use and with a little practice most mistakes can easily be removed.

The file can be heard by returning to the start with the ***Rewind "<<"*** button and then using the ***Play*** button ***">".*** Errors, silence, and unwanted audio can then be edited out with the ***Edit, Delete*** command. Standard ***Cut and Paste*** commands also apply, but NOT from the mouse right-click function. Use the ***Edit*** command.

When completed use the ***File, Export as WAV*** command to save the segment. It is preferable to use the .WAV format rather than .MP3 format at this stage as it is easier to manipulate the file should further editing be required.

Saving the file

When satisfied that the file is complete, save as an MP3 file with ***File, Export as MP3*** in the simple format ***xxxxddmm.mp3*** where ***xxxx*** is the abbreviation of the club, and ***ddmm*** is the day/month of the proposed broadcast, (not the date prepared).

It is recommended that a ***My Documents*, *Uploads*** folder be made to store the current file, and a ***My Documents*, *Uploads*, *Archives*** folder be used to store all historical files. Once the MP3 file has been sent the WAV file can be deleted. The document file should also be saved for future reference, and can be used as a basis for the following broadcasts.

Testing the file
The MP3 file should be tested by clearing all waveforms on the screen by pressing ***"X"*** adjacent to each waveform first, and pressing ***Project, Import Audio, xxxxddmm.mp3*** from the appropriate file-save area. The file waveform can then be viewed in the ***Track*** area, and the audio quality heard in the headphones with the ***Play*** command.

Advanced editing features
For the more advanced users there are a number of useful editing features available.
Files can be imported via ***Project, Import Audio*** into new ***Track*** areas.
Audio portions can be highlighted, and then ***Cut-and-Pasted*** to suit. Be aware of the ***Sampling Rate*** of the file to be pasted, and if it is not 44100 Hz then it needs to be modified.
Note: ***Edit, Preferences, File Formats, MP3 Export Setup, Bit Rate*** = 64. This solves the problem of the 44.100/22.050 problem experienced earlier.

Sample Rate
The sample rate chosen for the broadcast is **44.1KHz.** This is the best compromise between quality and file-size. If all files received for compiling to the broadcast use this format then considerable editing time will be saved.

It has become apparent that there may be a fault within the program whereby the mp3 file produced presents itself as **22.050 KHz** instead of the required **44.100.** This can be overcome by highlighting the whole waveform, pressing the ***Down-Arrow*** adjacent to the waveform, and ***Set Rate, 44100 Hz***. This will then change the sample rate to **44.100KHz,** but in doing so will double the audio frequencies. This then can easily be corrected by highlighting the complete section again, and ***Effect, Change Speed, Percent Change = -50%*** (a minus percentage value).

Improving the Signal-to-Noise Ratio
Audacity has a very powerful noise-reduction facility built-in. To use this facility carefully highlight a small section of blank area, (the waveform may be expanded with the "+" button) and then ***Effect, Noise Removal, Get Noise Profile***.
Return to the start and highlight the whole file, then ***Effect, Noise Removal.*** Adjust the Slider to the first notch, that is just above the "s" in "Less", then ***Remove Noise***.

Have audio fun!

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