Multimedia Systems

Practical Class
13/04/2020

1. (if you still haven't) Download and Install Audacity

Go to https://www.audacityteam.org/, download and install Audacity, which we'll be using for manipulating audio files. If you have another audio editing tool (e.g Wave Editor, Audition, Sound Forge) that you prefer, you don't need to do this step.

Note: Sonic Visualiser is more suited for the analysis of music signals, and Audacity is better for manipulation (e.g. cutting, fade out, conversion) of general audio files.

2. Audio File Compression

Compressed audio file formats such as MP3 can be very useful for casual listening when file storage space is a concern. It is important to understand when these compression schemes can be useful and when it is better to use the original uncompressed file format such as WAV and AIF.

Included with this document are 2 WAV files. Start by listening to the "Uncompressed.wav" file, which is a raw PCM encoded WAV file straight from the audio CD. Now listen to the "Compressed.mp3" file. This is the same as the first file but has been compressed using MP3 compression at 128 kbps. In many listening scenarios, these two files will sound identical.

Do you hear any difference? (Compare listening to both on headphones and on the computer loudspeaker)

However, there is a huge difference in the amount of space each file uses up on your hard drive. The uncompressed WAV file takes up 3.8 MB while the MP3 uses only 340 KB. If you just plan to listen to this recording on your portable music player or in your car, it makes sense to go with the compressed file because it will leave more storage space available for other files and sound just as good to the average person listening in those environments. But when we need to process these files, whether just for sound and music analysis (like the one we'll be doing in this class) or just for a simple processing like a *fade out* we'll **always** use uncompressed file formats, in a way that we never lose vital information.

Let's look at what happens when your intended use for this recording involves more than just casual listening. Let's assume you want to perform some editing on this file and use it in a project or performance. This particular recording has a lot of echo and reverberation effects. Let's assume that you want to isolate all the material that is echoing and reverberating and remove the "dry" material. This can be done by simply subtracting the left channel from the right channel. Echo and reverberation are effects that cause the signal to bounce around between the left and right channels while the dry signal that is being affected sits equally in both channels (stereo center). Subtracting one channel from the

other will remove everything that is the same between the two channels and leave only the things that are unique. In this case the unique parts are the reverb and echo.
Your sound editor (Audacity or other) might have a subtraction function. If not, you can perform the following steps manually on the “Uncompressed.wav” file:

- Select the right channel and invert the polarity without affecting the left channel.
- Sum the two channels together to a mono file.
- Normalize the mono file to 0 dBFS peak. (Look for the “normalize” process in your sample editing program.)

You should now have a single-channel file that contains all the distant-sounding echo and reverb of the guitar along with some remnants of the ocean sound.

Now try the same process on the MP3 file!
Describe the resulting sound.