

Week 6: Audio Processing and Compression

# **DIGITAL ASSET DEVELOPMENT**

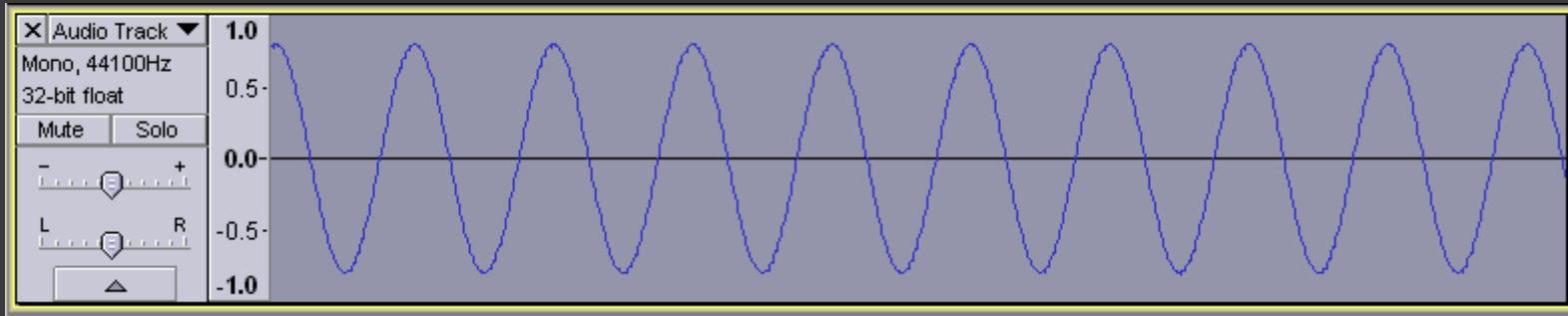
# Contents

- ⦿ Frequency-based audio analysis
- ⦿ Audio filtering
- ⦿ Audio compression

# Frequencies and Tones

- ⦿ As we saw last week, the tone or pitch of a sound relates to its frequency
  - Higher pitch implies higher frequency
- ⦿ Most audio consists of a combination of sounds with different tones
  - Known as **component frequencies**
- ⦿ The simplest sound is one that consists of a single tone
  - The corresponding sound wave has the appearance of a **sine wave**

# Sine Wave (Sinusoidal Curve)



- Any sound is made up of a combination of pure tones
  - In other words, it is made by combining sine waves of different frequencies
  - These waves are simply added together

# Decomposing Audio Data

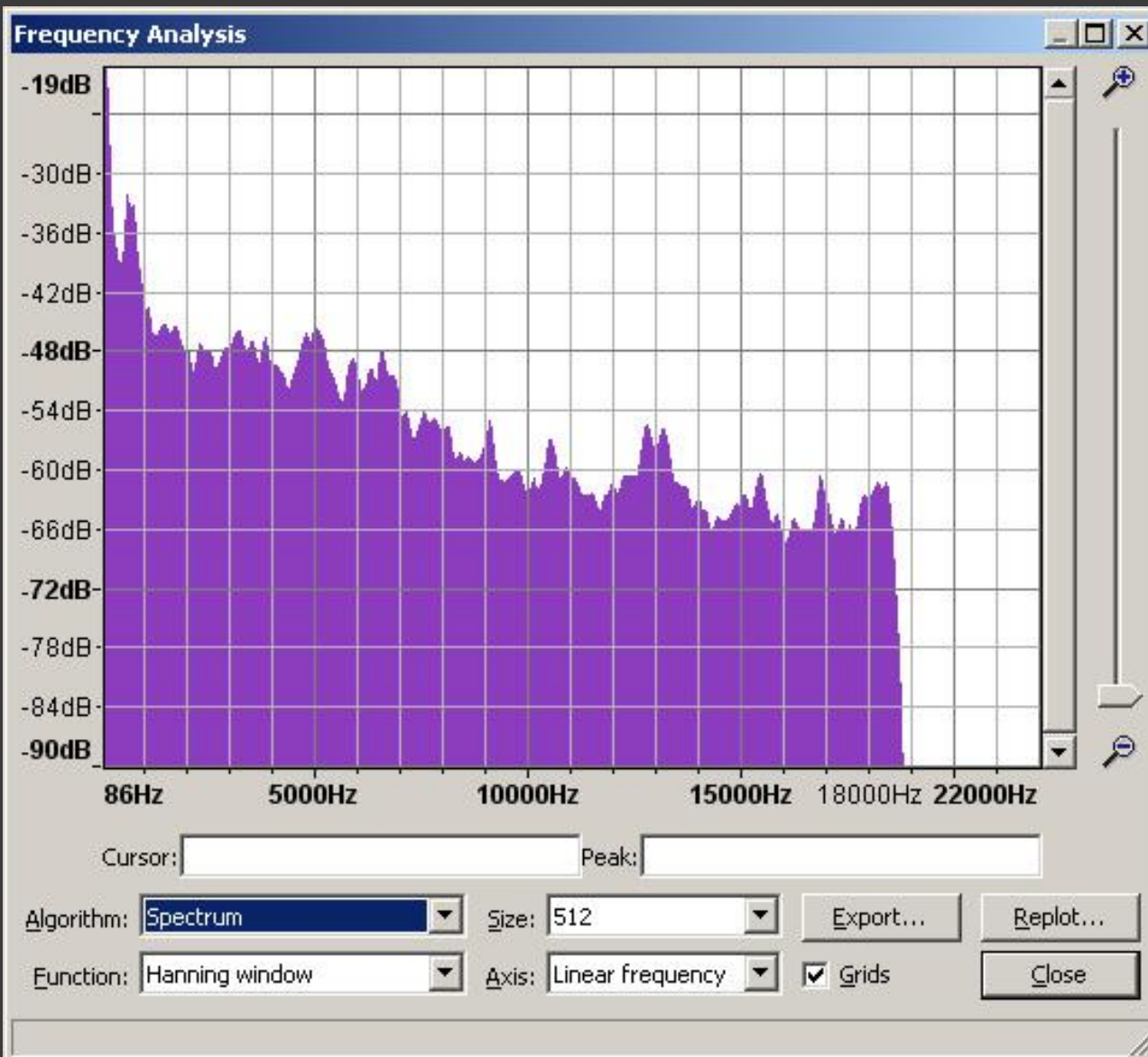
- ⦿ To process complex sounds, we need to access their **component frequencies**
  - This entails splitting a sound waveform into a collection of sine waves (**decomposition**)
  - We can then process specific frequencies individually within the overall sound
- ⦿ The **Fourier Transform** is a mathematical operation that does exactly this
  - Similar (though not identical) to the process used to compress a JPEG image

# Fourier Transform

- ⦿ The transform remaps the audio data into **frequency space**
- ⦿ Measures the contribution to the overall sound at specific frequencies
  - Parts of the sound wave that only change slowly contain low frequency energy
  - Parts that change rapidly represent high frequencies
- ⦿ Note that this process is reversible

# Frequency Spectrum

- ④ We can view the Fourier Transform of a sound as its **frequency spectrum**
- ④ **Plot Spectrum** function in Audacity
  - Plot frequencies along the x-axis (low to high)
  - Height for a given frequency corresponds to its contribution to the audio signal
- ④ Peaks in the graph represent dominant frequencies in the sound

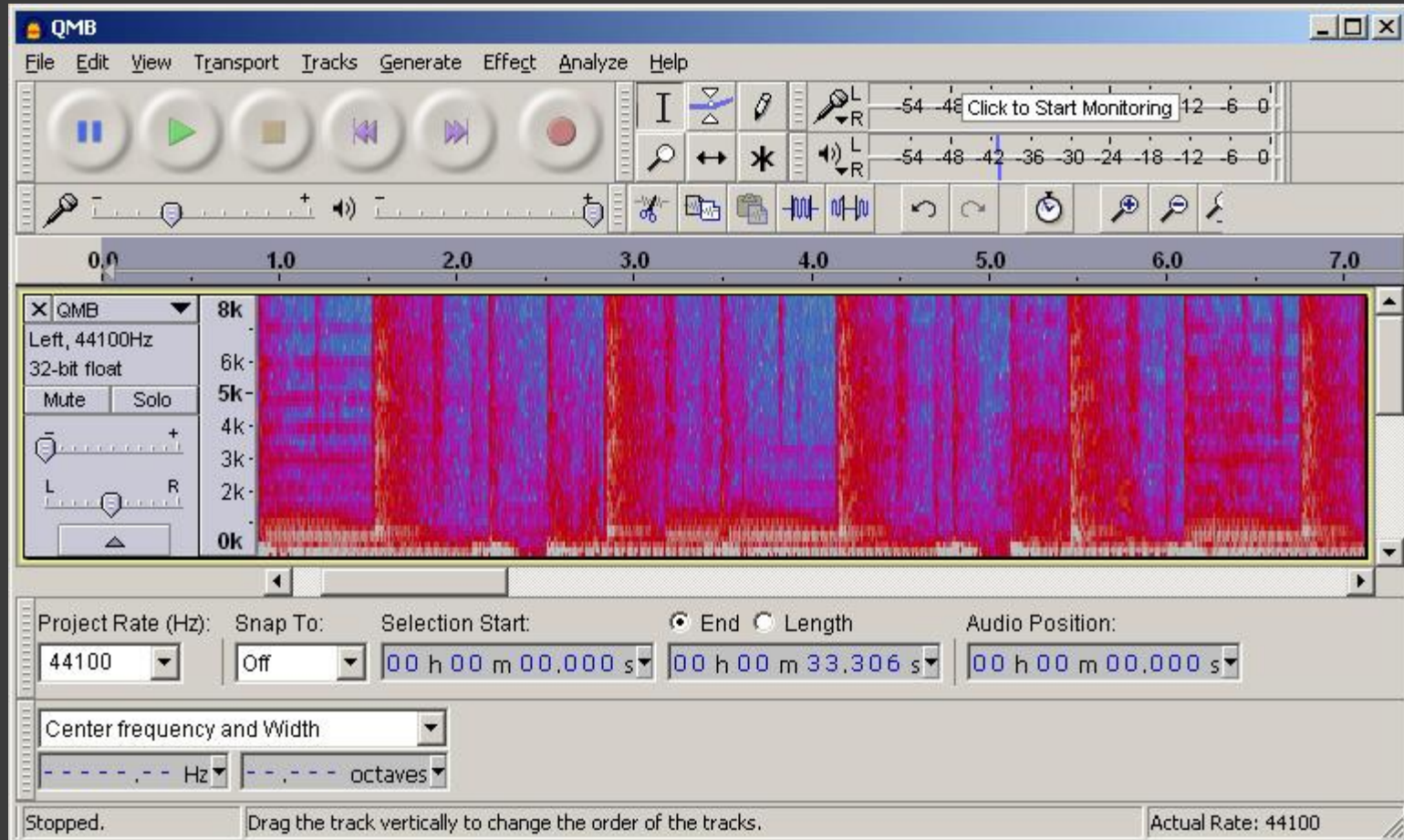




# Spectrogram

- ⦿ Sometimes it is more useful to see how the dominant frequencies vary over time
- ⦿ We can do this using an audio clip's **spectrogram**
- ⦿ This creates a 2D plot:
  - Time runs from left to right
  - Frequency is plotted vertically
  - Contribution (energy) at a given frequency and time is colour coded

# Spectrogram Example



# Using the Spectrogram

- The spectrogram is a very useful tool for processing audio files
- With experience, specific problems with audio files can be identified from the plot
  - Background noise at a specific frequency will appear as a horizontal bar
  - This is a common audio issue, often caused by electrical interference
  - Clicks and other glitches on an audio track may appear as a vertical spike

# Example (from lab exercise)



horizontal banding caused by (unwanted?)  
harmonics of a fundamental frequency

# Audio Cleanup

- ⦿ Using the spectrogram it is possible to isolate and rectify some audio problems
  - Select the affected set of frequencies
  - Apply some form of filter to reduce or eliminate the noise
  - Low pass, high pass and notch filters
- ⦿ Other approaches are possible, including “redrawing” the waveform
  - Useful for fine-tuning audio files

# Frequency-based Filters

- ⦿ A **low pass** filter suppresses high frequencies from a sound
  - Lets low frequencies “pass”
- ⦿ A **high pass** filter does the opposite
- ⦿ These are the audio equivalents of blur and edge detection filters for images
- ⦿ A **notch** or **band pass** filter suppresses frequencies within a set range
  - Useful for targeting specific types of noise

# Compressing Audio Data

- ⦿ As mentioned last week, it is convenient to be able to compress audio files
- ⦿ Ideally, we should do this without hurting overall audio quality
- ⦿ Mostly use frequency-based approach
  - Identify **component frequencies**
  - Remove or suppress those frequencies which humans will not hear (much!)
  - Store **quantised** frequency data

# Telephony

- ⦿ An early use of digital audio encoding was in telephone systems
- ⦿ Base standards for digital telecoms are  $\mu$ -law (N America / Japan) and A-law
  - Variations on pulse code modulation (PCM)
  - Use statistical properties of human voice patterns to improve data quality
  - Reduce effective 12 bit sample resolution to 8 bit requirement for transmission
- ⦿ Still the basis for telephone audio



# Psychoacoustic Models

- ◎ For more sophisticated compression, we need to study human hearing factors
  - Known as **psychoacoustic modelling**
- ◎ Main characteristics:
  - Frequency limits (typically 20 Hz – 20 kHz)
  - Our sensitivity peaks around 1-5 kHz
  - **Simultaneous masking**: sounds “merge” if played together (frequency dependent)
  - **Temporal masking**: louder sounds mask others immediately before or after them

# Audio Formats

- ⦿ Uncompressed audio formats include WAV and AIFF (Windows and Mac)
  - Used for raw recordings
  - Allow a choice of bit depths
- ⦿ A wide range of compression formats exist, though many are related
  - **MP3** is the most well known
- ⦿ MP3 supports a number of **codecs**
  - **Compressor/decompressor** algorithm

# The MP3 Format

- ⦿ MPEG-1 (or 2) Audio Layer 3
  - Part of the MPEG video standard
  - De facto standard for digital music
- ⦿ Follows frequency-based compression process described in earlier slide
- ⦿ Uses psychoacoustic modelling to optimise perceived quality
- ⦿ Choice of compression ratios allowed

# MP3 Compression

- ⦿ Most common bit rate is 128 kbits/sec
  - Approx 10:1 compression ratio
- ⦿ Higher bit rates should offer improved sound quality
- ⦿ This is for **constant bit rate** (CBR) files
- ⦿ MP3 also allows for **variable bit rate** (VBR) compression
  - Can be useful where the complexity of the audio signal varies greatly during a piece

# Choosing an Audio Format

- ⦿ For audio processing, it is best to use uncompressed formats
- ⦿ For distribution, key issues are sound quality and interoperability
- ⦿ MP3 is a reasonable default, though AAC and Vorbis reputedly sound better
  - Quality / file size also depends on codec
  - Some MP3 compression algorithms give better results than others