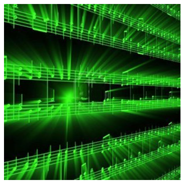
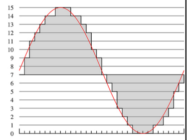
Penn Engineering ESE





Lecture #6 – Psychoacoustic Model/Compression/MP3

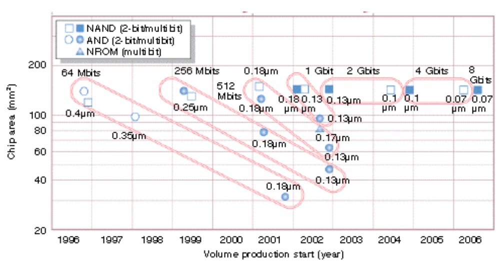
ESE 150 – Digital Audio Basics

ESE150 Spring 2018

First MP3 Player

- MpMan -- 1998
- SaeHan Information Systems
  - South Korea
- 32MB of Flash memory

Flash Scaling



PCM and CD Audio

- **PCM – Pulse Code Modulation**
  - CD-Quality Audio – state-of-the-art 1990s
    - Filtering/Sampling/Quantizing/Encoding using ADC
    - DAC/Reconstruction Filter
  - CD Quality Digital Audio uses PCM (uncompressed, lots of storage!)
    - 44,100 samples per second, each sample 16-bits
    - 1 sec. of music: 44,100 x 16bits = 705,600 bits or 86 kB
    - 60 seconds of music: 705,600 x 60 = 42,336,000 bits = 5167 kB = 5 MB
    - 3 minute song: 42,336,000 x 3 = 127,008,000 bits = 15 MB !
    - You want it in stereo??? 15 MB x 2 = 30 MB! (no compression!)

Observe

- **If we kept the CD Audio encoding format**
  - Could hold **one** song on the 1998 MpMan
  - (maybe 2 on the 64MB version)
- **For solid-state audio to be viable**
  - Needed more compact encoding for music

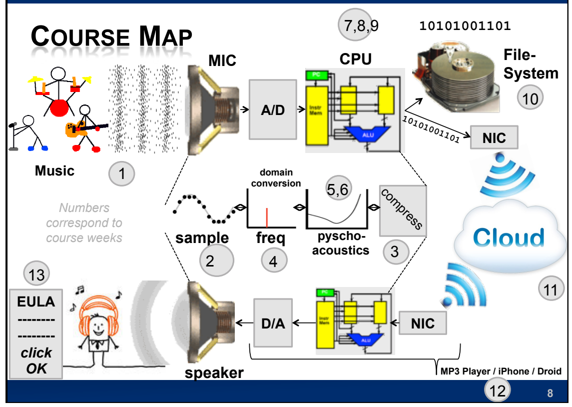
First MP3 Player

- **MpMan -- 1998**
- **SaeHan Information Systems**
  - South Korea
- **32MB of Flash memory**
- **Held 6 songs (MP3)**
- **(12 on 64MB version)**
- **3 years before Apple iPod**
  - October 2001
  - Initially hard disk
- **Diamond Rio later 1998**

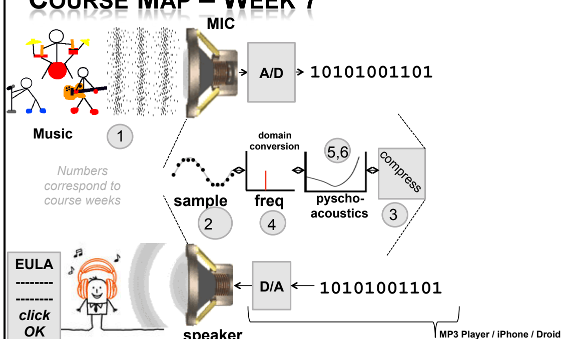
Lecture Topics

- Teaser
- Where are we?
- Preclass
- How do we take advantage of psychoacoustics in MP3
  Achieve this 6—12x reduction from CD Audio
  - Review Tricks
  - Formulate Optimization
  - Adaptation
- Next Lab
- References

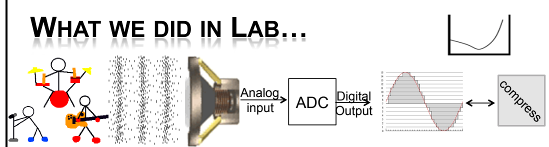
Course Map



Course Map – Week 7



What we did in Lab…



- Week 1: Converted Sound to analog voltage signal
  - a "pressure wave" that changes air molecules w/ respect to time
  - a "voltage wave" that changes amplitude w/ respect to time
- Week 2: Sampled voltage, then quantized it to digital sig.
  - Sample: Break up independent variable, take discrete 'samples'
  - Quantize: Break up dependent variable into n-levels (need 2^n bits to digitize)
- Week 3: Compress digital signal
  - Use even less bits without using sound quality!
- Week 4: Before we compress…
  - Put our 'digital' data into another form…BEFORE we compress…less stuff to compress!
- Week 5: Psychoacoustics
  - Measured limits of human hearing; measured masking

Preclass

Preclass

- 4 critical bands
- 10 frequencies
- 16b amplitude
- Preclass 1: Bits to represent (no further encode)?

PRECLASS

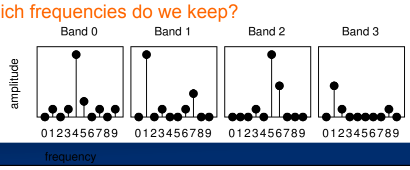
- **4 critical bands**
- **10 frequencies**
- **16b amplitude**
- **80b encoding budget**
- **Preclass 2: amplitude quantization necessary to achieve budget?**
- **Preclass 3: frequencies (reduced sampling rate) can keep to achieve budget?**

PRECLASS

- **4 critical bands**
- **10 frequencies**
- **16b amplitude**
- **80b encoding budget**
- **Preclass 4**
  - Bits to represent which frequency?
  - Bits to encode (frequency, amplitude) pair?
  - Number of (frequency,amplitude) pairs fit within budget?

PRECLASS

- **4 critical bands**
- **10 frequencies**
- **16b amplitude**
- **80b encoding budget**
- **Preclass 5**
  - Which frequencies do we keep?



PRECLASS

- **4 critical bands**
- **10 frequencies**
- **16b amplitude**
- **80b encoding budget**
- **Preclass 6: which likely to sound best?**
  - Amplitude quantization
  - Frequency quantization (reduce sampling rate)
  - Frequency selection based on masking

TRICKS FOR COMPRESSION

- **Quantization**
- **Sampling Rate / Frequency Quantization**
- **Critical Band Masking**
  - Selective frequency dropping
- **Other tricks?**

BACKGROUND

The MPEG-1 Standard

- **ISO (International Standards Organization)**
  - Looking for ways to reduce transmission requirements for digital video and audio (low bandwidth transmission of digital media)
  - 1988 – establishes a sub-committee of ISO:
    - *Moving* Picture Experts Group (MPEG)
    - Goal: Develop common standard for coding/compressing audio/video
      - To reduce size of data to transmit without sacrificing quality
      - Fraunhofer Institute and German University of Erlangen
        - Lots of basic research in Digital Audio Broadcast, tapped to be MPEG
  - Result: 1992: Finalized Standard called: MPEG-1 (Phase I)
    - 3 Parts: Audio/Video/System
    - Audio component: defined 3 layers: 1, 2, 3
      - Increasing levels of compression and complexity
      - MPEG-1, layer 3 achieves 12:1 compression ratio! *(for short MP3)*

The MPEG-1 Standard

- **MPEG-1, (3) Layers for Audio Coding:**

Complexity ↓

Coding	Ratio	Required bitrate
PCM CD Quality	1:1	1.4 Mbps
Layer I	4:1	384 kbps
Layer II	8:1	192 kbps
Layer III (MP3)	12:1	128 kbps

- **Why is PCM CD Quality 1.4 Mbps?**
  - Recall: 1 sec. of music: 44,100 x 16bits = 705,600 bits
  - Don't forget stereo (R/L): 2 x 706,600 = 1,413,200 (1.4Mbs)
    - Defines bandwidth requirements of network
  - Notice: 128 kbps was just about double modem speed in 1992
    - Enables transmission of audio (MP3) via modem! (Napster!)

The MPEG-1 Standard

- **How did MP3 achieve a 12:1 compression ratio?**
  - Psychoacoustics was the key
    - Becomes a scheme for lossy encoding
    - But we don't lose things human's can actually here!
    - Makes human a part of the "decompression" scheme
  - CD Audio PCM is lossless coding of digital audio
    - But human beings can't hear all sounds
  - MP3 Encoding is lossy coding scheme
    - But we don't lose things average human's can actually here!
    - We don't sacrifice high frequencies (sampling rate preserved)
    - The trick: drop quantization level when it doesn't matter!
      - The key is taking advantage of masking (frequency/temporal)
  - MP3 Combines: Psychoacoustics (lossy compression)
    - and Huffman coding (lossless compression)

They Kept on Going: MPEG-1, 2, 4, 7

MPEG is actually family of encoding standards for digital multimedia information
- *They all use Psychoacoustics to achieve high compression*

- **MPEG-1:** **a standard for storage and retrieval of moving pictures and audio on storage media (e.g., CD-ROM).**
  - *MPEG-1 Layer 3: (MP3)*
- **MPEG-2:** **standard for digital television, including high-definition television (HDTV), and for addressing multimedia applications.**
  - Advanced Audio Coding (AAC)
- **MPEG-4:** **a standard for multimedia applications, with very low bit-rate audio-visual compression for those channels with very limited bandwidths (e.g., wireless channels).**
- **MPEG-7:** **a content representation standard for information search**

Some Knowledge of Digital Audio Formats

- **.WAV & .AIFF**
  - Uncompressed PCM (lossless) CD-Audio Quality
- **.MPA**
  - MPEG-1 Layer 2
  - Compressed PCM using Psychoacoustics (lossy)
- **.MP3**
  - MPEG-1 Layer 3
  - Compressed PCM using Psychoacoustics (lossy)
- **.M4A (Audio only Apple's iTunes format)**
  - MPEG-4 Part 14 (MP4)
  - Compressed PCM using Psychoacoustics

Some Terminology

- **Psychoacoustics**
  - Research into how brain interprets sounds from ear
- **Perceptual Coding**
  - An audio encoding technique that takes advantage of psychoacoustics
  - CD delivers all recorded sounds to your ear…
    - …but your brain can't actually interpret them
  - Perceptual coded audio delivers sounds to your ear…
    - …that your brain can actually interpret!
- **CODEC**
  - Coder/Decoder
- **MP3 is a standard for a perceptual audio codec…**
  - That takes advantage of frequency/time masking to encode audio data!

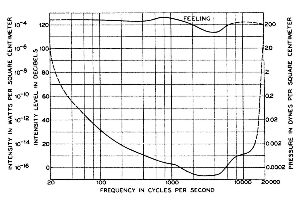
Optimizing Encoding

Knobs we can turn

- **Amplitude quantization**
- **Frequency quantization**
- **Frequencies kept (per critical band)**
  - Too soft
  - Masked
- **…and can perform lossless compression**

Bands vary in importance

- **Not equally sensitive across bands**
- **If quantize bands differently, where want finer resolution?**
- **Where tolerate more quantization?**



Knobs we can turn

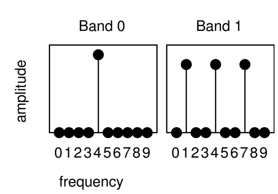
- **Amplitude quantization**
  - Per band
- **Frequency quantization**
  - Per band?
- **Frequencies kept (per critical band)**
  - Per band
- **…and can perform lossless compression**

Opportunities/Challenges

- **Trying to hit fixed bit rate, what challenge does lossless compression impose?**
  - Encounter many common frequencies?
  - Encounter many uncommon frequencies?

Opportunities/Challenges

- **What challenge/opportunities might these band spectra represent?**



Suggest

- May want to do something smarter than
  - Allocating fixed number of frequencies per band
  - Allocating fixed quantization to a band
- Like to adapt our encoding to the data
  - If more Huffman compressible, we get more frequencies
  - If fewer frequencies suffice for one band,
    - Allow more frequencies for another
    - …or allocate less quantization

Optimization Problem

- How fit in the resource constraints (128Kb/s) while maximizing goodness (sound quality)?
- Optimization problems central to engineering

Optimization Problem

- How fit in the resource constraints (128Kb/s) while maximizing goodness (sound quality)?
- Quantify bits used $\sum_{bands}\sum_{freq} Bits(freq)$
- Quantify goodness

$$\sum_{freq} Error(freq) \times W(freq)$$

Quantifying Bit cost

- Simple, fixed sample: Frequencies × Bits/freq
- Fixed frequencies per Band:
  - Bands × (Frequencies/Band) × Bits/freq
- Variable Frequencies per Band:

$$\sum_{bands}\left(frequencies(band) \times bits/freq\right)$$

- Variable Frequencies and quantization per Band:

$$\sum_{bands}\left(frequencies(band) \times bits(band)\right)$$

- Huffman means different bits/frequency $\sum_{bands}\sum_{freq} Bits(freq)$

Goodness/Sound Quality

- Error(freq) = |OrigFreq Amplitude – Encoded|
  - Whole OrigFreq if dropped
  - |OrigFreq-Quantize(OrigFreq,bits)| if quantized
- W(freq)
  - 0 if below hearing threshold
  - 0 if masked
  - Value between 0 and 1 if partially masked in critical band
  - Really depend on what already encoded $\sum_{freq} Error(freq) \times W(freq)$

Optimization Problem

- How fit in the resource constraints (128Kb/s) while maximizing goodness (sound quality)?
- Quantify bits used $\sum_{bands}\sum_{freq} Bits(freq)$
- Quantify goodness

$$\sum_{freq} Error(freq) \times W(freq)$$

Approach

$$\sum_{bands}\sum_{freq} Bits(freq)$$

$$\sum_{freq} Error(freq) \times W(freq)$$

- **Start with nothing**
- **Start with bitbudget**
- **While(bitbudget>0)**
  - Identify Largest Error component: Error(freq)×W(freq)
  - What do to reduce error?

Approach (Greedy)

- **Start with nothing**
- **Start with bitbudget**
- **While(bitbudget>0)**
  - Identify Largest Error component
  - Allocate some bits to reduce error
    - Add frequency
    - Add quantization bits to band
    - Pick one to most reduce the error

Approach (Refine Greedy)

- **Start with nothing**
- **Start with bitbudget**
- **While(bitbudget>0)**
  - Identify Largest Δerror/Δbits
- **Why might prefer?**
- **Large error, might take many bits to improve**
  - Maybe could be better spending those bits to fix many problems?

Adaptive Refinement

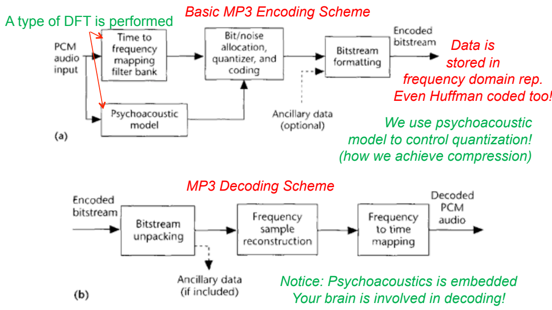
- **Rediscovering where to allocate everything every time may be laborious**
- **Maybe we can get close and adjust?**

Approach (Adaptive)

- **Start with budget guess**
  - Quantization in bands
  - Frequencies to keep in each band
- **Encode, compress**
- **What do if haven't used up all bits?**
- **What do if over budget?**

PERCEPTUAL CODING & MP3

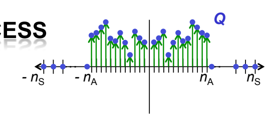
MP3 Flow Chart (Encoding/Decoding)



MP3 Encoding Process

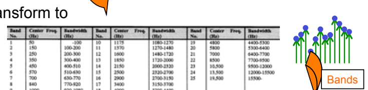
- **All MP3 files broken into "Frames"**
  - Each frame stores 1152 Audio Samples
  - Lasts for 26 ms
  - Frame also divided further into 2 "granuels"
    - Each granuel contains 576 samples
- **Bitrate determines size of each sample**
  - Increase the bitrate, increase the size of the frame!
- **Sampling rate also impacts frame size**
- **Formula:** $\frac{144 * bitrate}{samplefrequency} + Padding$ [bytes]
  - Example: frame size = 144 x 128kbs / 44.1kHz = 417

MP3 Encoding Process



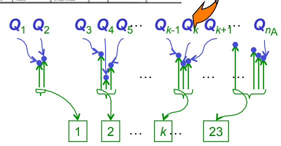
acquire & transform "frame"

- Bring in 1152 PCM samples
- Use DFT/MDCT to transform to frequency-domain



assign frequencies to bands

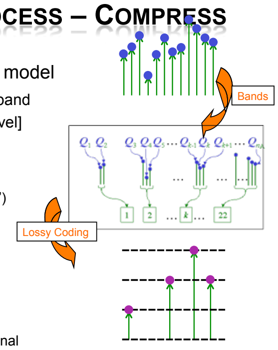
- use psychoacoustic model lookup to determine frequency bandwidth of each critical band
- separate samples into bands



MP3 Encoding Process – Compress

Compress based off masking model

- Goal: minimize bits per critical band
  - $C_{band\ k(j)}$ = Round[$Q_{band\ k(j)}$, Level]
- by appeal to "masking" models
  - $Q_{band\ k} = C_{band\ k} + D_{band\ k}$
- where distortion ("perceptual noise")
  - $D_{band\ k}$
- between retained signal
  - $C_{band\ k}$
- and actual signal
  - $Q_{band\ k}$
- should be "masked" by retained signal



Compress – Up close

E.g., Look at k[th] band

$Q_{band\ k} = (Q_{band\ k(1)}, \cdots, Q_{band\ k(m)})$

Determine masking paradigm:

- tone-masking-noise
- noise-masking-tone
- noise-masking-noise

E.g., for tone masker,

- Pick tone frequency, band k (j)
- at maximal amplitude in the band

Choose quantization level and compute compressed signal

- $C_{band\ k(j)}$ = Round[$Q_{band\ k(j)}$, Level]
- $C_{band\ k} = (0_{band\ k(1)}, \cdots, C_{band\ k(j)}, \cdots, 0_{band\ k(m)})$

Assess noise magnitude, $|D_{band\ k}|$

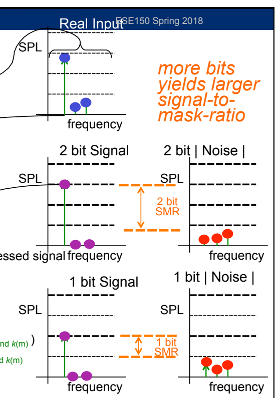
$D_{band\ k}$ =

$(Q_{band\ k(1)}, \cdots, Q_{band\ k(j)}, \cdots, Q_{band\ k(m)})$

$- (0_{band\ k(1)}, \cdots, C_{band\ k(j)}, \cdots, 0_{band\ k(m)})$

Use psychoacoustic model

- to determine whether compressed signal will mask the distortion noise for that band



Formal MP3 Perceptual Coding Algorithm

Commit to Observation Window

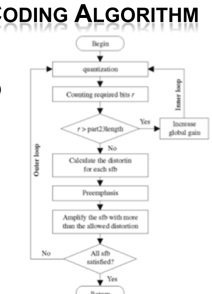
- "Long" (complex sound; frequency resolution)
- "Short" (transient sound; temporal resolution)

Estimate Perceptual Entropy

- Analyze Each Critical Band
- Characterize Masker
- Estimate Mask-to-Noise Threshold
- Update "bit reservoir"

Spectral Quantization/Coding Loop

- Allocate bits per critical band
- Quantize band to bits-allowed levels
- Run Huffmann & count actual bits



[You & Chen. *Multim.Tools & Appl.*, 40(3):341–359, 2008.]

Big Ideas

- **Can use pyschoacoustics to compress audio**
  - Eliminate portions of signal that human's don't notice
- **Optimization**
  - Identify Design Space (knobs)
  - Identify Costs and Constraints
  - Formulate quantitatively
  - Algorithms to approach
  - Iterative/adaptive approach
    - Deal with effects that aren't completely predictable

Learn More

- **Optimization –**
  - continuous mathematical optimization ESE204, ESE504, ESE605
  - discrete optimization CIS121, CIS320
- **Signal processing – ESE224**

Next Week

- **Monday start of 2 week lab:**
  - Perform perceptual compression
    - Convergence of everything seen first 6 weeks
  - Formal lab report on this 2 week lab
  - No weekly lab report Friday
- **Wednesday: midterm (class time)**
- **Office Hours:**
  - Cancel Wed. and Thursday evening
  - Add Tuesday 5-9pm
  - Wed. 1-3pm remain

Midterm – In Class

- **Closed book, no notes**
- **Calculators allowed**
- **4:35pm-5:50pm**
- **5% of grade**
  - prepare for final
- **Last year's final on syllabus**
  - Will be a bit different

Topics

- Data representation in bits
- Sounds waves
- Sampling
- Quantization
- Nyquist
- Lossy/lossless compression
- Common case
- Frequency domain
- Psychoacoustics
- Perceptual coding

References

Tutorials on Psychoacoustic Coding (in increasing order of abstraction and generality)

D. Pan, M. Inc, and I. L. Schaumburg. A tutorial on MPEG/audio compression. IEEE multimedia, 2(2):60–74, 1995.

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J. D. Johnston. Transform coding of audio signals using perceptual noise criteria. IEEE Journal on selected areas in communications, 6(2):314–323, 1988.