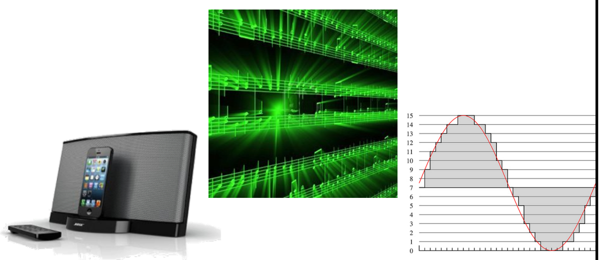


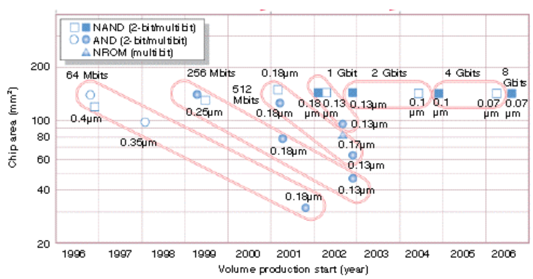
Lecture #11 – Psychoacoustic Model/Compression/MP3

ESE 150 – Digital Audio Basics

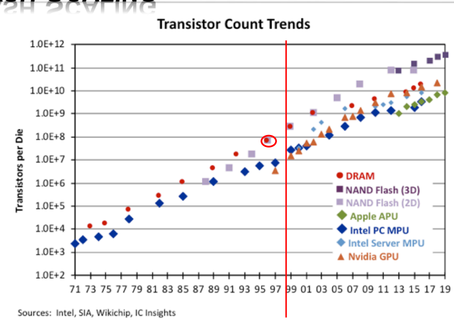
First MP3 Player

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

Flash Scaling



Flash Scaling



https://www.icinsights.com/news/bulletins/Transistor-Count-Trends-Continue-To-Track-With-Moores-Law/

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

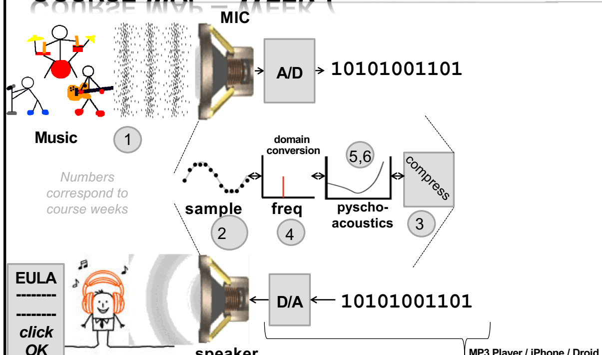
First MP3 Player

- 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

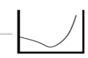
Lecture Topics

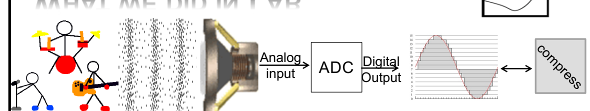
- Teaser
- Where are we?
- Preclass
- How do we take advantage of psychoacoustics in MP3 Achieve this 6—12x reduction from CD Audio
  - Review Tricks
  - Formulate Optimization
  - Adaptation (next Friday)
- Next Lab
- References

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

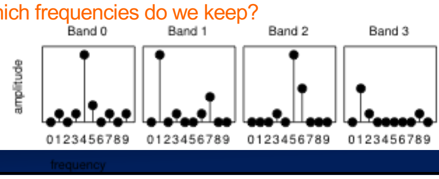
- 4 critical bands
- 10 frequencies
- 16b amplitude
- 107b encoding budget
- Preclass 2: amplitude quantization necessary to achieve budget?
- Preclass 3: frequencies (reduced sampling rate) can keep to achieve budget?

Preclass

- 4 critical bands
- 10 frequencies
- 16b amplitude
- 107b 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
- 107b encoding budget
- Preclass 5
  - Which frequencies do we keep?



Preclass

- 4 critical bands
- 10 frequencies
- 16b amplitude
- 107b 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

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!

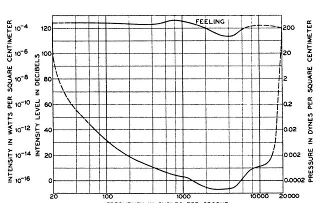
Optimizing Encoding

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

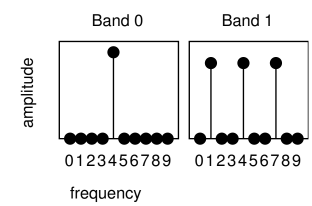
- **Amplitude quantization**
  - Per band
- **Frequency quantization**
  - Per band?
- **Frequencies kept (per critical band)**
  - Per band
- **…and can perform lossless compression**

Opportunities/Challenges

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

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** $\sum_{bands}\sum_{f \in freqs} Bits(f)$
- **Quantify goodness**

$$\sum_{f \in freqs} Error(f) \times W(f)$$

Quantifying Bit cost

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

$$\sum_{b \in bands} frequencies(b) \times (\text{bits/freq})$$

- **Variable Frequencies and quantization per Band:**

$$\sum_{b \in bands} frequencies(b) \times bit(b)$$

- **Huffman means different bits/frequency** $\sum_{bands}\sum_{f \in freqs} Bits(f)$

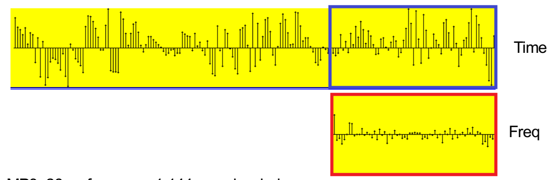
Goodness/Sound Quality

- **Error(Amp) = |Orig Amplitude – Encoded|**
  - Whole OrigAmplitude if dropped
  - |Orig Amplitude-Quantize(OrigAmplitude,bits)| if quantized
- **W(freq)**
  - 0 if below hearing threshold
  - 0 if masked
  - Value between 0 and 5 if partially masked in critical band
    - More critical bands, higher
  - Really depend on what already encoded

$$\sum_{f \in freqs} Error(f) \times W(f)$$

Reminder: A Window Operation

- **In the example below, we traverse the signal but only look at 64 samples at a time**



MP3: 26ms frames or 1,144 sample window

(sepwww.stanford.edu/oldsep/hale/FftLab.html)

Optimization Problem

- **How fit in the resource constraints (128Kb/s) while maximizing goodness (sound quality)?**
- **Quantify bits used:**
  - Cannot exceed 128Kb/s
  - = 3,408 b / 26ms frame

$$\sum_{bands} \sum_{f \in freqs} Bits(f)$$

- **Quantify goodness: minimize**

$$\sum_{f \in freqs} Error(f) \times W(f)$$

(as time permits)

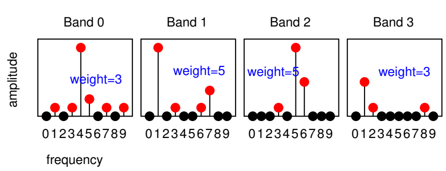
Approach

Approach

$$\sum_{bands} \sum_{f \in freqs} Bits(f)$$

- **Start with nothing**
- **Start with bitbudget**

$$\sum_{f \in freqs} Error(f) \times W(f)$$



Example Weight Function W(f)

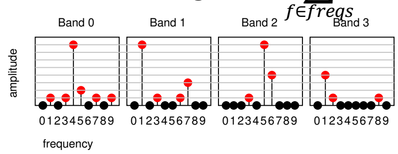
- **W(f)=CBWeight*Mask**
- **Mask = 0 if MaxAmp-FreqAmp<4**
- **(MaxAmp-FreqAmp)/MaxAmap otherwise**

Approach

$$\sum_{bands}\sum_{f \in freqs} Bits(f)$$

- Start with nothing
- Start with bitbudget

$$\sum_{f \in freqs} Error(f)\times W(f)$$



Bits = 0
Error = 190

	11 8 2 11	8 1 1 3	1 8 4	41 1
sum	14	13	13	6
weight	3	5	5	3
weighted	42	65	65	18

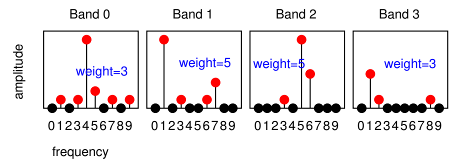
Approach

$$\sum_{bands}\sum_{f \in freqs} Bits(f)$$

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

$$\sum_{f \in freqs} Error(f)\times W(f)$$

Bits = 0
Error = 190

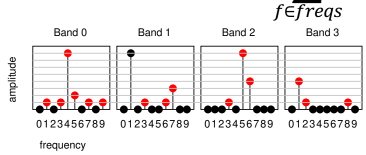


Approach

$$\sum_{bands}\sum_{f \in freqs} Bits(f)$$

- Start with nothing
- Start with bitbudget

$$\sum_{f \in freqs} Error(f)\times W(f)$$



Bits = 3+6=9
Error = 125

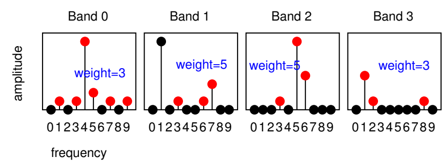
	11 8 2 11	0 0 0 0	1 8 4	41 1
sum	14		13	6
weight	3	5	5	3
weighted	42	0	65	18

Approach

$$\sum_{bands}\sum_{f \in freqs} Bits(f)$$

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

$$\sum_{f \in freqs} Error(f)\times W(f)$$

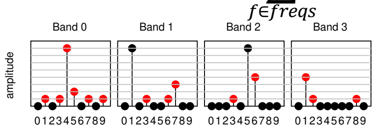


Approach

$$\sum_{bands}\sum_{f \in freqs} Bits(f)$$

- Start with nothing
- Start with bitbudget

$$\sum_{f \in freqs} Error(f)\times W(f)$$



Bits = 18
Error = 60

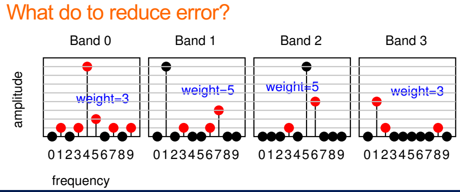
	11 8 2 11	0 0 0 0	0 00	41 1
sum	14			6
weight	3	5	5	3
weighted	42	0	0	18

Approach

$$\sum_{bands}\sum_{f \in freqs} Bits(f)$$

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

$$\sum_{f \in freqs} Error(f)\times W(f)$$

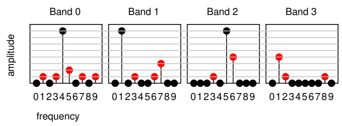


Approach

$$\sum_{bands} \sum_{f \in freqs} Bits(f)$$

- Start with nothing
- Start with bitbudget

$$\sum_{f \in freqs} Error(f) \times W(f)$$



Bits = 27
Error = 18

		0 0 0 0		0 0 0		41 1			
sum	0					6			
weight	3	5		5		3			
weighted	0	0		0		18			

Next Time

- Continue to refine into algorithm

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

Coming Up

- Feedback
- Lab 5 writeup today
- Lab 6 start on Monday
  - On syllabus
  - Start of 2 week lab
- Midterm on Wednesday
  - Rules/policies on syllabus now

References

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

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