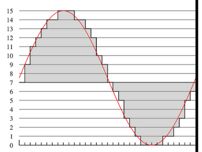
Penn Engineering

ESE



Lecture #6 – Psychoacoustic Model/Compression/MP3

ESE 150 – DIGITAL AUDIO BASICS

ESE150 Spring 2019

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FIRST MP3 PLAYER

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

2

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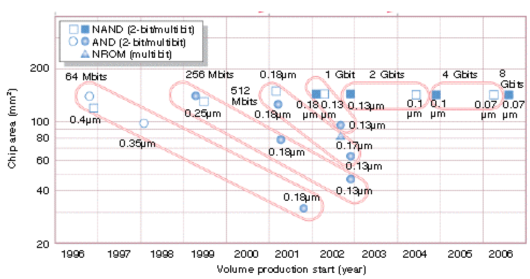
FLASH SCALING



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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!)

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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

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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

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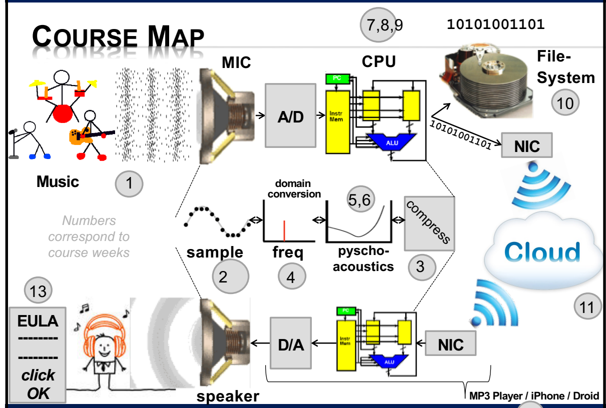
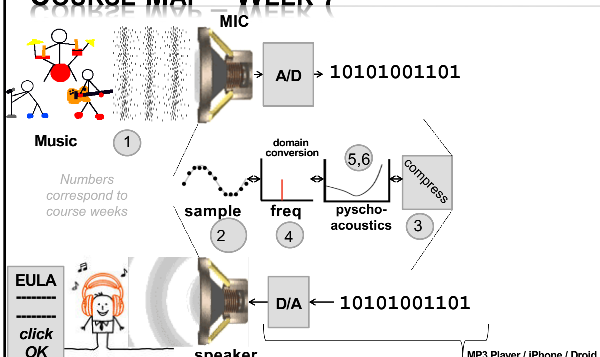
Lecture Topics

- 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

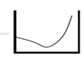
Course Map

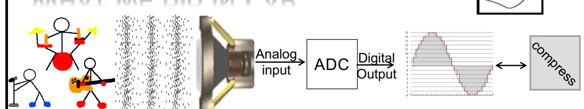


Course Map – Week 7



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

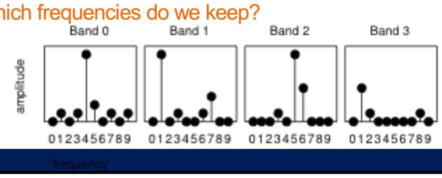
- 4 critical bands
- 10 frequencies
- 16b amplitude
- 80b encoding budget
- Preclass 2: amplitude quantization necessary to achieve budget?
- Preclass 3: frequencies (reduced sampling rate) can keep to achieve budget?

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!)

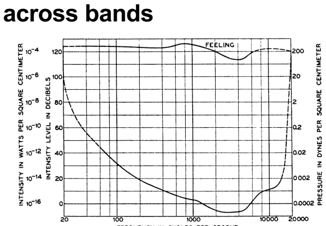
OPTIMIZING ENCODING

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

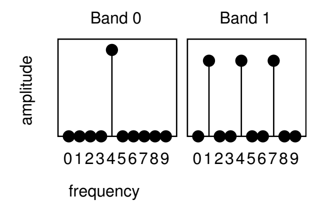
- **Amplitude quantization**
  - Per band
- **Frequency quantization**
  - Per band?
- **Frequencies kept (per critical band)**
  - Per band
- **…and can perform lossless compression**

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

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

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**

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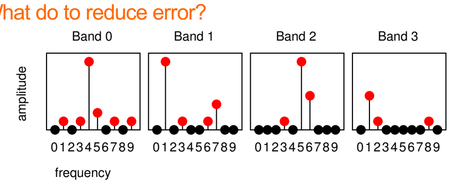
Approach

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

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

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



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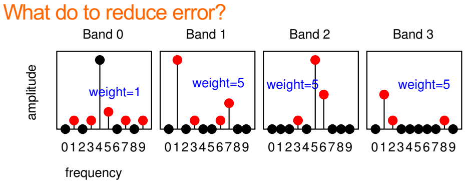

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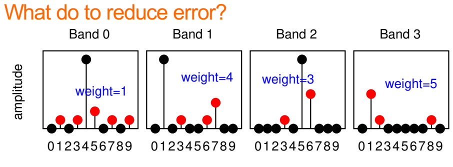


Approach (Greedy)

- **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

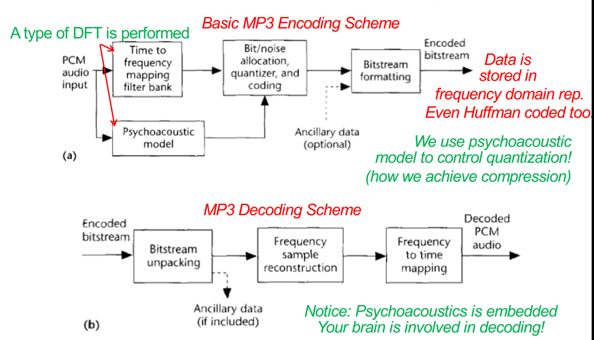
- **Rediscovering where to allocate everything every time may be laborious**
- **Maybe we can get close and adjust?**

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

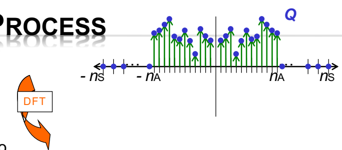
MP3 Flow Chart (Encoding/Decoding)



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

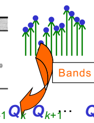
MP3 Encoding Process



acquire & transform "frame"

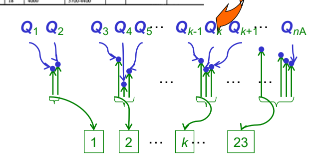
- Bring in 1152 PCM samples
- Use DFT/MDCT to transform to frequency-domain

Band No.	Center Freq. (Hz)	Bandwidth (Hz)	Band No.	Center Freq. (Hz)	Bandwidth (Hz)	Band No.	Center Freq. (Hz)	Bandwidth (Hz)
1	50	-100	10	1175	1080-1270	19	4800	4400-5300
2	150	100-200	11	1370	1270-1480	20	5800	5300-6400
3	250	200-300	12	1600	1480-1720	21	7000	6400-7700
4	350	300-400	13	1850	1720-2000	22	8500	7700-9500
5	450	400-510	14	2150	2000-2320	23	10,500	9500-12000
6	570	510-630	15	2500	2320-2700	24	13,500	12000-15500
7	700	630-770	16	2900	2700-3150	25	19,500	15500-
8	840	770-920	17	3400	3150-3700			
9	1000	920-1080	18	4000	3700-4400			



assign frequencies to bands

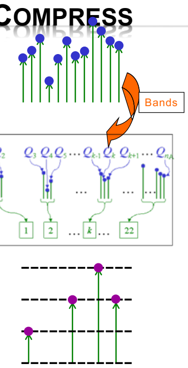
- use psychoacoustic model lookup to determine frequency bandwidth of each critical band
- separate samples into bands

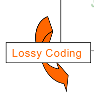


MP3 Encoding Process – Compress

Compress based off masking model

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





Big Ideas

- **Can use pschoacoustics 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**

After Spring Break

- **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
  - (not out, yet)
- **Wednesday: midterm (class time)**
- **Office Hours:**
  - Cancel Thursday office hours
  - Add Tuesday 7-10pm

Midterm – In Class

- **Closed book, no notes**
- **Calculators allowed**
- **4:35pm-5:50pm**
- **5% of grade**
  - prepare for final
- **Last year's final and answers**
  - on 2018 syllabus

Topics

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

Feedback

- **Note 2 sides**
  - 2 labs
  - 2 lectures

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.
  - Nikil Jayant, James Johnston, and Robert Safranek. Signal compression based on models of human perception. Proceedings of the IEEE, 81(10):1385–1422, 1993.
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- Scientific Basis of MP3 Coding Standard
  - J. D. Johnston. Transform coding of audio signals using perceptual noise criteria. IEEE Journal on selected areas in communications, 6(2):314–323, 1988.