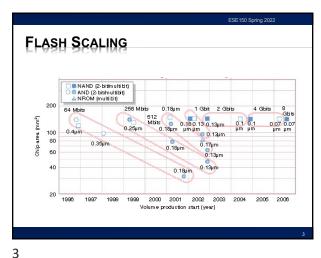


FIRST MP3 PLAYER MpMan -- 1998 SaeHan Information Systems South Korea 32MB of Flash memory

1



FLASH SCALING Transistor Count Trends 1.0E+12 1.0E+11 1.0E+10 1.0E+9 1.0E+8 1.0E+7 1.0E+6 NAND Flash (3D) 1.0E+5 1.0E+4 1.0E+2 71 73 75 77 79 81 83 85 87 89 91 93 95 97 99 01 03 05 07 09 11 13 15 17 19 https://www.icinsights.com/news/bulletins/Transistor-Count-Trends-Continue-To-Track-With-Moores-Law/

4

PCM AND CD AUDIO reconstruction filter 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

5 6



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)

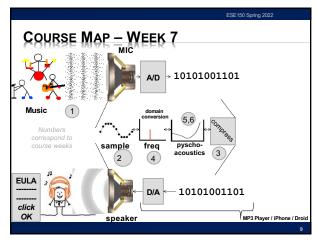
* Next Lab

* References

8

10

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

**a "voltage wave" that changes amplitude 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, lade discrete 'samples'

Quantize; Break up dependent variable into n-levels (need 2* bits to digitize)

**Week 3: Compress digital signal

**e see ven 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

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COMPARE TO LAB

* Lab 6

- Capture spirit of reducing frequencies
- Simplified – only taking loudest fraction in each band
- Rather than being rigorous about masking
- Or trying to hit some fixed rate

* Lecture 11 & 12 (this week)
- Help understand more what real MP3 encoding looks like
- Thinking about fixed rate
- And adaptation for variable rate encoding from Huffman
- Formulating masking more explicitly
- But still simplistic
- Illustrating Optimization Approaches

PRECLASS

11 12

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?

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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?

* 4 critical bands

* 10 frequencies

* 16b amplitude

* 107b encoding budget

* Preclass 5

- Which frequencies do we keep?

Band 0

Band 1

Band 2

Band 3

Band 3

Band 3

15

PRECLASS

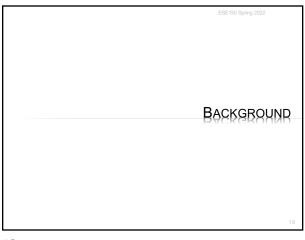
- × 4 critical bands
- × 10 frequencies
- × 16b amplitude
- × 107b encoding budget
- * Preclass 6: which likely to sound best?
 - + Amplitude quantization
 - + Frequency quantization (reduce sampling rate)
 - + Frequency selection based on masking

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TRICKS FOR COMPRESSION

- × Quantization
- x Sampling Rate / Frequency Quantization
- × Critical Band Masking
 - Selective frequency dropping

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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)

19 20

THE MPEG-1 STANDARD

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

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

Why is PCM CD Quality 1.4 Mbps?

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- 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!

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BIG PICTURE: WHY CARE × MP3 shows We can reduce bits needed to adequately represent music by 12x over PCM samples × 1998 Necessary to make MP3 player viable at all on 32MB Flash memory Small, affordable by masses Also meant lower bandwidth to stream × 2006 Necessary to make Spotify streaming viable 2021 With 64GB phones, maybe not essential for music on iPhone,

BIG PICTURE: WHY CARE Networks / Internet bandwidth / Cell capacity Allows us to reduce bandwidth needed for audio Alternative is Building out more wires and capacity Fewer cell calls supported simultaneously in a region Being efficient with bits Reduces physical investment needed in wires Same ideas enable video Viable to store (many) hours of video on phone/tablet Of increasing resolution: HD, 4K.... Viable to stream video into homes Zoom, Netflix, AppleTV, ... all enabled by compression

Penn Engineering Part 2 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?

OPPORTUNITIES/CHALLENGES

lossless compression impose?

Trying to hit fixed bit rate, what challenge does

+ Encounter many common frequencies, amplitudes?

+ Hint: what did we see in Lab 3 for time samples?

+ Encounter many uncommon frequencies, amplitudes?

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KNOBS WE CAN TURN

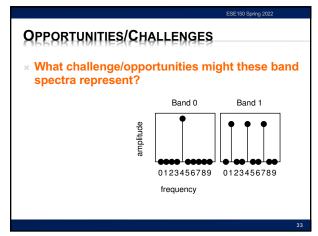
x Amplitude quantization

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

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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

33 34

OPTIMIZATION PROBLEM

How fit in the resource constraints (128Kb/s) while maximizing goodness (sound quality)? REMINDER: A WINDOW OPERATION In the example below, we traverse the signal but only look at 64 samples at a time Time Freq MP3: 26ms frames or 1,144 sample window (sepwww.stanford.edu/oldsep/hale/FftLab.html)

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OPTIMIZATION PROBLEM

How fit in the resource constraints (128Kb/s) while maximizing goodness (sound quality)?

- 128K/s * 0.026 s = 3408 bits per 26ms frame
- 3408/2 = 1704 per stereo channel

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OPTIMIZATION PROBLEM

How fit in the resource constraints (128Kb/s = 3048 bits/frame) while maximizing goodness (sound quality)?

× Quantify bits used

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

Quantify goodness

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

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OPTIMIZATION PROBLEM

How fit in the resource constraints (128Kb/s) while maximizing goodness (sound quality)?

- + 128K/s * 0.026 s = 3408 bits per 26ms frame
- 3408/2 = 1704 per stereo channel
- Optimization problems central to engineering

QUANTIFYING BIT COST

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

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

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

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

* Quantify goodness: minimize $\sum_{f \in freqs} \sum_{error(f) \times W(f)} Error(f) \times W(f)$

41 42

PROBLEM FORMULATION

* 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

* Quantify goodness: minimize $\sum_{f \in freqs} Error(f) \times W(f)$

FORMULATION VS. SOLUTION

* Formulation – goal – formally what want to achieve
+ Today's focus

* Solution – how go about solving it
+ How achieve (minimize) constraints (costs) in formulation
+ More on Wednesday

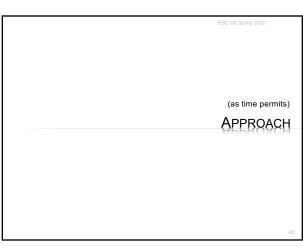
43 44

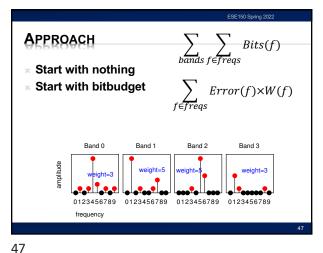
MOVES

* Select a frequency to encode

- Main example today

* Assign more bits to a band for quantization





EXAMPLE WEIGHT FUNCTION W(F) W(f)=CBWeight*Mask Mask = 0 if MaxAmp-FreqAmp>3 1 otherwise

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APPROACH $\sum_{bands} \sum_{f \in freqs} Bits(f)$ x Start with nothing $\sum_{i} Error(f) \times W(f)$ Start with bitbudget Band 0 Bits = 0 Error = 190 frequency 11 8 2 11 8 1 1 3 1 8 4 13

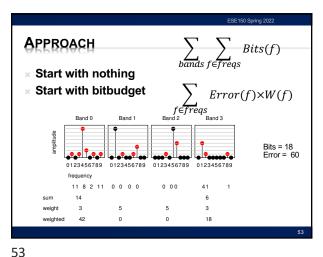
 $\sum_{bands} \sum_{f \in freqs} Bits(f)$ **APPROACH** Start with nothing $\sum_{f \in freqs} Error(f) \times W(f)$ Start with bitbudget While(bitbudget>0) + Identify Largest Error component: Error(freq)×W(freq) + What do to reduce error? Bits = 0 Error = 190 Band 0 Band 1 Band 2 Band 3 0123456789 0123456789 0123456789 0123456789 frequency

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 $\sum_{bands} \sum_{f \in freqs} Bits(f)$ **APPROACH** x Start with nothing $\sum_{f \in freqs} Error(f) \times W(f)$ × Start with bitbudget Band 0 Band 2 Assume 8 levels Bits = 3+6=9 Error = 125

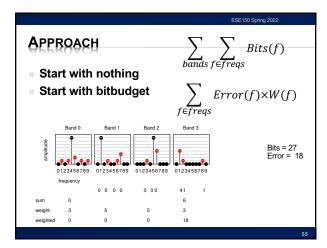
 $\sum_{bands} \sum_{f \in freqs} Bits(f)$ **APPROACH** Start with nothing $\sum_{f \in freqs} Error(f) \times W(f)$ Start with bitbudget While(bitbudget>0) Identify Largest Error component: Error(freq)×W(freq) + What do to reduce error? Band 0 Band 1 Band 2 Band 3 frequency

51 52



 $\sum_{bands} \sum_{f \in freqs} Bits(f)$ **APPROACH** × Start with nothing $\sum_{f \in freqs} Error(f) \times W(f)$ Start with bitbudget While(bitbudget>0) + Identify Largest Error component: Error(freq)×W(freq) + What do to reduce error? Band 0 Band 1 Band 2 Band 3 0123456789 0123456789 0123456789 frequency

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NEXT TIME Continue to refine into algorithm / solution

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ESE150 Spring 2022	
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	
	57

COMING UP Feedback including Lab Lab 5 writeup today × Lab 6 start on Wednesday On syllabus Start of 2 week lab Simplified version compression algorithm × Midterm Next Monday

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ESE150 Spring 202

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