CELT: A Low-latency, High-quality Audio Codec

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The Xiph.Org Foundation

Outline

- **Introduction and Motivation**
- CELT Design
- libcelt
- Demo
- Conclusion

Introduction

- Two common types of lossy audio codecs
  - Speech/communication (G.72x, GSM, AMR, Speex)
    - Low delay (15-30 ms)
    - Low sampling rate (8 kHz to 16 kHz): limited fidelity
    - No support for music
  - General purpose (MP3, AAC, Vorbis)
    - High delay (> 100 ms)
    - High sampling rates (44.1 kHz or higher)
    - "CD-quality" music
  - We want both: high fidelity with *very* low delay

Introduction

- Low delay is critical to live interaction
  - Prevents collisions during conversation
  - Reduce need for echo cancellation
    - Good for small, embedded devices without much CPU
  - Higher sense of presence
  - Allows synchronization for live music
    - Need less than 25 ms *total* delay to synchronize (Carôt 2006)
    - Equivalent to sitting 8 m apart (farther requires a conductor)
- Lower delay in the codec increases range
  - 1 ms = 200 km in fiber

High delay (~250 ms)

Low delay (~15 ms)

Introduction

- No entrenched standard in this space
  - G.722.1C (ITU-T) [40 ms delay, up to 32 kHz]
  - AAC-LD (MPEG) [20-50 ms delay, up to 48 kHz]
  - ULD (Fraunhofer) [< 10 ms delay, up to 48 kHz]
- CELT is already ahead of the competition
  - Delay: Configurable, 1.3 ms to 24 ms, ~8 ms typical
  - Quality (at equivalent rates): Much better than G.722.1C, as good as or better than AAC-LD, better than ULD
  - Flexibility: 24 kbps to 160+ kbps, 32 kHz to 96 kHz, configurable delay, low-complexity mode
  - Freedom: Open source (BSD), no patents

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CELT: "Constrained Energy Lapped Transform"

- Transform codec (MDCT, like MP3, Vorbis)
  - Short windows (~8 ms) → poor frequency resolution
- *Explicitly* code energy of each band of the signal
  - Coarse shape of sound preserved no matter what
- Code remaining details using vector quantization
- Also uses pitch prediction with a time offset
  - Similar to linear prediction used by speech codecs
  - Helps compensate for poor frequency resolution

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  - **"Lapped Transform"**
  - "Constrained Energy"
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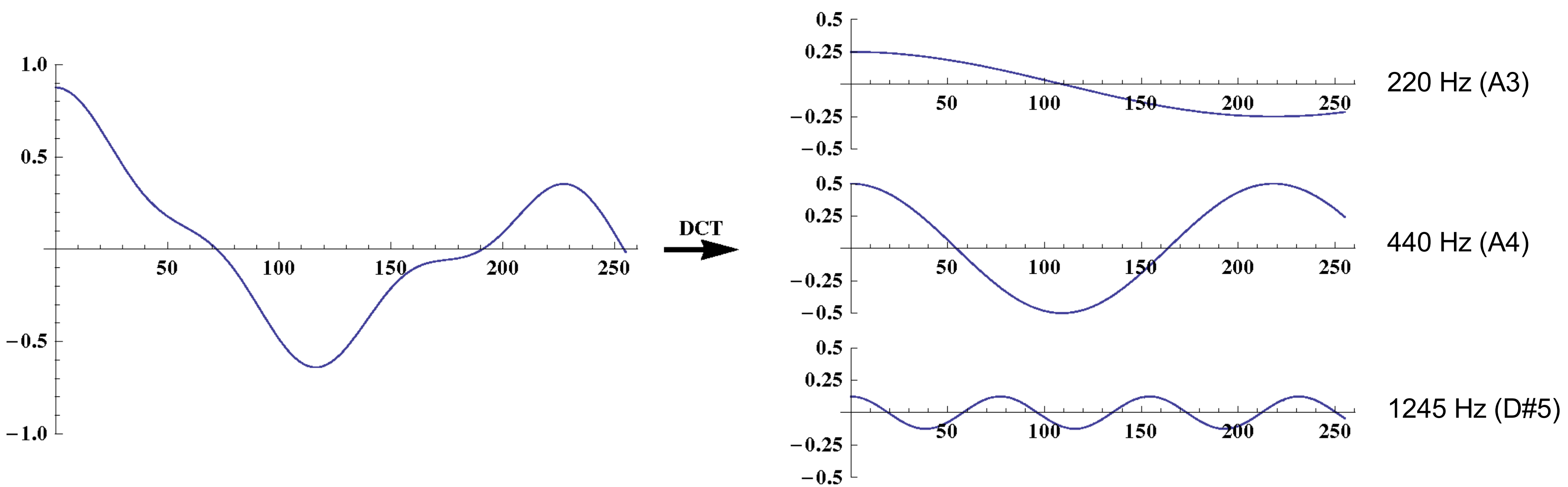
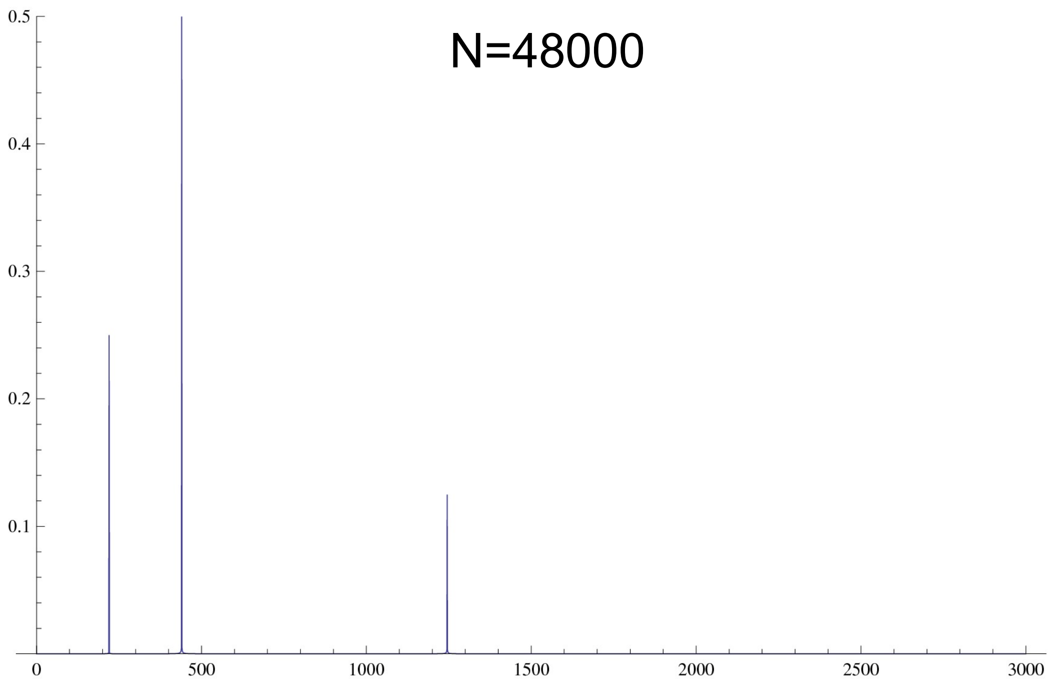
"Lapped Transform" Time-Frequency Duality

- *Any* signal can be represented as a weighted sum of cosine curves with different frequencies
- The Discrete Cosine Transform (DCT) computes the weights for each frequency



"Lapped Transform" Discrete Cosine Transform

- The "Discrete" in DCT means we're restricted to a finite number of frequencies
  - As the transform size gets smaller, energy "leaks" into nearby frequencies (harder to compress)



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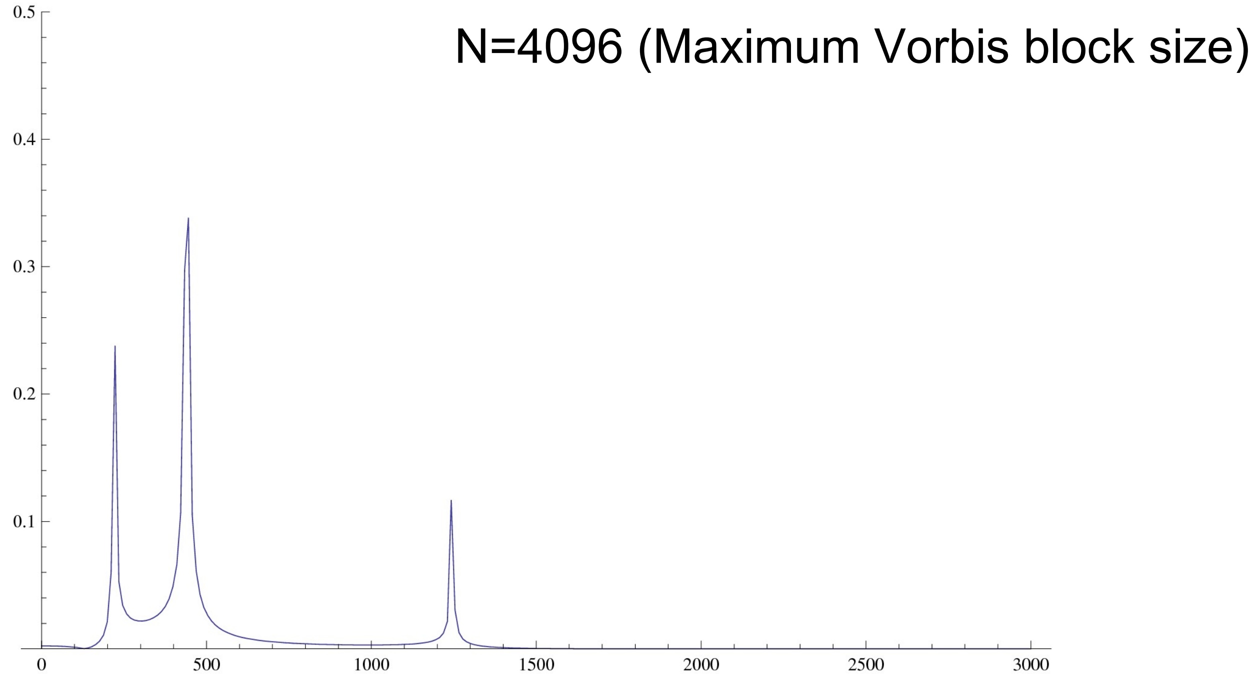

"Lapped Transform"
Discrete Cosine Transform

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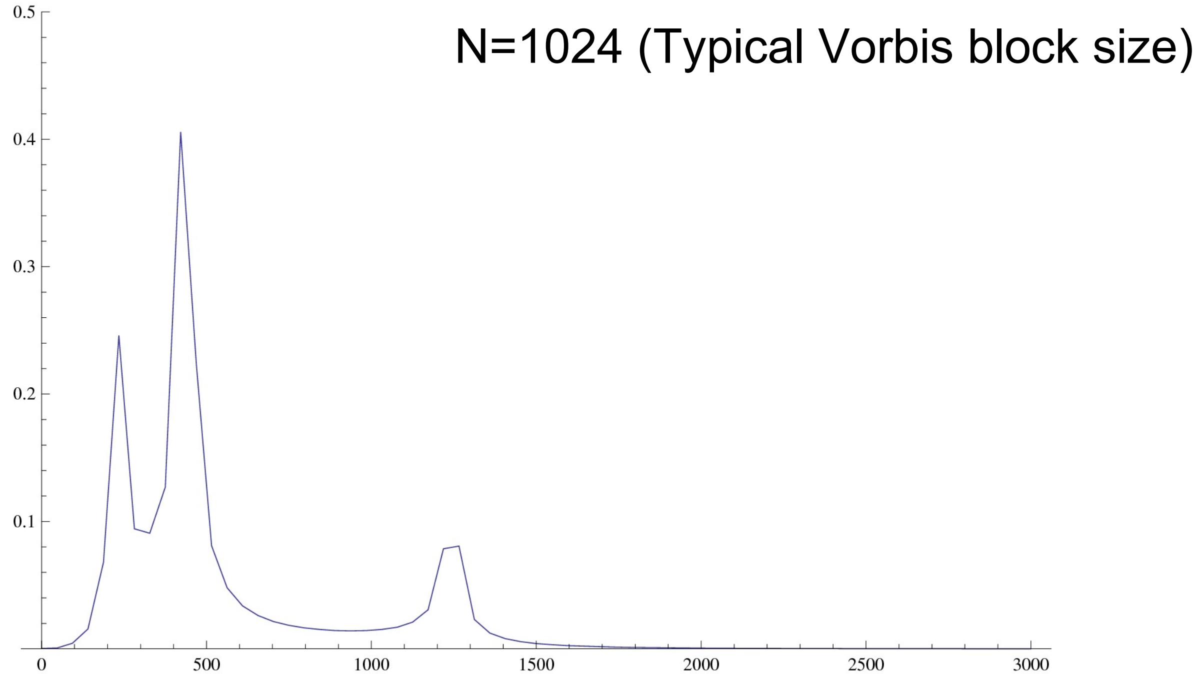

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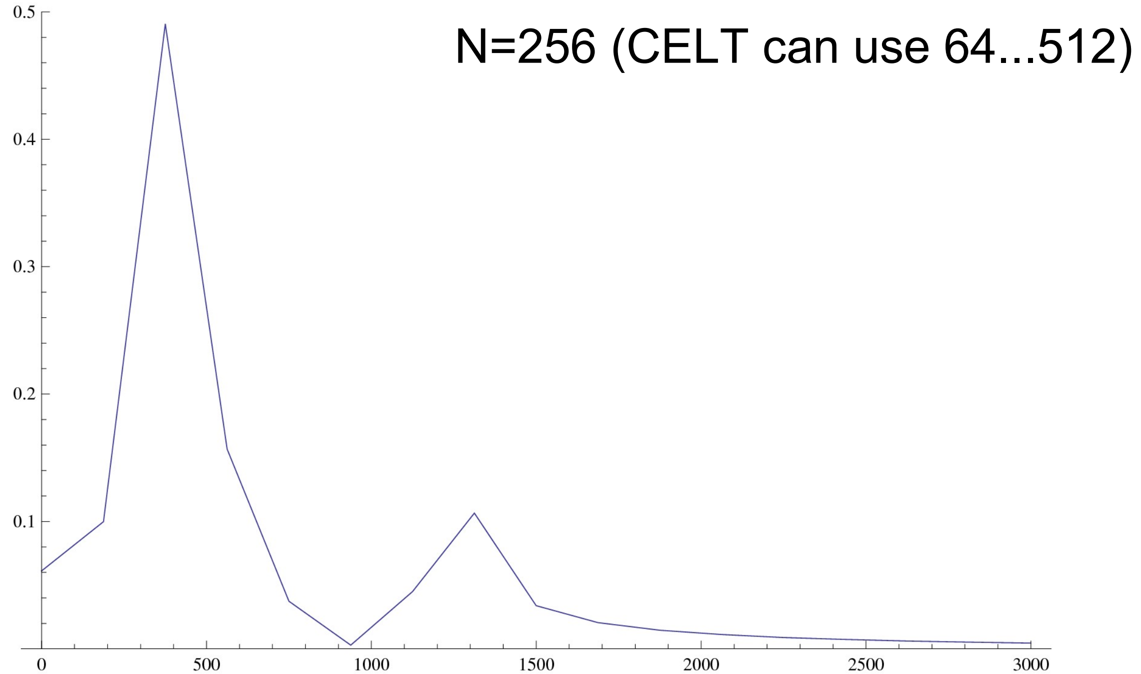
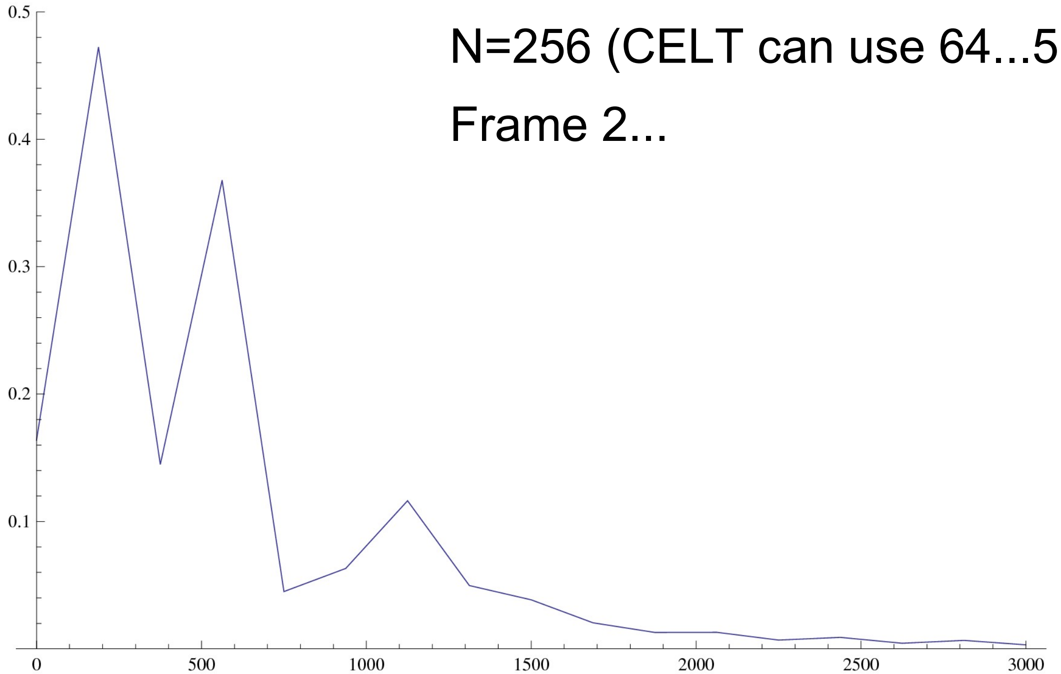


"Lapped Transform" Discrete Cosine Transform

- The "Discrete" in DCT means we're restricted to a finite number of frequencies
  - As the transform size gets smaller, energy "leaks" into nearby frequencies (unstable over time)



N=256 (CELT can use 64...512)

Frame 2...

"Lapped Transform" Discrete Cosine Transform

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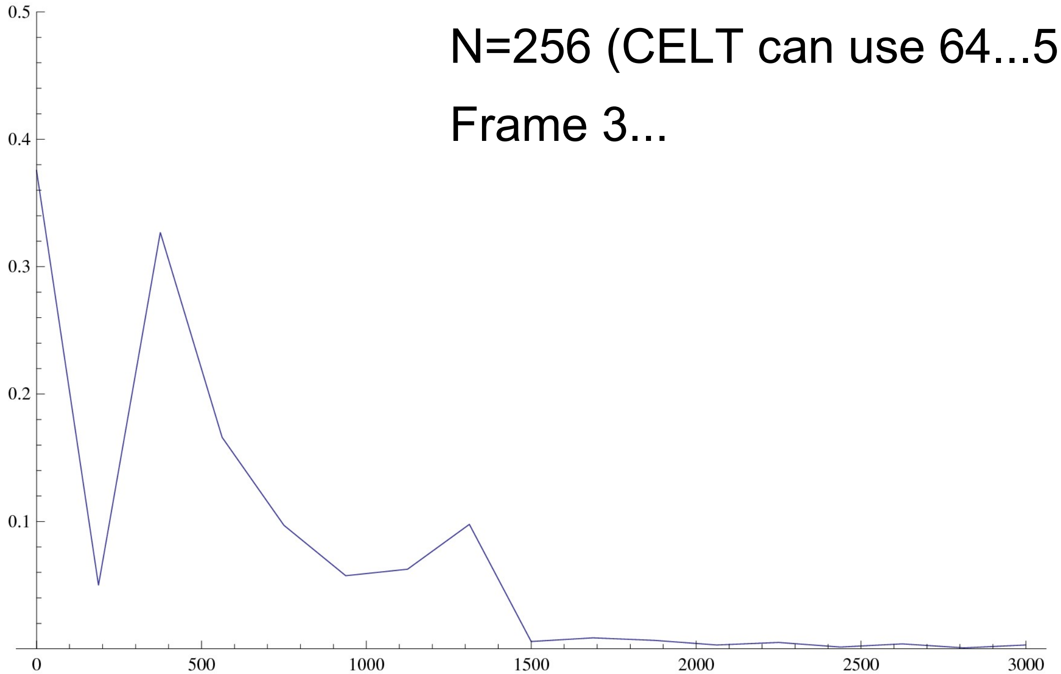
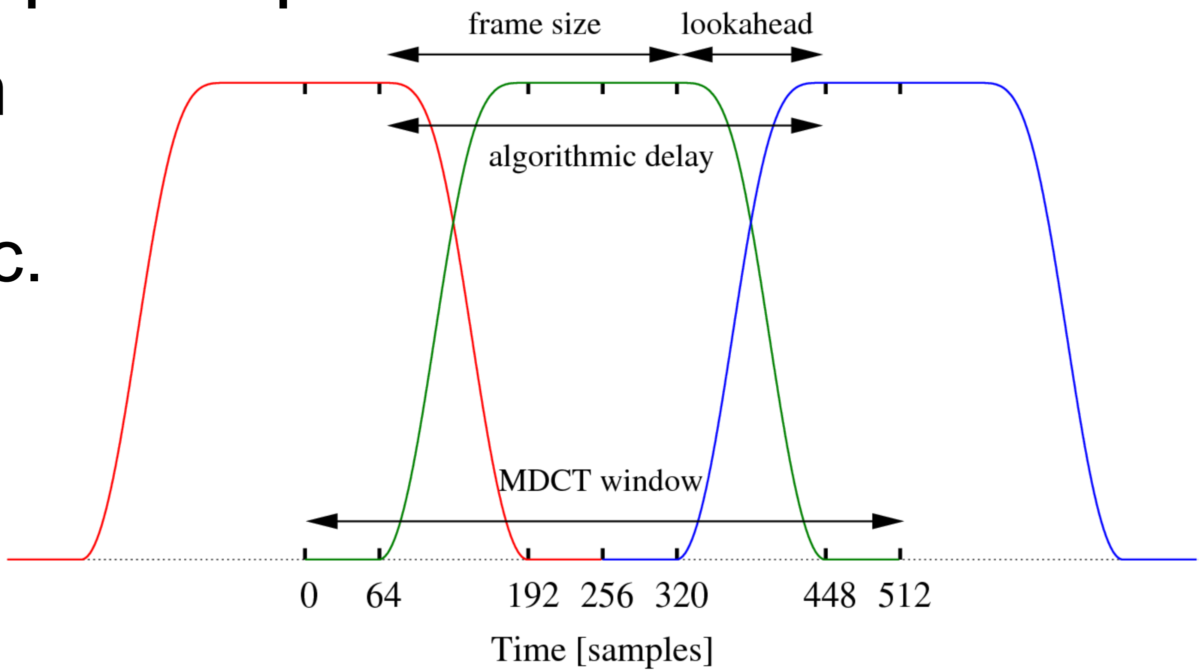


N=256 (CELT can use 64...512)

Frame 3...

"Lapped Transform" Modified DCT

- The normal DCT causes coding artifacts (sharp discontinuities) between blocks, easily audible
- The "Modified" DCT (MDCT) uses a decaying window to overlap multiple blocks
  - Same transform used in MP3, Vorbis, AAC, etc.
  - But with much smaller blocks, less overlap



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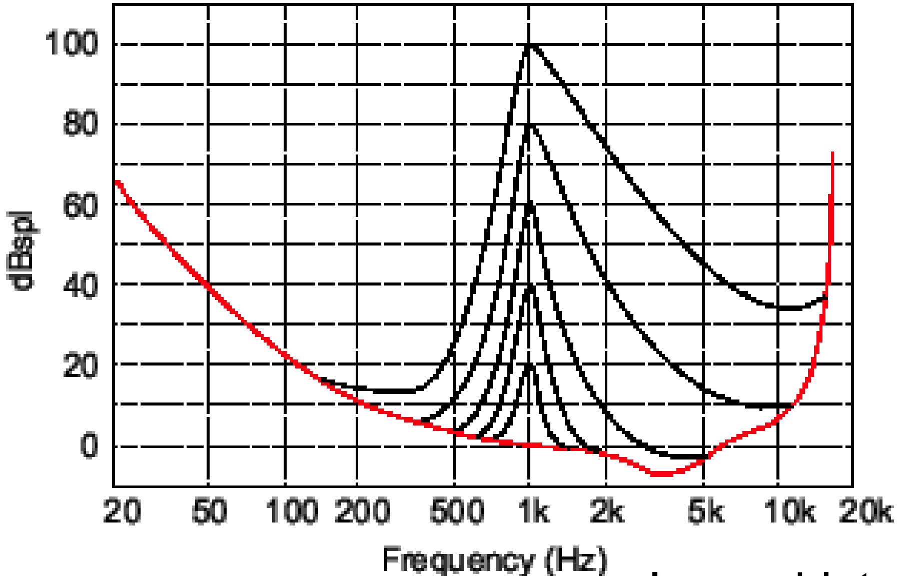
"Constrained Energy" Critical Bands

- The human ear can hear about 25 distinct "critical bands" in the frequency domain
  - Psychoacoustic masking within a band is much stronger than between bands

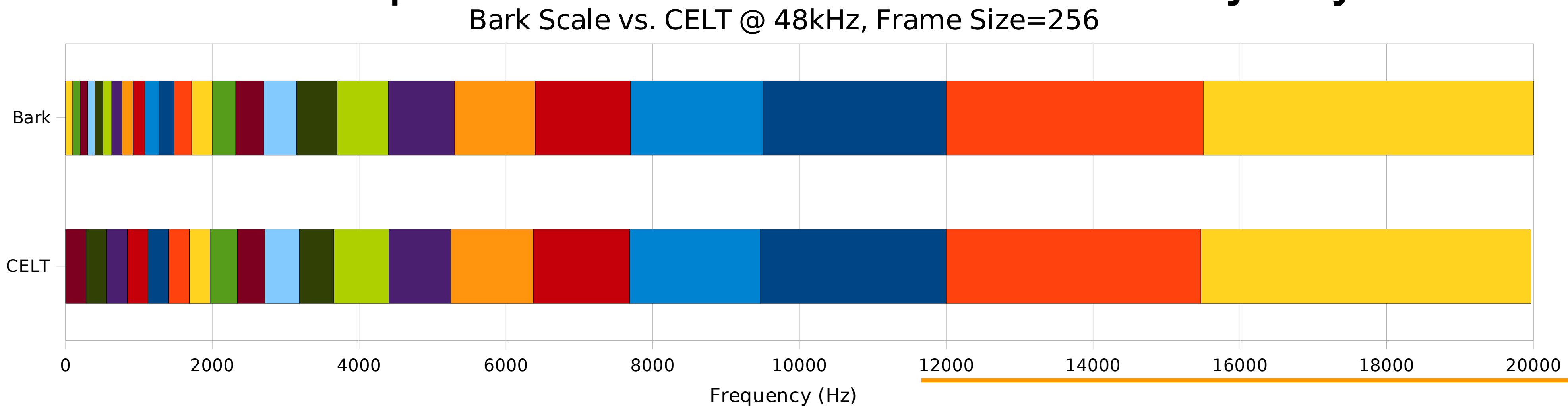


Threshold of detection in the presence of masker at 1kHz with a bandwidth of 1 critical band and various levels.

Image blatantly stolen from
http://www.tonmeister.ca/main/textbook/node331.html

"Constrained Energy" Critical Bands

- Group MDCT coefficients into bands approximating the critical bands (Bark scale)
  - We limit bands to contain at least 3 coefficients to minimize per-band overhead
  - Insufficient frequency resolution for all the bands
  - But we spend most of our bits on LFs anyway



"Constrained Energy" Coding Band Energy

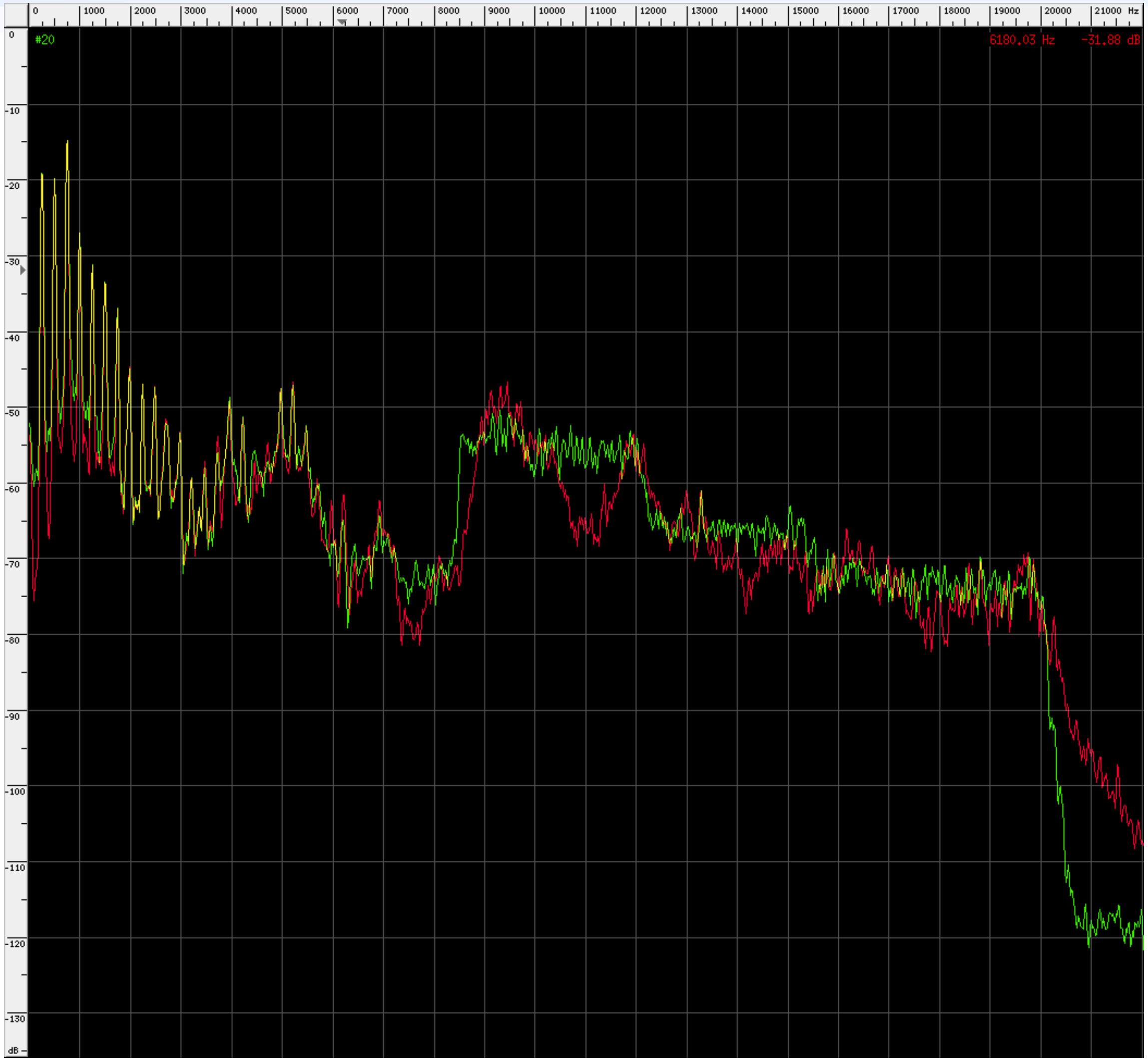
- Most important psychoacoustic lesson learned from Vorbis:

Preserve the energy in each band

- Vorbis does this implicitly with its "floor curve"
- CELT codes the energy explicitly
  - Coarse energy (6 dB resolution), predicted from previous frame and from previous band
    - Prediction saves 28 bits/frame, 5.6 kbps with 5 ms frames
  - Fine energy, improves resolution where we have available bits, not predicted

"Constrained Energy" Coding Band Energy

- CELT (green) vs original (red)
  - Notice the quantization between 8.5 kHz and 12 kHz
  - Speech is intelligible using coarse energy alone (~9 kbps for 5.3 ms frame sizes)



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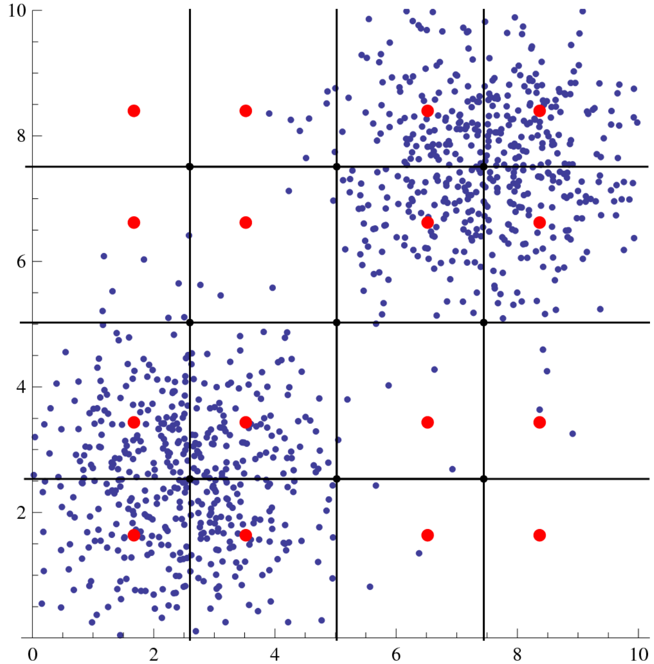
Coding Band Shape

- After normalizing, each band is represented by an *N*-dimensional unit vector
  - Point on an *N*-dimensional sphere
  - Describes "shape" of energy within the band
- Code this shape using two pieces:
  - An *adaptive codebook* using previously decoded signal content to predict the current frame
  - A *fixed codebook* to handle the part of the signal that can't be predicted (the "innovation")
- Latter uses *vector quantization*

Coding Band Shape
Vector Quantization

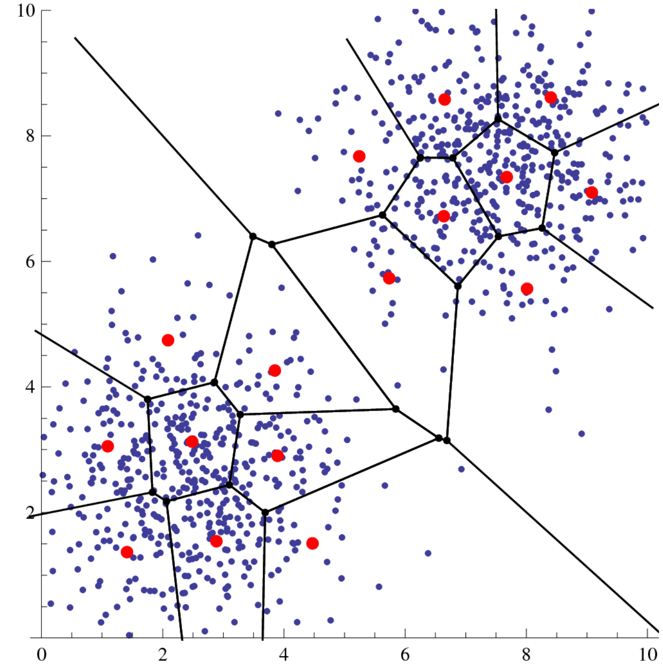
- Approximates a multidimensional distribution with a finite number of codewords (vectors)

Scalar Quantization (2 bits/dim)



RMS error = 0.89

Vector Quantization (2 bits/dim)

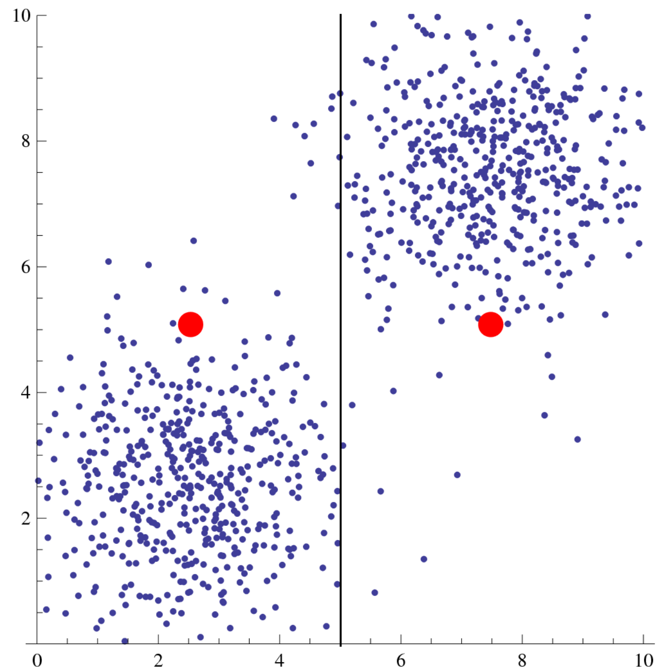


RMS error = 0.71
(20% better)

Coding Band Shape
Vector Quantization

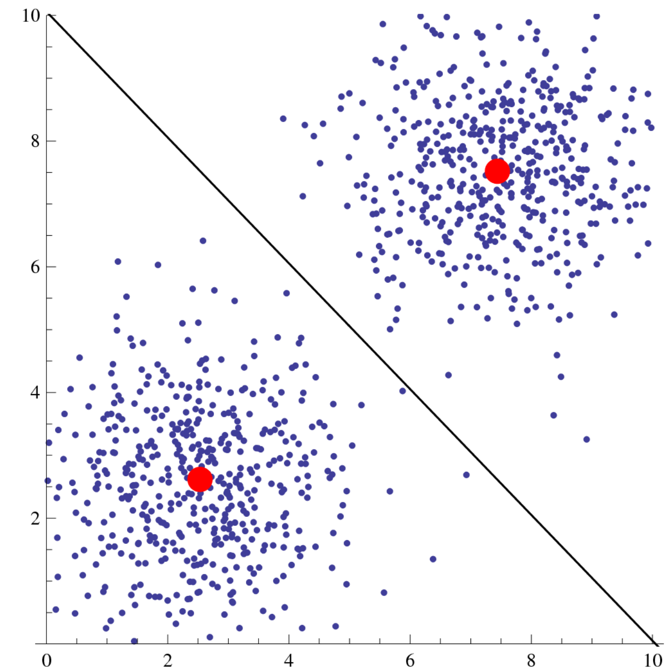
- Easily scales to less than 1 bit per dimension (very important for HF bands: 50-200 dims)

Scalar Quantization (0.5 bits/dim)



RMS error = 2.93

Vector Quantization (0.5 bits/dim)



RMS error = 1.63
(44% better)

Coding Band Shape
Algebraic Vector Quantization

- CELT requires a *lot* of codebooks
  - Every band can have a different # of dimensions
  - Exact number of bits available for each band varies from packet to packet
- CELT requires *large* codebooks
  - Exponential in # of dimensions: 50 dims at 0.6 bits/dim. requires over a billion codebook entries
  - We couldn't even store one codebook that large
  - And even if we could, it'd take forever to search
- But we have uniformly distributed unit vectors

Coding Band Shape
Algebraic Vector Quantization

- Use a regularly structured, algebraic codebook: Pyramid Vector Quantization (Fischer, 1986)
  - We want evenly distributed points on a sphere
    - Don't know how to do that for arbitrary dimension
  - Use evenly distributed points on a pyramid instead
- For *N* dimensional vector, allocate *K* "pulses"
- Codebook consists vectors with integer coordinates whose magnitudes sum to *K*

$$S(N,K) = \{\mathbf{y} \in \mathbb{Z}^N : \sum_{i=1}^{N} |y_i| = K\}$$

Coding Band Shape
Pyramid Vector Quantization

- PVQ codebook has a fast enumeration algorithm
  - Converts between vector and integer codebook index
  - $O(N+K)$ (lookup table, muls) or simpler $O(NK)$ (adds)
  - Latter great for embedded processors, often faster
- Fast codebook search algorithm: $O(N \cdot \min(N,K))$
  - Divide by L_1 norm, round down: at least K-N pulses
  - Place remaining pulses (at most N) one at a time
- Codebooks larger than 32 bits
  - Split the vector in half and code each half separately

Coding Band Shape
Pitch Prediction

- Short block sizes → poor frequency resolution
  - Speech/music have periodic, stationary content
  - Can't represent the period accurately via short MDCT
- Pitch prediction compensates for poor resolution
  - Search the past 1024 decoded samples in the time domain, code the offset of the best match
    - Resolution equal to the sampling rate
    - Range (48 kHz, FS=256): $\frac{48000}{1024}$ to $\frac{48000}{384} = 46.875\,\mathrm{Hz}$ to $125\,\mathrm{Hz}$
  - Apply an MDCT to convert to the freq. domain
  - Confine prediction to bands below 8kHz

Coding Band Shape Mixing

- Scale each band of pitch MDCT to unit norm: **p**
- Compute a *pitch gain*, $g_a \in [0...1]$ for each band
- Mix with the fixed codebook vector **y** via

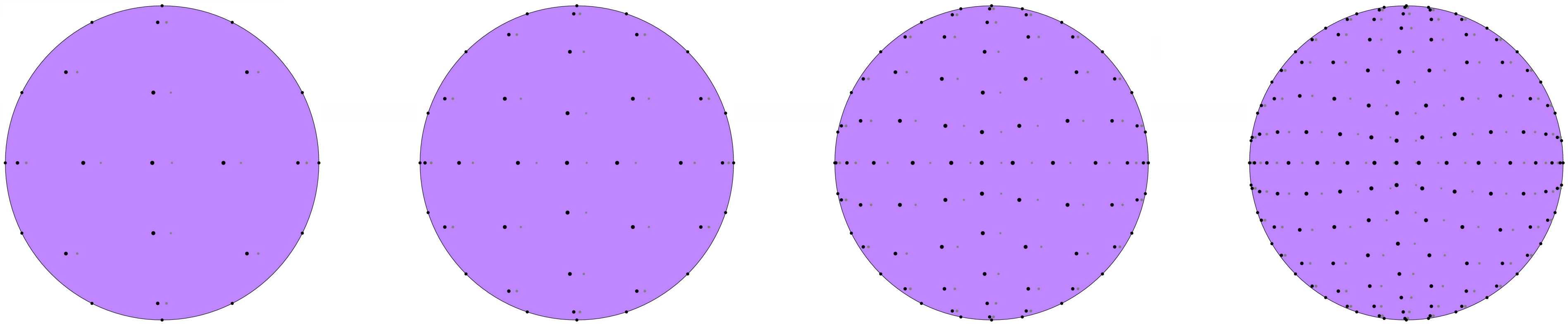
$$\tilde{\mathbf{x}} \triangleq \tilde{g}_a \mathbf{p} + g_f \mathbf{y}$$

- Output must have unit norm, so g_f is completely determined:

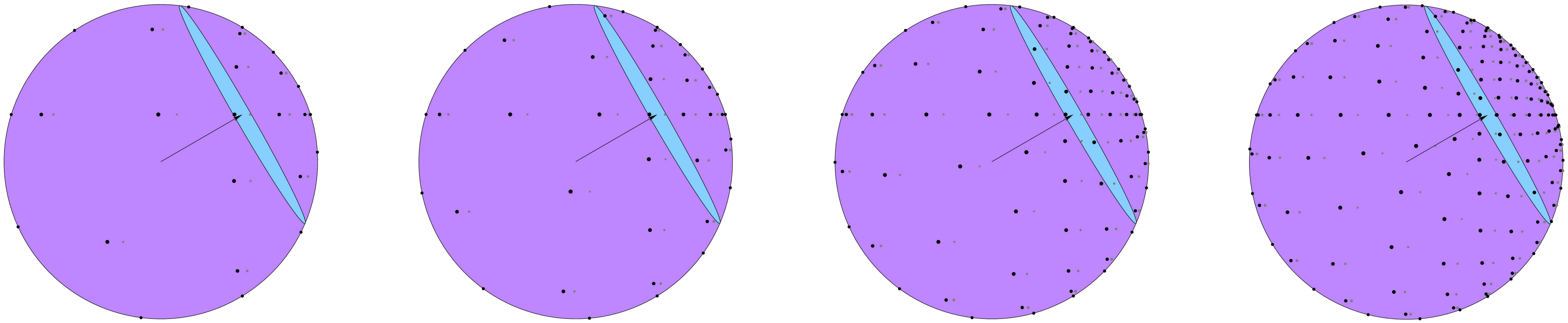
$$g_f \triangleq \frac{\sqrt{\tilde{g}_a^2 (\mathbf{y}^T \mathbf{p})^2 + \mathbf{y}^T \mathbf{y} (1 - \tilde{g}_a^2)} - \tilde{g}_a \mathbf{y}^T \mathbf{p}}{\mathbf{y}^T \mathbf{y}}$$

Coding Band Shape
Adaptive vs. Fixed Codebooks

Before applying pitch prediction



5.25 bits | 6.04 bits | 7.19 bits | 8.01 bits

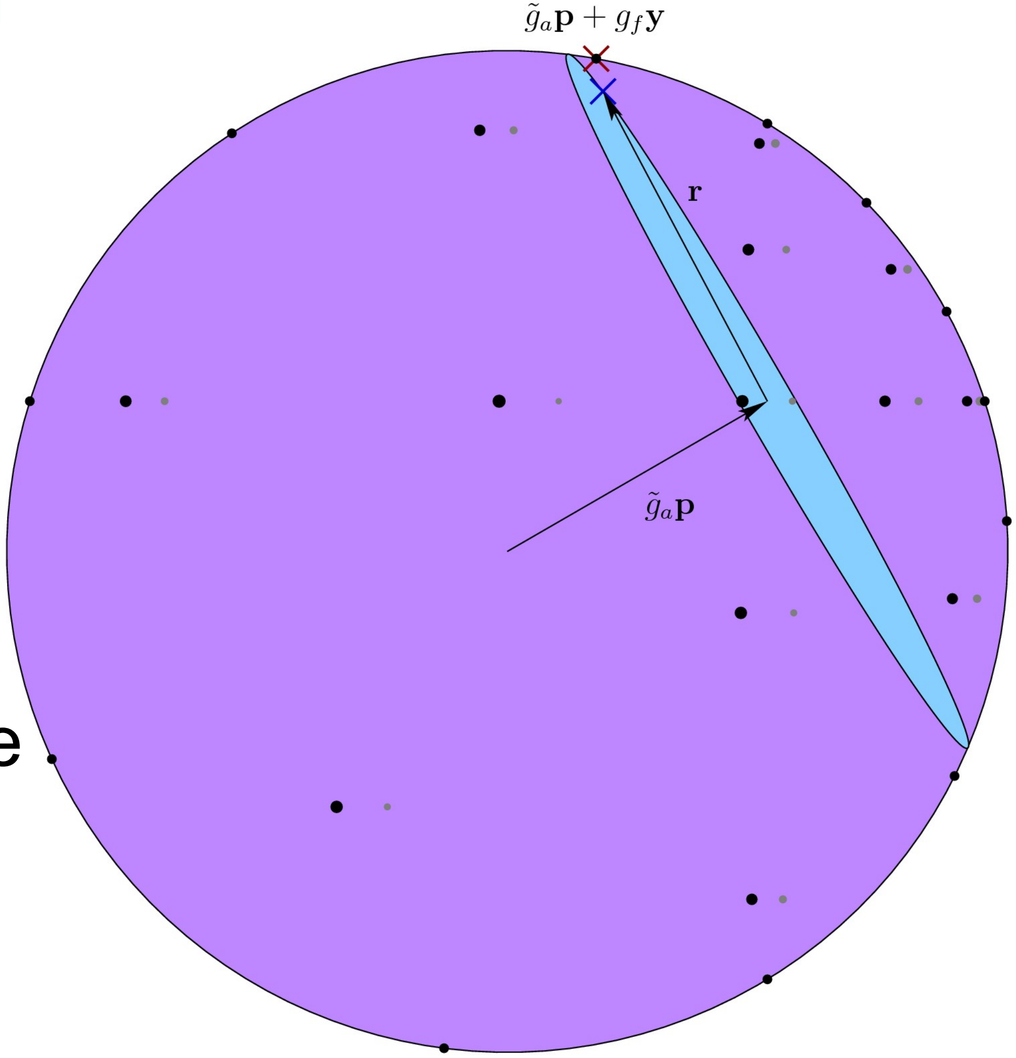


After applying pitch prediction

- Tried stronger adaptation, but required more CPU for no perceptible gain

Coding Band Shape
Mixing

- Ideal g_a chosen so that residual **r** = **x**-g_a**p** orthogonal to **p**
- Quantizing g_a means orthogonality not exact
  - Used to use basic VQ to code all g_a values at once
  - Now use 1 bit per band, g_a is either 0 or 0.9

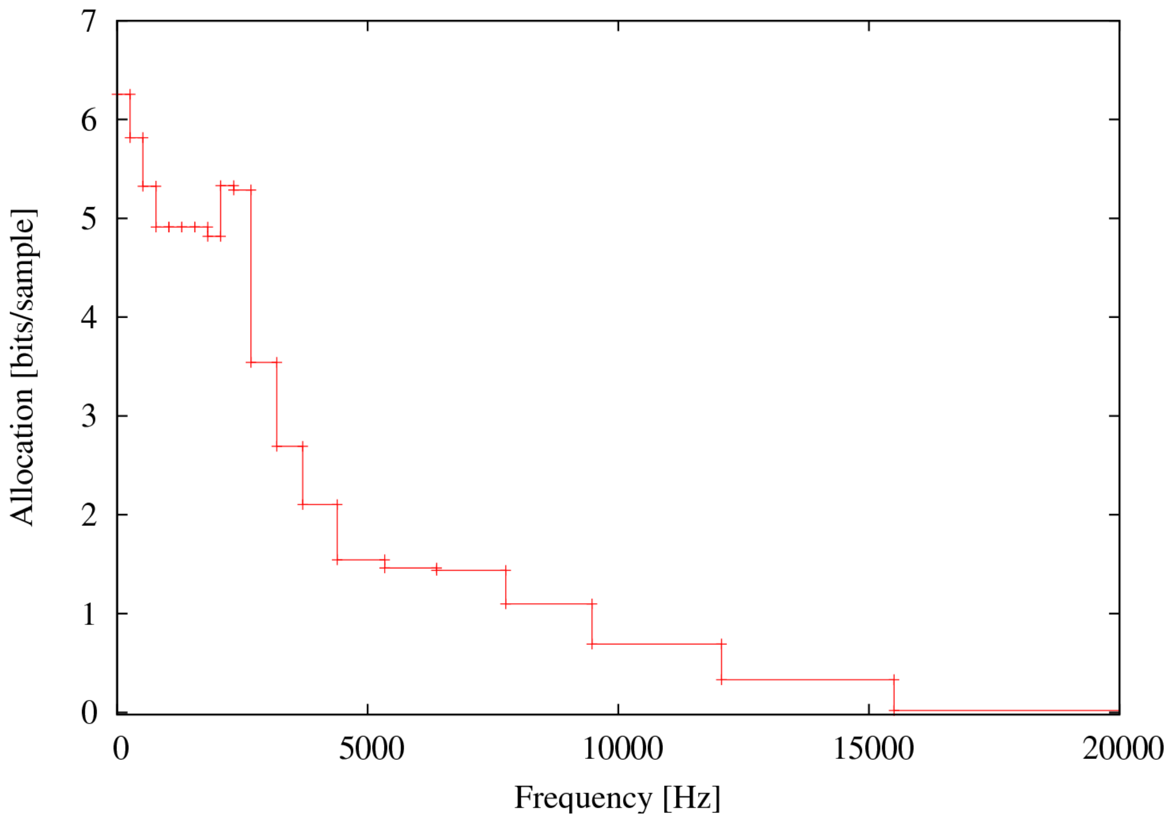


Rate Allocation

- Only CBR supported
  - VBR requires buffering, and buffering means delay
  - User specifies the exact number of bytes to encode each packet into
  - Can change from packet to packet, to adapt to channel statistics
- Only a few things are variable-sized
  - Coarse energy (entropy coded)
  - Pitch parameters (can be omitted if not useful)
  - PVQ codewords over 32 bits (rare)

Rate Allocation

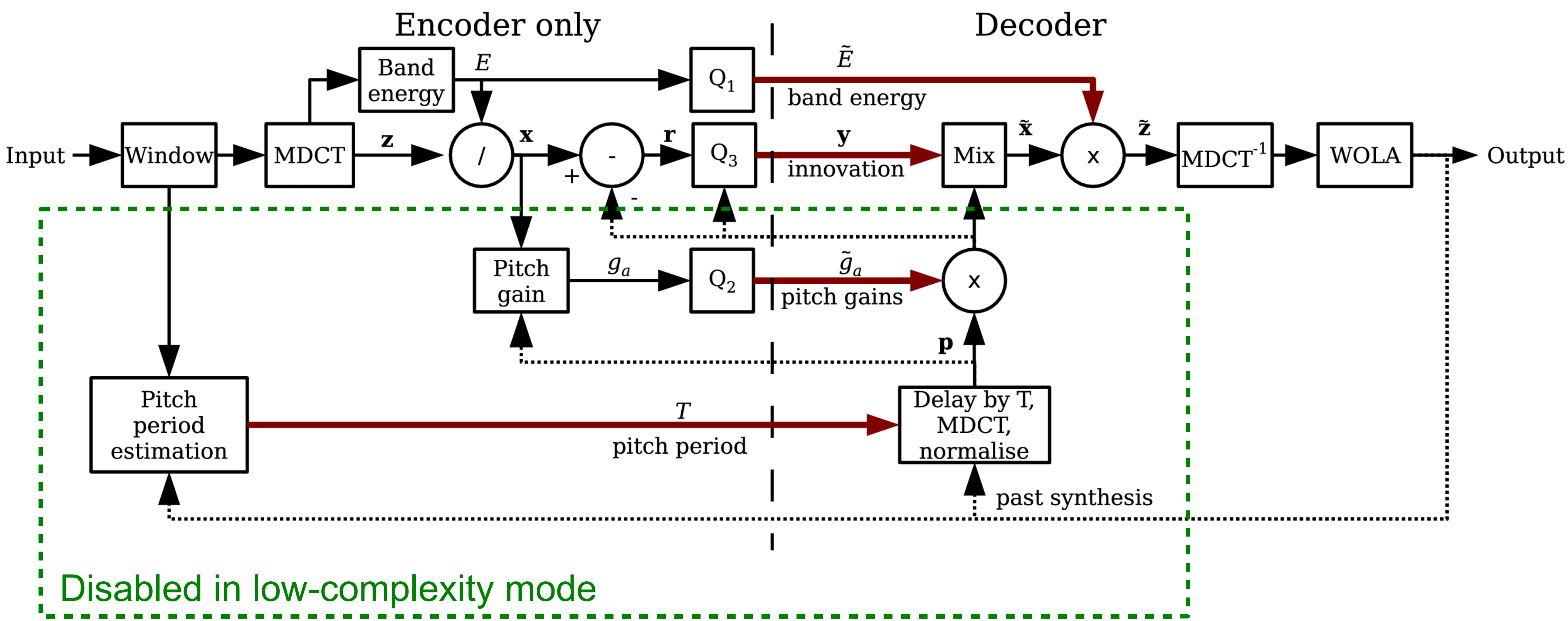
- Each band's share of available bits is *fixed*
  - CELT transmits *no* side information for allocation
  - Equivalent to modeling within-band masking
    - "Signal-to-mask" ratio for each band is roughly constant
  - Ignores inter-band masking and tone vs. noise effects



Psychoacoustic Tricks

- Avoiding "birdie" artifacts
  - *K* may be small, giving a sparse spectrum > 8 kHz
  - Use *spectral folding*, a scaled copy of lower-frequency MDCT coefficients, in place of **p**
    - Acts as a cheap source of time-localized noise
    - Mix using a small value for g_a (a function of K)
- Avoiding "pre-echo" artifacts
  - When a strong transient is detected, split the frame and do a smaller MDCT on each piece
  - Interleave the results and continue as normal

Block Diagram



Future Work

- Freeze bitstream format
  - No side information for allocation means *many* details of the encoding become normative
- Dynamic rate allocation
  - Hard to do psychoacoustic analysis without delay
  - Almost any per-band overhead uses a lot of bits
- Improve stereo coupling
  - Currently using PVQ to handle phase vs. magnitude
- Improve pitch prediction

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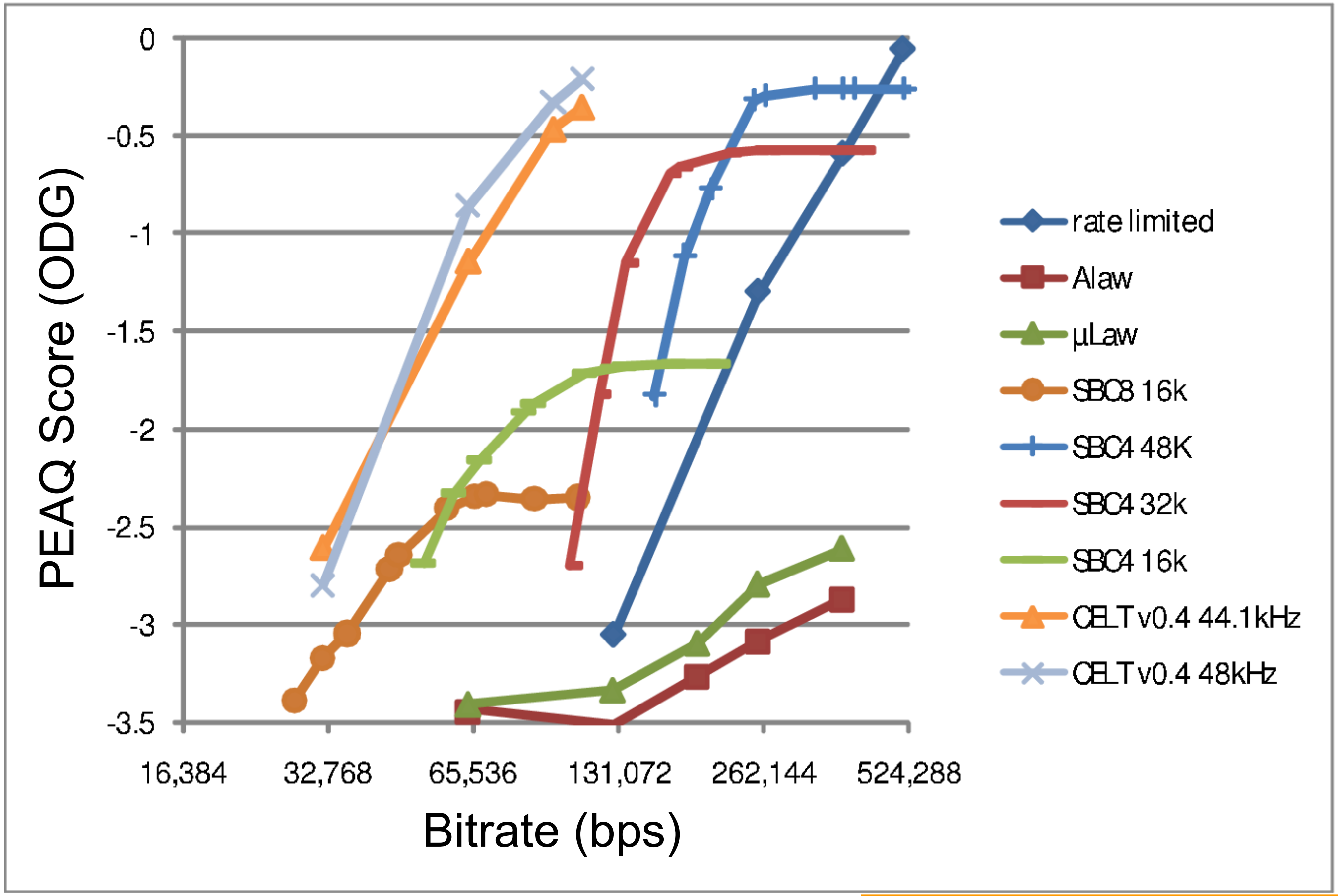
CELT vs. The Competition

- Results from Dr. Christian Hoene for ITU-T Workshop last September



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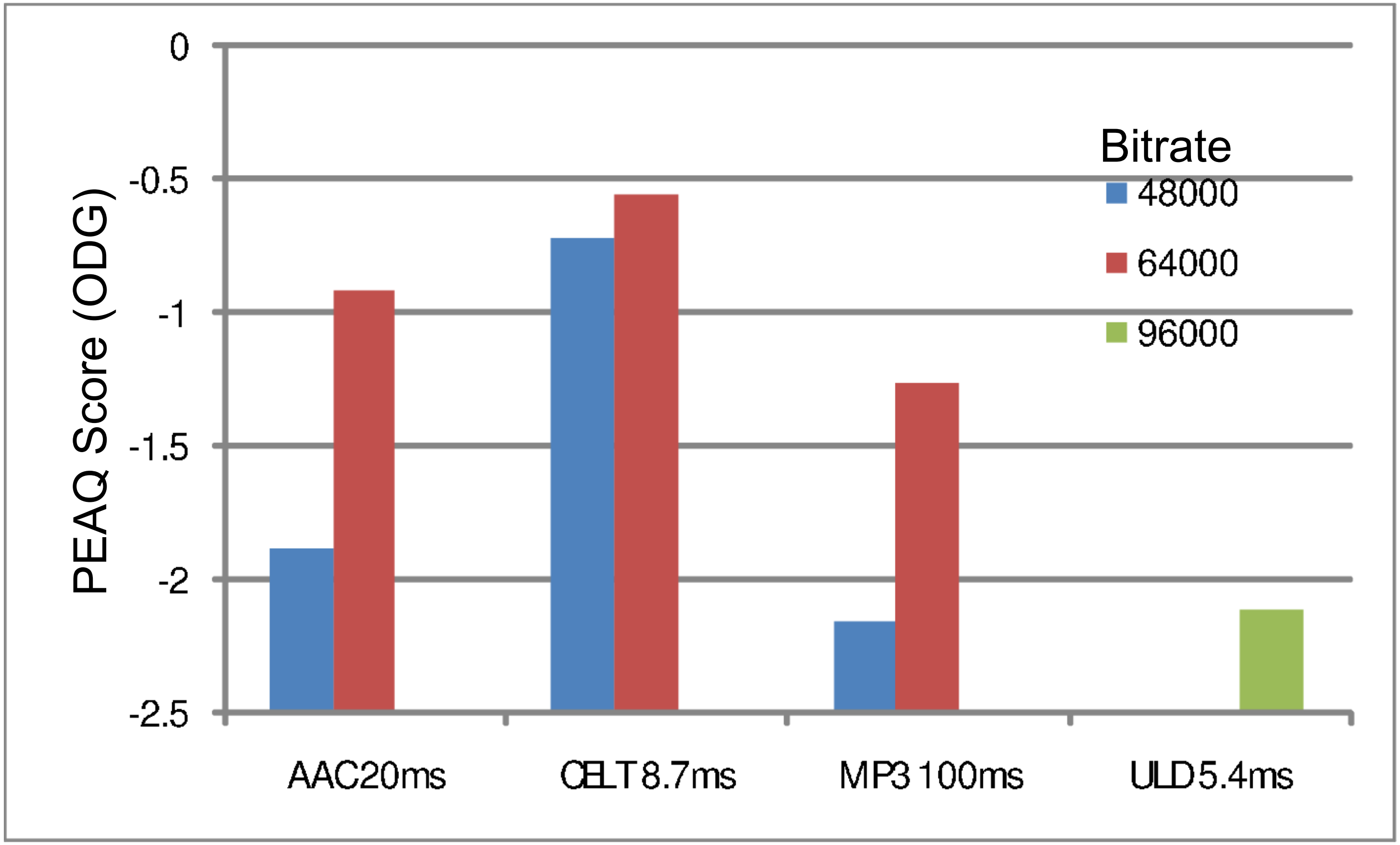
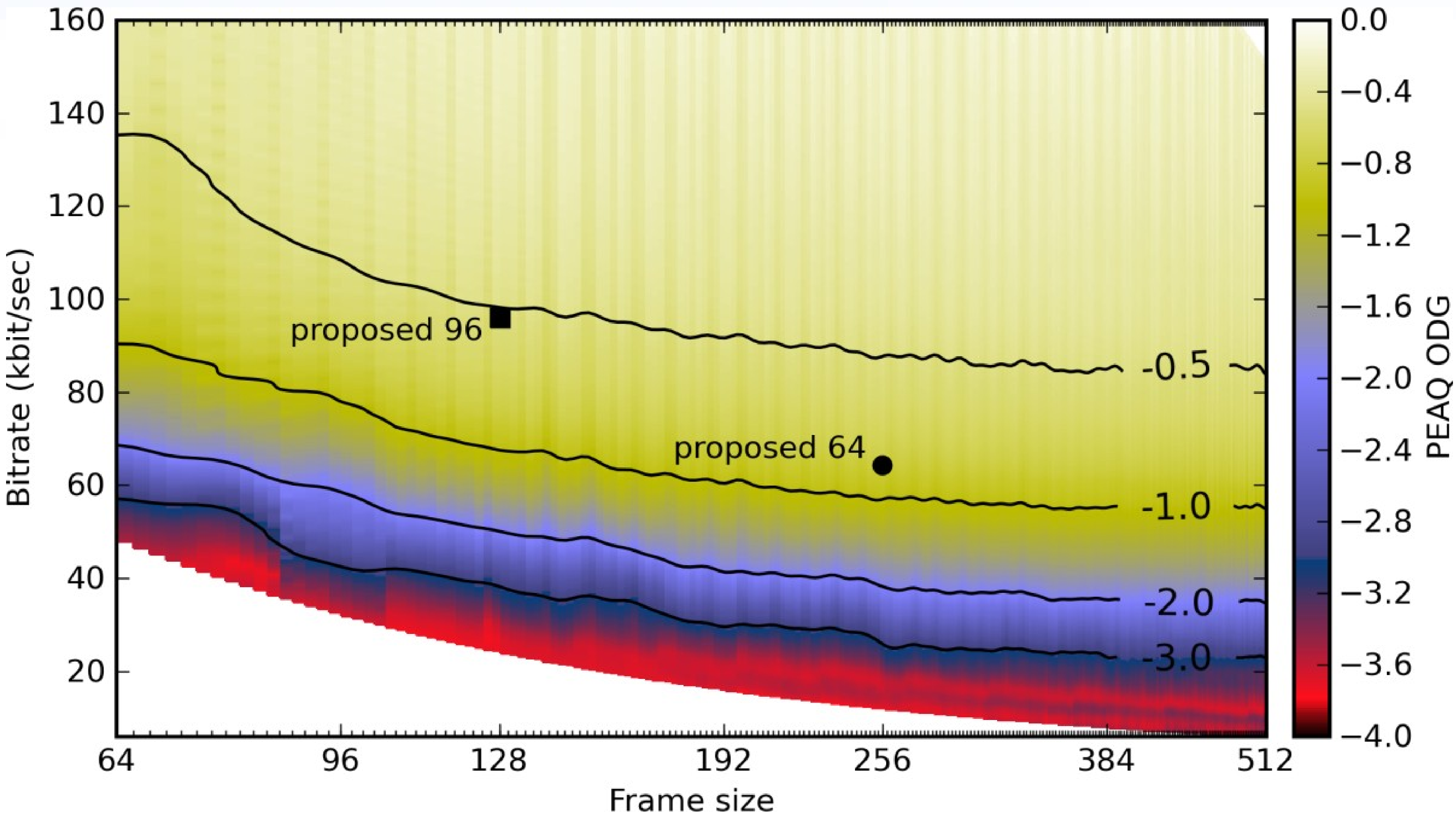
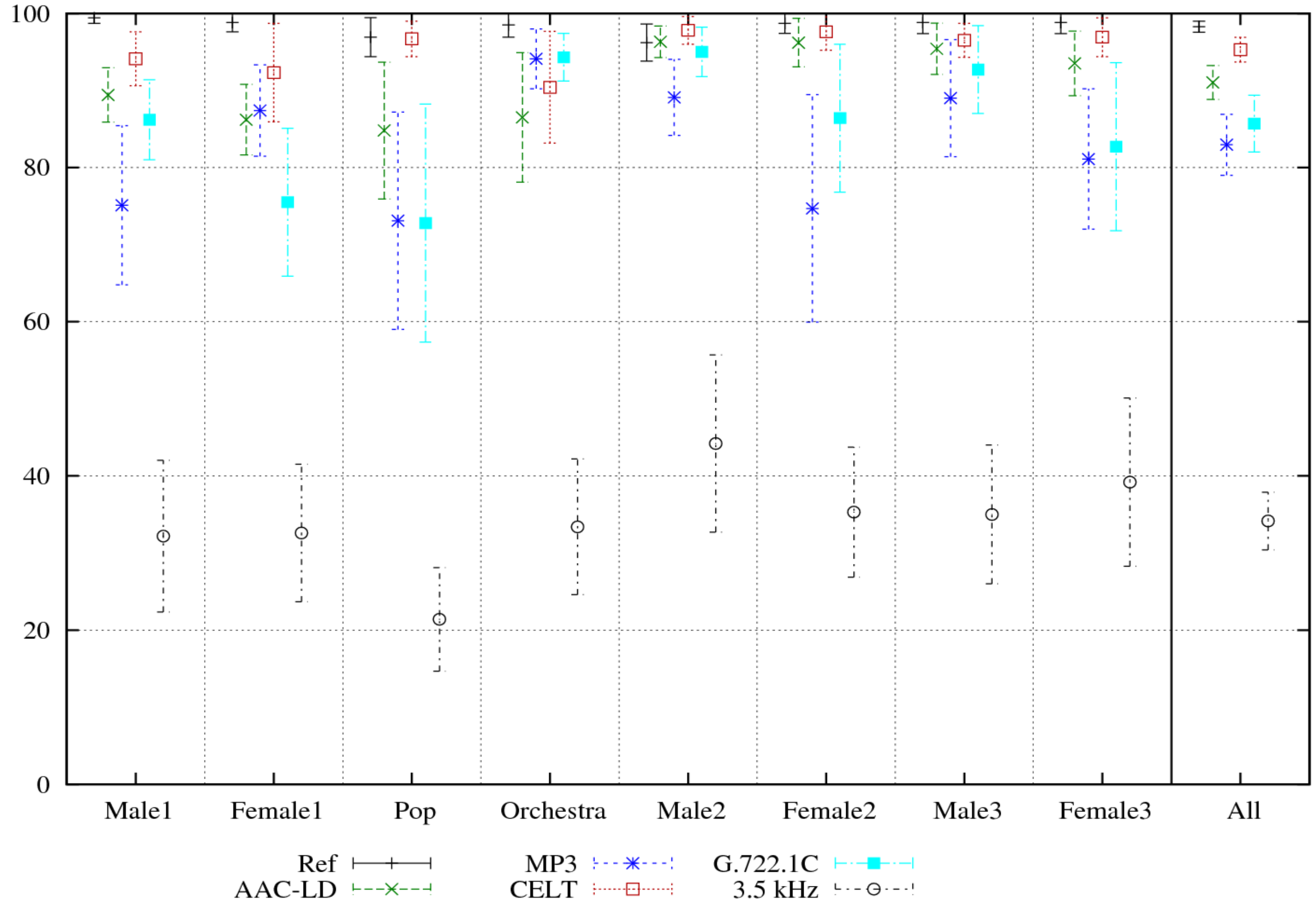


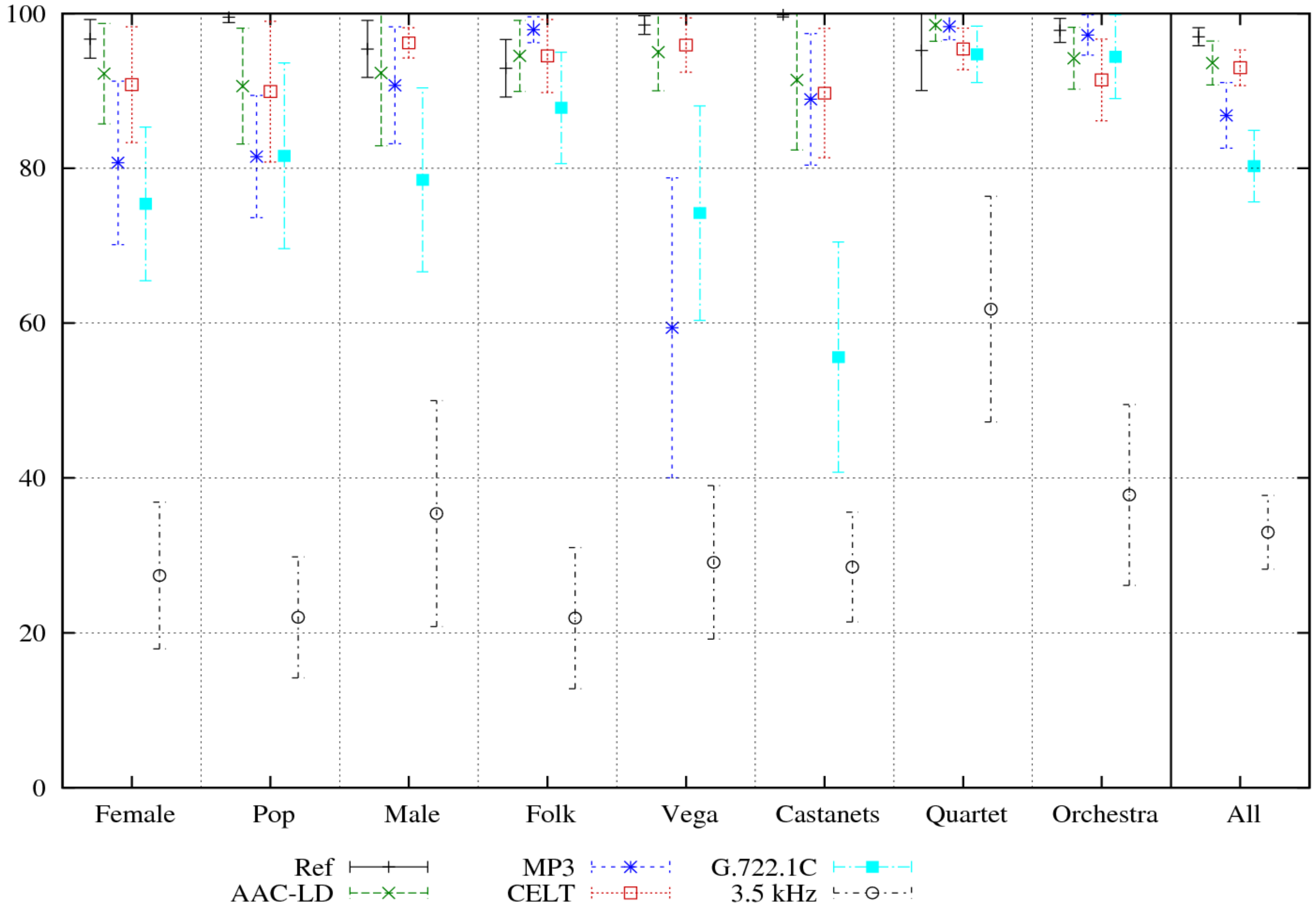
Quality vs. Delay (v0.5, no pitch)



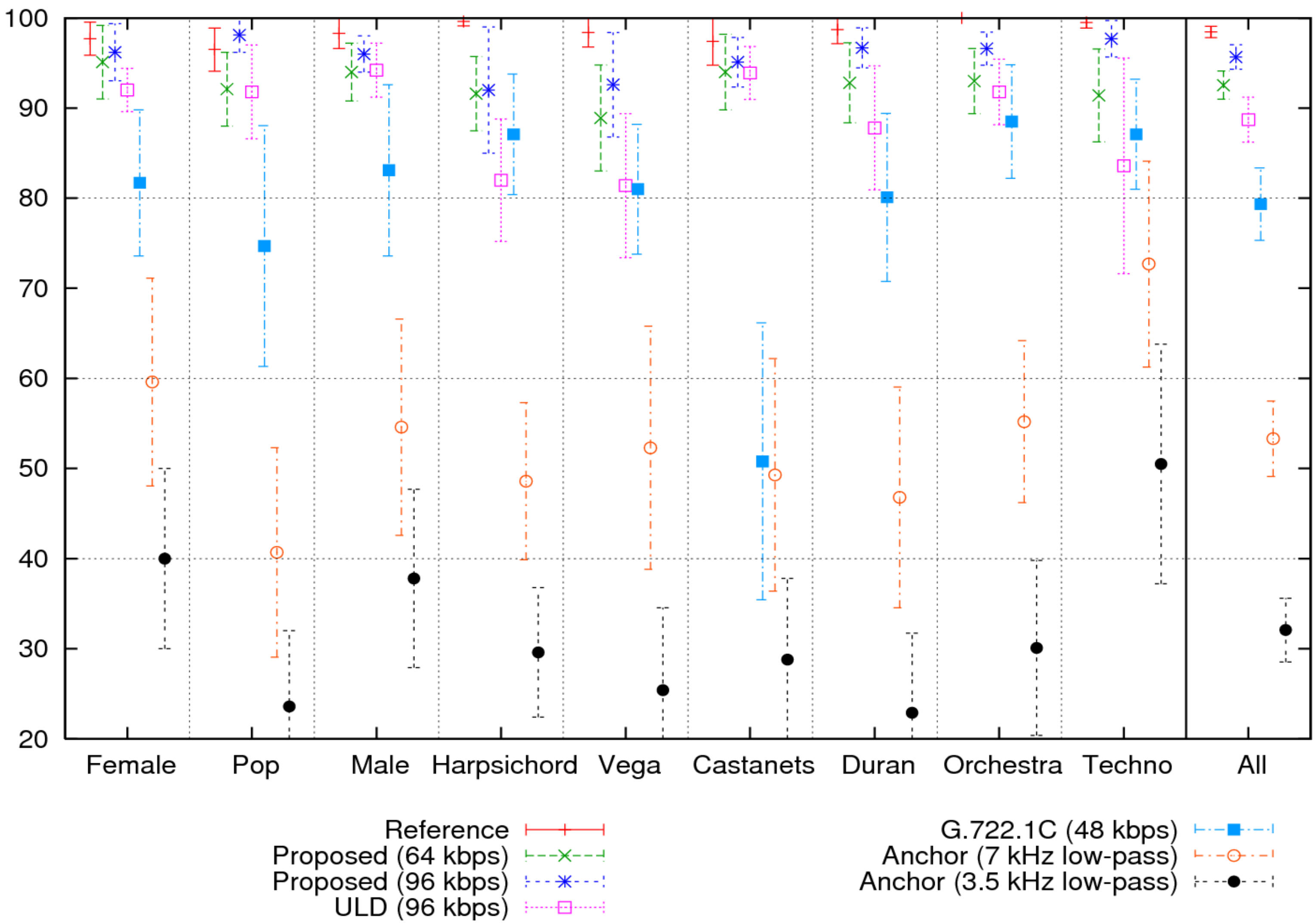
Listening Tests – 48 kbps (v0.3.2, with pitch)



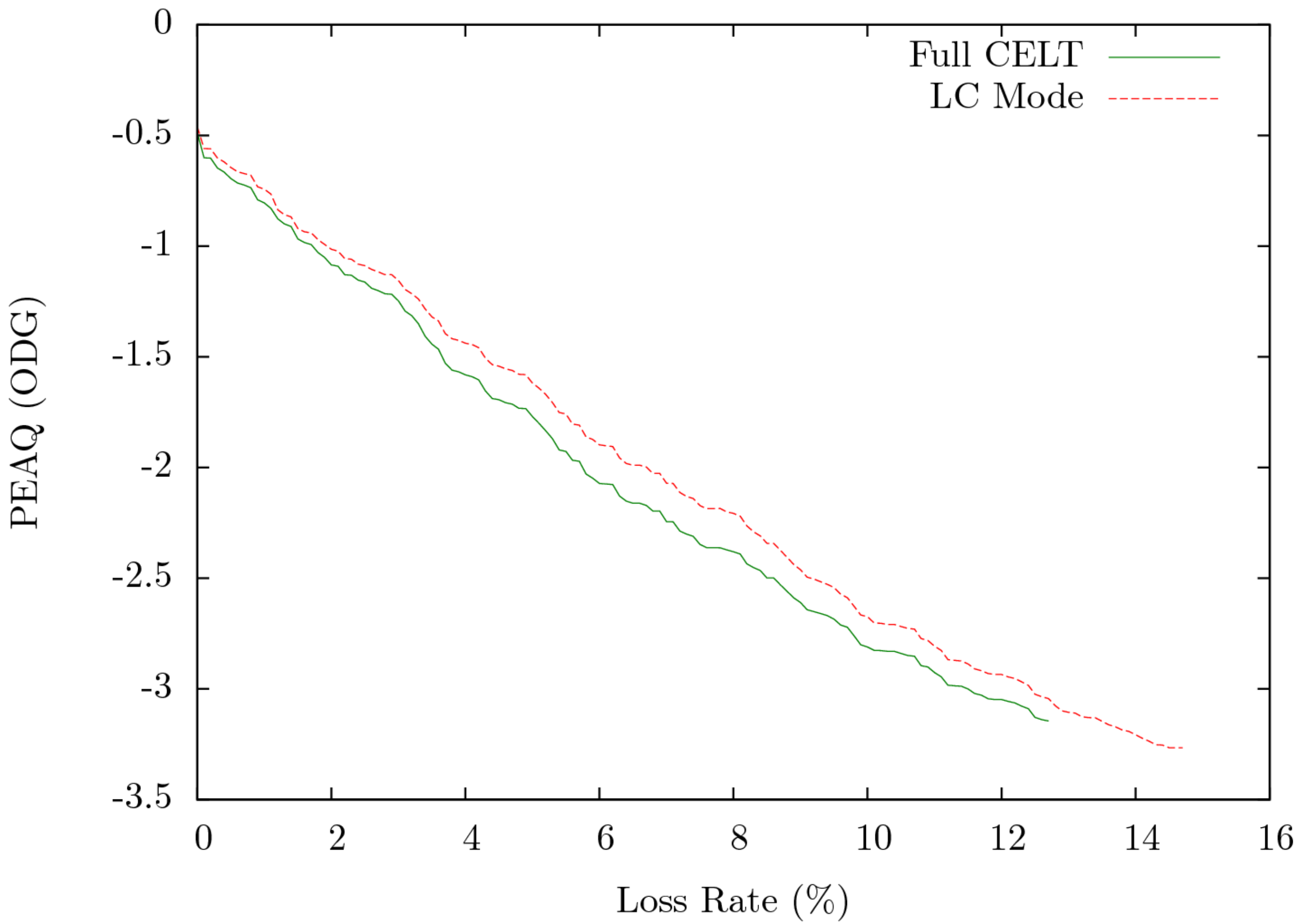
Listening Tests – 64 kbps (v0.3.2, with pitch)



Listening Tests – LC Mode (v0.5, no pitch)



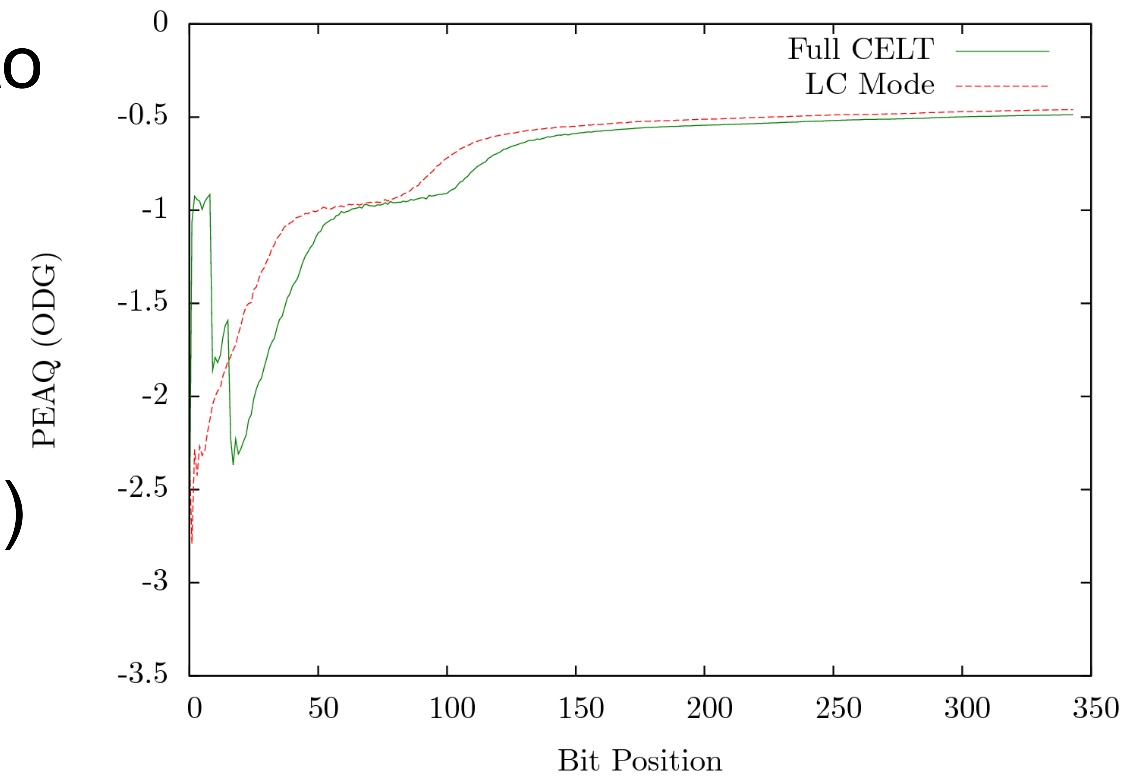
Packet Loss



0
-0.5
-1
-1.5
-2
-2.5
-3
-3.5
PEAQ (ODG)
0 2 4 6 8 10 12 14 16
Loss Rate (%)
Full CELT
LC Mode

Bit Errors vs. Position

- Wireless transmission means individual bits can be corrupted without causing packet loss
  - Quality loss due to bit errors varies with location in a packet
  - Trellis Coded Modulation (TCM) can give better protection to earlier bits



Example

- Original file (706 kbps)
- Scalar Quantization (227 kbps, SNR=20.9 dB)
  - 5.15 bits per sample
- Encoded with CELT (64.8 kbps, SNR=20.9 dB)
  - 1.47 bits per sample (Frame Size=256)
- Scalar Quantizaion Residual (amplified 2×)
- CELT Residual (amplified 2×)
  - Throw away information only where it's masked by something else in the signal

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libcelt

- *Extremely* light-weight fixed-point impl.

	Full CELT	***LC mode***
Enc/Dec State (each)	4.5 kB	0.5 kB
Required Stack	11-13 kB	7 kB
Table Data (ROM)	5.5 kB	5.5 kB
CPU (TI-C55x DSP)	60 MIPS (enc)+ 30 MIPS (dec)	~30 MIPS (enc)+ ~15 MIPS (dec)

- Also has a floating-point implementation
  - Requires twice the RAM for CELT-LC, an extra 0.5 kB for full CELT.
  - 0.9% of one core on a 3 GHz Core2

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  - **The API**
  - Low-latency Linux Audio
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libcelt API

```
CELTMode     *celt_mode_create(celt_int32_t Fs,int channels,int frame_size,
                               int *error);
int           celt_mode_info(const CELTMode *mode,int request,
                             celt_int32_t *value);
             • CELT_GET_FRAME_SIZE, CELT_GET_LOOKAHEAD,
               CELT_GET_NB_CHANNELS, CELT_GET_BITSTREAM_VERSION

CELTEncoder *celt_encoder_create(const CELTMode *mode);
int          celt_encoder_ctl(CELTEncoder *st,int request,...);
             • CELT_SET_COMPLEXITY_REQUEST, CELT_SET_COMPLEXITY(x) /*0-10 (int)*/
             • CELT_SET_LTP_REQUEST,        CELT_SET_LTP(x)       /*0 or 1 (int)*/
int          celt_encode(CELTEncoder *st,const celt_int16_t *pcm,
                         celt_int16_t *optional_synthesis,
                         unsigned char *compressedBytes,int nbCompressedBytes);
void         celt_encoder_destroy(CELTEncoder *st);

CELTDecoder *celt_decoder_create(const CELTMode *mode);
int          celt_decode(CELTDecoder *st,unsigned char *compressedBytes,
                         int nbCompressedBytes,celt_int16_t *pcm);
void         celt_decoder_destroy(CELTDecoder *st);

void         celt_mode_destroy(CELTMode *mode);
```

Hello Encoder

```c
#include <stdio.h>
#include <stdlib.h>
#include <celt/celt.h>

int main(int argc,const char *argv[]){
  celt_int16_t   in[256];
  unsigned char  out[43];
  CELTMode      *mode;
  CELTEncoder   *enc;
  mode=celt_mode_create(48000,1,256,NULL);
  if(mode==NULL)return EXIT_FAILURE;
  enc=celt_encoder_create(mode);
  if(enc==NULL)return EXIT_FAILURE;
  while(fread(in,sizeof(celt_int16_t),256,stdin)>=256){
    if(celt_encode(enc,in,NULL,out,43)<0)return EXIT_FAILURE;
    fwrite(out,sizeof(unsigned char),43,stdout);
  }
  celt_encoder_destroy(enc);
  celt_mode_destroy(mode);
  return EXIT_SUCCESS;
}
```

Hello Decoder

```
#include <stdio.h>
#include <stdlib.h>
#include <celt/celt.h>

int main(int argc,const char *argv[]){
  unsigned char  in[43];
  celt_int16_t   out[256];
  CELTMode      *mode;
  CELTDecoder   *dec;
  celt_int32_t   skip;
  mode=celt_mode_create(48000,1,256,NULL);
  if(mode==NULL)return EXIT_FAILURE;
  celt_mode_info(mode,CELT_GET_LOOKAHEAD,&skip);
  dec=celt_decoder_create(mode);
  if(dec==NULL)return EXIT_FAILURE;
  while(fread(in,sizeof(unsigned char),43,stdin)>=43){
    if(celt_decode(dec,in,43,out)<0)return EXIT_FAILURE;
    fwrite(out+skip,sizeof(celt_int16_t),256-skip,stdout);
    skip=0;
  }
  celt_decoder_destroy(dec);
  celt_mode_destroy(mode);
  return EXIT_SUCCESS;
}
```

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- libcelt
  - The API
  - **Low-latency Linux Audio**
- Demo
- Conclusion

Low-latency Linux Audio

- Audio hardware often doesn't work with small buffer sizes
  - 256 samples (5.3 ms) sometimes fails
  - Even 512 samples (10.6 ms) occasionally fails
  - I don't know how often this is a Linux driver problem vs. a hardware problem, but...
- There's no easy way to tell if it will work other than to try it and fail
  - And this *is* Linux's problem

Low-latency Linux Audio

- Even if small buffers work, scheduling delays can prevent us from filling them on time
  - Loading/unloading drivers still causes huge delays, even with RT patches
    - Hot-plugging some USB devices virtually guarantees deadline miss
- Network latency is also critical
  - Some drivers will attempt to throttle interrupts when sending hundreds of packets a second
    - This only makes latency worse
  - Some wi-fi drivers have weird spikes over 100ms (OpenMoko FreeRunner)

Low-latency Linux Audio

- Library support also important
  - On x86-64, glibc's exp() takes substantially longer than average for some arguments
    - Turns out it uses a generic C implementation
    - Includes its own custom multi-precision arithmetic library to compute hundreds of digits of intermediate results if necessary so that the rounding is exactly right
  - expf() is even slower than exp()
    - Changes exception handling mode of FPU, even if it's already set correctly, then changes it "back"
- Now imagine all the dependencies of a video-conferencing app...

Outline

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Conclusion

- CELT brings CD-quality sound to VoIP-style low-delay applications
  - Better than MP3 *and* <10 ms delay
- Better than emerging proprietary standards
  - As good or better than AAC-LD with half the delay
  - Better quality and error robustness than ULD
  - Supports wider range of bitrates, sampling rates

Early Adopters

- CELT is already being used by a number of projects
  - Soundjack (Alexander Carôt)
    http://virtualsoundexchange.net/node/21
  - NexGenVoIP (Dr. Christian Hoene)
    http://www.nexgenvoip.org/
  - FreeSWITCH (Anthony Minessale II, Brian K. West)
    http://www.freeswitch.org/ (source code available)
  - jack-audio-connection-kit (netjack) (Torben Hohn)
    http://jackaudio.org/ (source code available)
  - Radio CHNC (Jonathan Thibault, http://navigue.com)
    http://www.radiochnc.com/

Questions?