AUDIO

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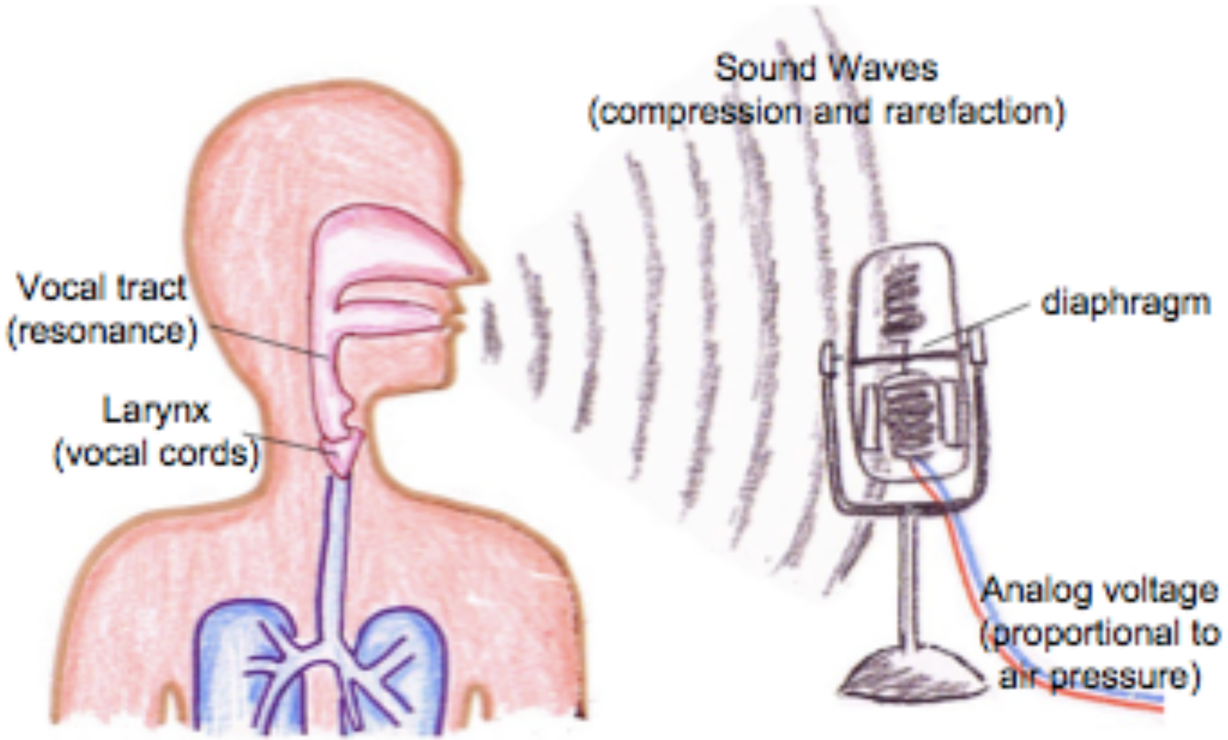
Key objectives

- How do humans generate and process sound?
- How does digital sound work?
- How fast do I have to sample audio?
- How can we represent time domain signals in the frequency domain? Why?
- How do audio codecs work?
- How do we measure their quality?
- What is the impact of networks (packet loss) on audio quality?

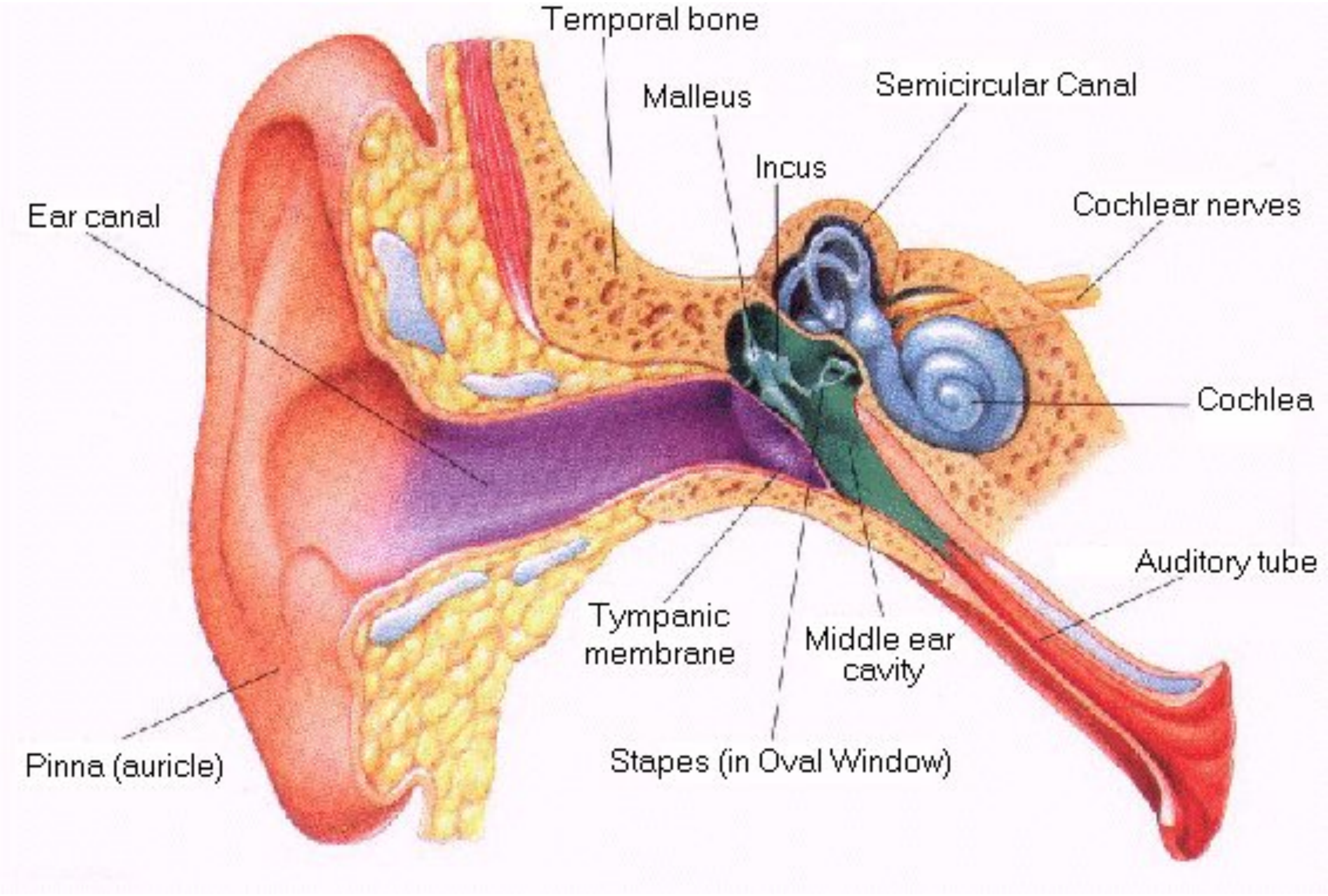
Human speech



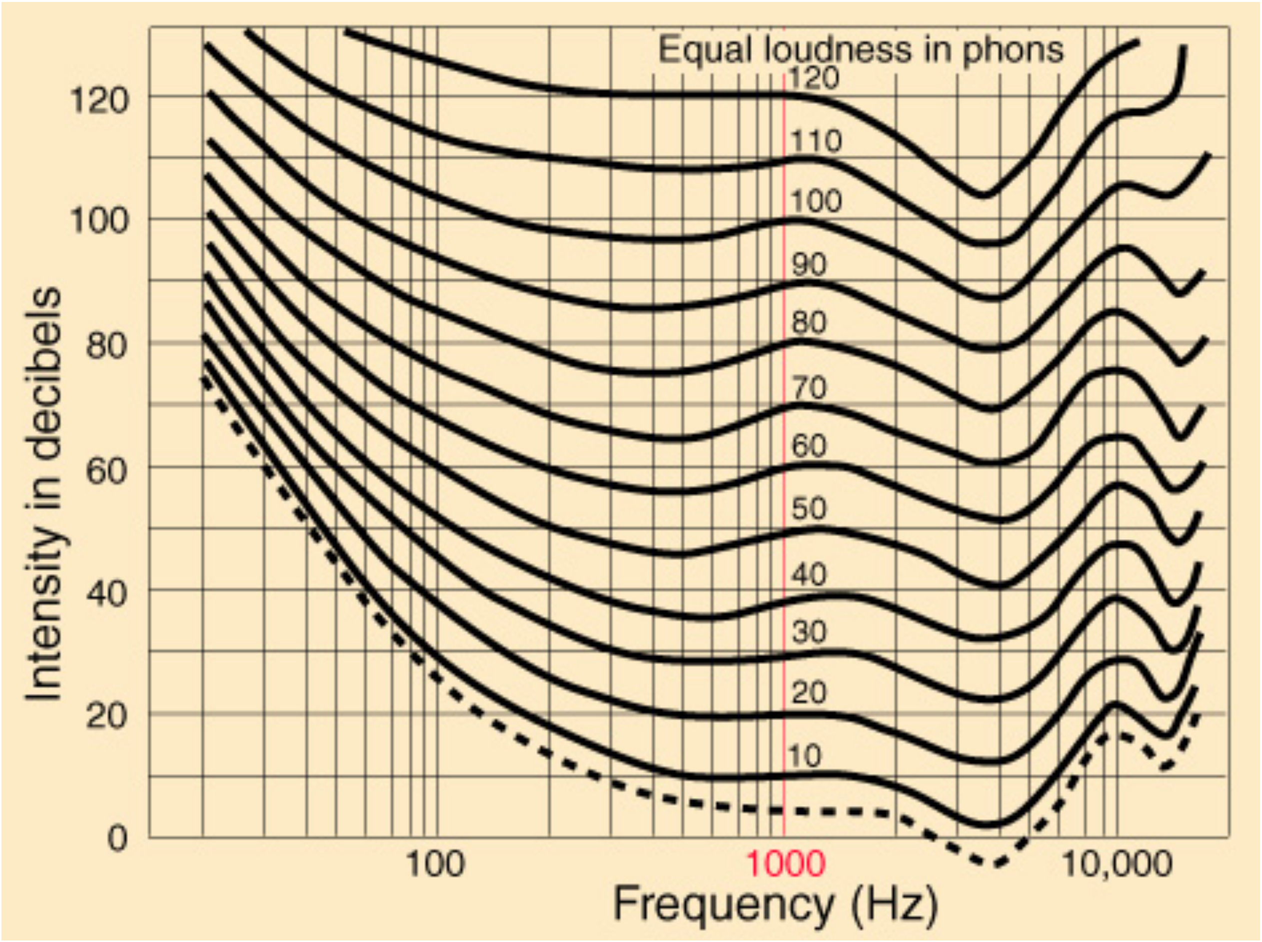
Human speech

- voiced sounds: vocal cords vibrate (e.g.,A4 [above middle C] = 440 Hz
  - vowels (a, e, i, o, u, …)
  - determines pitch
- unvoiced sounds:
  - fricatives (f, s)
  - plosives (p, d)
- filtered by vocal tract
- changes slowly (10 to 100 ms)
- air volume → loudness (dB)

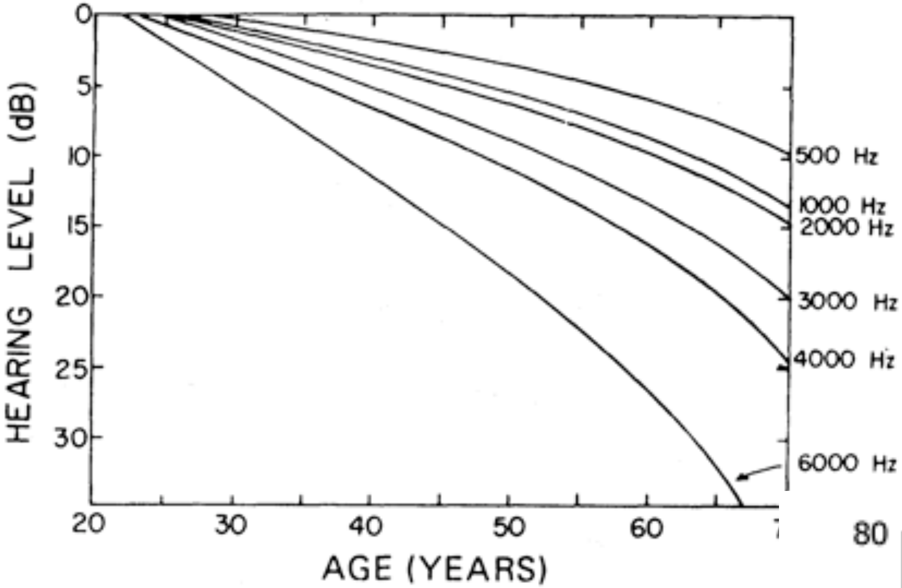
Human hearing

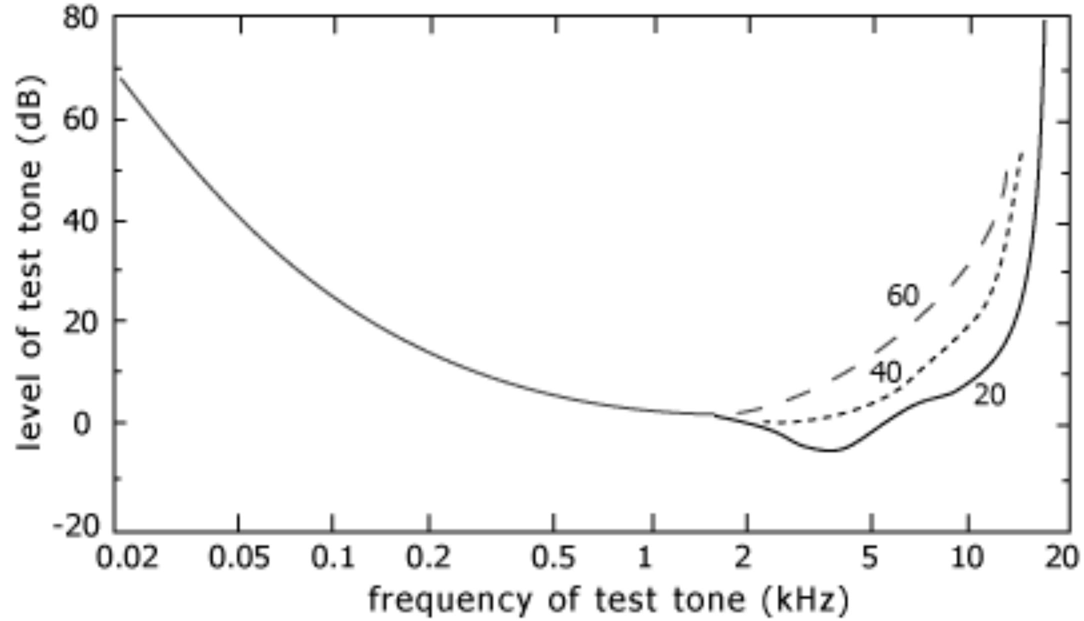


Human hearing

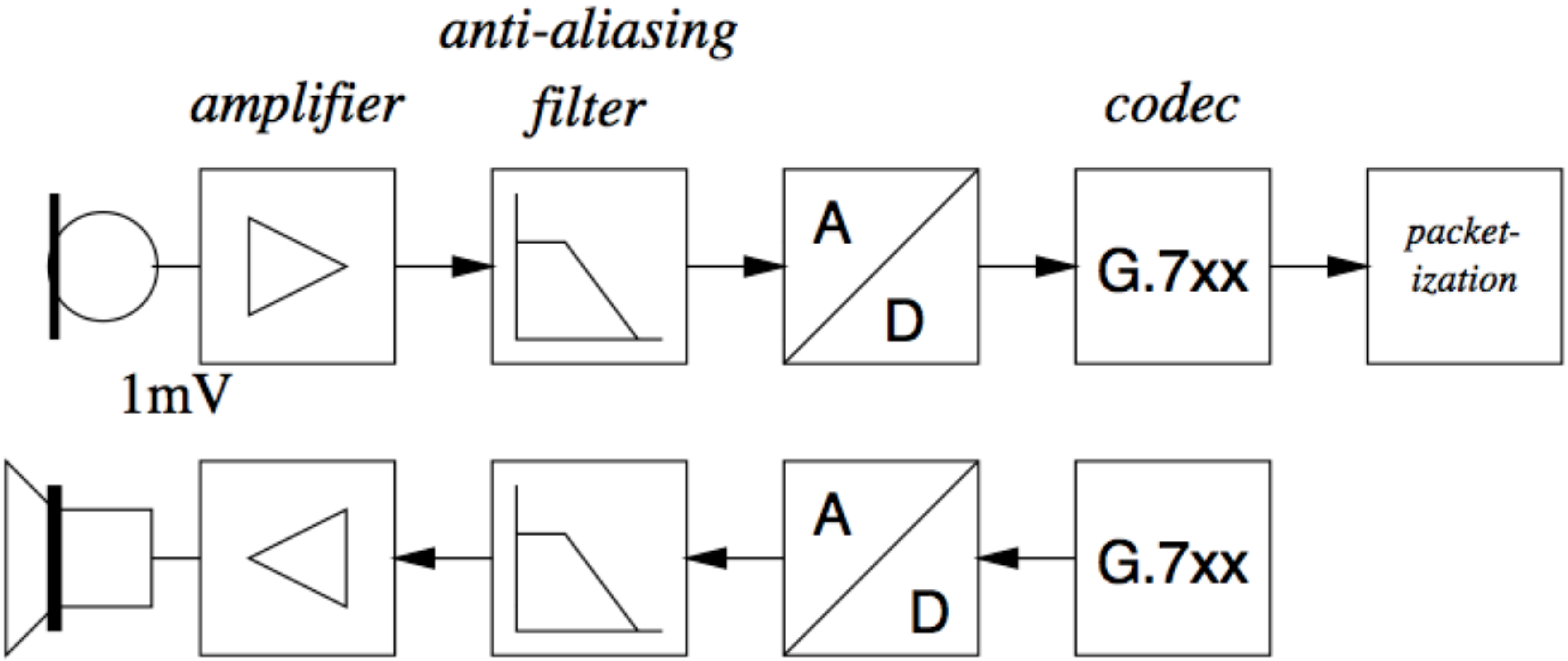


Human hearing & age



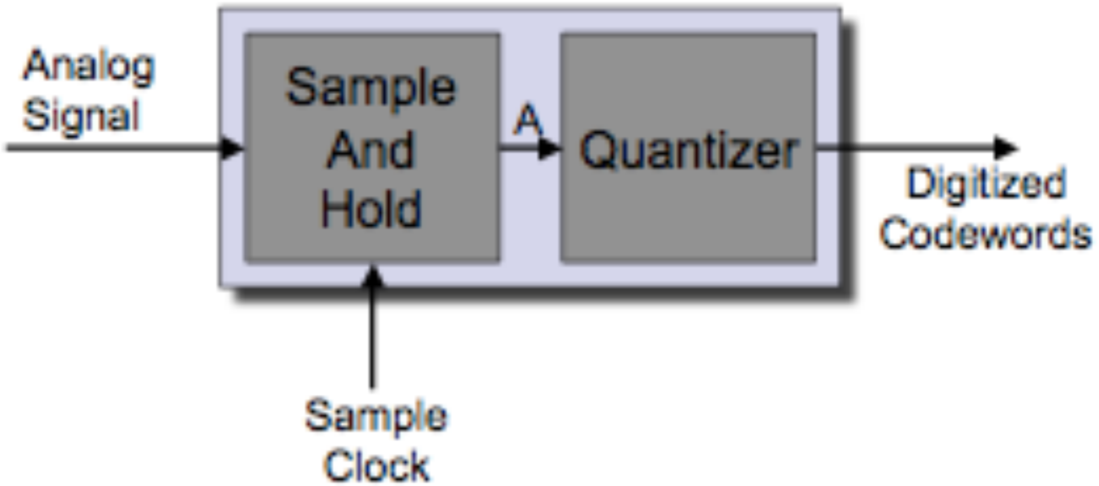


Digital sound

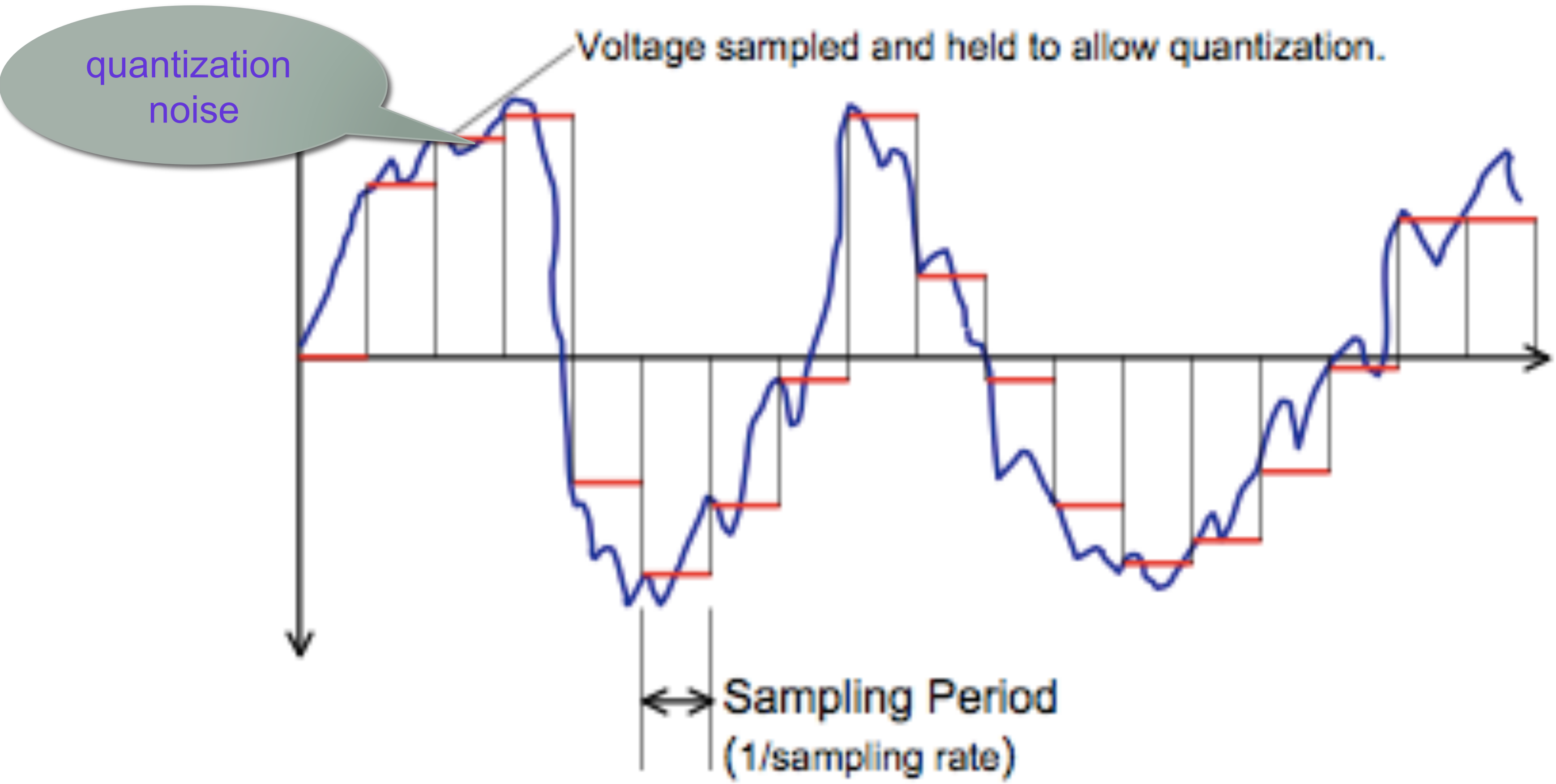


Analog-to-digital conversion

- Sample value of digital signal at f_s (8 – 96 kHz)
- Digitize into 2^B discrete values (8-24)



Sample & hold



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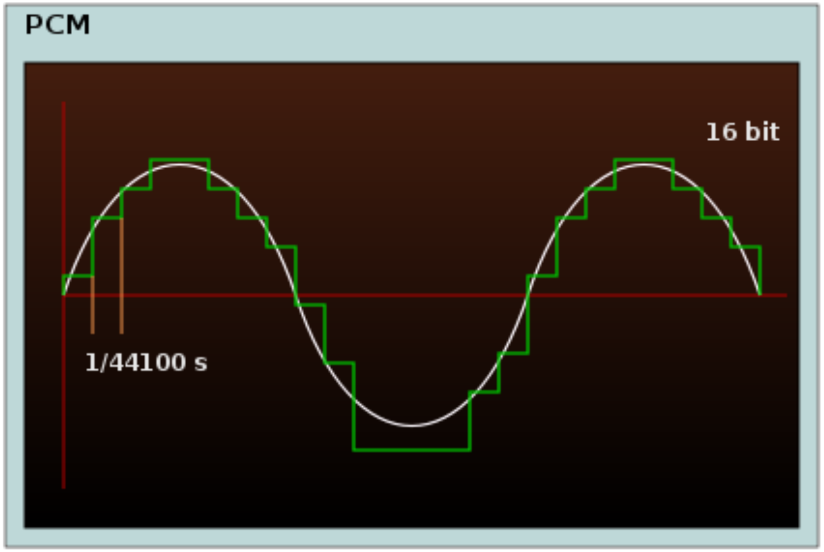
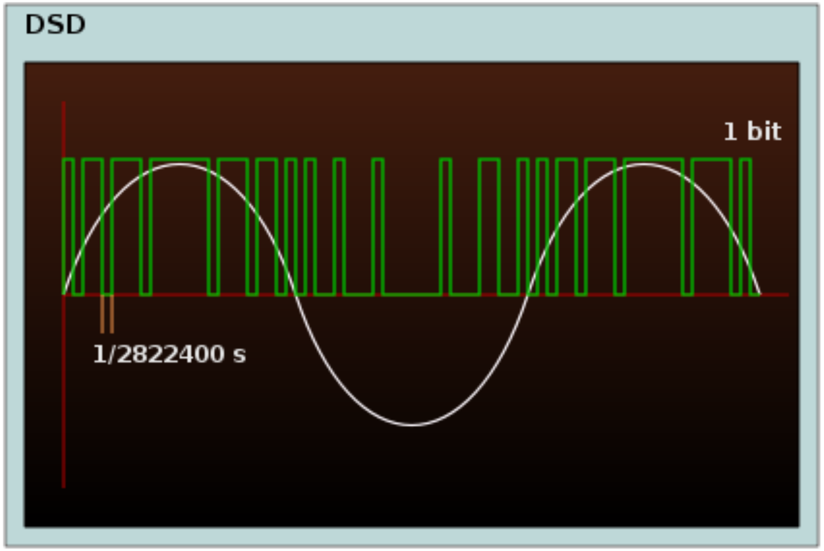
Direct-Stream Digital





Delta-Sigma coding

How fast to sample?

- Harry Nyquist (1928) & Claude Shannon (1949)
  - no loss of information → sampling frequency ≥ 2 * maximum signal frequency
- More recent: compressed sensing
  - works for *sparse* signals in some space

Audio coding

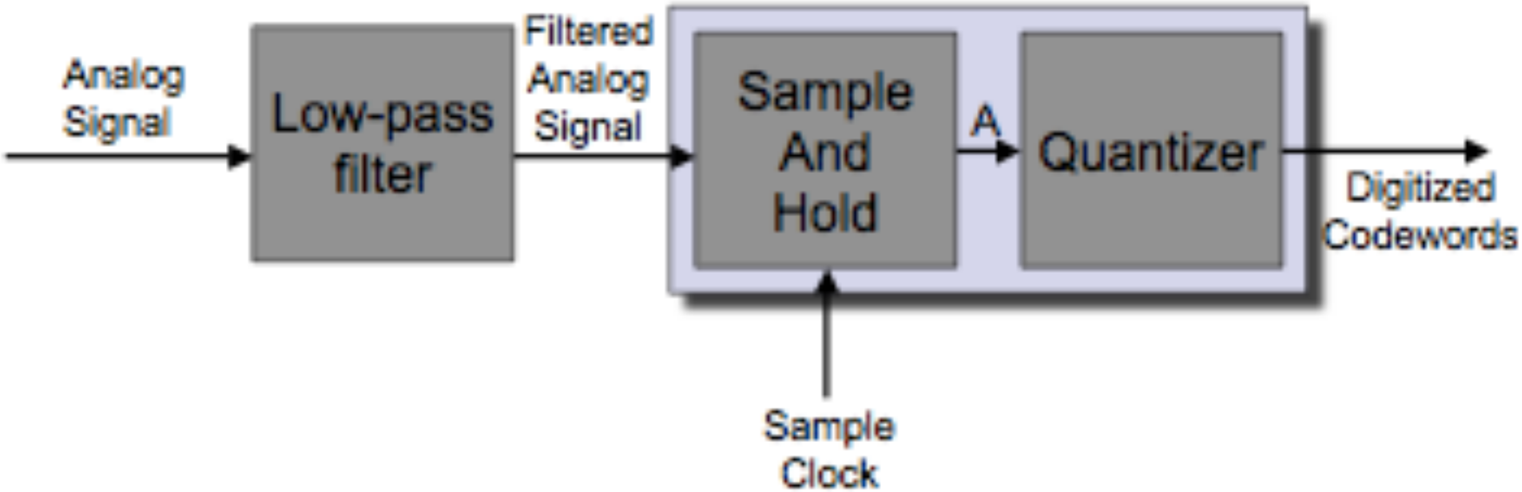
application	frequency	sampling	quantization
telephone	300-3,400 Hz	8 kHz	12-13
wide-band	50-7,000 Hz	16 kHz	14-15
high quality	30-15,000 Hz	32 kHz	16
	20-20,000 Hz	44.1 kHz	16
	10-22,000 Hz	48 kHz	≤ 24

CD

DAT

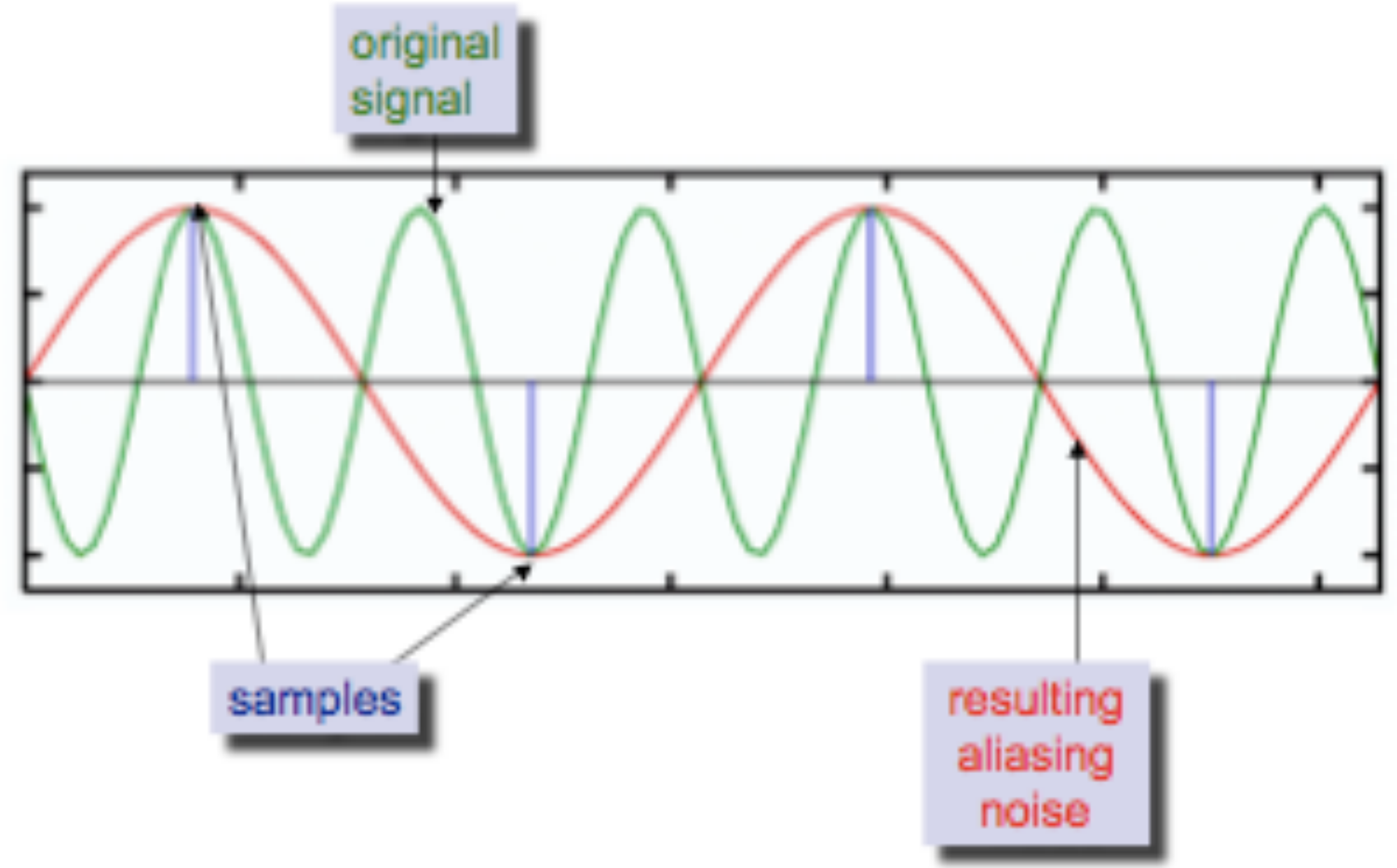
24 bit, 44.1/48 kHz

Complete A/D



Analog Signal
Low-pass filter
Filtered Analog Signal
Sample And Hold
A
Quantizer
Digitized Codewords
Sample Clock

Aliasing distortion



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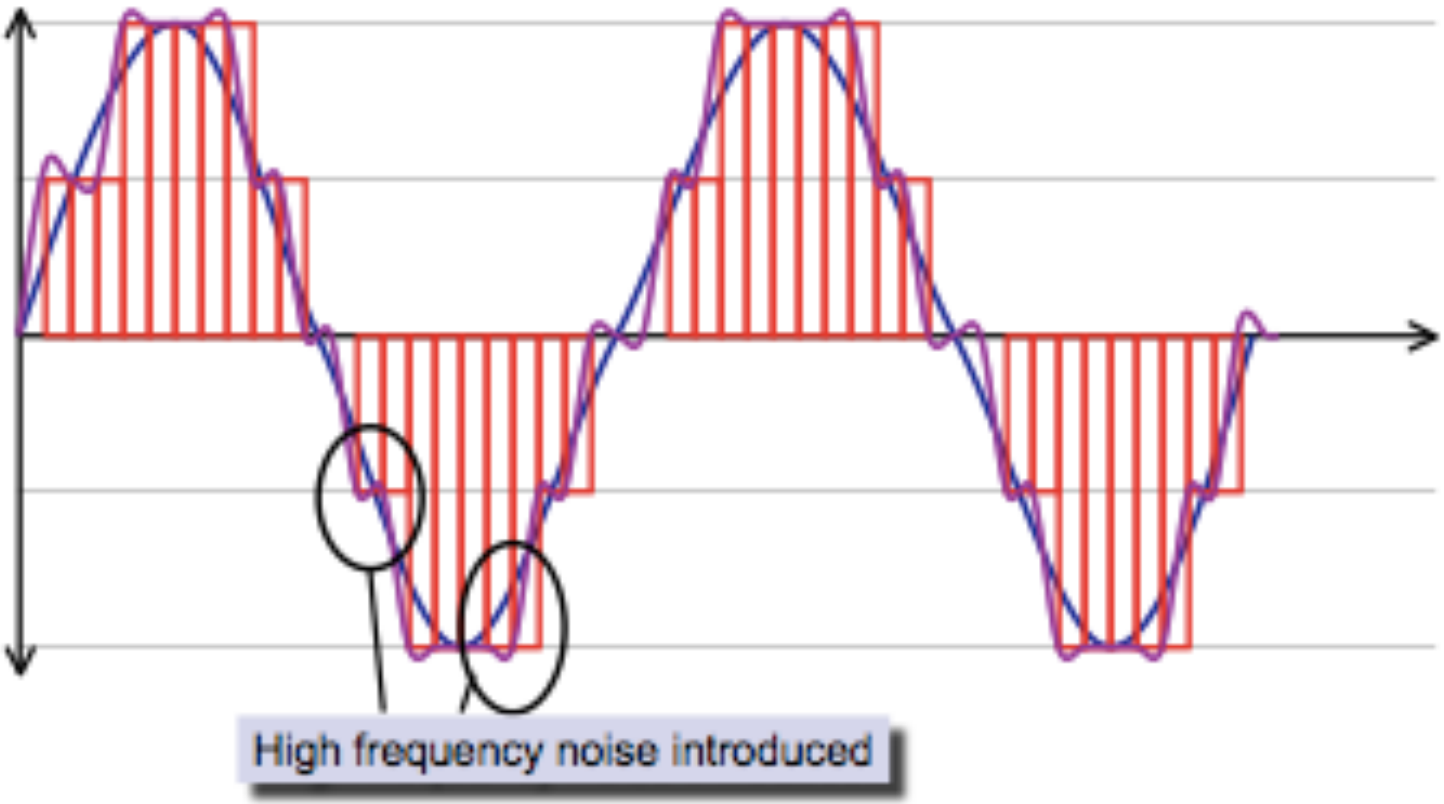
Quantization

- CDs: 16 bit → lots of bits
- Professional audio: 24 bits (or more)
- 8-bit linear has poor quality (noise)
- Ear has logarithmic sensitivity → "companding"
  - used for Dolby tape decks
  - quantization noise ~ signal level

Quantization noise

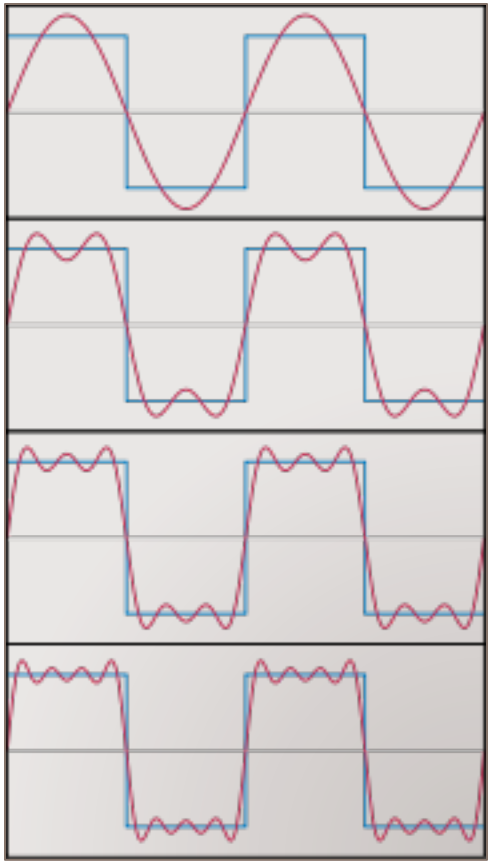


Fourier transform

- Fourier transform: time series → series of frequencies
  - complex frequencies: amplitude & phasess
- Inverse Fourier transform: frequencies (amplitude & phase) → time series
- Note: also works for other basis functions

Fourier series

- Express periodic function as sum of sines and cosines of different amplitudes
  - iff band-limited, finite sum
- Time domain → frequency domain
  - no information loss
    - and no compression
  - but for periodic (or time limited) signals
- http://www.westga.edu/~jhasbun/osp/Fourier.htm



Fourier series of a periodic function

$$f(x) = \frac{1}{2}a_0 + \sum_{n=1}^{\infty}[a_n \cos(nx) + b_n \sin(nx)]$$

$$a_n = \frac{1}{\pi}\int_{-\pi}^{\pi} f(x)\cos(nx)dx$$

$$b_n = \frac{1}{\pi}\int_{-\pi}^{\pi} f(x)\sin(nx)dx$$

continuous
time,
discrete
frequencies

Fourier transform

forward transform (time x, real frequency k)

$$F(k) = \int_{-\infty}^{\infty} f(x) e^{-2\pi i k x} dk$$

inverse transform

$$f(k) = \int_{-\infty}^{\infty} F(k) e^{2\pi i k x} dk$$

continuous time, continuous frequencies

$$e^{xi} = cos(x) + isin(x)$$

Discrete Fourier transform

- For sampled functions, continuous FT not very useful → DFT

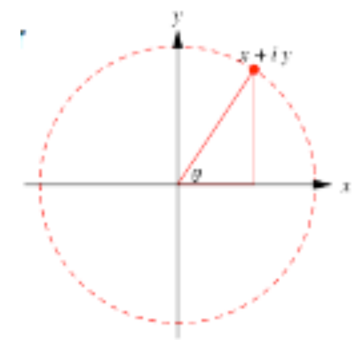
$$F_n = \sum_{k=0}^{N-1} f_k e^{-2\pi i n k/N}$$

complex numbers → complex coefficients

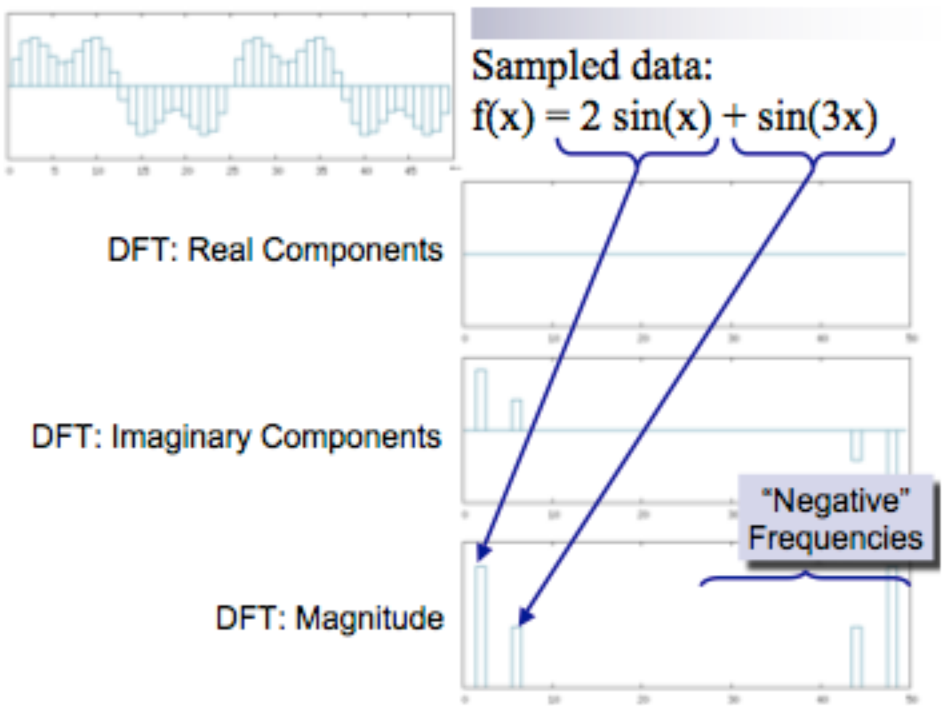
$$f_k = \frac{1}{N} \sum_{n=0}^{N-1} F_n e^{2\pi i n k/N}$$

DFT example

- Interpreting a DFT can be slightly difficult, because the DFT of real data includes complex numbers.
  - The magnitude of the complex number for a DFT component is the power at that frequency.
  - The phase θ of the waveform can be determined from the relative values of the real and imaginary coefficients.
  - Also both positive and “negative” frequencies show up.

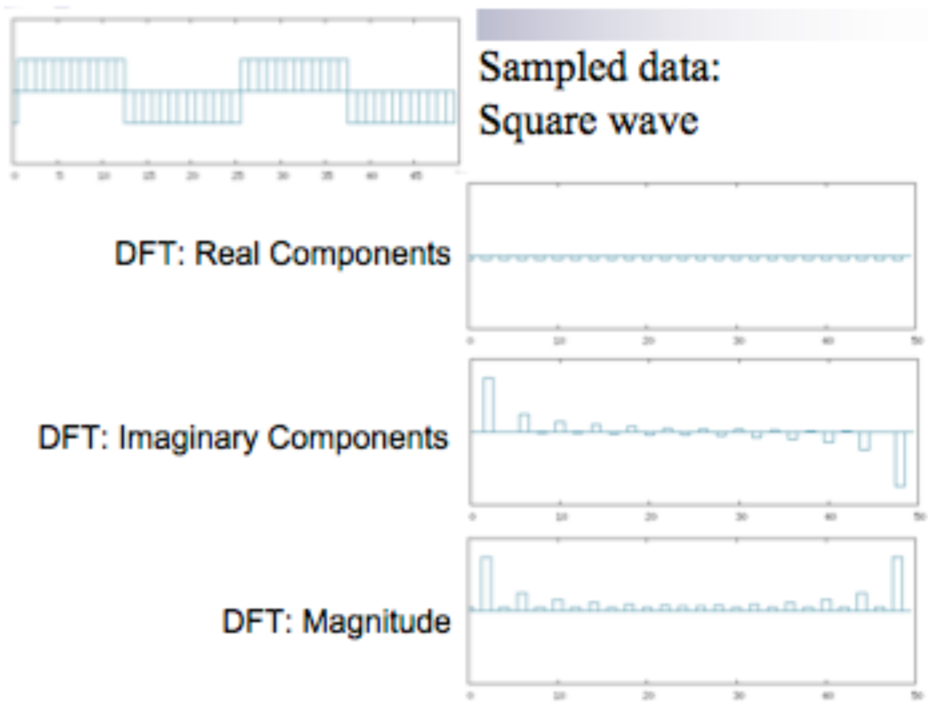


DFT example



Sampled data:
f(x) = 2 sin(x) + sin(3x)
DFT: Real Components
DFT: Imaginary Components
"Negative" Frequencies
DFT: Magnitude

DFT example



Sampled data:
Square wave
DFT: Real Components
DFT: Imaginary Components
DFT: Magnitude

Fast Fourier Transform (FFT)

- Discrete Fourier Transform would normally require *O(n2)* time to process for *n samples:*

$$F_n = \sum_{k=0}^{N-1} f_k e^{-2\pi i n k / N}$$

- Don’t usually calculate it this way in practice.
  - Fast Fourier Transform takes *O(n log(n)) time.*
  - Most common algorithm is the Cooley-Tukey Algorithm.

Fourier Cosine Transform

- Split function into odd and even parts:

$$f(x) = \frac{1}{2}[f(x) + f(-x)] + \frac{1}{2}[f(x) - f(-x)] = E(x) + O(x)$$

- Re-express FT:

$$F(k) = \int E(x)\cos(2\pi kx)dx - i\int O(x)\sin(2\pi kx)dx$$

- Only real numbers from an even function → DFT becomes DCT

DCT (for JPEG)

$$f_j = \sum_{n=0}^{N-1} x_n \cos[\frac{\pi}{N} j(n + \frac{1}{2})]$$

other versions exist (e.g., for MP3, with overlap)

Why do we use DCT for multimedia?

- For audio:
  - Human ear has different dynamic range for different frequencies.
  - Transform to from time domain to frequency domain, and quantize different frequencies differently.
- For images and video:
  - Human eye is less sensitive to fine detail.
  - Transform from spatial domain to frequency domain, and quantize high frequencies more coarsely (or not at all)
  - Has the effect of slightly blurring the image - may not be perceptible if done right.

Why use DCT/DFT?

- Some tasks easier in frequency domain
  - e.g., graphic equalizer, convolution
- Human hearing is logarithmic in frequency (→ octaves)
- Masking effects (see MP3)

Example: DCT for image

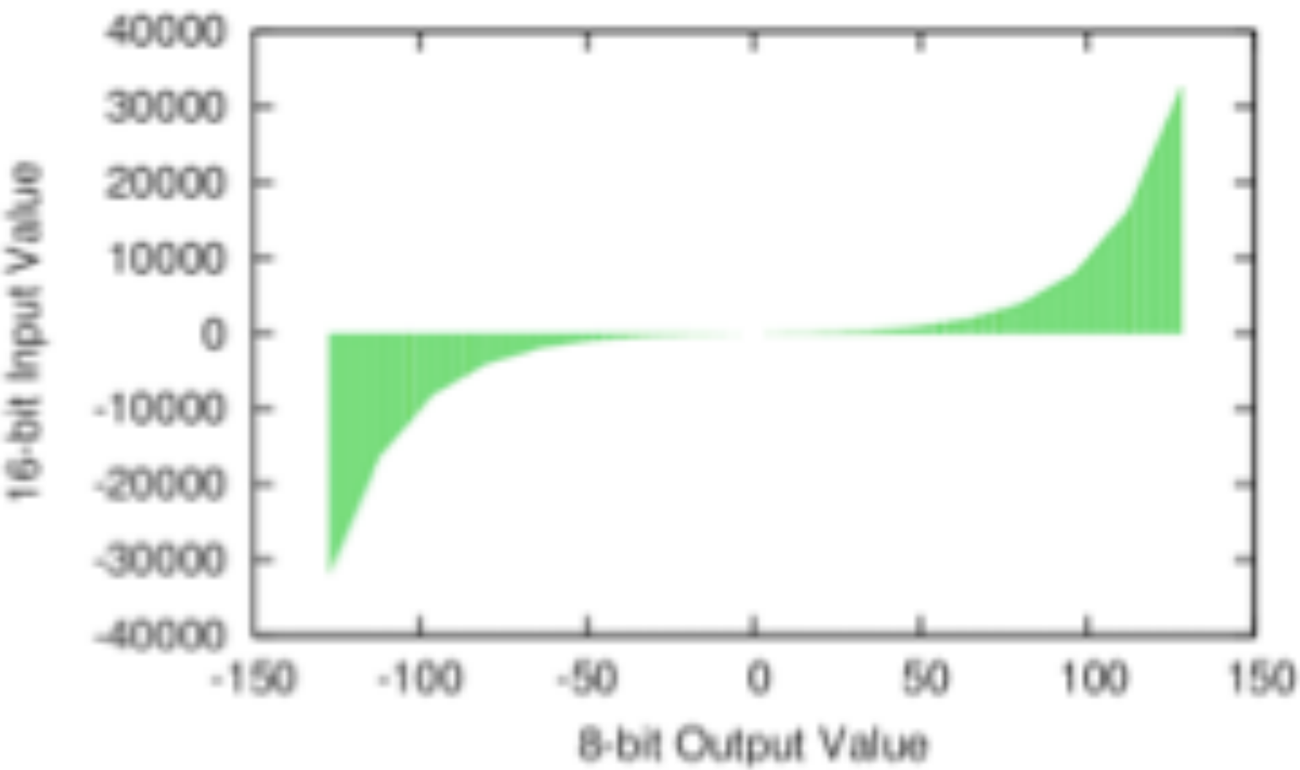
PICTURE MATRIX

$$\begin{bmatrix} 40 & 24 & 15 & 19 & 28 & 24 & 19 & 15 \\ 38 & 34 & 35 & 35 & 31 & 28 & 27 & 29 \\ 40 & 47 & 49 & 40 & 33 & 29 & 32 & 43 \\ 42 & 49 & 50 & 39 & 34 & 30 & 32 & 46 \\ 40 & 47 & 46 & 35 & 31 & 32 & 35 & 43 \\ 38 & 43 & 42 & 31 & 27 & 27 & 28 & 33 \\ 39 & 33 & 25 & 17 & 14 & 15 & 19 & 26 \\ 29 & 16 & 6 & 1 & -4 & 0 & 7 & 18 \end{bmatrix}$$

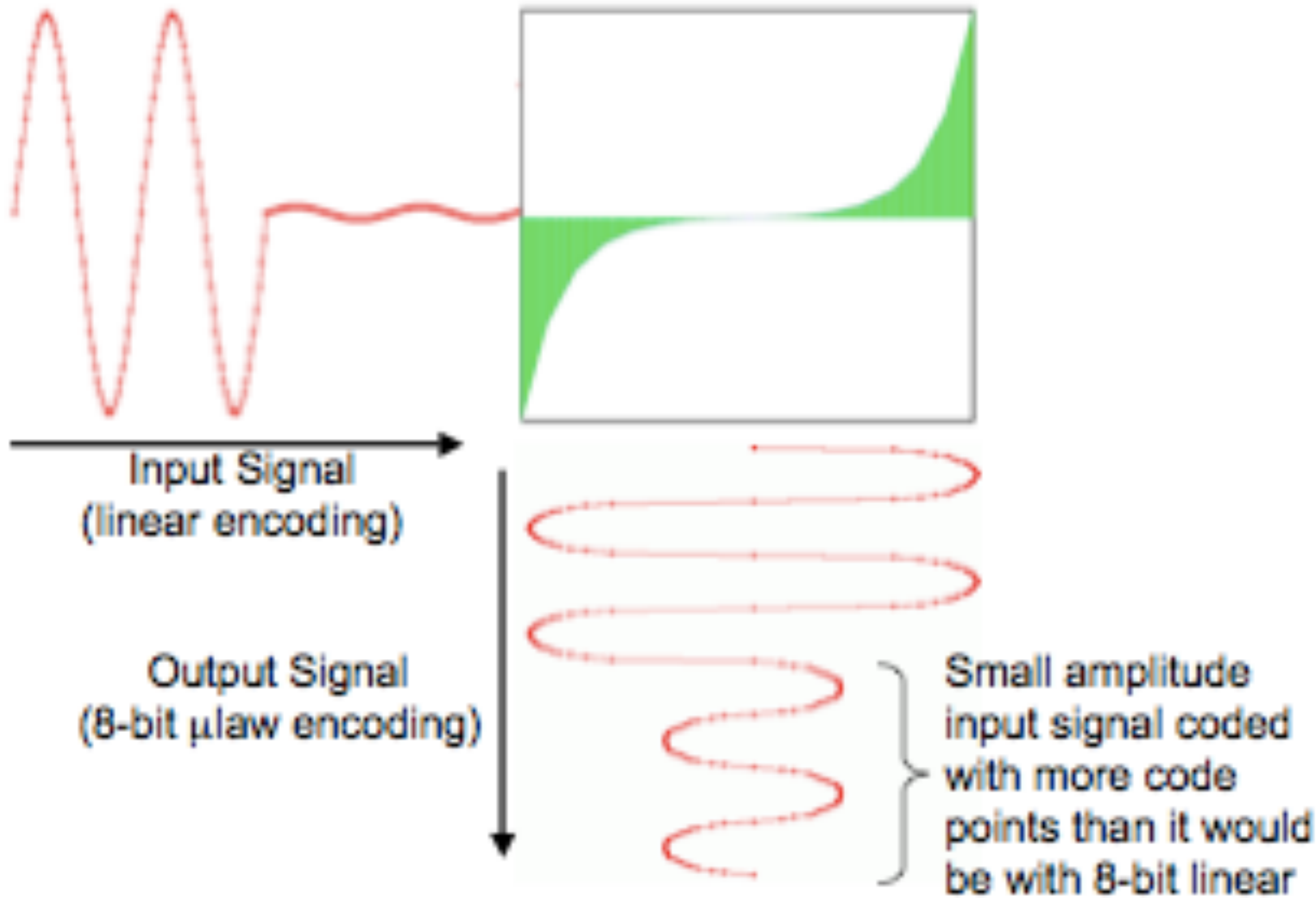
DCT COEFFICENTS

$$\begin{bmatrix} 239 & 32 & 27 & -12 & 3 & -5 & 3 & 1 \\ 34 & -3 & -19 & 6 & 3 & 0 & -1 & 1 \\ -70 & 2 & 8 & 23 & 9 & 6 & -1 & -1 \\ 5 & 0 & -6 & 11 & -2 & 0 & -1 & 1 \\ -17 & -3 & 6 & 6 & 3 & -1 & 0 & 0 \\ 2 & 4 & 2 & 2 & 1 & -2 & 0 & 1 \\ -3 & 0 & 0 & -1 & -1 & -1 & 0 & 0 \\ 1 & -1 & 3 & 1 & 0 & 0 & 0 & 0 \end{bmatrix}$$

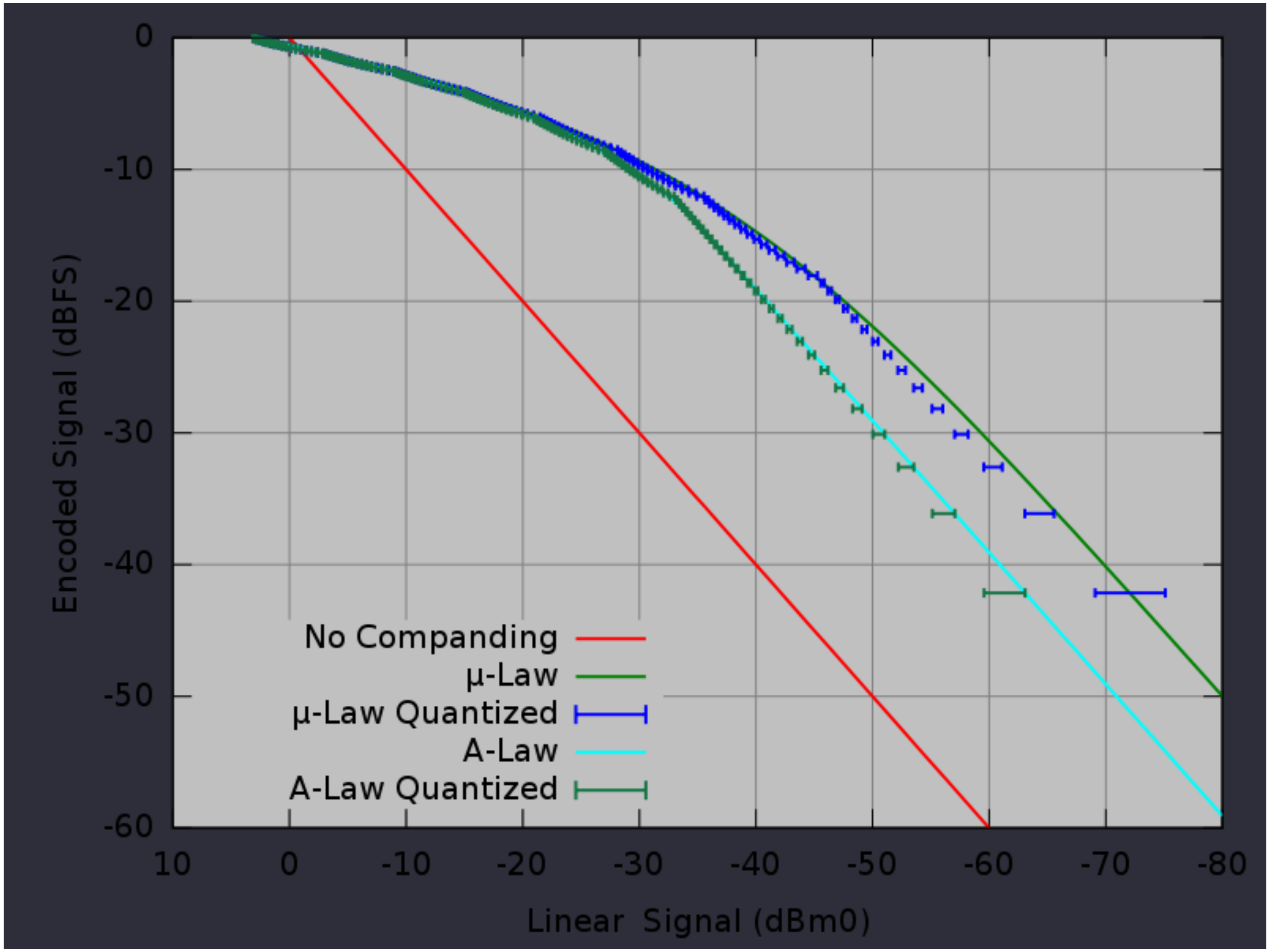
μ-law encoding



µ-law encoding



Companding



Wikipedia

μ-law & A-law

μ-law

$$F(x) = \frac{sgn(x) ln(1 + \mu|x|)}{ln(1 + \mu)} \qquad 0 \leq |x| \leq 1$$

μ is 255 in US/Japan

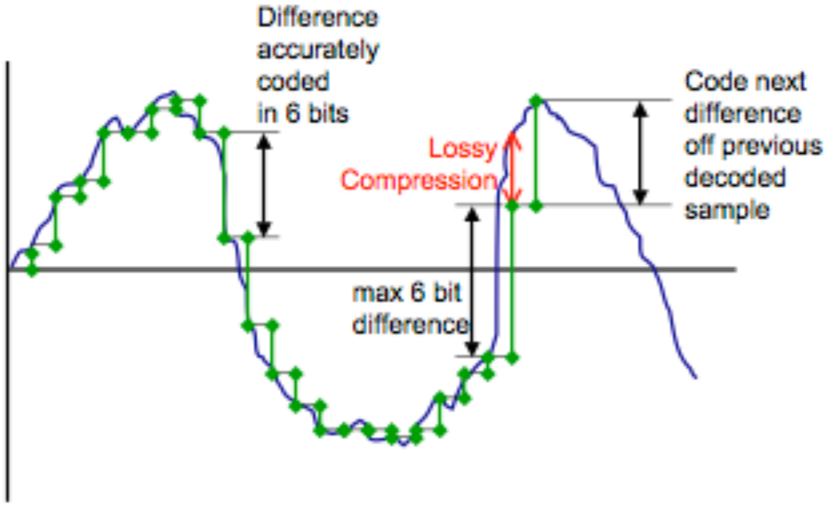
A-law

$$F(x) = \frac{A|x|}{ln(1 + A)} \qquad 0 \leq |x| < \frac{1}{A}$$

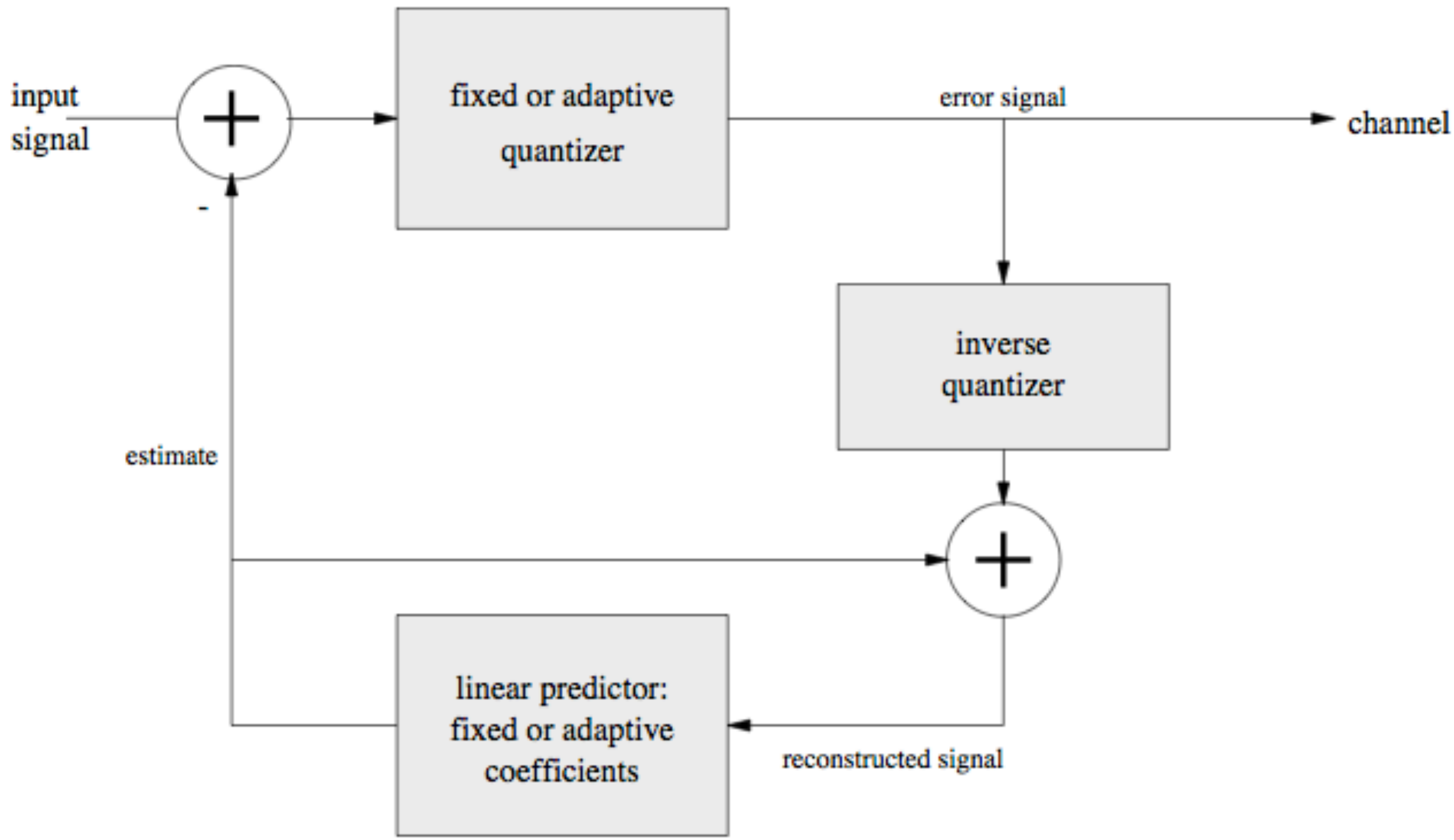
$$F(x) = \frac{sgn(x) ln(1 + A|x|)}{ln(1 + A)} \qquad \frac{1}{A} \leq |x| \leq 1$$

A = 87.7 in Europe

Differential codec



(Adaptive) Differential Pulse Code Modulation



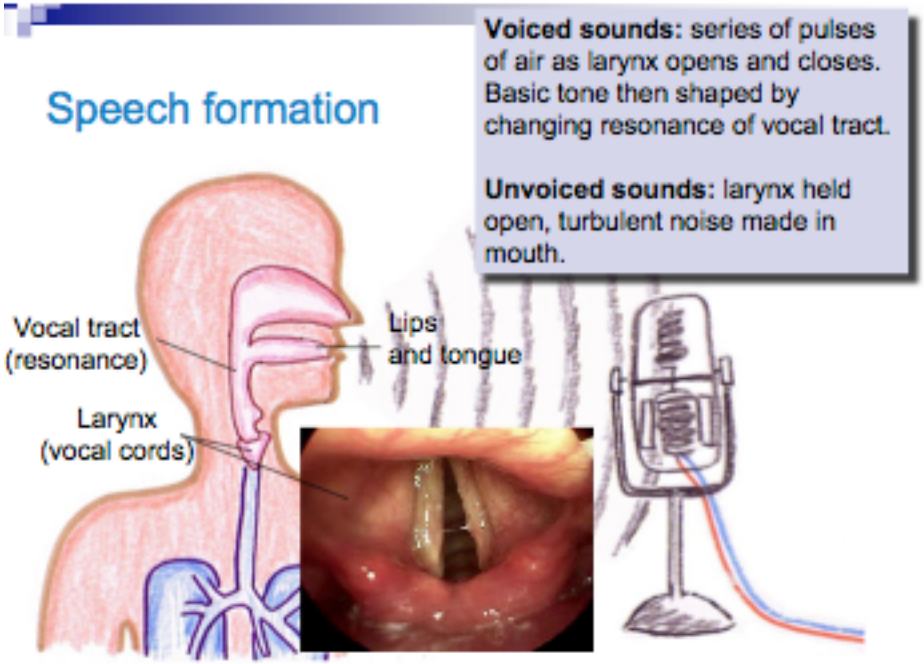
ADPCM

- Makes a simple prediction of the next sample, based on weighted previous *n* samples.
- For G.721, previous 8 weighted samples are added to make the prediction.
- Lossy coding of the difference between the actual sample and the prediction.
  - Difference is quantized into 4 bits ⇒ 32Kb/s sent.
  - Quantization levels are adaptive, based on the content of the audio.
  - Receiver runs same prediction algorithm and adaptive quantization levels to reconstruct speech.

Model-based coding

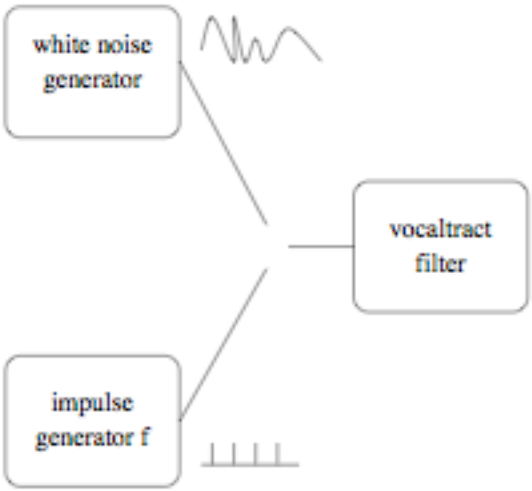
- PCM, DPCM and ADPCM directly code the received audio signal.
- An alternative approach is to build a *parameterized model of the sound source (i.e., human voice).*
- *For each time slice (e.g., 20ms):*
  - Analyze the audio signal to determine how the signal was produced.
  - Determine the model parameters that fit.
  - Send the model parameters.
  - At the receiver, synthesize the voice from the model and received parameters.

Speech formation

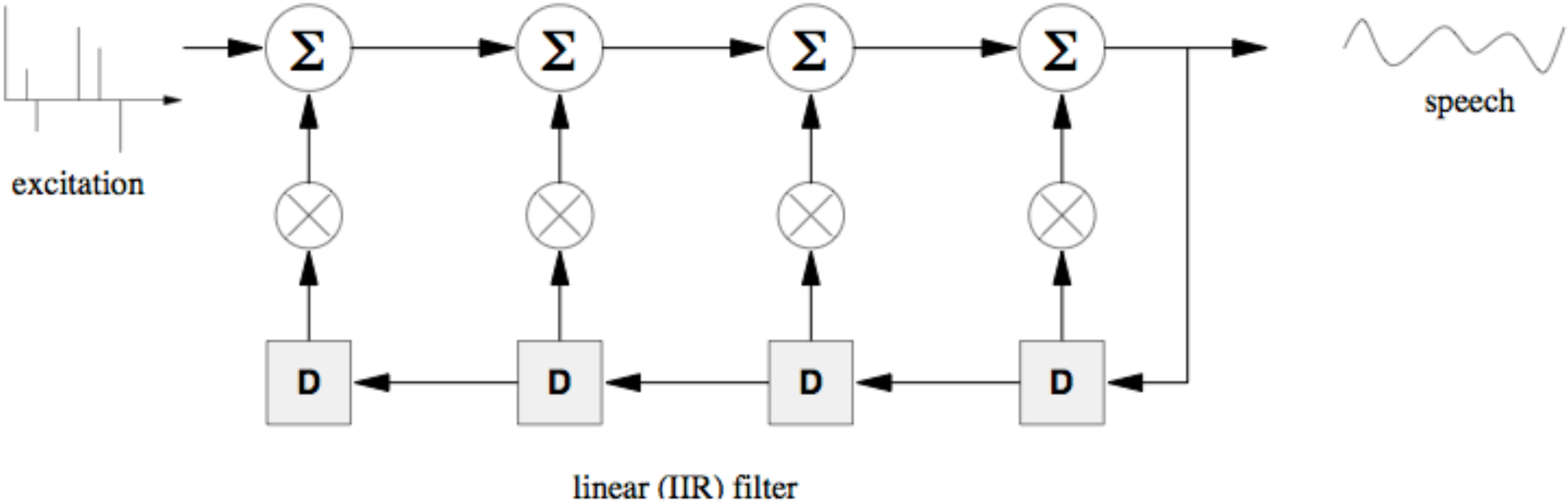


Linear predictive codec

- Earliest low-rate codec (1960s)
- LPC10 at 2.4 kb/s
  - sampling rate 8 kHz
  - frame length 180 samples (22.5 ms)
  - linear predictive filter (10 coefficients = 42 bits)
  - pitch and voicing (7 bits)
  - gain information (5 bits)



Linear predictive codec



infinite impulse response (IIR) vs. finite impulse response

Code Excited Linear Prediction (CELP)

- Goal is to efficiently encode the residue signal, improving speech quality over LPC, but without increasing the bit rate too much.
- CELP codecs use a codebook of typical residue values. (→ *vector quantization*)
- Analyzer compares residue to codebook values.
  - Chooses value which is closest.
  - Sends that value.
  - Receiver looks up the code in its codebook, retrieves the residue, and uses this to excite the LPC formant filter.

CELP (2)

- Problem is that codebook would require different residue values for every possible voice pitch.
  - Codebook search would be slow, and code would require a lot of bits to send.
    - One solution is to have two codebooks.
    - One fixed by codec designers, just large enough to represent one pitch period of residue.
    - One dynamically filled in with copies of the previous residue delayed by various amounts (delay provides the pitch)
  - CELP algorithm using these techniques can provide pretty good quality at 4.8Kb/s.

Enhanced LPC usage

- GSM (Groupe Speciale Mobile)
  - Residual Pulse Excited LPC
  - 13 kb/s
- LD-CELP
  - Low-delay Code-Excited Linear Prediction (G.728)
  - 16 kb/s
- CS-ACELP
  - Conjugate Structure Algebraic CELP (G.729)
  - 8 kb/s
- MP-MLQ
  - Multi-Pulse Maximum Likelihood Quantization (G.723.1)
  - 6.3 kb/s

Distortion metrics

- error (noise) *r(n)* = *x(n)* – *y(n)*
- variancesσx2,σy2,σr2
- power for signal with pdf *p(x)* and range $-V \ldots +V$

$$\sigma_x^2 = \int_{-V}^{+V} (x - \bar{x})^2 p(x) dx$$

- SNR = 6.02N − 1.73 for uniform quantizer with N bits

Distortion measures

- SNR *not* a good measure of perceptual quality
- ➠ segmental SNR: time-averaged blocks (say, 16 ms)
- frequency weighting
- *subjective measures:*
  - A-B preference
  - subjective SNR: comparison with additive noise
  - MOS (mean opinion score of 1-5), DRT, DAM, . . .

Quality metrics

- speech vs. music
- communication vs. toll quality

score	MOS	DMOS	understanding
5	excellent	inaudible	no effort
4	good, toll quality	audible, not annoying	no appreciable effort
3	fair	slightly annoying	moderate effort
2	poor	annoying	considerable effort
1	bad	very annoying	no meaning

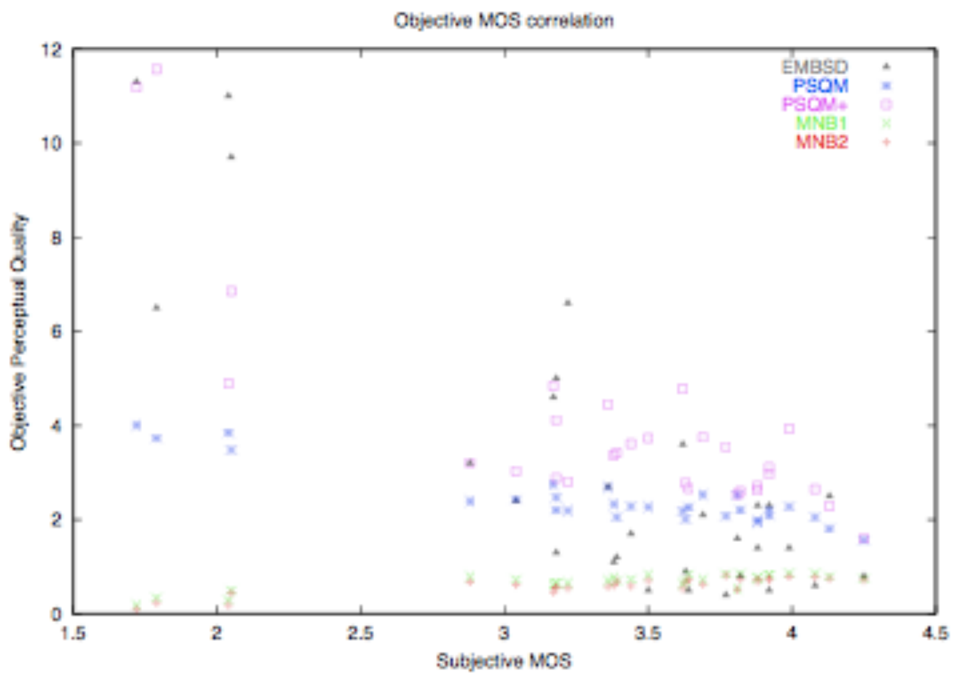
Subjective quality metrics

- Test phrases (ITU P.800)
  - *You will have to be very quiet.*
  - *There was nothing to be seen.*
  - *They worshipped wooden idols.*
  - *I want a minute with the inspector.*
  - *Did he need any money?*
- Diagnostic rhyme test (DRT)
  - 96 pairs like dune vs. tune
  - 90% right → toll quality

Objective quality metrics

- approximate human perception of noise and other distortions
- distortion due to encoding and packet loss (gaps, interpolation of decoder)
- examples: PSQM (P.861), PESQ (P.862), MNB, EMBSD – compare reference signal to distorted signal
- either generate MOS scores or distance metrics
- much cheaper than subjective tests
- only for telephone-quality audio so far

Objective vs. subjective quality



Common narrowband audio codecs

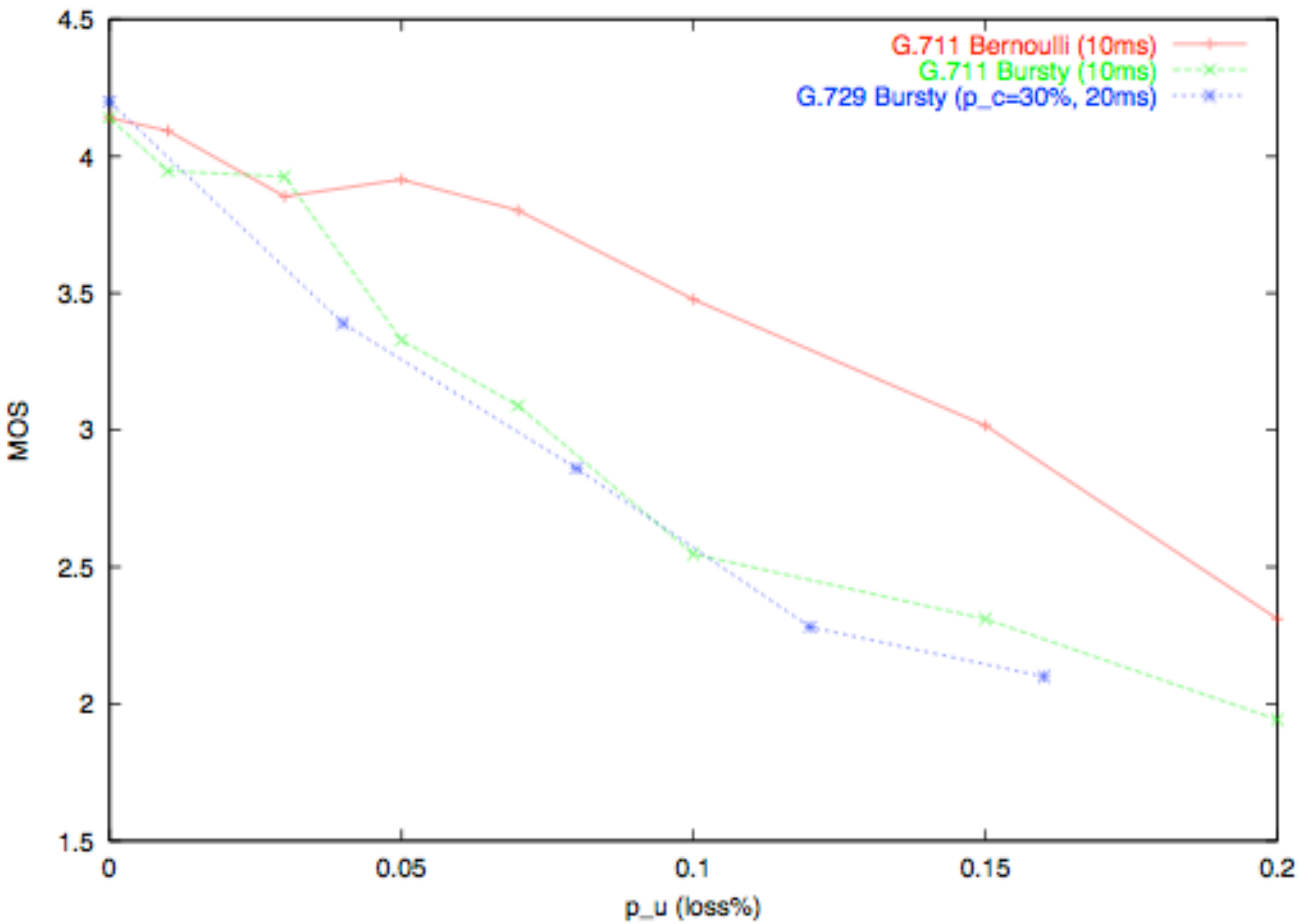
Codec	rate (kb/s)	delay (ms)	multi-rate	em-bedded	VBR	bit-robust/ PLC	remarks
iLBC	15.2 13.3	20 30				--/X	quality higher than G.729A no licensing
Speex	2.15--24.6	30	X	X	X	--/X	no licensing
AMR-NB	4.75--12.2	20	X			X/X	3G wireless
G.729	8	15				X/X	TDMA wireless
GSM-FR	13	20					GSM wireless (Cingular)
GSM-EFR	12.2	20				X/X	2.5G
G.728	16 12.8	2.5				X/X	H.320 (ISDN videconferencing)
G.723.1	5.3 6.3	37.5 37.5				X/--	H.323, videoconferences

Common wideband audio codecs

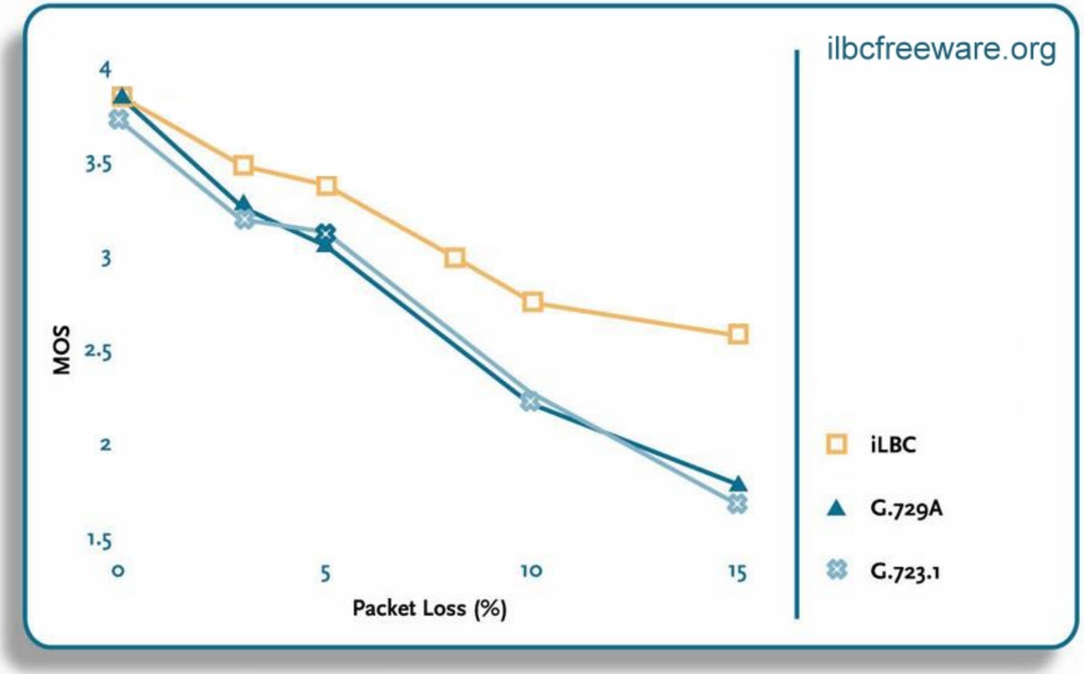
Codec	rate (kb/s)	delay (ms)	multi-rate	em-bedded	VBR	bit-robust/ PLC	remarks
Speex	4—44.4	34	X	X	X	--/X	no licensing
AMR-WB	6.6—23.85	20	X			X/X	3G wireless
G.722	48, 56, 64	0.125 (1.5)				X/--	2 sub-bands now dated

http://www.voiceage.com/listeningroom.php

MOS vs. packet loss



iLBC – MOS behavior with packet loss



The tests were performed by Dynstat, Inc., an independent test laboratory.
Score system range: 1 = bad, 2 = poor, 3 = fair, 4 = good, 5 = excellent

Courtesy of GLOBAL IP SOUND

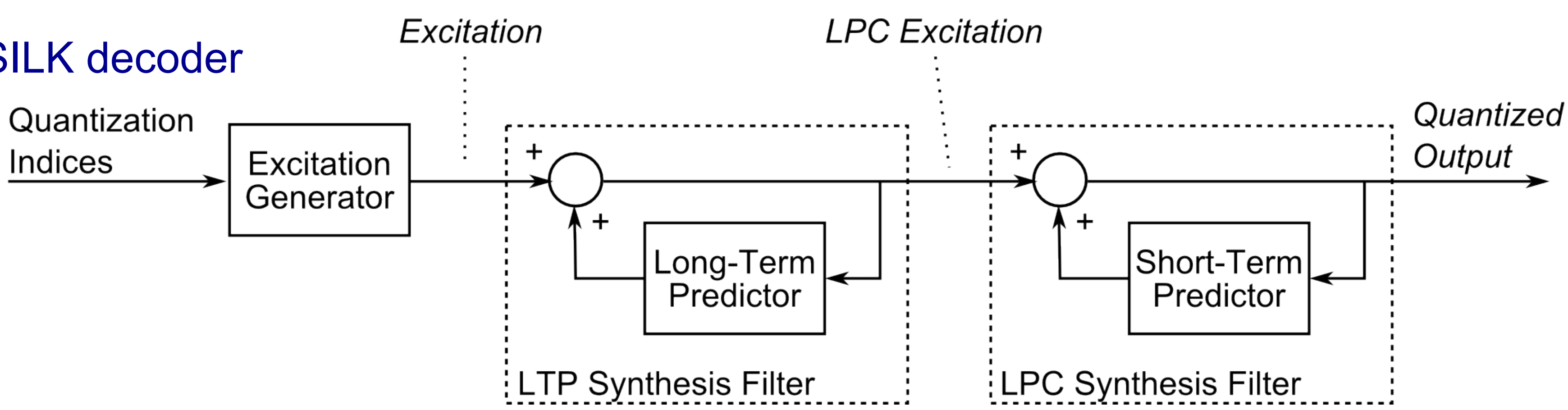
Recent audio codecs

- iLBC: optimized for high packet loss rates (frames encoded independently)
- AMR-NB
  - 3G wireless codec
  - 4.75-12.2 kb/s
  - 20 ms coding delay

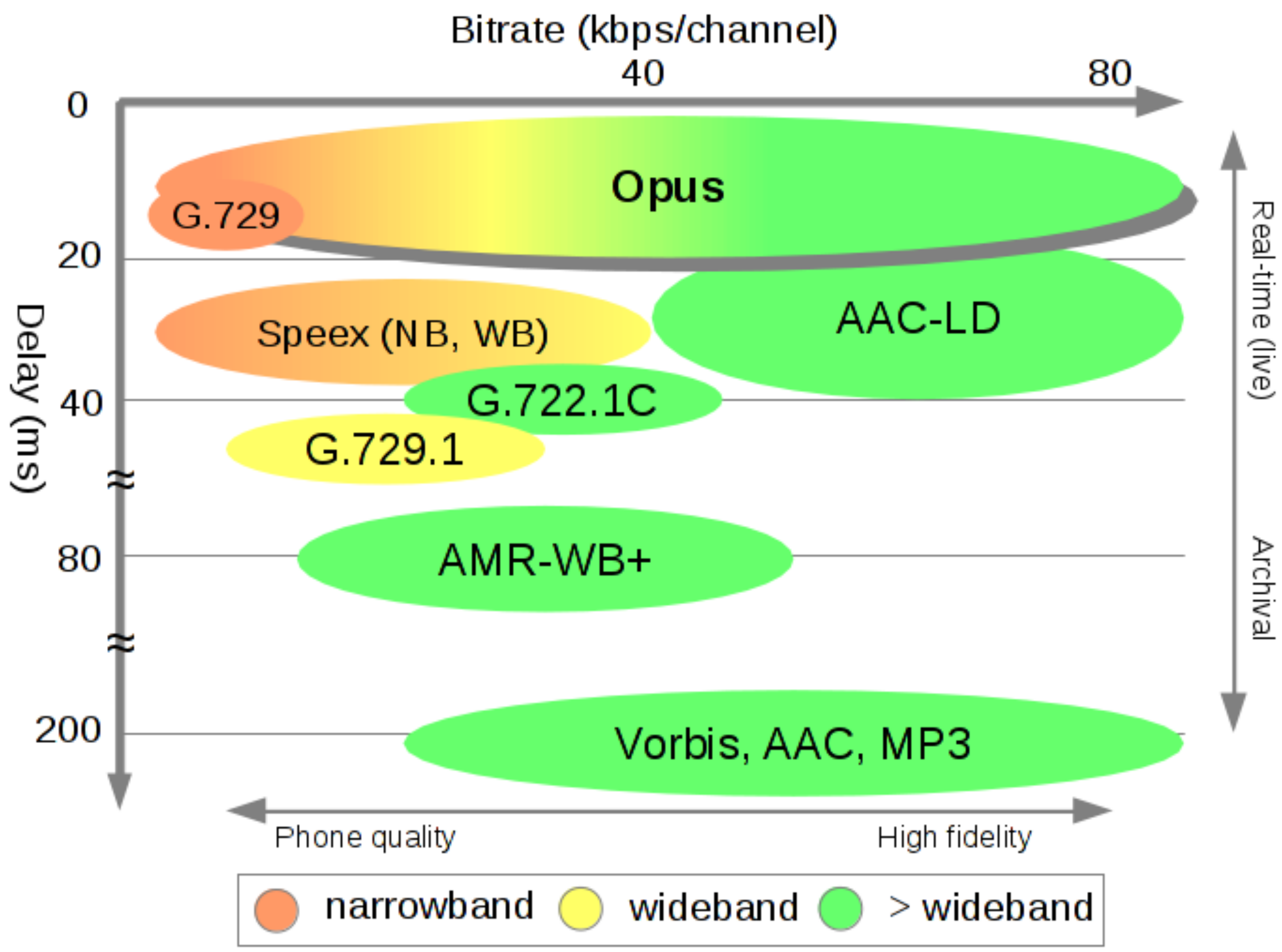
Opus audio codex (RFC 6716)

- interactive speech & (stereo) music
- 6 kb/s … 510 kb/s (music)
- frame size: 2.5 ms … 60 ms
- Linear prediction + MDCT
- SILK
  - Developed by Skype
  - Based on Linear Prediction
  - Efficient for voice
  - Up to 8 kHz audio bandwidth
- CELT
  - Developed by Xiph.Org
  - Based on MDCT
  - Good for universal audio/music

SILK decoder



Comparison



Bitrate (kbps/channel)
40
80
0
20
40
80
200
Delay (ms)
Opus
G.729
Speex (NB, WB)
AAC-LD
G.722.1C
G.729.1
AMR-WB+
Vorbis, AAC, MP3
Real-time (live)
Archival
Phone quality
High fidelity
narrowband
wideband
> wideband

Audio traffic models

- talkspurt: *typically*, constant bit rate:
  - one packet every 20. . . 100 ms ➠ mean: 1.67 s
- silence period: usually none
  - (maybe transmit background noise value) ➠ 1.34 s
- ➠ for telephone conversation, both roughly *exponentially distributed*
- double talk for “hand-off”
- may vary between conversations
  - ➠ only in aggregate

Sound localization

- Human ear uses 3 metrics for stereo localization:
  - intensity
  - time of arrival (TOA) – 7 μs
  - direction filtering and spectral shaping by outer ear
- For shorter wavelengths (4 – 20 kHz), head casts an acoustical shadow giving rise to a lower sound level at the ear farthest from the sound sources
- At long wavelength (20 Hz - 1 KHz) the, head is very small compared to wavelengths
  - In this case localization is based on perceived Interaural Time Differences (ITD)

Audio samples

- http://www.cs.columbia.edu/~hgs/audio/codecs.html
- Opus: http://opus-codec.org/examples/
  - both narrowband and wideband