4: Speech Compression

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Data Rates

- Telephone quality voice:
  - 8000 samples/sec, 8 bits/sample, mono
  - 64Kb/s
- CD quality audio:
  - 44100 samples/sec, 16 bits/sample, stereo
  - ~1.4Mb/s
- Communications channels and storage cost money (although less than they used to)
  - What can we do to reduce the transmission and/or storage costs without sacrificing too much quality?

Speech Codec Overview

- PCM - send every sample
- DPCM - send differences between samples
- ADPCM - send differences, but adapt how we code them
- SB-ADPCM - wideband codec, use ADPCM twice, once for lower frequencies, again at lower bitrate for upper frequencies.
- LPC - linear model of speech formation
- CELP - use LPC as base, but also use some bits to code corrections for the things LPC gets wrong.

PCM

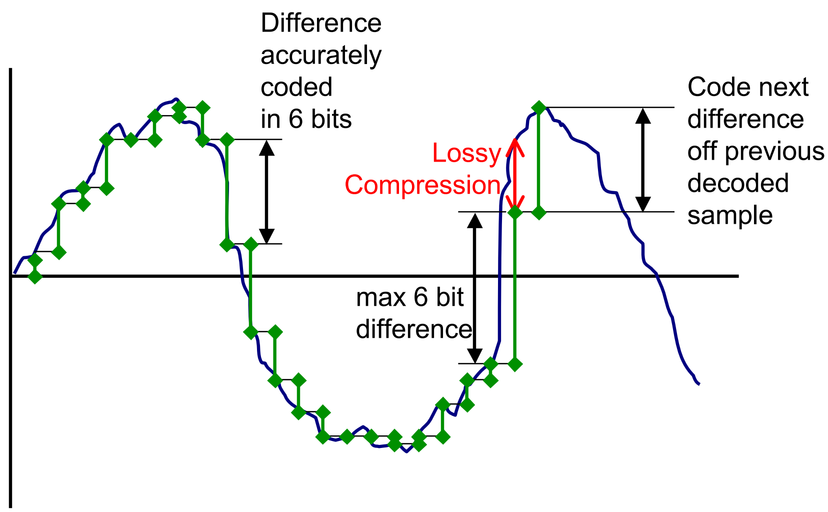
- μ-law and a-law PCM have already reduced the data sent.
- Lost frequencies above 4KHz.
- Non-linear encoding to reduce bits per sample.

- However, each sample is still independently encoded.
  - In reality, samples are correlated.
  - Can utilize this correlation to reduce the data sent.

Differential PCM

- Normally the difference between samples is relatively small and can be coded with less than 8 bits.
- Simplest codec sends only the differences between samples.
  - Typically use 6 bits for difference, rather than 8 bits for absolute value.
- Compression is *lossy, as* not all differences can be coded
  - Decoded signal is slightly degraded.
  - Next difference must then be encoded off the previous *decoded* sample, so losses don't accumulate.

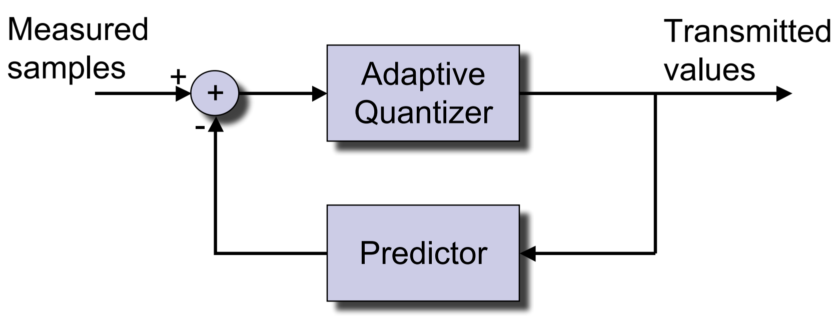
Differential PCM



ADPCM (Adaptive Differential PCM)

- 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.

ADPCM



ADPCM

- Adaptive quantization cannot always exactly encode a difference.
  - Shows up as quantization noise.
- Modems and fax machines try to use the full channel capacity.
  - If they succeed, one sample is not predictable from the next.
  - ADPCM will cause them to fail or work poorly.

- ADPCM not normally used on national voice circuits, but commonly used internationally to save capacity on expensive satellite or undersea fibres.
  - May attempt to detect if it's a modem, and switch back to regular PCM.

Predictor Error

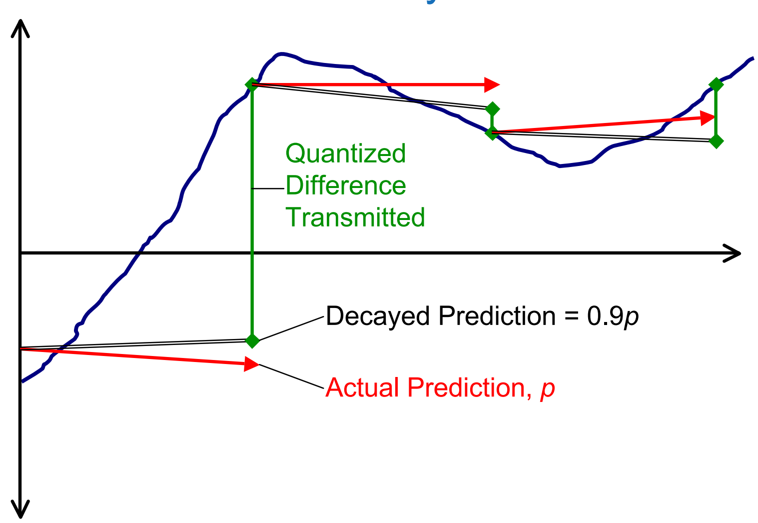
- What happens if the signal gets corrupted while being transmitted?
  - Wrong value will be decoded.
  - Predictor will be incorrect.
  - All future values will be decoded incorrectly!

- Modern voice circuits have low but non-zero error rates.
  - But ADPCM was used on older circuits with higher loss rates too. How?

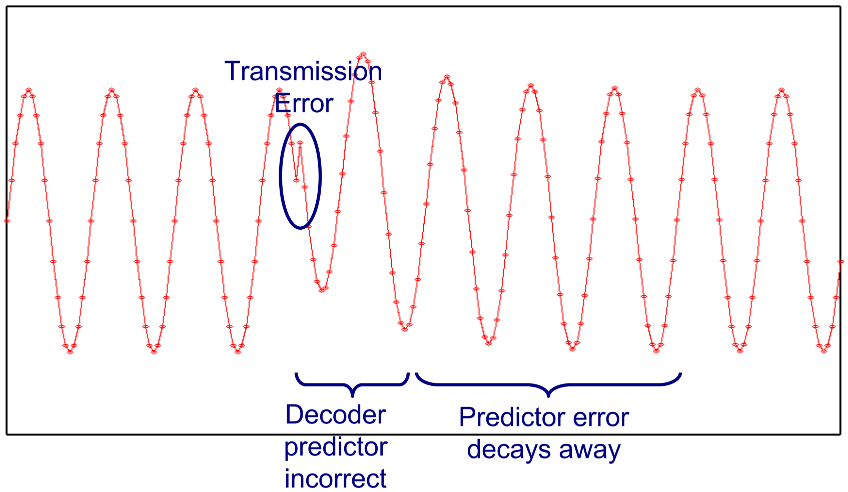
ADPCM Predictor Error

- Want to design a codec so that errors do not persist.
- Build in an automatic decay towards zero.
  - If only differences of zero were sent, the predictor would decay the predicted (and hence decoded) value towards zero.
- Differences have a mean value of zero (there are as many positive differences as negative ones).
  - Thus predictor decay ensures that any error will also decrease over time until it disappears.

ADPCM Prediction Decay



ADPCM Predictor Error



Sub-band ADPCM

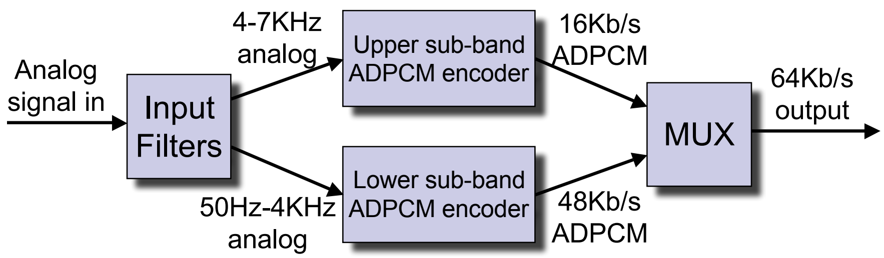
- Regular ADPCM reduces the bitrate of 8KHz sampled audio (typically 32Kb/s).
- If we have a 64Kb/s channel (eg ISDN), we could use the same techniques to produce better that toll-quality.

- Could just use ADPCM with 16KHz sampled audio, but not all frequencies are of equal importance.
  - 0-3.5KHz important for intelligibility
  - 3.5-7KHz helps speaker recognition and conveys emotion
- Sub-band ADPCM codes these two ranges separately.

Sub-band ADPCM

Filter into two bands:

- □ 50Hz - 3.5 KHz: sample at 8KHz, encode at 48KB/s
- □ 3.5KHz - 7KHz: sample at 16KHz, encode at 16KB/s



Sub-band ADPCM

- Practical issue:
  - □ Unless you have dedicated hardware, probably can't sample two sub-bands separately at the same time.
  - □ Need to process digitally.
    - Sample at 16KHz.
    - Use digital filters to split sub-bands and downsample the lower sub-band to 8KHz.

Key point of Sub-band ADPCM:

- □ Not all frequencies are of equal importance (quantization noise is more disruptive to some parts of the signal than others)
- □ Allocate the bits where they do most good.

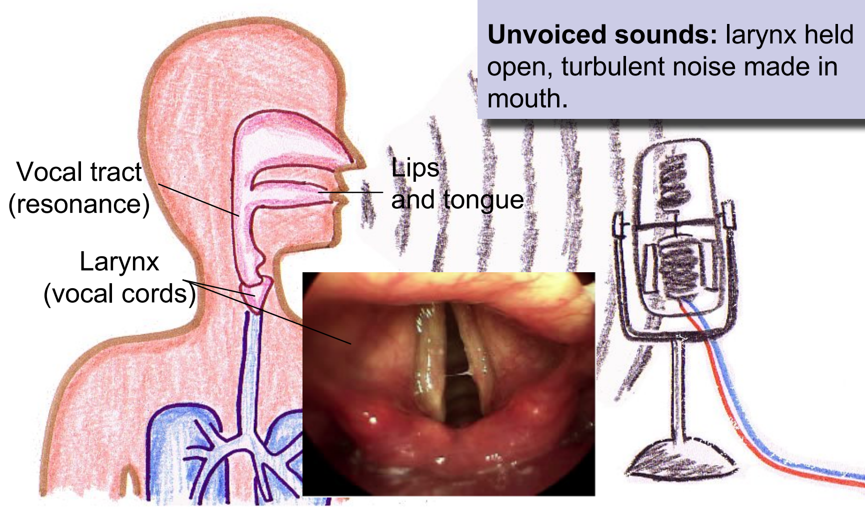
Model-based Coding

- PCM, DPCM and ADPCM directly code the received audio signal.
- An alternative approach is to build a *parameterized model of the sound source* (ie. Human voice).
- For each time slice (eg 20ms):
  - Analyse 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

Voiced sounds: series of pulses of air as larynx opens and closes. Basic tone then shaped by changing resonance of vocal tract.

Unvoiced sounds: larynx held open, turbulent noise made in mouth.



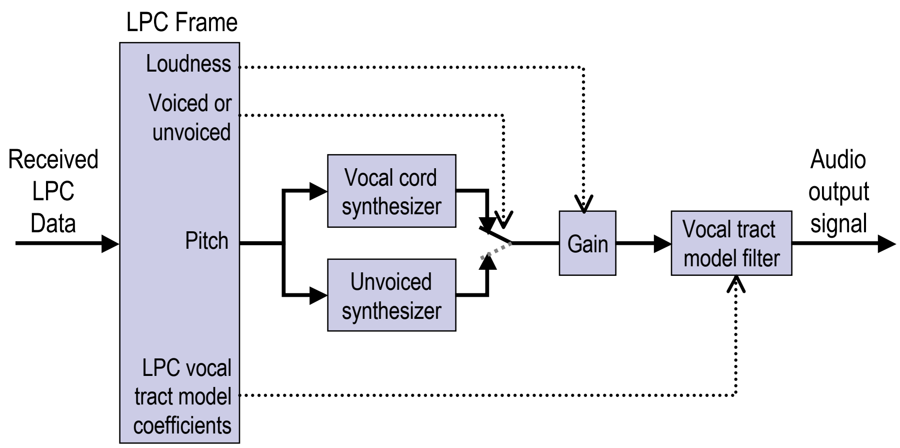
Linear Predictive Coding (LPC)

- Introduced in 1960s.
- Low-bitrate encoder:
  - 1.2Kb/s - 4Kb/s
- Sounds very synthetic
  - Basic LPC mostly used where bitrate really matters (eg in miltary applications)
  - Most modern voice codecs (eg GSM) are based on enhanced LPC encoders.

LPC

- Digitize signal, and split into segments (eg 20ms)
- For each segment, determine:
  - Pitch of the signal (ie basic formant frequency)
  - Loudness of the signal.
  - Whether sound is voiced or unvoiced
    - Voiced: vowels, “m”, “v”, “l”
    - Unvoiced: “f”, “s”
  - Vocal tract excitation parameters (LPC coefficients)

LPC Decoder



LPC Decoder

- Vocal chord synthesizer generates a series of impulses.
- Unvoiced synthesizer is a white noise source.

- Vocal tract model uses a linear predictive filter.
  - n^{th} sample is a linear combination of the previous p samples plus an error term:

    $$x_n = a_1 x_{n-1} + a_2 x_{n-2} + .. + a_n x_{n-p} + e_n$$

  - e_n comes from the synthesizer.
  - The coefficients $a_1 .. a_p$ comprise the vocal tract model, and shape the synthesized sounds.

LPC Encoder

- Once pitch and voice/unvoiced are determined, encoding consists of deriving the optimal LPC coefficients ($a_1 .. a_p$) for the vocal tract model so as to minimize the mean-square error between the predicted signal and the actual signal.
- Problem is straightforward in principle. In practice it involves:
  1. The computation of a matrix of coefficient values.
  2. The solution of a set of linear equations.
  - Several different ways exist to do this efficiently (autocorrelation, covariance, recursive latice formulation) to assure convergence to a unique solution.

Limitations of LPC Model

- LPC linear predictor is very simple.
  - For this to work, the vocal tract "tube" must not have any side branches (these would require a more complex model).
  - OK for vowels (tube is a reasonable model)
  - For nasal sounds, nose cavity forms a side branch.
- In practice this is ignored in pure LPC.
  - More complex codecs attempt to code the residue signal, which helps correct this.

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