SPEECH PROCESSING
Speech Coding

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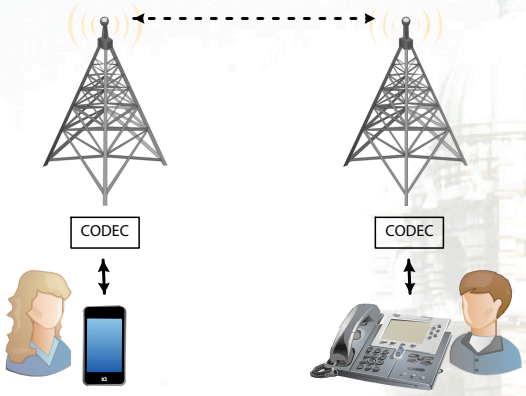
Aims of this module

- In this lecture we will study techniques for coding speech signals such that the bit-rate of speech transmission/storage can be reduced
  - Objectives of Speech Coding
  - Quality versus bit rate
  - Uniform Quantization
    - Quantization Noise
  - Non-uniform Quantization
  - Adaptive Quantization

- Simplified schematic



Speech Coding Objectives

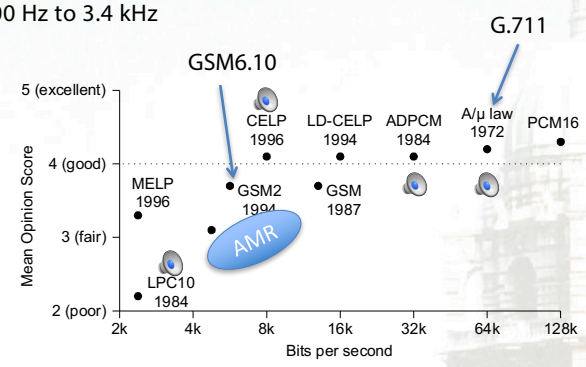
- High perceived quality
- High measured intelligibility
- Low bit rate (bits per second of speech)
- Low computational requirement (MIPS)
- Robustness to successive encode/decode cycles (transcoding)
- Robustness to transmission errors
- Objectives for real-time only:
  - Low coding/decoding delay (ms) - latency
  - Work with non-speech signals (e.g. touch tone - DTMF)

Subjective Quality Assessment

- MOS-LQS (Mean Opinion Score – Listening Quality Subjective):
  - a panel of test listeners rates the quality 1 to 5
    1=bad, 2=poor, 3=fair, 4=good, 5=excellent
  - Typical landline ~ 4
  - Typical mobile ~ 3
  - Limit of intelligibility ~ 1.5
- DRT (Diagnostic Rhyme Test):
  - listeners choose between pairs of rhyming words (e.g. veal/feel)
- DAM (Diagnostic Acceptability Measure):
  - Trained listeners judge various factors e.g. muffledness, buzziness, intelligibility

MOS vs. bit rate

- Narrow band speech coders
  - 300 Hz to 3.4 kHz



Objective Quality Assessment

- **Segmental SNR**
  - Average value of $10\log_{10}(E_{\text{speech}}/E_{\text{error}})$ evaluated in 20 ms frames
    - Not good for coding schemes that introduce delays as even a one-sample time shift causes a large change in SNR
- **Spectral distances**
  - Based on the power spectrum of the original and coded-decoded speech signals, $P(\omega)$ and $Q(\omega)$
  - $<\ldots>$ denotes the average over the frequency range $0 \ldots 2\pi$

Itakura Distance

$$\log\left(\left\langle\frac{P(\omega)}{Q(\omega)}\right\rangle\right)-\left\langle\log\left(\frac{P(\omega)}{Q(\omega)}\right)\right\rangle$$

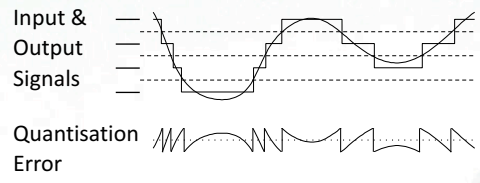
Itakura-Saito Distance

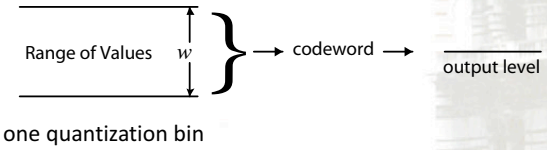
$$\left\langle\frac{P(\omega)}{Q(\omega)}-\log\left(\frac{P(\omega)}{Q(\omega)}\right)-1\right\rangle$$

  - Both distances are evaluated over 20 ms frames and averaged
  - Both may also be calculated directly from the LPC coefficients of original and coded speech
  - Neither is a true distance metric since they are asymmetrical $d(P,Q)\neq d(Q,P)$

Quantization Error

- Consider a finite number of quantization levels





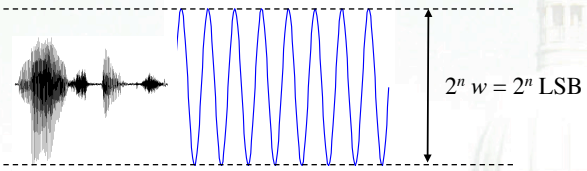
- Coding/Decoding introduces a quantization error of $\pm w/2$
- If input values are uniformly distributed within the bin, the mean square quantization error is

$$\int_{-w/2}^{+w/2} x^2 \frac{dx}{w} = \left[\frac{x^3}{3w}\right]_{-w/2}^{+w/2} = \frac{w^2}{12}$$

or in RMS: $0.289w$

- If the quantization levels are uniformly spaced, w is the separation between adjacent output values
  - called a Least Significant Bit (LSB)

Linear Pulse Code Modulation (PCM)

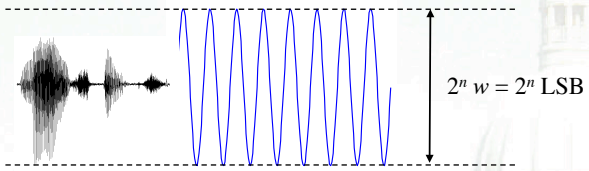


- RMS quantization noise is $0.289w = 0.289$ LSB
- **Sine Wave SNR**
  - Full scale sine wave has peak of $0.5\ w$ x 2^n and RMS value of $0.354\ w$ x 2^n

$$\text{SNR} = 20\log_{10}\left(\frac{\text{rms of maximal sine wave}}{\text{rms of quantisation noise}}\right)$$

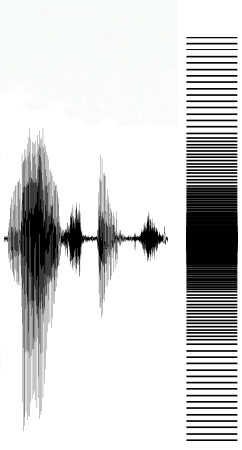
$$= 20\log_{10}\left(\frac{0.354}{0.289}\times 2^n\right) = 20\log_{10}(1.22) + 20n\log_{10}(2)$$

$$= 1.76 + 6.02n \text{ dB}$$



- **Speech SNR**
  - RMS value of speech or music signal is 10 to 20 dB less than a sine wave with the same peak amplitude
  - SNR is 10 to 20 dB worse
  - Need to consider time-varying envelope in order to design an efficient quantization scheme
  - Much of a speech signal has relatively small amplitude

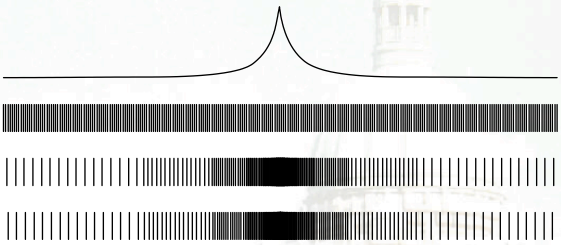
Concept of Non-uniform Quantization



- RMS quantization error = 0.29 × Quantization step
  - As defined in two previous slides
- Low amplitude samples are more common than high amplitude samples
  - The pdf of speech samples is not a uniform distribution
  - Super-Gaussian distributions are more correctly representative, such as Laplacian
- Concept:
  - Make quantization steps closer together for more common samples amplitudes

Non-uniform PCM @ 64 kbits/s (fs = 8 kHz)

- Speech pdf
- Uniform Quantization
- A-law
- mu-law



Ticks indicate quantization levels

Byte Structure for Non-uniform Coding

- Instead of linear coding, use floating point with smaller intervals for more frequently occurring signal levels:
  - s = sign bit (1 = +ve)
  - e = 3-bit exponent
  - m = 4-bit mantissa Range of values ≈ ±4095
  - 0 dB = 2002 (μ-law) or 2017 (A-law)

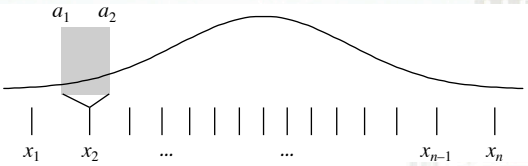
s	e	e	e	m	m	m	m

8 bits making up 1 sample

Usually called G.711

- μ-law (US standard):
  - Bin centres at $\pm\{(m+16½)\ 2^e - 16½\}$
  - Bin widths: 2^e
- A-law (European standard):
  - Bin centres at $\pm(m+16½)\ 2^e$ for $e>0$ and $\pm(2m+1)$ for $e=0$
  - Bin widths: 2^e for $e>0$ and 2 for $e=0$
- Performance: MOS=4.3, DRT=95, DAM=73, SNR ≤38dB
  - For sine waves > –30 dB, SNR is almost constant at 38 dB
    At very low signal levels (–40dB) μ-law≈13 bit, A-law ≈12 bit linear

Quantization of a Signal with Gaussian pdf



- Input values from a_{i-1} to a_i are converted to x_i
- For minimum error we must have $a_i = \frac{1}{2}(x_i + x_{i+1})$
- For this range of x, the quantization error is $q(x) = x_i - x$

- The mean square quantization error is

$$E = \int_{-\infty}^{+\infty} p(x)q^2(x)dx = \sum_{i=1}^{n} \int_{a_{i-1}}^{a_i} p(x)(x_i - x)^2 dx$$

- Differentiating w.r.t. x_i

$$\frac{\partial E}{\partial x_i} = \int_{a_{i-1}}^{a_i} -2p(x)(x - x_i)dx$$

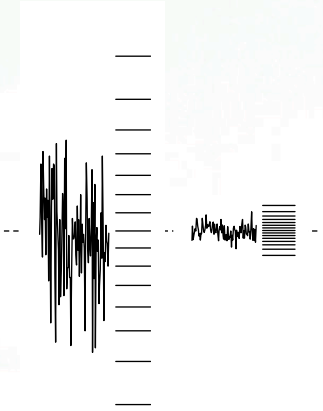
$$= 2x_i \int_{a_{i-1}}^{a_i} p(x)dx - 2\int_{a_{i-1}}^{a_i} xp(x)dx$$

- Find minimum error by setting the derivative to zero

$$x_i = \frac{\int_{a_{i-1}}^{a_i} xp(x)dx}{\int_{a_{i-1}}^{a_i} p(x)dx} \quad \Leftrightarrow x_i \text{ is at the centroid of its bin}$$

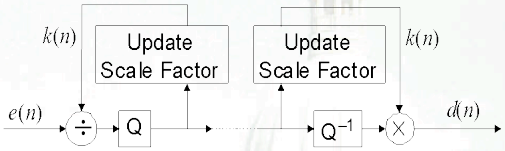
- To find optimal $\{x_i\}$, take an initial guess and iteratively use the above equation to improve their estimates.
  - For 15 bins, SNR = 19.7dB

Adaptive Quantization



- We want to use smaller steps when the signal level is small
- We can adjust the step sizes automatically according to the signal level:
  - If almost all samples correspond to the central (small amplitude) quantization levels, we must reduce the step size
  - If many samples correspond to the outer (high amplitude) quantization levels, we must increase the step sizes

- Transmitter:
  - Divide input signal by k
  - Q quantizes to 15 levels
  - Transmit one of 15 code words
  - Update k to new value



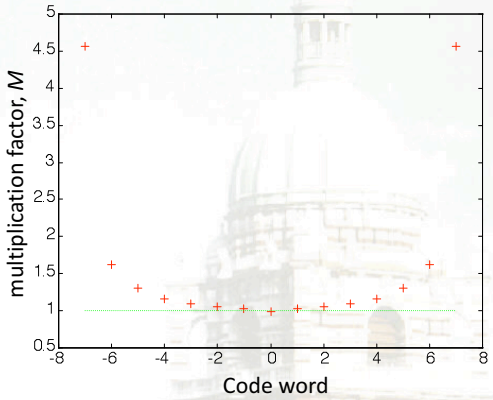
- Receiver:
  - Inverse quantizer, Q^{-1} converts received code word to a number
  - Multiply by k to give output value
  - Update k to new value
- Update Algorithm
  - Decrease k whenever a small code word is transmitted/received
  - Increase k whenever a large code word is transmitted/received
  - Transmitter and Receiver will always have the same value of k

Scale Factor Updates

- k is decreased slightly for "zero" code word (× 0.98)
- k is increased rapidly for extreme code words (× 4.6)
- k is increased slightly for other code words
- Calculation done in log domain:

$$\log(k_n) = 0.97\log(k_{n-1}) + \log(M)$$

- "leakage factor" of 0.97 allows recovery from transmission errors.



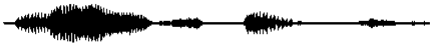
Adaptive Quantization Example

- Small signal sections ⇒ small k

Original

Quantised (15 levels)

Error

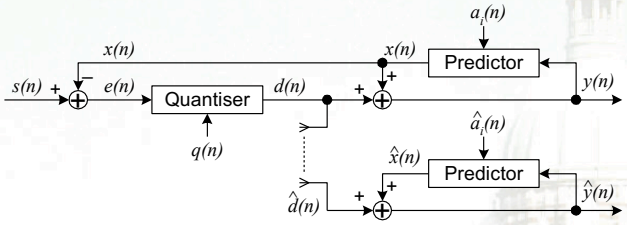


k

Part 2 – Adaptive Differential PCM

- In this lecture, we study a widely used method of waveform coding
- Differential Speech Coding
  - ADPCM coding
  - Prediction Filter
  - Filter Adaptation
  - Stability Triangle

Differential PCM



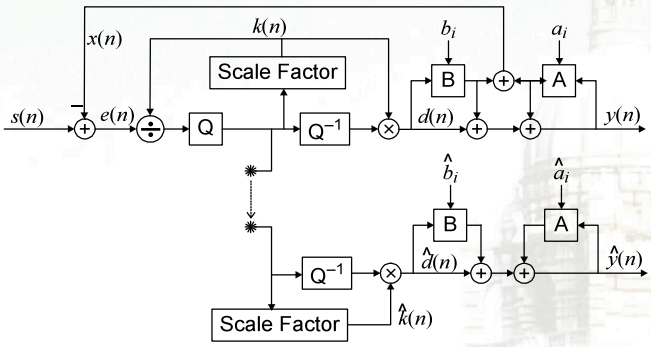
- The encoder contains a complete copy of the decoder
- If no transmission errors $\hat{y}(n) \equiv y(n)$
- Quantization noise, $q(n) = d(n) - e(n)$
- Output $y(n) = x(n) + d(n) = s(n) - e(n) + d(n) = s(n) + q(n)$

Good prediction leads to good SNR

$$SNR = \frac{E_s}{E_q} = \frac{E_s}{E_e} \times \frac{E_e}{E_q} = G_p \times \frac{E_e}{E_q}$$

- The predictor is chosen to maximize the prediction gain:
  - G_p = signal energy / prediction error energy.
- E_e/E_q is the quantizer SNR
- The a_i can be one of:
  - fixed constants
  - transmitted separately
  - deduced from $d(n)$ and/or $y(n)$

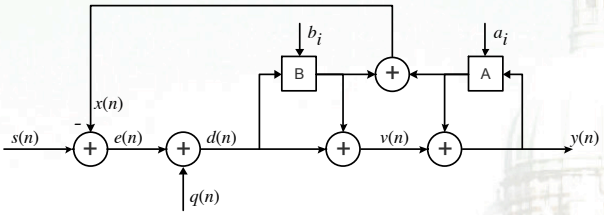
Adaptive Differential PCM (ADPCM) @ 32 kbit/s



Performance: MOS=4.1, DRT=94, DAM=68, 2 MIPS

- Quantizer has 15 levels ⇒ 4 bits x 8 kHz = 32 kbit/s
- Quantization levels assume $e(n)$ / $k(n)$ is Gaussian with unit variance.
  - Scale factor $k(n)$ is adjusted to make this true
- The LPC filter, A, is only of order 2 so that it is easy to ensure stability.
- The FIR filter, B, is of order 6 and partially compensates for the low order of A.

Prediction Filter



- Filters: $A(z) = a_1 z^{-1} + a_2 z^{-2}$

  $B(z) = b_1 z^{-1} + b_2 z^{-2} + b_3 z^{-3} + b_4 z^{-4} + b_5 z^{-5} + b_6 z^{-6}$

- Note that: $\frac{\partial A}{\partial a_i} = \frac{\partial B}{\partial b_i} = z^{-i}$

  and $A(z)$ and $B(z)$ filters involve only past values

- Signals $Y = D + BD + AY \Rightarrow Y = \frac{1+B}{1-A} D$

  $X = BD + AY = Y - D = \frac{A+B}{1-A} D$

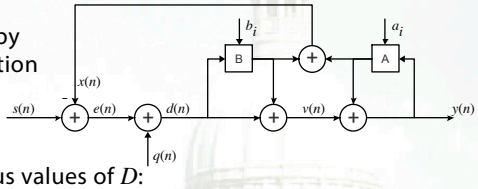
  $D = S + Q - X = S + Q - Y + D \Rightarrow Y = S + Q$

- In the absence of transmission errors, the $y(n)$ output at the receiver and transmitter will be identical and equal to the input speech plus the quantization noise

Filter Adaptation

- Adaptation of b_i
  - We adjust the b_i to reduce $d^2(n)$ by moving them a little in the direction

$$-\frac{\partial}{\partial b_i}\left(d^2(n)\right) = -2d(n)\frac{\partial d(n)}{\partial b_i}$$



  - We express D in terms of previous values of D:

$$D = S + Q - X = S + Q - \frac{A+B}{1-A}D$$

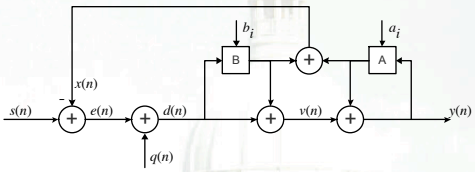
$$\frac{\partial D}{\partial b_i} = -z^{-i}\frac{1}{1-A}D \approx -z^{-i}D \;\Rightarrow\; \frac{\partial d(n)}{\partial b_i} \approx -d(n-i)$$

  - Hence we want to adjust b_i in the direction $d(n)d(n\text{-}i)$:

$$b_i \leftarrow 0.996b_i + 0.008 \times \mathrm{sgn}\left(d(n)d(n-i)\right)$$

    - The function sgn() takes the sign of its argument.
    - The 0.996 leakage factor assists in recovery from transmission errors and prevents b_i from exceeding ±2

- Adaptation of a_i



  - Adjustment of a_i is similar but more ad-hoc:

$$a_1 \leftarrow 0.996a_1 + 0.012 \times \mathrm{sgn}\left(v(n)v(n-1)\right)$$

$$a_2 \leftarrow 0.996a_2 + 0.008 \times \mathrm{sgn}\left(v(n)v(n-2)\right) - 0.03a_1 \times \mathrm{sgn}\left(v(n)v(n-1)\right)$$

Filter Stability

$$\frac{1}{1-a_1z^{-1}-a_2z^{-2}} \text{ has poles at } \left(a_1 \pm \sqrt{a_1^2+4a_2}\right)$$

- For stability, we make the poles have modulus less than R:
- Real: $a_1^2+4a_2>0$ and $\begin{cases}\frac{1}{2}\left(a_1+\sqrt{a_1^2+4a_2}\right)<R\\ \frac{1}{2}\left(a_1-\sqrt{a_1^2+4a_2}\right)>-R\end{cases}$

$$\Rightarrow \sqrt{a_1^2+4a_2}<2R\pm a_1$$

$$\Rightarrow a_1^2+4a_2<a_1^2\pm 4Ra_1+4R^2 \Rightarrow a_2<R^2\pm Ra_1$$

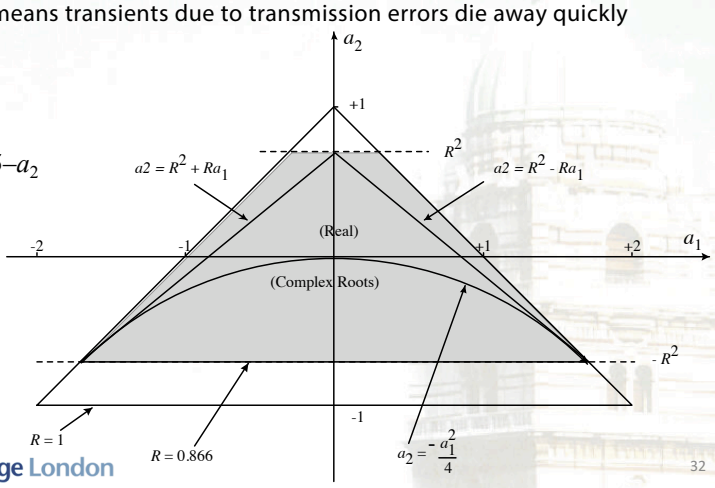
- Complex: $a_1^2+4a_2<0$ and $a_2>-R^2$

Stability Triangle

- Poles more or less kept within a radius of 0.866
  - Prevents huge gains and coefficient sensitivity.
  - Also means transients due to transmission errors die away quickly

Shading:

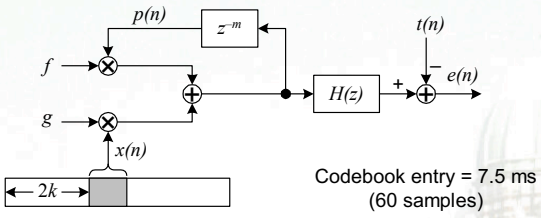
- $|a_2|<0.75$
- $|a_1|<0.9375-a_2$



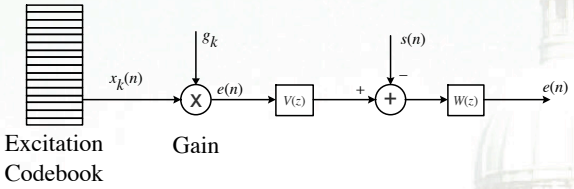
Part 3 – Code Excited Linear Prediction (CELP)

- CELP is one of the most widely used speech coding methods, adopted into the GSM6.10 narrow band speech coding standard. Derivates are also used in the adaptive multirate codecs for narrow and wide band speech.
  - Principles of CELP coding
  - Adaptive and Stochastic Codebooks
  - Perceptual Error Weighting
  - Codebook searching

4.8 kbits/s CELP Encoding



CELP Coder



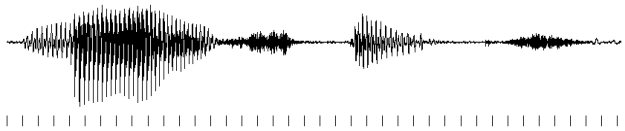
- Performance:
  - 16 kbit/s: MOS=4.2, Delay = 1.5 ms, 19 MIPS
  - 8 kbit/s: MOS=4.1, Delay = 35 ms, 25 MIPS
  - 2.4 kbit/s: MOS=3.3, Delay = 45 ms, 20 MIPS

Overview of Operation of CELP Coder

- Encoding
  - LPC analysis $\Rightarrow V(z)$
  - Define perceptual weighting filter. This permits more noise at formant frequencies where it will be masked by the speech
  - Define a codebook as a set of possible excitation signals. Codebook is stored both in the coder and decoder.
    - Synthesize speech using each codebook entry in turn as the input to $V(z)$
    - Calculate optimum gain to minimize perceptually weighted error energy in speech frame
    - Select codebook entry that gives lowest error
  - Transmit LPC parameters and codebook index
- Decoding
  - Receive LPC parameters and codebook index
  - Resynthesise speech using $V(z)$ and codebook entry

This approach is known as 'analysis by synthesis'

LPC Analysis for CELP



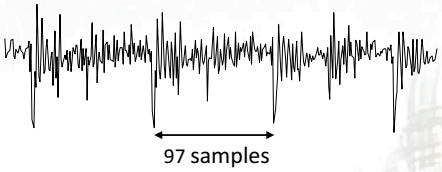
- Segment speech into 30 ms frames (240 samples) using Hamming window
- Perform 10th order autocorrelation LPC (no preemphasis)
- Bandwidth expansion: form $V(z/0.994)$ by multiplying a_i by 0.994^i
  - Moves all poles in towards the origin of the z plane
  - Pole on unit circle moves to a radius of 0.994 giving a 3 dB bandwidth of $\approx -\ln(r)/\pi \approx (1-r)/\pi = 0.0019$; an unnormalized frequency of 15 Hz @ fs = 8 kHz
  - Improves LSP quantization
  - Reduces parameter sensitivity of spectrum
  - Avoids chirps due to very sharp formant peaks

- Convert 10 LPC coefficients to 10 LSF frequencies
- Quantize using 4 bits for each of f_2 to f_5 and 3 bits for each of the others (34 bits in total) from empirically determined probability density functions.
- Smooth filter transitions by linearly interpolating a new set of LSP frequencies every ¼ frame (subframe).

LPC Residual

- If we were to filter the speech signal by $A(z)$, the inverse vocal-tract filter, we would obtain the LPC residual:

$x(n)$



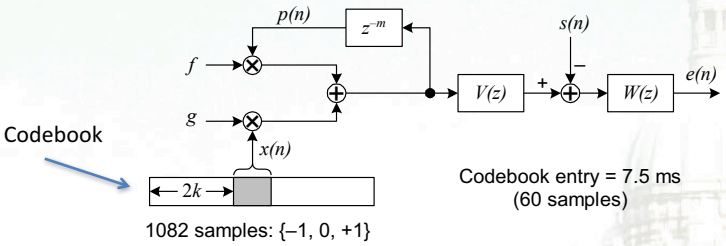
- We can represent this signal as the sum of a periodic component and a noise component.

- We can represent the periodic component by using a delayed version of $x(n)$ to predict itself: this is called the ***long-term predictor*** or *LTP*. Subtracting this delayed version gives us just the noise component:

$$x(n) - 0.76x(n-97)$$

- Having removed the periodic component, we search a codebook of noise signals (a *stochastic codebook*) for the best match to the residual excitation signal.

Excitation Codebooks



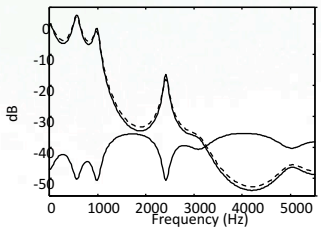
- Excitation signal is the sum of 60-sample-long segments (60 samples = 7.5 ms = ¼ frame) from two sources :
- Long-term predictor (also called ***Adaptive Codebook***) is a delayed version of previous excitation samples multiplied by a gain, f.
  - The value of m is in the range $20 \le m \le 147 \Rightarrow$ 7 bits ($400 \text{ Hz} > f_0 > 54 \text{ Hz}$)
  - Some coders use fractional m for improved resolution at high frequencies: this requires interpolation.
- ***Stochastic Codebook*** contains 1082 independent random values from the set {–1, 0, +1} with probabilities {0.1, 0. 8, 0.1}. The values of k is in the range $0 \le k \le 511 \Rightarrow$ 9 bits are needed.
- ***Encoding Process***
  - The values of m and f are determined first.
  - Then each possible value of k is tried to see which gives the lowest weighted error.

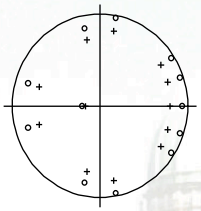
Perceptual Error Weighting

- We want to choose the LTP delay and codebook entry that gives the best sounding resynthesized speech
- We exploit the phenomenon of ***masking***
  - a listener will not notice noise at the formant frequencies because it will be overwhelmed ('masked') by the speech energy.
- We therefore filter the error signal by:

$$W(z) = \frac{V(z/0.8)}{V(z)} = \frac{A(z)}{A(z/0.8)}$$

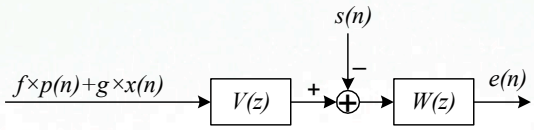
- This emphasizes the error at noticeable frequencies, causing the optimization to minimize the noticeable errors more strongly.



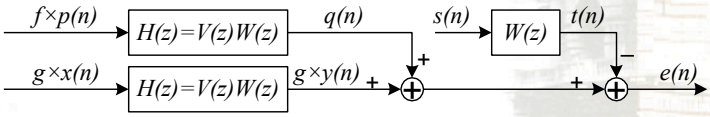


- Solid line is original $V(z)$
- Dashed line is bandwidth expanded $V(z/0.994)$ with slightly lower, broader peaks: more robust.
- Lower solid line is $W(z)$ shifted down (arbitrarily, to make it more visible). Errors are emphasized between formants.
- Pole-zero plot shows $W(z)$
  - Poles from $1/A(z/0.8)$;
  - Zeros from $A(z)$.

Codebook Search



- We can interchange the subtraction and $W(z)$ and also separate the excitation components:



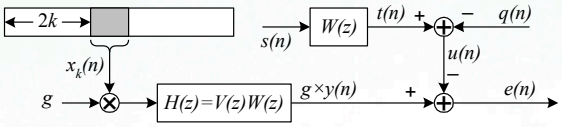
- Here $q(n)$ is the output from the filter $H(z)$ due to both the LTP signal $p(n)$ and the excitation in previous sub-frames
- $y(n)$ is the output due to the selected portion of the codebook
- If $h(n)$ is the impulse response of $H(z)$ then:

$$y(n) = \sum_{r=0}^{n} h(r)x(n-r) \quad \text{for } n = 0,1,\ldots,N-1$$

  where the sub-frame length, N, is 60 samples (7.5 ms)
  - Note that even though $h(r)$ is an infinite impulse response, the summation only goes to n since $x(n–r)$ is zero for $r > n$
- We want to choose the codebook entry that minimizes

$$E = \sum_{n=0}^{N-1} e^2(n) = \sum_{n=0}^{N-1} \big(gy(n) + q(n) - t(n)\big)^2$$
$$= \sum_{n=0}^{N-1} \big(gy(n) - u(n)\big)^2 \quad \text{where} \quad u(n) = t(n) - q(n)$$

Gain Optimization



- We can find the optimum g by setting $\frac{\partial E}{\partial g}$ to zero:

$$E = \sum_{n=0}^{N} e^2(n) = \sum_{n=0}^{N} \left(gy(n) - u(n)\right)^2$$

$$\Rightarrow 0 = \frac{\partial E}{\partial g} = \sum_n y(n)e(n) = \sum_n gy^2(n) - \sum_n y(n)u(n)$$

$$\Rightarrow g_{opt} = \frac{\sum_n y(n)u(n)}{\sum_n y^2(n)}$$

- Substituting this in the expression for E gives:

$$E_{opt} = \sum_n \left(g_{opt}y(n) - u(n)\right)^2 = \sum_n \left(g_{opt}^2 y^2(n) - 2g_{opt}u(n)y(n) + u^2(n)\right)$$

$$= g_{opt}^2 \sum_n y^2(n) - 2g_{opt}\sum_n u(n)y(n) + \sum_n u^2(n)$$

$$= \sum_n u^2(n) - \frac{\left(\sum_n u(n)y(n)\right)^2}{\sum_n y^2(n)}$$

Energy Calculation

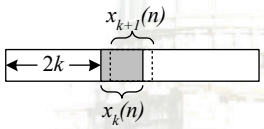
- We need to find k that minimizes

Try each k exhaustively to find the minimum error

$$E_{opt} = \sum_n u^2(n) - \frac{\left(\sum_n u(n)y_k(n)\right)^2}{\sum_n y_k^2(n)} \quad \text{where } y_k(n) = \sum_{j=0}^{n} h(j)x_k(n-j)$$

- Note that for $n \geq 2$

$$x_k(n) = x_{k+1}(n-2)$$



- Hence

$$\begin{aligned} y_k(n) &= \sum_{r=0}^{n} h(r)x_k(n-r) \\ &= h(n)x_k(0) + h(n-1)x_k(1) + \sum_{r=0}^{n-2} h(r)x_{k+1}(n-2-r) \\ &= h(n)x_k(0) + h(n-1)x_k(1) + y_{k+1}(n-2) \end{aligned}$$

Thus we can derive $y_k()$ from $y_{k+1}()$ with very little extra computation

We only need to perform the full calculation for the last codebook entry.

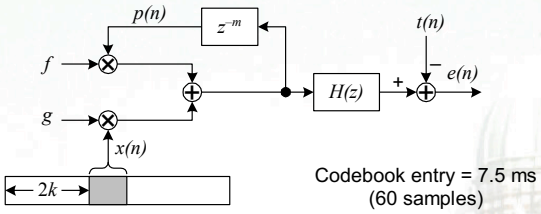
Note too that the multiplication $h(n)x_k(0)$ is particularly simple since

$$x_k(0) \in \{-1,0,+1\}.$$

No divisions are needed for comparing energies since:

$$\left(C - \frac{A_2^2}{B_2^2}\right) < \left(C - \frac{A_1^2}{B_1^2}\right) \Leftrightarrow A_1^2 B_2^2 < A_2^2 B_1^2$$

4.8 kbits/s CELP Encoding



- Do LPC analysis for 30 ms frame (34 bits transmitted)
- For each ¼-frame segment (7.5 ms) do :
  - Calculate $q(n)$ as the output of $H(z)$ due to the excitation from previous sub-frames
  - Search for the optimal LTP delay, m (4 × 7 bits transmitted per frame) and determine the optimal LTP gain, f (4 × 5 bits transmitted)
  - Recalculate $q(n)$ to equal the output of $H(z)$ when driven by the sum of the selected LTP signal as well as the inputs from previous frames
  - Search for the optimal stochastic codebook entry, k (4 × 9 bits transmitted) and gain, g (4 × 5 bits transmitted)
  - Calculate a Hamming error correction code to protect high-order bits of m (5 bits transmitted)
  - Allow one bit for future expansion of the standard
- Total = 144 bits per 30 ms frame
- Decoding is the same as encoding but without $W(z)$