Multimedia Communications

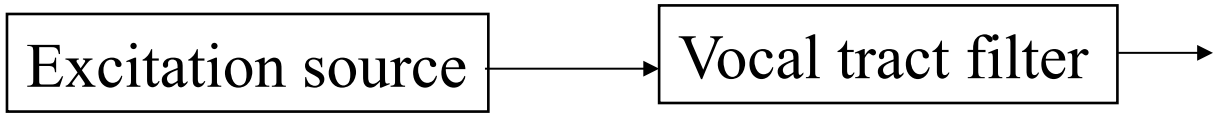
Speech Compression

Introduction

- No matter what language is being spoken, speech is generated using machinery that is not very different from person to person
- This machinery has to obey physical laws that substantially limit the behavior of the outputs
- Speech can be analyzed in terms of a model and model can be extracted and transmitted to the receiver
- At the receiver the speech is synthesized using the model

Introduction

- Speech is produced by forcing air first through an elastic opening, the vocal cords, laryngeal, oral, nasal and pharynx passages and finally through mouth and nasal cavity
- First sound is generated and is modulated into speech as it traverses through the vocal tract
- In order to generate a fragment of speech we have to generate a sequence of excitation signals and the corresponding sequence of vocal tract approximation



Introduction

- Many speech compression schemes:
  - Channel vocoder
  - Linear predictive coder (LPC)
  - Code excited linear prediction (CELP)
  - Sinusoidal coders
  - Mixed excitation linear prediction (MELP)

Vocoder

- Each segment of input speech is analyzed using a bank of band-pass filters called the analysis filters
- Energy at the output of each filter is estimated at fixed time intervals and transmitted to the receiver
- A decision is made as to whether the speech in that segment is voiced or unvoiced
- Voiced sound tend to have a pseudo-periodic structure
- The period of the fundamental harmonic is called the pitch period
- Transmitter also forms an estimate of the pitch period which is transmitted to the receiver

Vocoder

- Unvoiced sounds tend to have a noise like structure
- At the receiver, the vocal tract filter is implemented by a bank of band-pass filters (identical to the filters at transmitter)
- The input to the filters is noise source (for unvoiced segments) or periodic pulse (for voiced)

Vocoder

- Variations of vocoder: formant vocoder, voice excited vocoder
- Vocal tract is a tube of non-uniform cross section that resonates at a number of different frequencies known as formants
- Formant vocoder transmits an estimate of the formant values (usually 4) and an estimate of the bandwidth of each formant
- At the receiver the excitation signal is passed through tunable filters tuned to the formant frequency and bandwidth

Vocoder

- In voice excited channel vocoder, the voice is first filtered using a narrow-band low-pass filter
- Output of the filter is sampled and transmitted to the receiver
- At the receiver, the low-pass signal is passed through a nonlinearity to generate higher order harmonics that together with the low pass signal are used as the excitation signal
- Voice excitation removes the problem of pitch extraction and declaring every segment voiced or unvoiced

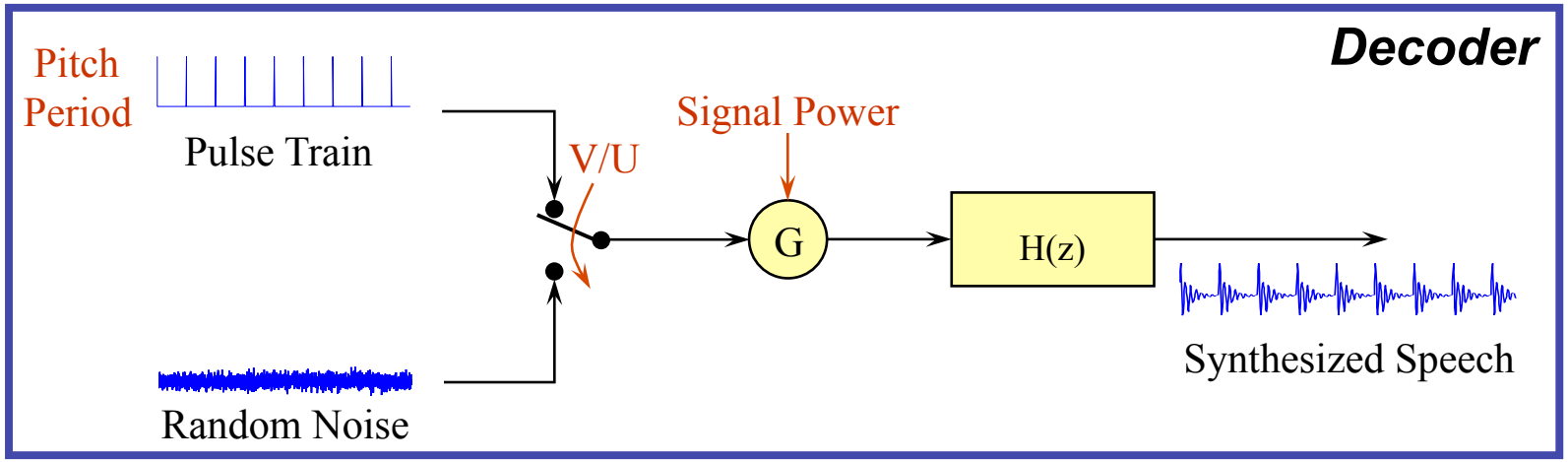
Linear Predictive Coder (LPC-10)

- Instead of the vocal tract being modeled by a bank of filters, in LPC, it is modeled as a single linear filter:

$$y[n] = \sum_{i=1}^{M} b_i y[n-i] + Ge[n]$$

- The input to the vocal tract filter is either the output of a random noise generator or a periodic pulse generator
- At the transmitter a segment of speech is analyzed to make a decision on voiced/unvoiced, the pitch period and the parameters of the vocal tract filter
- In LPC-10 input speech is sampled at 8000 samples per second which is broken into 180 sample segments
- Rate: 2.4 kbps

LPC-10



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Multi-pulse linear predictive coding (MP-LPC)

- One of the most important factors in generating natural sounding speech is the excitation signal
- Human ear is especially sensitive to pitch errors
- Using a single pulse per pitch period leads to a buzzy twang
- Multi-pulse linear predictive coding (MP-LPC): several pulses were used during each segment
- Spacing of these pulses is determined by evaluating a number of patterns from a codebook of patterns

MP-LPC

- Each entry in codebook is an excitation sequence that consists of a few nonzero values separated by zeros
- Codebook entry generating minimum average weighted error is declared the best match.
- Index of the best match is sent to the receiver.

Regular pulse excitation (RPE) coding

- RPE is a modification of MP-LPC
- Instead of using excitation vectors in which the nonzero values are separated by an arbitrary number of zero values, the nonzero values occur at regularly spaced intervals.
- A variation of RPE called regular pulse excitation with long term prediction (RPE-LTP) was adopted as a standard for digital cellular phones by GSM.

Code excited linear prediction (CELP)

- In CELP instead of having a codebook of pulse patterns we allow a variety of excitation signals
- Given a segment, encoder obtains the vocal tract filter
- Encoder then excites the vocal tract filter with the entries of the codebook
- Difference between original speech segment and the synthesized speech is fed to a perceptual weighting filter
- Codebook entry generating minimum average weighted error is declared to the best match
- Two examples will be reviewed: Federal Standard 1016 (FS 1016), and G.728

FS 1016

- Vocal tract filter: $y[n] = \sum_{i=1}^{10} b_i y[n-i] + \beta y[n-P] + Ge[n]$
- Input speech is sampled at 8000 samples per second and divided into 30 ms frames containing 240 samples.
- Each frame is divided into four subframes of length 7.5 ms.
- The coefficients b_i are obtained using the auto-correlation method
- Pitch period is calculated once every subframe
- FS 1016 uses two codebooks: stochastic and adaptive
- An excitation sequence is generated for each subframe by adding one scaled element from the stochastic codebook and one scaled element from the adaptive codebook.

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FS 1016

- Scale factors and indices are selected to minimize the perceptual error between the input and synthesized speech
- Stochastic codebook contains 512 entries which are generated using a Gaussian random number generator
- Adaptive codebook consists of the excitation vectors from the previous frame
- FS 1016 provides excellent reproduction in both quiet and noisy environment at rates of 4.8 kbps and above

G.728

- Coding delay: the time between when a speech sample is encoded to when it is decoded if there was no transmission delay
- Coding delay consists of buffering delay (to store a segment), processing delay.
- Long delays are not acceptable in some applications.
- G728 is a CELP coder with a coder delay of 2 ms operating at 16 kbps (2 bits per sample).
- Reduce delay: reduce the segment size
- Segment size: 5 samples

G.728

- The algorithm obtains vocal tract filter parameters in a backward adaptive manner: vocal tract filter coefficients to be used to synthesize the current segment are obtained by analyzing the previous decoded segment
- G.728 algorithm does not use pitch filter instead it uses a 50^{th} order vocal tract filter
- Since the vocal tract filter is completely determined in a backward adaptive manner, we have all 10 bits available to encode the excitation sequence
- 10 bits => 1024 excitation sequences => too many to analyze in 0.625 ms

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G.728

- G.728 uses a product codebook: each excitation is represented by the product of a normalized sequence and a gain term
- 3 bits used to encode the gain using predictive encoding and 7 bits form the index to a codebook containing 127 sequences

G.728

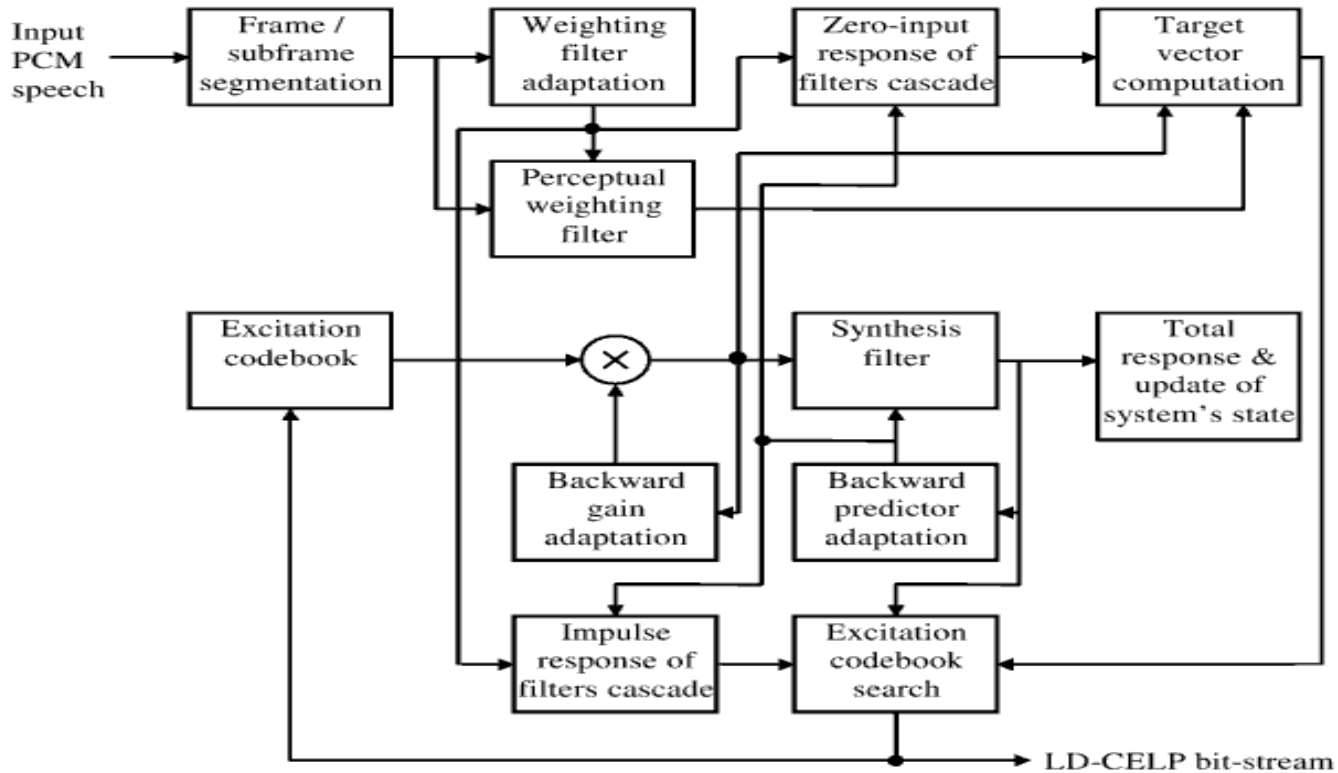

Figure 14.9 Block diagram of the G.728 encoder.

Sinusoidal Coders

- Main problem with the LPC coder: excitation signal
- CELP coder solves this problem using a codebook of excitation signals
- Sinusoidal coders solve this problem by using an excitation signal that is the sum of sine waves of arbitrary amplitudes, frequencies and phases.
- Since the vocal tract is a linear system, the synthesized speech (based on a sinusoidal excitation) will also be sinusoidal
- Sinusoidal coders directly estimate the parameters required to synthesize the speech at the decoder

Sinusoidal Coders

- Sinusoidal coders divide the input speech into frames and obtain the parameters of the speech separately for each frame
- Sinusoidal coder uses interpolation algorithms to smooth discontinuities at frame boundaries.

Sinusoidal coder

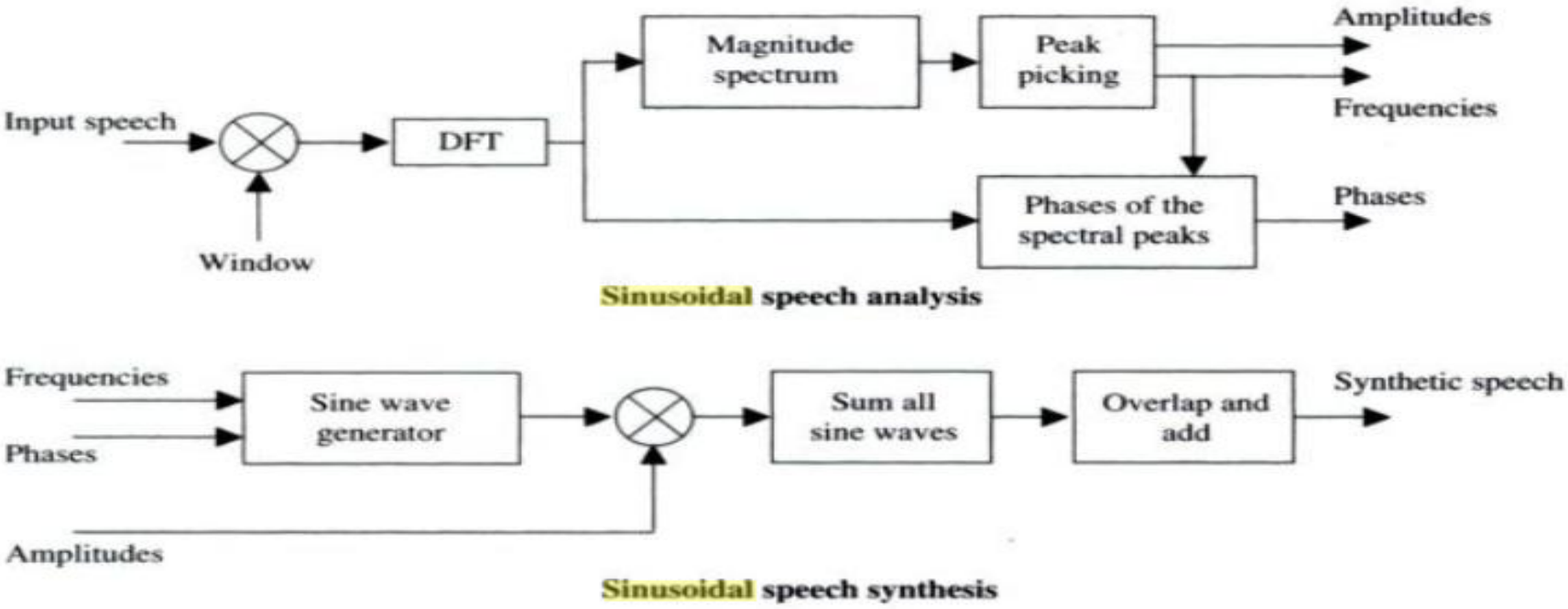

Figure 8.1 General sinusoidal analysis and synthesis

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Mixed excitation linear prediction (MELP)

- MELP is the new federal standard for speech coding at 2.4 kbps
- MELP uses LPC filter to model the vocal tract and a much more complex approach to the generation of the excitation signal
- Excitation signal is a multiband mixed excitation
- Mixed excitation contains both a filtered signal from a noise generator as well as a contribution depending on the input signal
- First step in constructing the excitation signal is pitch extraction

MELP

- Input is also subjected to a multiband voicing analysis using five filters with passband 0-500, 500-1000, 1000-2000, 2000-3000 and 3000-4000 Hz.
- The goal of the analysis is to obtain the voicing strength for each band used in the shaping filters

MELP

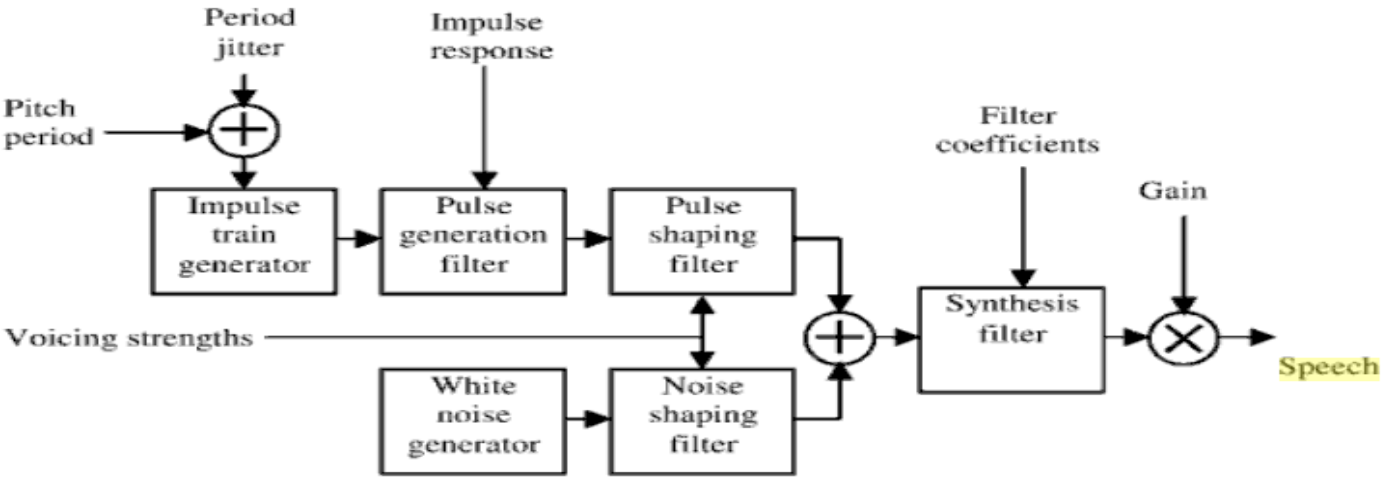

Figure 17.1 The MELP model of speech production.