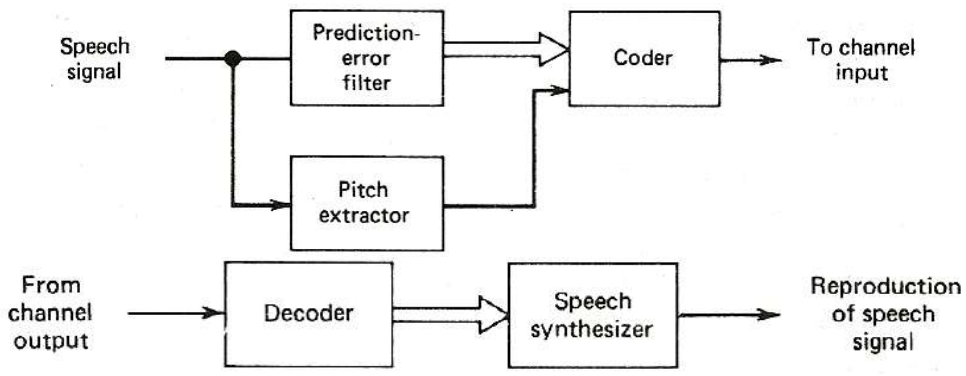

Figure 2.21 Block diagram of linear predictive vocoder. *(a)* Transmitter. *(b)* Receiver.

The transmitter, shown in Fig. *2.21a* first performs **analysis** on the input speech signal, block by block. Typically, each block is 10-30 ms long, for which the speech-production process may be treated as stationary. The parameters resulting from the analysis, namely, the prediction-error filter (analyzer) coefficients, a voiced/unvoiced parameter, and the pitch period, provides a complete description for the particular segment of the input speech signal. A digital representation of the parameters of this complete description constitutes the transmitted signal. The receiver, shown in Fig. *2.21b,* first performs decoding, followed by **synthesis** of the speech signal which utilizes the model of Fig. 2.19. The standard result of this analysis/synthesis is an artificial-sounding reproduction of the original speech signal. This poor reproduction quality of a linear predictive vocoder is tolerated for secure military communications where very low bit rates (4 *kb/s* or less) are required.

2 MARKS

1. Mention two merits of DPCM.

i) Bandwidth requirement of DPCMis less compared to PCM.
ii) Quantization error is reduced because of prediction filter.
iii) Number of bits used to represent one sample value are also reduced compared to PCM.

2. What is the main difference in DPCM and DM?

DM encodes the input sample by only one bit. It sends the information about $+\delta \ or \ -\delta$, i.e. step rise or fall. DPCM can have more than one bit for encoding the sample. It sends the information about difference between actual sample value and predicted sample value.

3. Mention the use of adaptive quantizer in adaptive digital waveform coding schemes.

Adaptive quantizer changes its step size according to variance of the input signal. Hence quantization error is significantly reduced due to adaptive quantization. ADPCM uses adaptive quantization. The bit rate of such schemes is. reduced due to adaptive quantization.

4. What do you understand from adaptive coding?

In adaptive coding, the quantization step size and prediction filter coefficients are changed as per properties of input signal. This reduces the quantization error and number of bits used to represent the sample value. Adaptive coding is used for speech coding at low bit rates.

5. What is meant by adaptive delta modulation ?

In adaptive delta modulation, the step size is adjusted as per the slope of 1the input signal. Step size is made high if slope of the input signal is high. This avoids slope overload distortion.

6. What is the advantageof delta modulation over pulse modulation schemes ?

Delta modulation encodes one bit per sample. Hence signaling rate is reduced in DM.

7. What should be the minimum bandwidth required to transmit a PCM channel ?

The minimum transmission bandwidth in PCM is given as,

$$B_T = vW$$

Here v is number of bits used to represent one pulse.
W is the maximum signal frequency.

8. What is the advantage of delta modulation over PCM ?

Delta modulation uses one bit to encode one sample. Hence bit rate of delta modulation is low compared to PCM.

9. What are the two types of quantization errors that occur in delta modulation ?

1) Slope over load error: The step size of quantization is not enough to follow the large changes in input signal. Hence there is difference between approximated signal and input signal. It is called slope overload error.

2) Granular noise: The step size is to large, hence approximated signal cannot follow the small variations in input Signal. For every small variation in input signal, there is large change in approximated signal. It is called hunting or granular noise.

10. State the advantages of adaptive delta modulation over delta modulation?

1) ADM eliminates slope overload error and granular noise.
2) ADM has wide dynamic range.
3) Bandwidth utilization is better.

11. PWM is referred to as --------- and -----------

PWM is basically pulse width modulation. Width of the pulse changes according to amplitude of the modulating signal. It is also referred as pulse duration modulation or PDM.

12. How the message can be recovered from PAM?

The message can be recovered from PAM by passing the PAM signal through reconstruction filter. The reconstruction filter integrates amplitudes of PAM pulses. Amplitude smoothing of the reconstructed signal is done to remove amplitude discontinuities due to pulses.

13. How is PDM wave converted into PPM systems?

The PDM signal is given as a clock signal to monostable multivibrator. The multivibrator triggers on falling edge. Hence a PPM pulse of fixed width is produced after falling edge of PDM pulse. PPM represents-the input signal amplitude in the form of width of the pulse. A PPM pulse is produced after this 'width' of PDM pulse. In other words, the position of the PPM pulse depends upon input signal amplitude.

14. Define pulse amplitude modulation?

The amplitude of the pulse is directly proportional to the amplitude of modulating signal at the sampling instant. The width of the pulse remains constant.

15. What is meant by distortionless transmission ?

For distortionless transmission, the transfer function of the system if given as,

$$H(\omega) = K e^{-j\omega t_0} \leftarrow \text{linear phase shift}$$

$$K = \text{Constant magnitude response}$$

Above transfer function imposes two requirements on the system :

1) The system response must have constant magnitude response.

2) The systems phase shift response must be linear with the frequency.

16. Differentiate: Noise and Fading.

Sr. No.	Noise	Fading
1.	It is an unwanted signal that tends to interfere with the required signal.	The signal is randomly attenuated due to random or semiperiodic variations in the channel. This is called fading.
2.	The noise can have particular range of frequencies depending upon its source.	Fading is frequency dependent and different frequency components are affected unequally.
3.	Effect of noise can be minimized by using appropriate filters.	Effects of fading can be reduced by employing Automatic Gain Control (AGC).

17. Which parameter is called figure of merit of a digital communication system and why?

The ratio *Eb IN* 0 or bit energy to noise power spectral density is called figure of merit for digital communication systems.

Reasons i) Energy is calculated for bit. Hence E_b/N_0 allows us to compare different systems at bit level.

ii) E_b/N_0 is also unit less as does *S/N*.

iii) Bit energy *(E_b)* can be calculated easily for digital data.

16 MARKS

1. Explain Delta modulation systems
2. Draw the block diagram of adaptive delta modulation systems and explain. (8)
3. Explain the two types of quantization noise in delta modulation systems (8)
4. Describe the principle and operational procedure for Delta Modulation. Draw appropriate waveforms. (8)
5. Explain the process of quantization, encoding and decoding in PCM?. In what way DPCM is better than PCM (16)
6. Using a block schematic of transmitter and receiver and discuss Delta Modulation scheme (16)
7. Describe various classification of encoding techniques for analog sources. (16)
8. (i) Explain the concept of spectral waveform encoding. (8)
   (ii) Compare the performance of various speech encoding techniques (8)
9. Discuss the Delta modulation transmitter and receiver and derive the SNR expression for it.