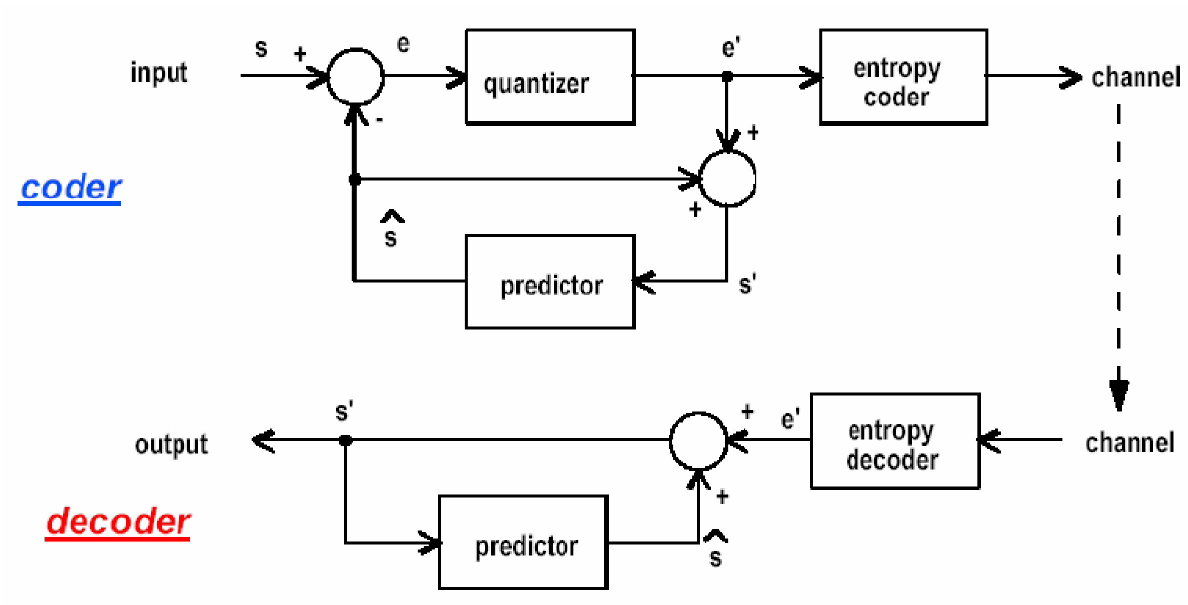
2.3 Differential Pulse Code Modulation (DPCM) Block diagrams



Prediction error	Reconstruction	Reconstruction error = quantization error
$e = s - \hat{s}$	$s' = e' + \hat{s}$	$s' - s = e' - e = q$

Figure 2.6. Coder and decoder block diagram for DPCM

2.4 Adaptive differential pulse code modulation (ADPCM)

Pulse code modulation (PCM) samples an input signal using a fixed quantizer to produce a digital representation. This technique, although simple to implement, does not take advantage of any of the redundancies in speech signals. The value of the current input sample does not have an effect on the coding of future samples. Adaptive differential PCM (ADPCM), on the other hand, uses an adaptive predictor, one that adjusts according to the value of each input sample, and thereby reduces the number of bits required to represent the data sample from eight (nonadaptive PCM) to four.

ADPCM does not transmit the value of the speech sample, but rather the difference between a predicted value and the actual sample value. Typically, an ADPCM transcoder is inserted into a PCM system to increase its voice channel capacity. Therefore, the ADPCM encoder accepts PCM values as input, and the ADPCM decoder outputs PCM values.

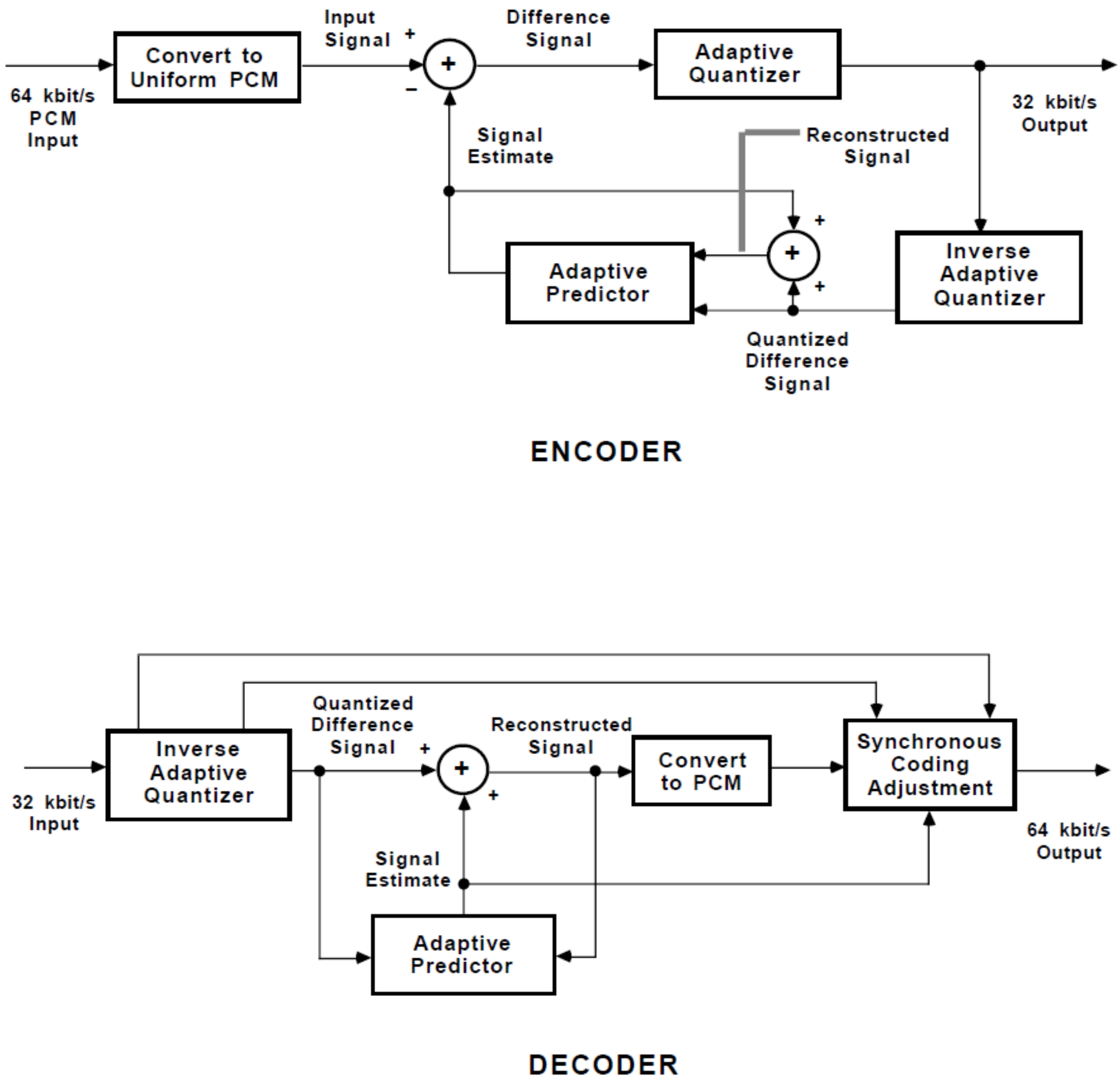


Figure 2.7. Coder and decoder block diagram for ADPCM

Both the encoder and decoder update their internal variables based on only the generated ADPCM value. This ensures that the encoder and decoder operate in synchronization without the need to send any additional or sideband data. A full decoder is embedded within the encoder to ensure that all variables are updated based on the same data.
In the receiving decoder as well as the decoder embedded in the encoder, the transmitted ADPCM value is used to update the inverse adaptive quantizer, which produces a dequantized version of the difference signal. This dequantized value is added to the value generated by the adaptive predictor to produce the reconstructed speech sample. This value is the output of the decoder.

2.4 Coding of speech signal at low bit rates (Vocoders, LPC)

Speech coding is the process of reducing the data rate of digital voice to manageable levels. Parametric speech coder or (Vocoders) utilise a priori information about the mechanism by which speech is produced in order to achieve extremely efficient compression of speech signal (as low a 1kbps).
The channel Vocoders employs a bank of bandpass filters, typically 16-20 filters are used, and each has a bandwidth between 100 and 300 Hz.

The output of each filter is rectified and lowpass filtered.

- The bandwidth of the lowpass filter is selected to match the time variations in the characteristics of the vocal tract.
- For measurement of the spectral magnitudes, a voicing detector and a pitch estimator are included in the speech analysis.
- 16-20 linear-phase FIR filters
- Covering 0-4 kHz
- Each having a bandwidth between 100-300 Hz
- 20-ms frames, or 50 Hz changing of spectral magnitude.
- LPF bandwidth: 20-25 Hz
- Sampling rate of the output of the filters: 50 Hz

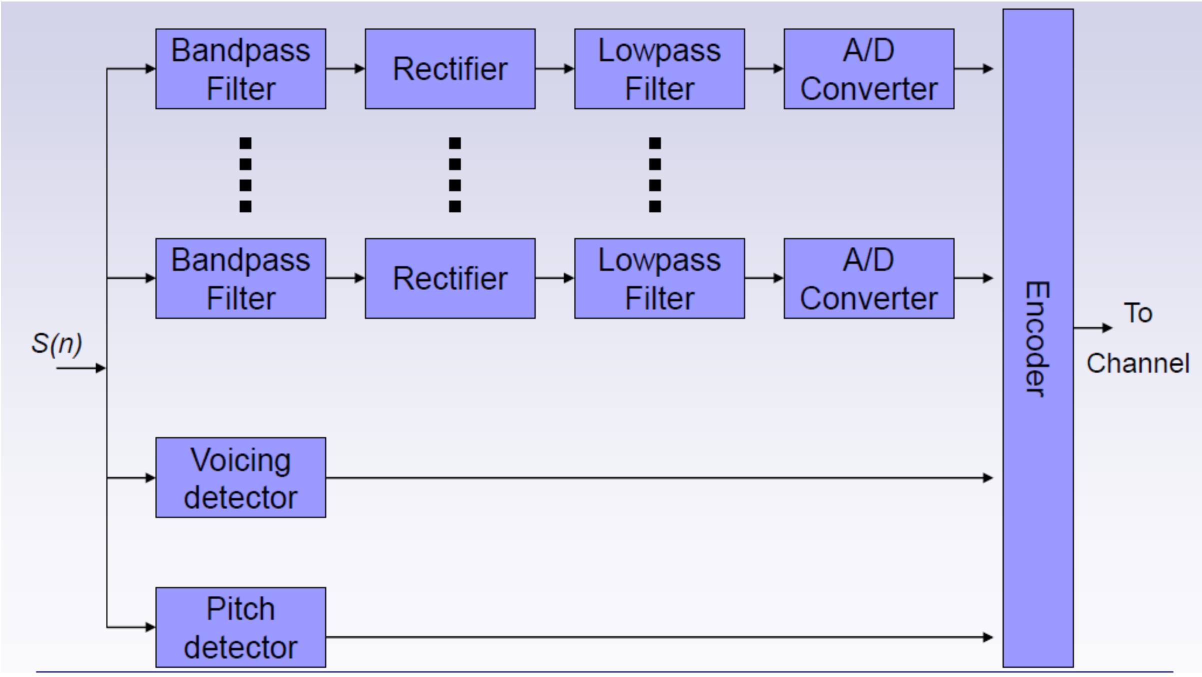


Figure 2.8. The channel Vocoders (Transmitter).

Bit rate:

- 1 bit for voicing detector.
- 6 bits for pitch period.
- For 16 channels, each coded with 3-4 bits, updated 50 times per second.
- Then the total bit rate is 2400-3200 bps.
- Further reductions to 1200 bps can be achieved by exploiting frequency correlations of the spectrum magnitude.

- At the receiver, the signal samples are passed through D/A converters.
- The outputs of the D/As are multiplied by the voiced or unvoiced signal sources.
- The resulting signal are passed through bandpass filters.
- The outputs of the bandpass filters are summed to form the synthesized speech signal.

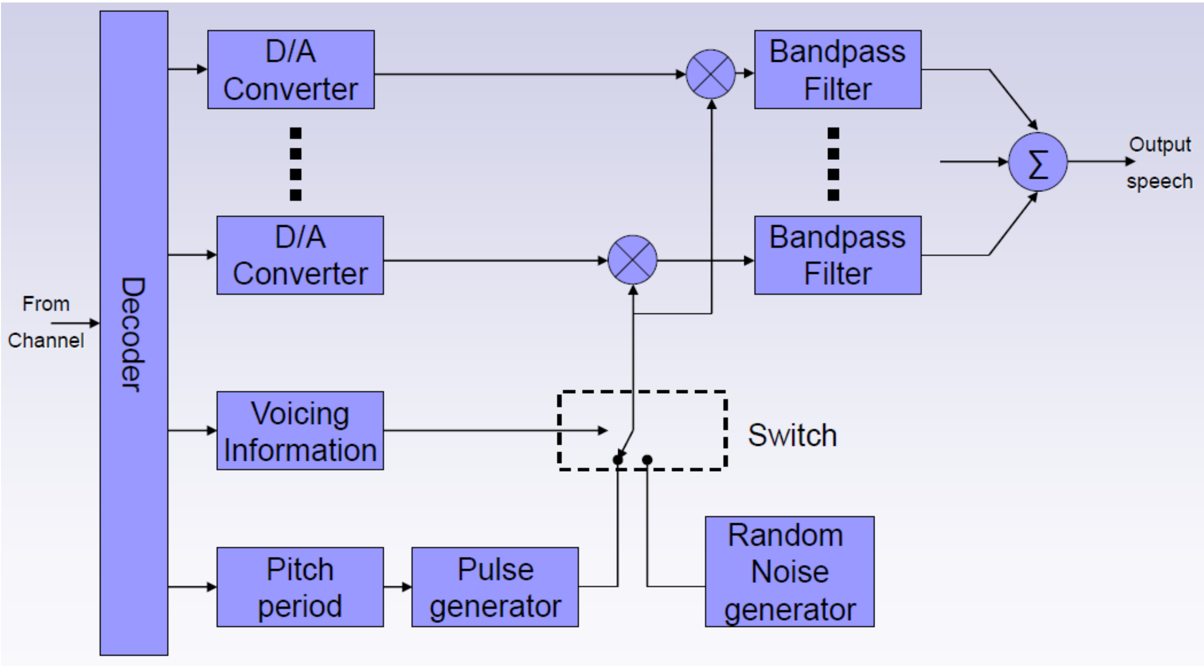


Figure 2.9. The channel Vocoders (Receiver).