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Service

Digital Telephony

Second Edition

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Frequency domain voice coders provide improved coding efficiencies by encoding the most important components of the spectrum on a dynamic basis. As the sounds change different portions (formants) of the frequency band are encoded. The period between formant updates is typically 10 to 20 msec. Instead of using periodic spectrum measurements, some higher quality vocoders continuously track gradual changes in the spectral density at a higher rate. Frequency domain vocoders often provide lower bit rates than the time domain coders, but typically produce more unnatural sounding speech.

3.4 DIFFERENTIAL PULSE CODE MODULATION

Differential pulse code modulation (DPCM) is designed specifically to take advantage of the sample-to-sample redundancies in a typical speech waveform. Since the range of sample *differences* is less than the range of individual *samples*, fewer bits are needed to encode difference samples. The sampling rate is often the same as for a comparable PCM system. Thus the bandlimiting filter in the encoder and the smoothing filter in the decoder are basically identical to those used in conventional PCM systems.

A conceptual means of generating the difference samples for a DPCM coder is to store the previous input sample directly in a sample-and-hold circuit and use an analog subtractor to measure the change. The change in the signal is then quantized and encoded for transmission. The DPCM structure shown in Figure 3.27 is more complicated, however, because the previous input value is reconstructed by a feedback loop that integrates the encoded sample differences. In essence, the feedback signal is an estimate of the input signal as obtained by integrating the encoded sample differences. Thus the feedback signal is obtained in the same manner used to reconstruct the waveform in the decoder.

The advantage of the feedback implementation is that quantization errors do not accumulate indefinitely. If the feedback signal drifts from the input signal, as a result of an accumulation of quantization errors, the next encoding of the difference signal automatically compensates for the drift. In a system without

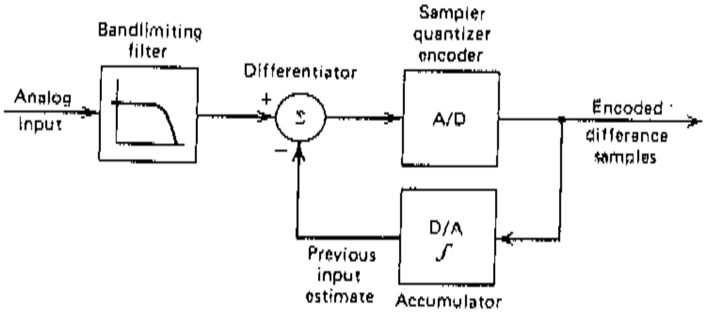

Figure 3.27. Functional block diagram of differential PCM.