1.7 PCM Quality Versus Required Rate

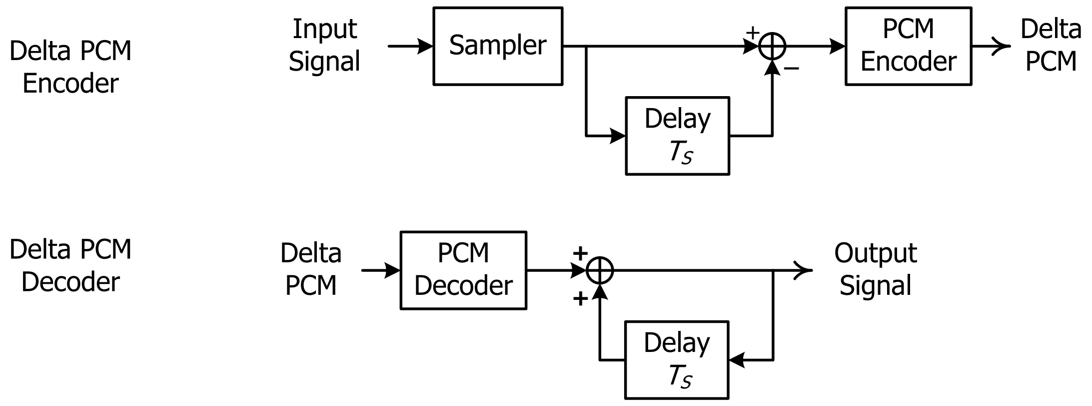
In addition to the channel effects, PCM performance depends primarily on the quantization noise. To make the reconstructed signal like the original baseband signal we must reduce the quantizing noise by increasing L. Now, we learned that increasing the number of quantization levels requires more bits per sample to be transmitted. Large l is not perfect for crowded channels and due to some channel's limitations. So, beside the non-uniform quantization, several techniques are used to reduce the quantization noise at the same number of the L.

1.8 Bandwidth Reduction Techniques

The channel bandwidth is limited, and it is a valuable resource. A frequent objective of the communications engineer is to transmit the maximum information rate via the minimum possible bandwidth. This is especially true for radio communications in which radio spectrum is a scarce, and therefore valuable, resource. The following systems are used to maintain the same coding fidelity using fewer bits per sample.

1.8.1 Delta PCM

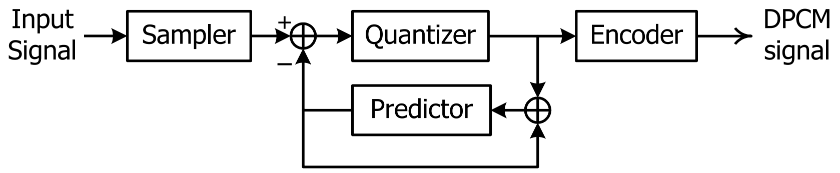
Because the samples of most of the baseband signals are highly correlated, it is possible to transmit the information about the changes between samples instead of sending the sample values themselves. A simple way for such systems is the Delta PCM. This method transmits the difference between adjacent samples through code words. This difference is significantly less than the actual sample values, hence it is coded using fewer binary symbols per word than the conventional PCM. However, Delta PCM systems cannot accommodate rapidly varying transient signals.



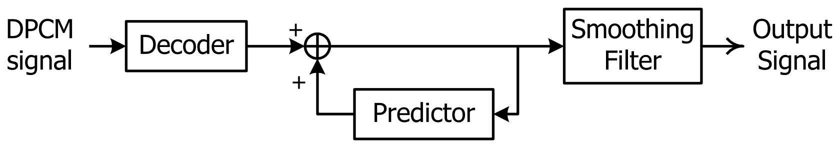
1.8.2 Deferential PCM

Since neighbor samples within many information signals are highly correlated, Deferential PCM (DPCM) uses an algorithm to predict future values. Such algorithms monitor the trend of the baseband samples and use some models to predict the value of the incoming samples. Then DPCM waits until the actual value becomes available for examination and transmits the correction to the already predicted value. The correction signal represents the information signal's unpredictable part. By this means, DPCM reduces the redundancy in signal and allows the information to be transmitted using fewer symbols, less spectrum, and shorter time.

DPCM Encoder



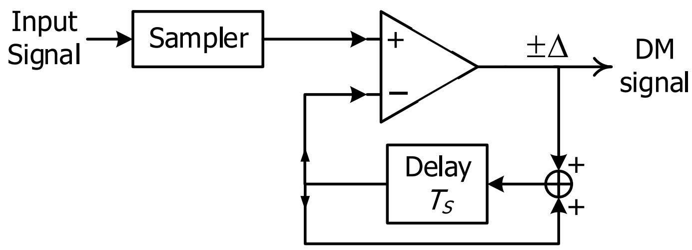
DPCM Decoder



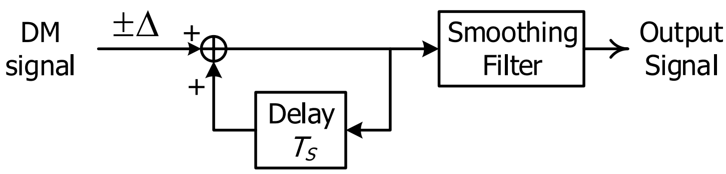
1.9 DELTA MODULATION (DM)

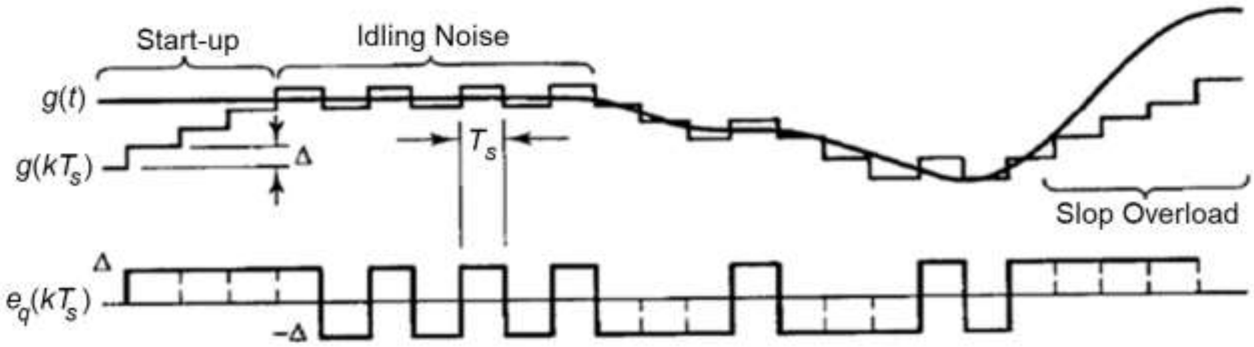
If the quantizer of the DPCM system is restricted to one bit (i.e. the two levels only: $\pm\Delta$) and the predictor to one sample delay, then the resulting scheme is called DM. The information signal is represented by a stepped waveform. The resolution of this waveform depends on Δ & T_s values.

DM Encoder



DM Decoder

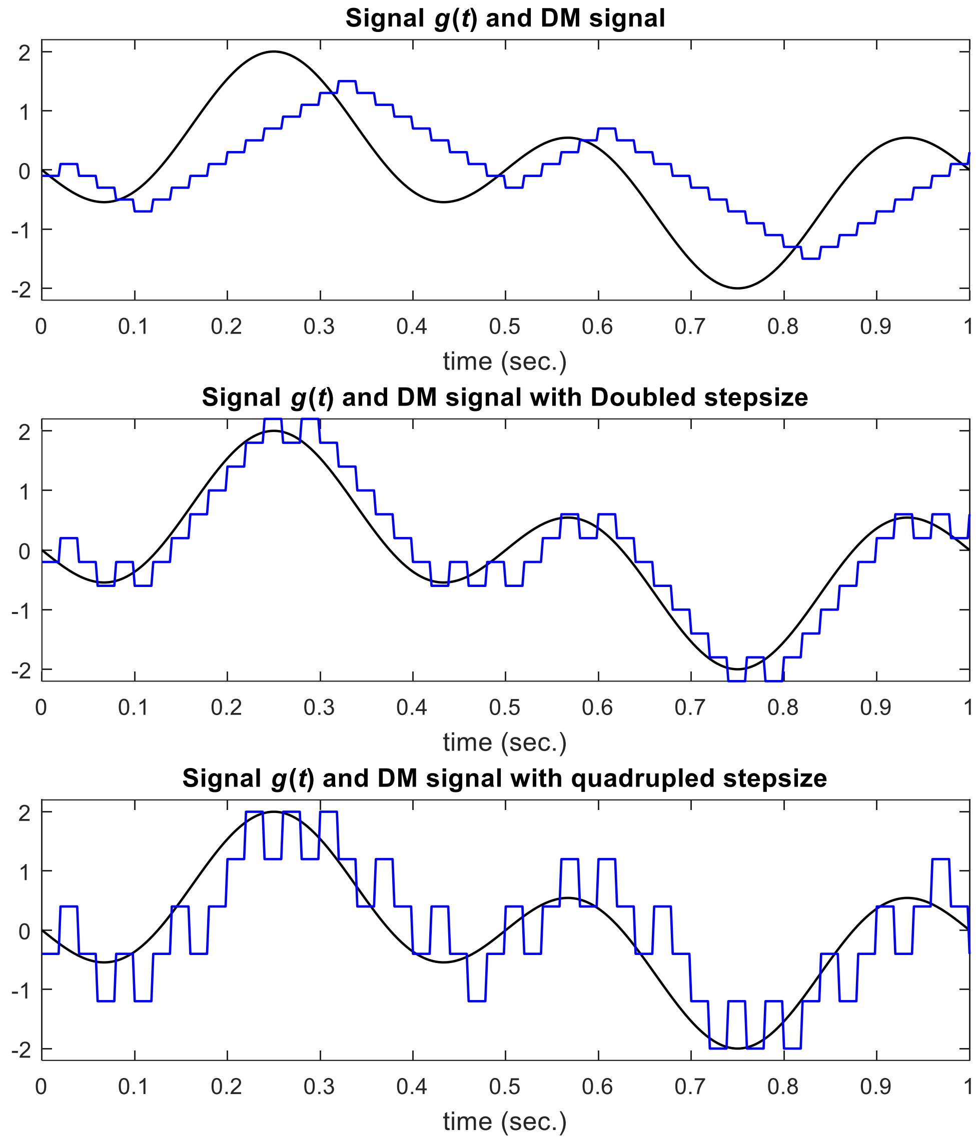




We observe from the figure above that:

- The system requires notable time for the start-up to catch the required level of the input.
- If the input remains constant, the reconstructed DM waveform exhibits ripple known as *idling noise*, which almost filtered out at the receiver.
- The rate-of-rise overload problem occurs when the input changes too rapidly for the stepped waveform to follow.

To resolve these problems, we are clearly in a dilemma about how to choose the best values for Δ and T_s. A small step size (Δ) is desirable for accuracy, but the clock rate ($f_s = 1/T_s$) should be fast to avoid *slop-overload*, which is not recommended. Inversely, as f_s get smaller, many small details of the information signal will be lost. This dilemma is depicted in the figures below:



However, we can calculate the optimum values for these parameters that overcome the slop-overload problem. An estimate of the rate-of-rise condition for DM may be obtained quit easily for sinusoidal modulation. Let the input be $g(t) = b\cos(\omega_m t)$, so that:

$$\max\left[\frac{dg(t)}{dt}\right] = 2\pi b f_m = \text{ maximum slop of this signal.}$$

Since the maximum rate-of-rise $= \frac{\Delta}{T_s} = \Delta f_s$, then

$$\Delta f_s \geq 2\pi b f_m \quad \Leftrightarrow f_s \geq \frac{2\pi b f_m}{\Delta} \quad \Leftrightarrow \quad b \leq \frac{\Delta f_s}{2\pi f_m}$$

The above f_s condition may be applied to band-limited signals by letting f_m be the highest frequency component, and $b = \max|g(t)|$.

Since the quantization noise: $\overline{e^2} = \int_{-\Delta}^{\Delta} \frac{e^2}{2\Delta} de = \frac{\Delta^2}{3}$

And by filtering this noise to a bandwidth B, we get: $N_q = \overline{n_q^2} = \frac{\Delta^2 B}{3f_s}$

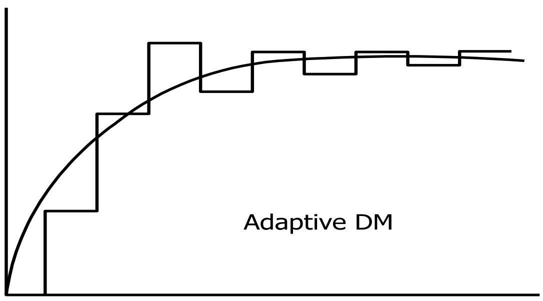
The mean-square value of the information signal is:

$$S = \overline{g^2(t)} = \left(\frac{V_p}{\sqrt{2}}\right)^2 = \frac{b^2}{2} = \frac{1}{8}\left(\frac{\Delta f_s}{\pi f_m}\right)^2$$

$$\therefore \mathrm{SN_qR} = \frac{3f_s^3}{8\pi^2 f_m^2 B}$$

Adaptive Delta Modulation

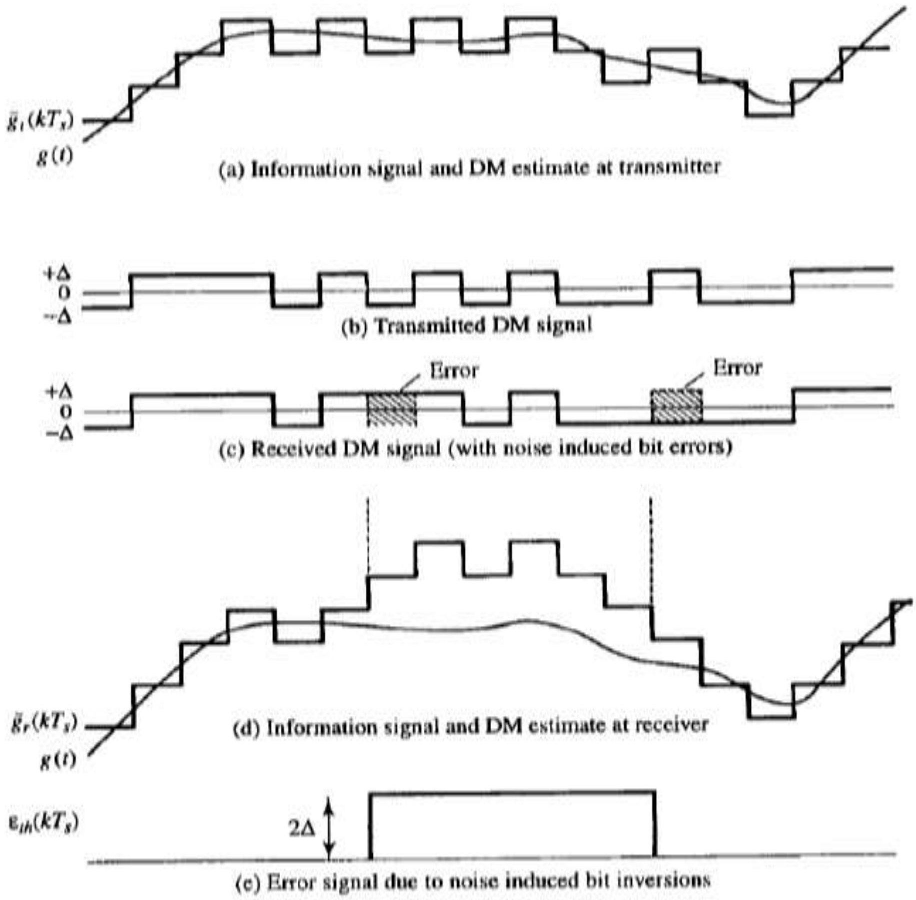
We have seen that: a large step size causes unacceptable quantization noise, and a small step size results in sample-overload distortion. This means that a good choice for Δ is a "medium" value, but in some cases, the performance of the best "medium" values is not satisfactory. An approach that works well in these cases is to change the step size according to changes in the input: if the input tends to change rapidly, the step size is chosen to be large (and vice versa). So, the output can follow the input quickly without distortion.



Errors in DM

If the SNR is not sufficiently high, then the DM receiver will occasionally interpret a received symbol in error (i.e. $+\Delta$ instead of $-\Delta$ or the converse), and this is equivalent to the addition of an error of 2Δ to the accumulated signal at the DM receiver, as shown below. This situation continues until another error occurs which either cancels the first error or double it!

DM is primarily used for telemetry systems and speech transmission in telephone. It has been found that: PCM is preferable for high quality speech transmission, whereas DM is easier to implement and yields transmission of acceptable quality.



(a) Information signal and DM estimate at transmitter

(b) Transmitted DM signal

(c) Received DM signal (with noise induced bit errors)

(d) Information signal and DM estimate at receiver

(e) Error signal due to noise induced bit inversions