Analog Data, Digital Signals

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Analog data, digital signals

- Convert analog data into digital data : digitization
- The device used for converting analog data into digital form for transmission, and subsequently recovering the original analog data from the digital, is known as a codec (coder-decoder).

Techniques used in codecs

- Pulse Code Modulation (PCM)
- Delta Modulation (DM)

Pulse Code Modulation (PCM)

Sampling Theorem

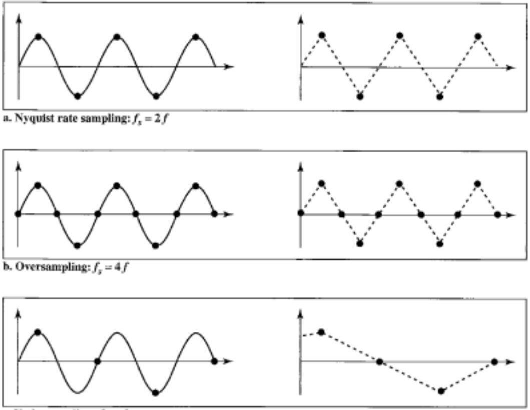
If a signal f(t) is sampled at regular intervals of time and at a rate higher than twice the highest signal frequency (Nyquist sampling rate), then the samples contain all the information of the original signal. The function f(t) may be reconstructed from these samples by the use of a lowpass filter.

A low-pass filter (LPF) is a filter that passes signals with a frequency lower than a selected cutoff frequency and attenuates signals with frequencies higher than the cutoff frequency.

High-pass and band-pass filters

- A high-pass filter (HPF) is a filter that passes signals with a frequency higher than a selected cutoff frequency and attenuates signals with frequencies lower than the cutoff frequency.
- A band-pass filter (BPF) is a filter that passes frequencies within a certain range and attentuates frequencies outside this range

Recovery of a sine wave for different sampling rates



a. Nyquist rate sampling: $f_s = 2f$

b. Oversampling: $f_s = 4f$

c. Undersampling: $f_s = f$

Question

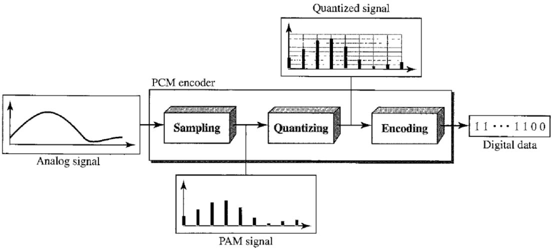
If voice data is limited to frequencies below 4kHz, ______ samples would be sufficient to characterize the voice signal.

(A) 2000

(B) 8000

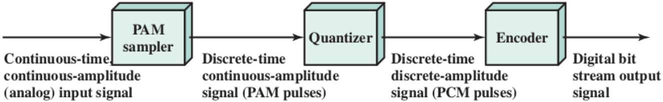
(C) 4000

Components of a PCM encoder



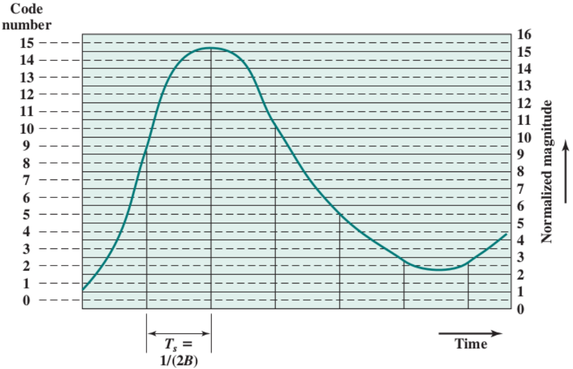
(A) Sample the analog signal (to get Pulse Amplitude Modulation samples)
(B) Quantize the sampled signal
(C) Encode the quantized values as streams of bits

Components of a PCM encoder



PCM Example

The highest frequency of the signal is B



PAM value	1.1	9.2	15.2	10.8	5.6	2.8	2.7
Quantized code number	1	9	15	10	5	2	2
PCM code	0001	1001	1111	1010	0101	0010	0010

By quantizing the PAM pulse, the original signal is now only approximated and cannot be recovered exactly. This effect is known as quantizing error

Steps in quantization

(A) Assume that the original analog signal has instantaneous amplitudes between V_{min} and V_{max}.

(B) Divide the range into L zones, each of height Δ

$$\Delta = \frac{V_{max} - V_{min}}{L}$$

(C) Assign quantized values of 0 to L-1 to the midpoint of each zone

(D) Approximate the value of the sample amplitude to the quantized values

Quantization and channel capacity

- Quantization error changes SNR of a signal, reducing channel capacity

$$SNR_{dB} = 6.02n + 1.76dB$$

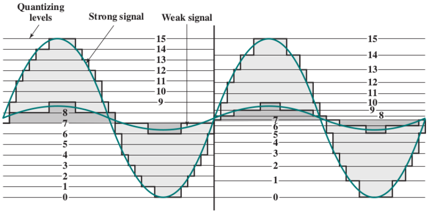
  where n is the number of bits

- Each additional bit for quantization increases SNR by about 6dB

Question

Assume that voice signals, bandlimited with a bandwidth of 4kHz is sampled at the Nyquist rate. Assuming an 8-bit sample, how many quantizing levels are present? What is the expected data rate?

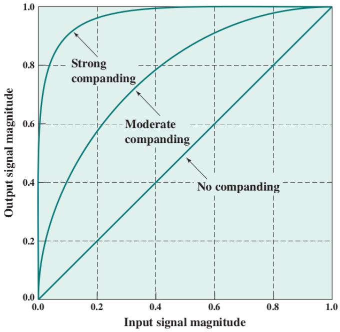
Nonlinear encoding



(a) Without nonlinear encoding (b) With nonlinear encoding

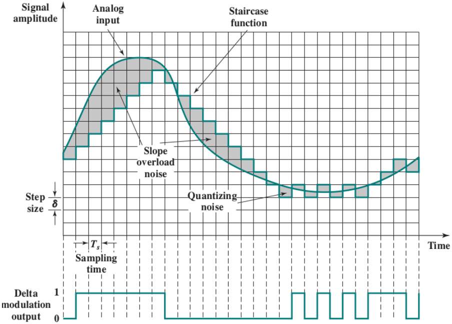
- The mean absolute error for each sample is the same, regardless of signal level
- Consequently, lower amplitude values are relatively more distorted
- Solved by non-linear encoding
- use a greater number of quantizing steps for signals of low amplitude
- use a smaller number of quantizing steps for signals of large amplitude
- Nonlinear encoding can significantly improve the PCM SNR. For voice signals, improvements of 24 to 30 dB have been achieved.

Companding



- Companding (compressing-expanding) is a process that compresses the intensity range of a signal by imparting more gain to weak signals than to strong signals on input
- At the output, the reverse process is performed
- With a fixed number of quantizing levels, more levels are available for lower-level signals

Delta Modulation



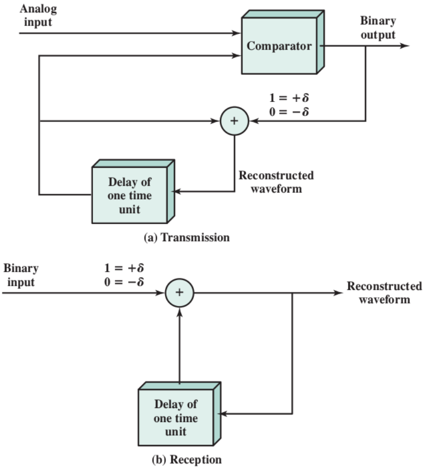
At each sampling time, the analog input is compared to the most recent value of an approximating function (called a staircase function)

The figure above shows the staircase function that is *generated* due to the analog signal

Delta Modulation

- The approximating function is a staircase function — moves up or down by one quantization level (δ) at each sampling interval (T_s)
- Output signal: 1 if the value of the sample $>$ the most recent value of approximating function
- Output signal: 0 if the value of the sample $<$ the most recent value of approximating function

Delta PCM: Transmission and Reception



Delta PCM: Paramaters

- Size of step $= \delta$ and number of samples
- δ must be chosen to balance between two types of error or noise:
  - Quantizing noise: if the analog waveform is changing only slowly
  - - increases as δ increases
  - Slope overload noise: when the analog waveform is changing more rapidly than the staircase can follow
  - - noise increases as δ is decreased
- Increased sampling rate improves the accuracy of DM
- - but increases the data rate of the output signal
- DM is simpler to implement than PCM
- But PCM exhibits better SNR at the same data rate

Performance

- A voice signal (bandlimited at 4kHz) is sampled at the Nyquist rate. If 128 quantization levels are used, a data rate of 7*8000 = 56 kbps is required for PCM encoded digital data
- Using the Nyquist criterion, a digital signal of 56 kbps requires using a bandwidth of the order of 28 kHz (assuming M=2)
- Thus if voice is converted to data signals and sent, more bandwidth (than 4 kHz) is required
- If voice is sent as a baseband signal (on a telephone line) only 4kHz is sufficient
- Yet, digital transmission is preferred