Analogue Data / Digital Signals

- Two steps involved:
  - Convert *analogue data* into *digital data*
  - Encode the *digital data* onto a *digital signal*
- A CODEC is used to convert analogue data to digital data and vice versa
- Two methods are employed:
  - Pulse Code Modulation
  - Delta Modulation

Analogue Data / Digital Signals

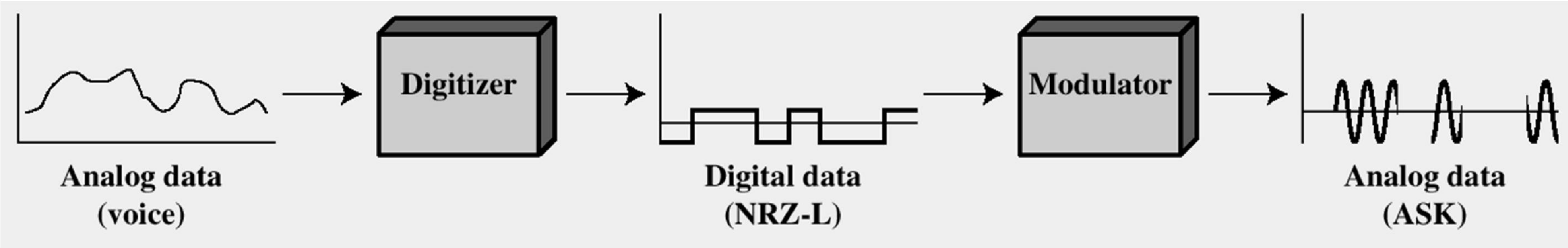
- The Sampling Theorem (from Nyquist):

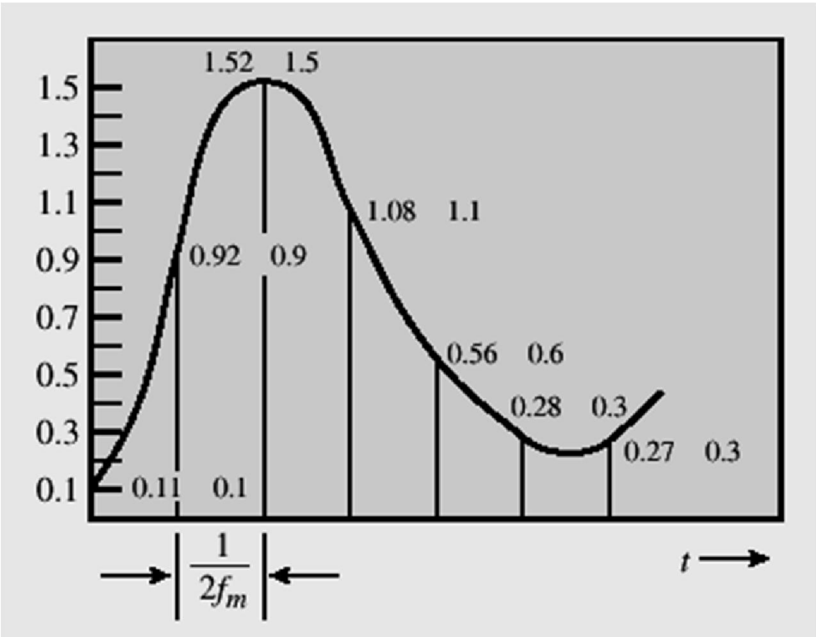
"If a signal f(t) is sampled at regular intervals of time and at a rate higher than twice the highest signal frequency, then the samples will contain all the information necessary for the reconstruction of the original signal."

Pulse Code Modulation

- The original analogue signal is *sampled* at regular intervals to produce *PAMS*
- The PAMS are *quantized* i.e. assigned a binary value
- *Quantization* gives rise to *Quantization error* or *Quantizing noise*
- Hence the original analogue signal can <u>never</u> be truely reproduced

Pulse Code Modulation



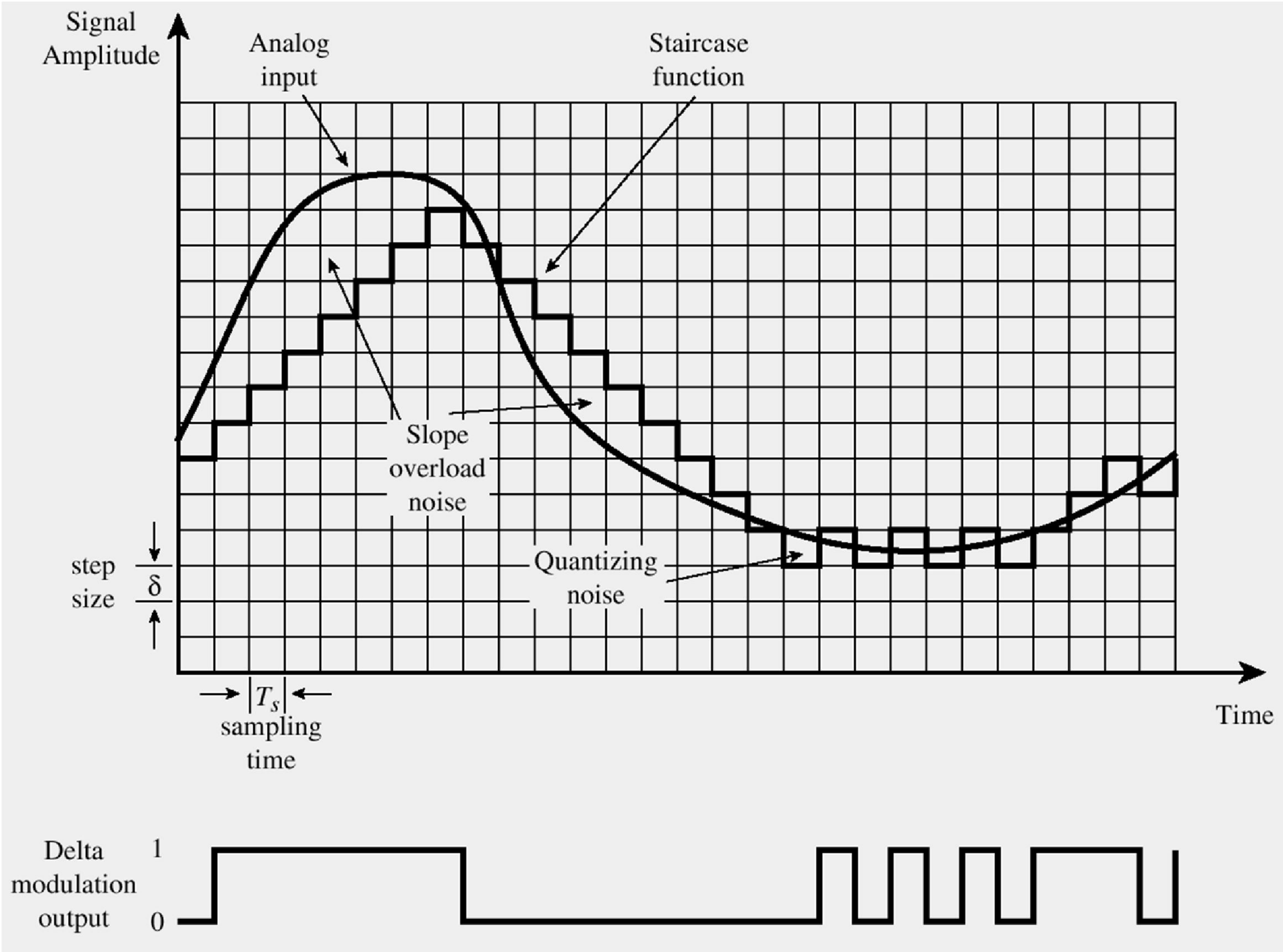


Digit	Binary Equivalent	PCM waveform
0	0000	
1	0001	
2	0010	
3	0011	
4	0100	
5	0101	
6	0110	
7	0111	
8	1000	
9	1001	
10	1010	
11	1011	
12	1100	
13	1101	
14	1110	
15	1111	

Delta Modulation

- The analogue signal is approximated by a *staircase function*
  - at each sampling interval a positive or negative *quantization step* is added to the output
- The result is a *single* binary digit for each sample indicating a positive or negative slope in the original analogue signal

Delta Modulation



Delta Modulation

- Two Significant Parameters:
  - Quantization step *delta* (δ)
    - large delta reduces *Slope Overload Noise*
    - small delta reduces *Quantization Noise*
  - Sampling rate
    - Large rate reduces noise but increases data rate of output signal
- DM compared to PCM:
  - DM is easier to implement
  - PCM produces better SNR characteristics

Voice data: Digital V's Analogue

- Digital (PCM):
  - Good voice reproduction with 256 quantisation levels i.e. 8-bit coding
  - Required sampling rate of 8000 per second for 4000hz voice i.e. 64Kbps voice channels
  - 64Kbps voice channel requires approx. 32KHz. of BW
- Analogue (PoTS):
  - Voice channel requires approx. 4KHz. of BW
- However, digital is still preferable to analogue for the following reasons (see next slide)

Analogue Data / Digital Signals

- Popularity increasing
  - Repeaters can be used so no additive noise
  - Time-division multiplexing (TDM) can be used eliminating intermodulation noise which occurs with FDM
  - Digital signalling allows use of digital switching techniques

Analogue Data / Analogue Signals

- Analogue data can be used to modulate an analogue signal
- This is often used to:
  - Obtain a more appropriate frequency for a particular transmission
  - Allow a number of analogue signals to share a transmission medium (Frequency Division Multiplexing) – will examine later
- Three basic techniques available

Analogue Modulation Techniques

- Amplitude Modulation
  - Carrier amplitude varies in proportion to the amplitude of the analogue data
- Frequency Modulation
  - Frequency deviation is proportional to the analogue data
- Phase Modulation
  - Phase is proportional to the analogue data
- These techniques were examined previously (see Digital Data/Analogue Signal slides)