Data and Signals

COMP312

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Lecture Outline

- Analogue Data on Analogue Signals
- Digital Data on Analogue Signals
- Analogue Data on Digital Signals
- Digital Data on Digital Signals

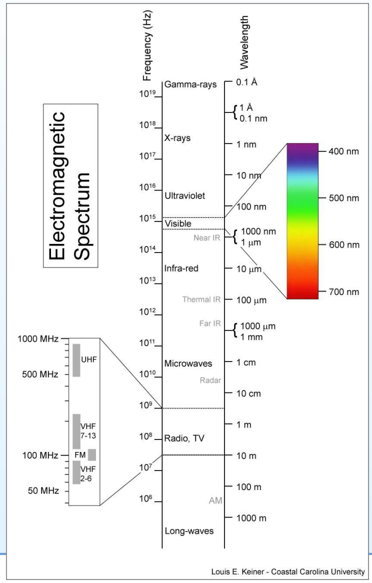
OSI Protocol Model

Working Here

Analogue Data on Analogue Signals

- Analogue Signals -review
- Fourier Analysis
- Modulation
  - Amplitude Modulation
  - Frequency Modulation

Spectrum - Review



Signal Components

Fourier Series

Periodic Signals

$$x(t) = \frac{A_0}{2} + \sum_{n=1}^{\infty} [A_n \cos(2\pi n f_0 t) + B_n \sin(2\pi n f_0 t)]$$

Where

$$\begin{aligned} A_0 &= \frac{2}{T} \int_0^T x(t)dt \\ A_n &= \frac{2}{T} \int_0^T x(t)\cos(2\pi n f_0 t)dt \\ B_n &= \frac{2}{T} \int_0^T x(t)\sin(2\pi n f_0 t)dt \end{aligned}$$

Fourier Transform

Aperiodic signals

$$X(f) = \int_{-\infty}^{\infty} x(t)e^{-j2\pi ft}df$$

Inverse:

$$x(t) = \int_{-\infty}^{\infty} X(f)e^{j2\pi ft}df$$

Discrete Fourier Transform

$$X(k) = \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} x(n) e^{\frac{-j2\pi kn}{N}}$$

Inverse:

$$x(t) = \frac{1}{\sqrt{N}} \sum_{k=0}^{N-1} X(k) e^{\frac{j2\pi nk}{N}}$$

Fast Fourier Transform

- Discrete Fourier Transform requires $O(N^2)$ complex multiplications
- Fast Fourier Transforms are a class of algorithms that require significantly less computation (often $O(NlogN)$).
- Choice of algorithm typically depends on N.
- Speedup for typical N is order of 100 times.

Modulation

Analog signals may be transmitted as is. This is *baseband* transmission and is used for ordinary telephones.
Often it is useful to transmit signals in a different frequency band. This may be because:

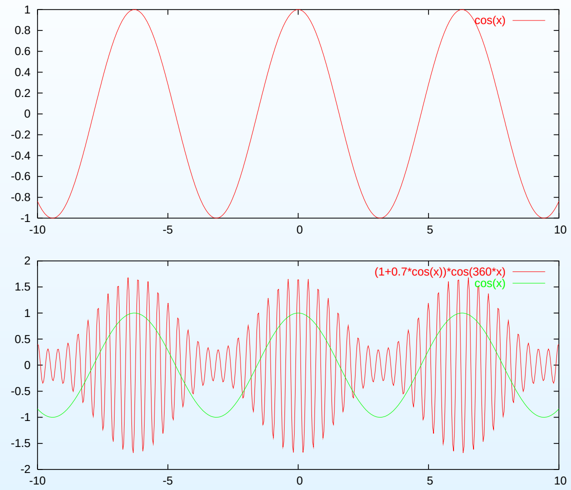
- The medium does not support baseband (e.g. radio).
- There are frequency restrictions.
- Multiple signals are to be transmitted at different frequencies on the same medium

Amplitude Modulation

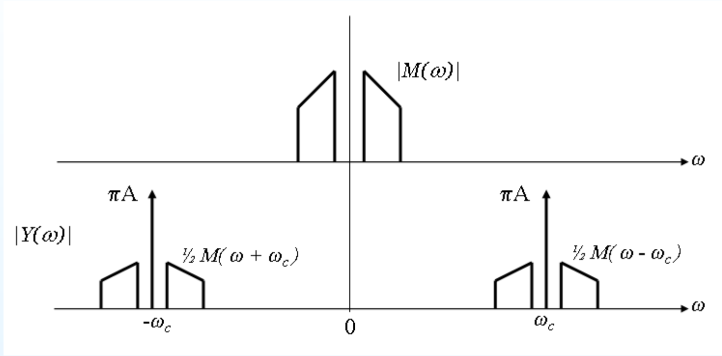
Amplitude Modulation is the carrier wave strength (amplitude) is proportional to (i.e. is multiplied by) the analoge data signal. i.e.

$$s(t) = [1 + n_a x(t)] \cos 2\pi f_c t$$

Amplitude Modulation



Amplitude Modulation



Angle Modulation

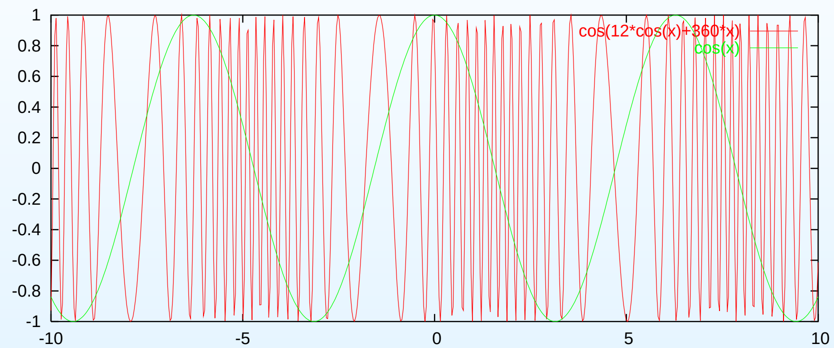
Phase Modulation is the carrier wave phase is offset by (i.e. is added to) the analoge data signal. i.e.

$$s(t) = \cos\left[2\pi f_c t + n_p m(t)\right]$$

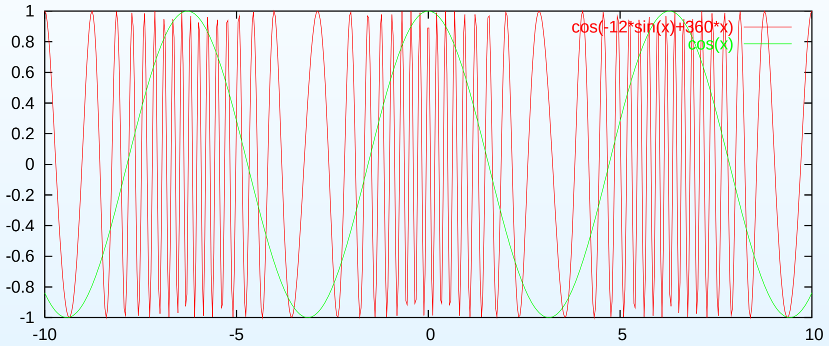
Frequency Modulation is the carrier wave instantaneous frequency is offset by (i.e. is added to) the analoge data signal. i.e.

$$s(t) = \cos\left[\left(2\pi f_c + n_f m(t)\right) t\right]$$

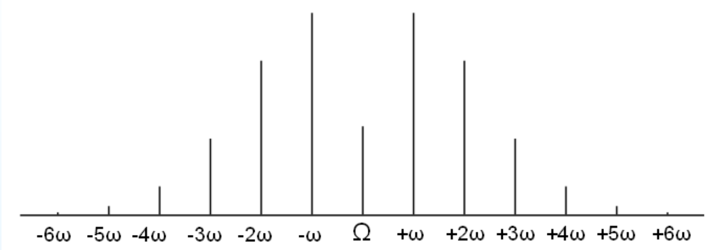
Phase Modulation



Frequency Modulation



Frequency Modulation



Use of Analogue Modulation

- Broadcast radio and television
- Low cost radio links
- Original analogue telephone trunks

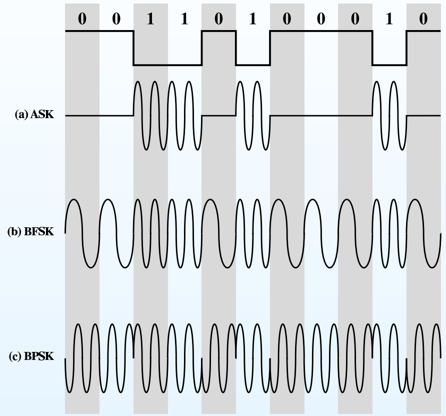
Digital Data on Analogue Signals

- Digital Data
- Digital Modulation Schemes
  - Amplitude Shift Keying
  - Frequency Shift Keying
  - Phase Shift Keying
  - Quadrature Amplitude Modulation
- Chanel Capacity Limits

Digital Modulation

- Digital Modulation is required when the medium will not carry (baseband) digital signals. e.g. telephone lines, radio
- Often analogue signals are encoded to digital then modulated onto an analogue carrier. The key advantage of this is the regeneration of digital signals
- The same basic carrier variables used in analogue modulation (amplitude, frequency, phase) can be used in digital modulation.

Digital Modulation

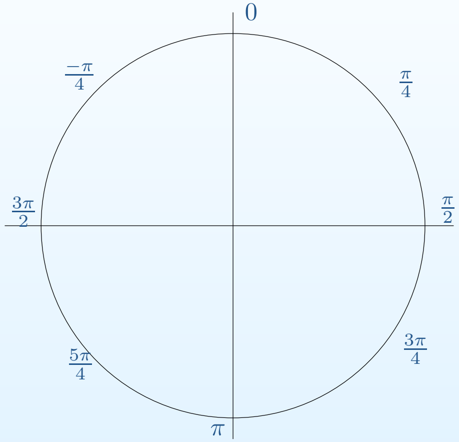


Nyquist

- Nyquist found that that the maximum rate of symbols carried on a channel is twice the bandwidth B.
- For binary symbols the maximum data rate is 2B.
- To increase the bandwidth we need to increase the number of bits per symbol. This is called multi-level signalling

Four Level PSK

Modulation using four different phases is called Quadrature Phase Shift Keying *QPSK*

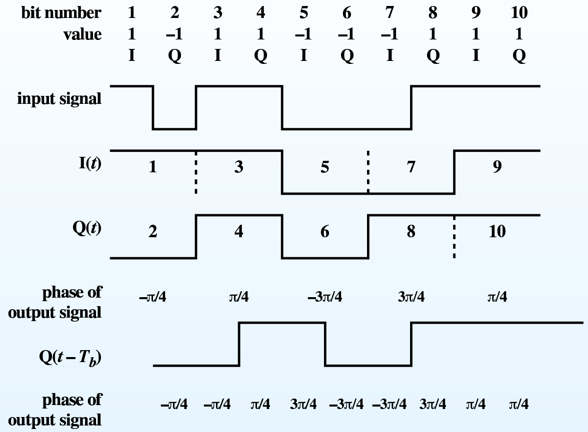


QPSK

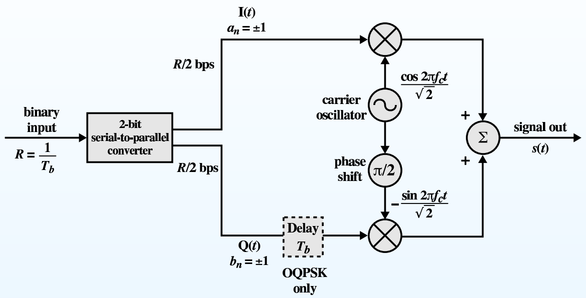
$$s(t) = \begin{cases} A\cos\left(2\pi f_c t + \frac{\pi}{4}\right) & 11 \\ A\cos\left(2\pi f_c t + \frac{3\pi}{4}\right) & 01 \\ A\cos\left(2\pi f_c t - \frac{3\pi}{4}\right) & 00 \\ A\cos\left(2\pi f_c t - \frac{\pi}{4}\right) & 10 \end{cases}$$

$$s(t) = \frac{1}{\sqrt{2}} I(t)\cos(2\pi f_c t) - \frac{1}{\sqrt{2}} Q(t)\sin(2\pi f_c t)$$

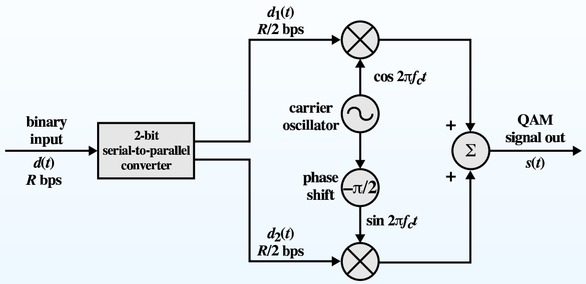
QPSK Waveforms



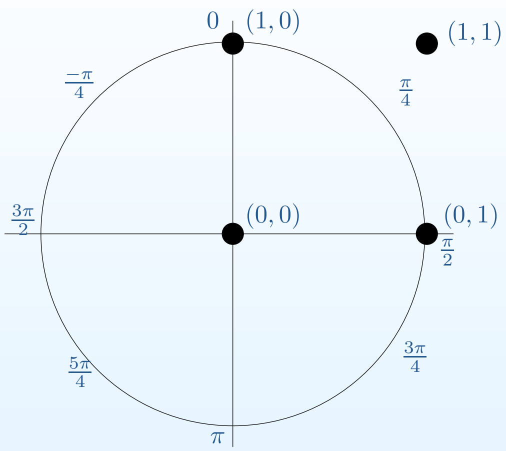
QPSK Modulator



Quadrature Amplitude Modulation



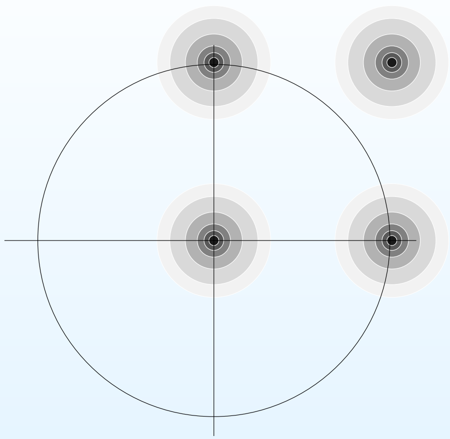
QAM Constellation



Capacity

- It would be nice if we could arbitrarily increase the data rate on a channel just by increasing the number of bits per symbol.
- In practice the number of symbols a receiver can distinguish is limited by noise in the channel.
- Noise blurs the received signal and they need to be spaced far enough apart so that different symbols can be distinguished.

Noise



Chanel Capacity Limit

Claude Shannon found a limit on the capacity of a channel in the presence of noise.

$$C = B \log_2(1 + SNR)$$

Where:

$$SNR = \frac{signalpower}{noisepower}$$

Return to ToC

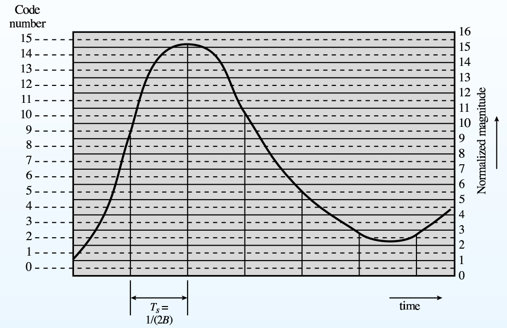
Analogue Data on Digital Signals

- Nyquist Sampling Theorum
- Data Types
  - Voice and Audio
  - Video
  - Data
- Network Performance Parameters
- Interactivity
- Requirements

Analogue to Digital Conversion

- Analogue signals are continuous in time. Digital data can only represent the signal at discrete points in time.
- The process of measuring the signal at discrete points in time is called *Sampling*.
- The sample is then converted to a (binary) digital value this is known as *Quantisation*.
- Analogue to Digital conversion is a combination of sampling and quantisation.
- The accuracy of the representation depends on the sampling rate and the quantisation precision.

PCM Coding



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

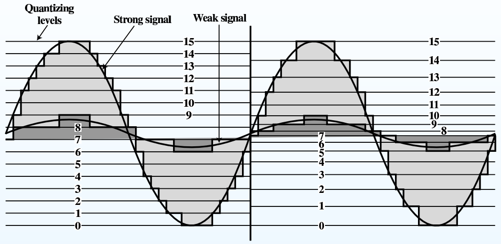
Nyquist Sampling Theorum

"A signal can be properly reconstructed if it is sampled at a frequency (rate) that is greater than *twice* the highest frequency component of the signal.

Voice and Audio

Media	Upper Limit	Lower Limit	Dynamic Range
Human Voice	100Hz	8kHz	>60dB
Telephone	300Hz	3kHz	30 - 40dB
AM Radio	50 Hz	5 - 10kHz	40 - 50dB
FM Radio	50 Hz	15kHz	50 - 60dB
CD	20Hz	20kHz	90dB
Human Hearing	20Hz	20kHz	140dB

Nonlinear Sampling



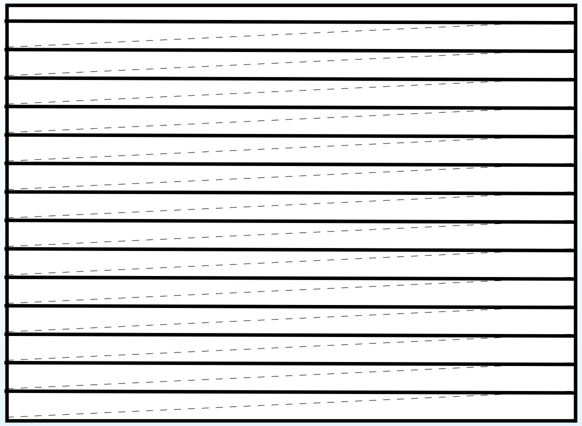
(a) Without nonlinear encoding

(b) With nonlinear encoding

Voice Compression

Codec Algorithm	Rate kb/s	Complexity MIPS	Delay ms	MOS score
G.711 PCM	64	<1	.25	4.4
G.726 ADPCM	32	1	.25	4.2
G.728 LD-CELP	16	30	3-5	4.2
G.729a CS ACELP	8	20	20	4.2
G.723.1 ACELP	5.3	18	30	3.6
GSM REP	13.2	4.5	40	3.7

Video



Television

- Standards include PAL, NTSC, SECAM.
- PAL 625 lines/frame, 25 frames /sec
- Alternate lines belong to two different fields: This is *interlacing*
- Main signal is luminance (B&W) - occupies 5.3MHz
- Two chrominance (colour difference) signals occupy 1.3 MHz each
- Sound is sent on a separate channel.

Video Compression

- Compression in Space and time

What is Data?

- Numerical or other information represented in a form suitable for processing by computer.
- Most important condideration is whether an error will make a difference.

Digital Data on Digital Signals

- Applications
- Digital Encoding Schemes
- Scrambling

Applications

- Modulating an analogue carrier is relatively complex and expensive
- It is simpler to just sent digital signals at baseband frequencies.
- Square waves occupy significant bandwidth due to the sharp corners at transitions
- Digital transmission is suitable for links with plenty of bandwidth where the cost of modem equipment is unwarranted. These are typically short copper connections or fibre optic connections.

Digital Encoding Scheme Performance

Evaluated in terms of

- Spectrum. High frequencies may be attenuated, DC results in power transfer.
- Clocking. Receiver needs to maintain synchronisation with transmitter.
- Error Detection. Can any errors be detected without additional techniques.
- Noise immunity. Will spikes cause errors in the signal.
- Cost/complexity. How difficult are the receiver and transmitter to build.

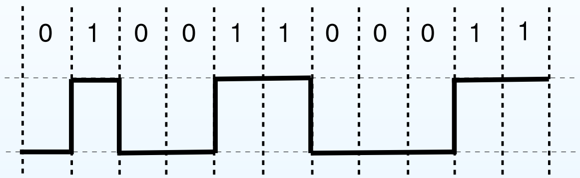
Digital Encoding Schemes

- Non-Return to Zero Schemes
- Multilevel Schemes
- Biphase Schemes

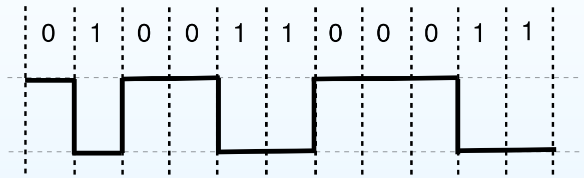
Non-Return to Zero Level

- Voltage levels are constant for each bit period
- Simplest schemes to engineer
- Bandwidth efficient
- Poor noise immunity
- Synchronisation problems with long strings of 1’s or 0’s

Non-Return to Zero

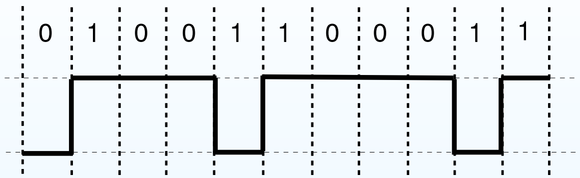


Non-Return to Zero Level



More commonly used in practice.

Non-Return to Zero Invert on Ones

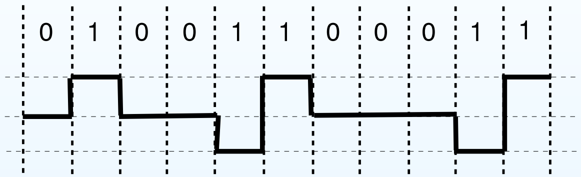


- Differential coding.
- Gives greater noise immunity as transitions are easier to detect than levels.
- No inherent polarity.

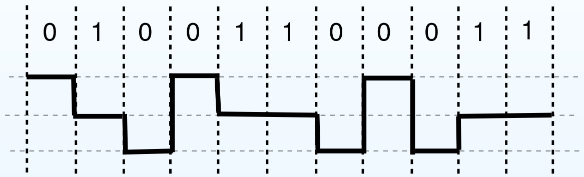
Multilevel Schemes

- Have some redundancy - can detect some errors
- Are very bandwidth efficient
- Provide synchronisation on "marks" but not on non-marks.
- Worse noise immunity due to multiple levels - higher bit error rates.
- More expensive than NRZ codes.

Bipolar Alternate Mark Inversion



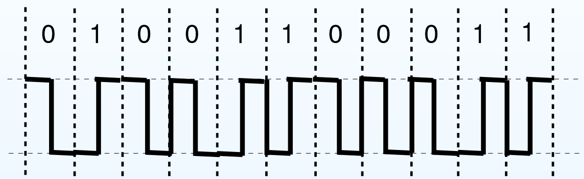
Pseudoterary



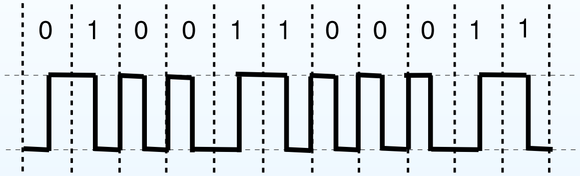
Biphase Schemes

- Always at least one transition per bit period
- Double the bandwidth requirements
- No DC component
- Very good synchronisation
- Greater noise immunity

Manchester



Differential Manchester



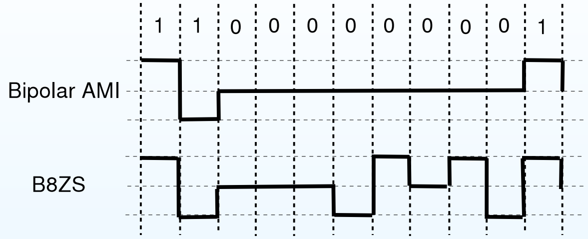
Scrambling

- Although biphase codes solve many problems, their bandwidth requirements are undesirable on long distance connections.
- Scrambling removes long strings of constant line levels with transitions.
- Removes potential DC components
- Provides synchronisation
- Can add error detection capability
- May reduce required line rate

Scrambling Schemes

- NRZ Codes are scrambled by using a mathematical transformation to produce a random looking bit stream with many transitions. The receiver reverses the transformation to produce the original data stream.
- Multilevel schemes can be scrambled by replacing long sequences of non-marks with defined patterns using polarity violations.
- Example is B8ZS based on Bipolar-AMI with replacement of strings of eight zeroes
- Following a positive mark use 000+-0-+
- Following a negative mark use 000-+0+-

B8ZS



Return to ToC