Lesson 18

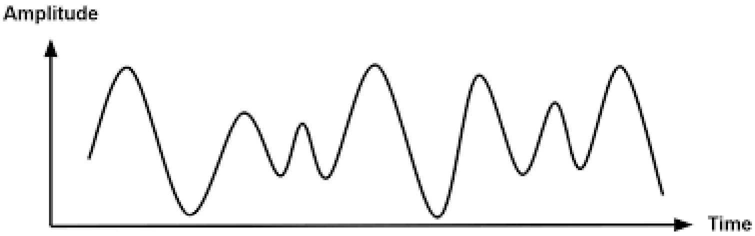
DIGITAL

ACMA Syllabus February 2024 Chapters 1.10 and 3.8

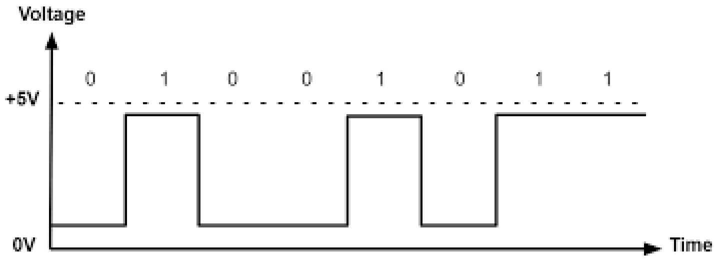
Contents

Digital Signals

An analogue signal is a signal that changes in amplitude over time when impacted by an external data.



A digital signal represents data as discrete values.



Binary Numbering

We count numbers to the base 10. This is a decimal counting system. Computer work on a binary counting system. A system to the base 2. In the binary number a number is represented by using only two digits (0 and 1). Each 1 or 0 is called a bit and a group of eight bits is called a byte. The value of each bit depends on its position in a byte.

Positions

1	2	3	4	5	6	7	8
1	2	4	8	16	32	64	128

Max value allocated to each position

A 1 in position one is one.

1	2	3	4	5	6	7	8
1	0	0	0	0	0	0	0
1	2	4	8	16	32	64	128

A 1 in position 5 is sixteen.

1	2	3	4	5	6	7	8
0	0	0	0	1	0	0	0
1	2	4	8	16	32	64	128

Counting in binary from 1 to 10 would look like this.

Number	P1	P2	P3	P4	P5	P6	P7	P8
1	1	0	0	0	0	0	0	0
2	0	1	0	0	0	0	0	0
3	1	1	0	0	0	0	0	0
4	0	0	1	0	0	0	0	0
5	1	0	1	0	0	0	0	0
6	0	1	1	0	0	0	0	0
7	1	1	1	0	0	0	0	0
8	0	0	0	1	0	0	0	0
9	1	0	0	1	0	0	0	0
10	0	1	0	1	0	0	0	0

Examples:

134 =	0	1	1	0	0	0	0	1
0 =	0	0	0	0	0	0	0	0
255 =	1	1	1	1	1	1	1	1

ASCII

Computers don’t know the alphabet, so letters, numbers and characters are allocated a binary number representing the letters, numbers and characters. ASCII (American Standard Code for Information Interchange) is the most common character encoding format for text data in computers and on the internet. The ASCII characters number 255, just what a byte can count too.

Check it out.

Hexadecimal System

Hexadecimal or hex is a numbering system to the base 16. Sixteen being the bits in two bytes on a computer.

Decimal	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
Hex	1	2	3	4	5	6	7	8	9	A	B	C	D	E	F

A to D and D to A

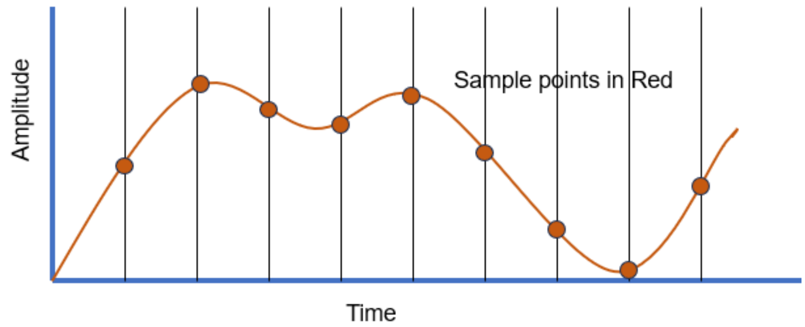
A method to convert analogue signals to digital and reconvert them back to analogue is an essential part of the digital process.

- The analogue to digital conversion is referred to as the ADC. (Analogue to Digital Conversion)
- The digital to analogue conversion is referred to as DAC (Digital to Analogue Conversion)

A **CODEC** (coder/decoder) performs both ADC and DAC functions in one.
The sampling rate of an analogue signal, for conversion to digital, should be at least twice the highest frequency of the analogue signal. This is called the Nyquist Theorem.

Digital Signal Processing (DSP)

The digitisation of an analogue signals involves sampling the voltage at predetermined points along the signal. (A to D)



These points are rounded to approximate the original signal by quantisation. The accuracy of the signal reproduction depends on the number of levels the original voltage was digitised too. (D to A)

Sampling

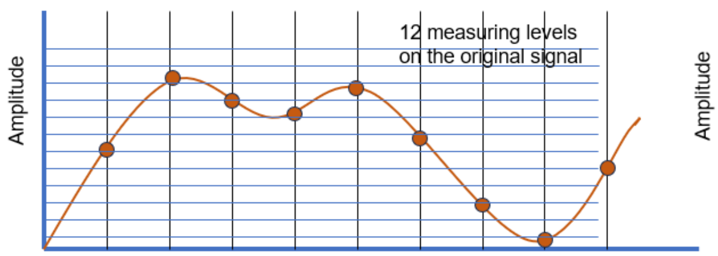
The number of sampling points on an analogue signal is determined by the Nyquist theorem which states that an analogue signal can be digitised, without aliasing errors, when the sampling rate is greater than or equal to twice the highest frequency component in the signal being digitised.

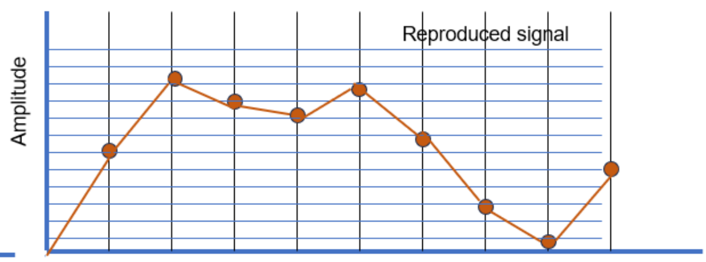
Aliasing refers to the incorrect measurement of a signal's frequency due to an inadequate digital sampling rate. If a signal is not sampled using enough data points, its true frequency will be underestimated.

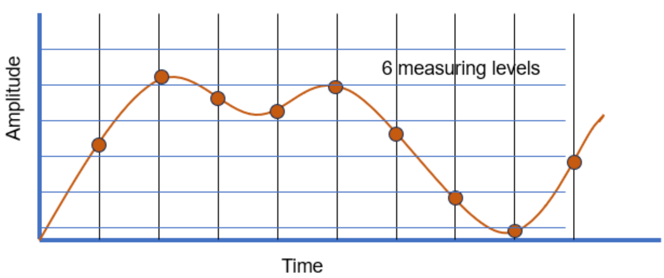
Example: Digitising an analogue audio signal and the highest frequency in the audio signal is 20 kHz. The minimum sampling rate is 40 kHz,

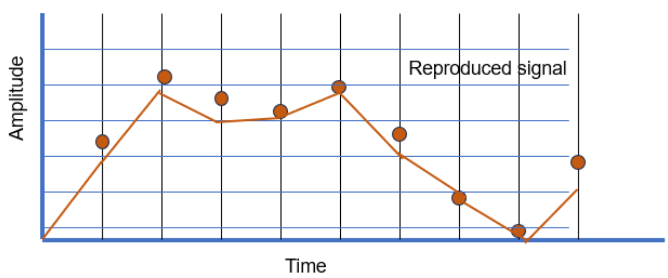
Quantisation

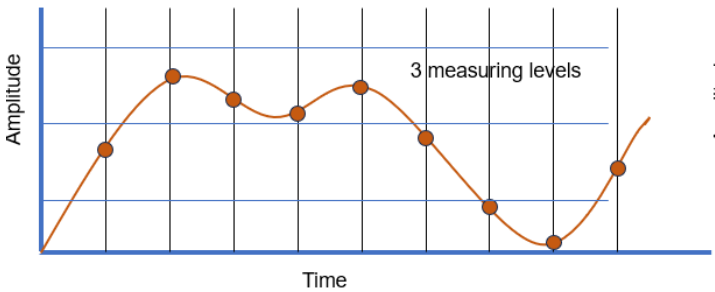
The level of sampling is import for the signal reproduction. Quantisation is the restoring of the signal from the samples. The number of levels in measuring the amplitude are important for the accurate reproduction of the signal. See examples below.

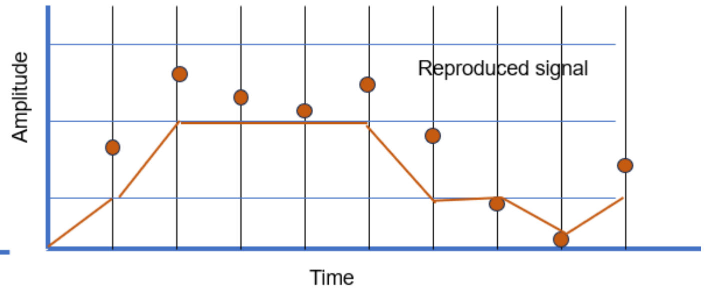












Anti-Aliasing Filter (AAF)

AAF is a device used before a signal is sampled to restrict the bandwidth of a signal. If the assumption was made that the highest frequency in the signal is X, then according to Nyquist theorem, sampling should be 2X. The AAF basically keeps the signal being sampled to a limit of 2X. A practical AAF makes a trade, off in order to ensure that all frequencies of interest can be reconstructed, a practice called oversampling is applied.

Digital Filtering

A digital filter is a mathematical algorithm to remove any unwanted information such as glitches or noise on a measured signal.

Digital filters are divided into the following two categories with each filter is categorised by the length of its impulse response.

- Infinite impulse response (IIR)
- Finite impulse response (FIR)

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IIR filters

IIR filters are generally chosen for applications where memory is limited. They have a low implementation cost but can be less numerically stable than their FIR (finite impulse response) counterparts, due to the feedback paths.

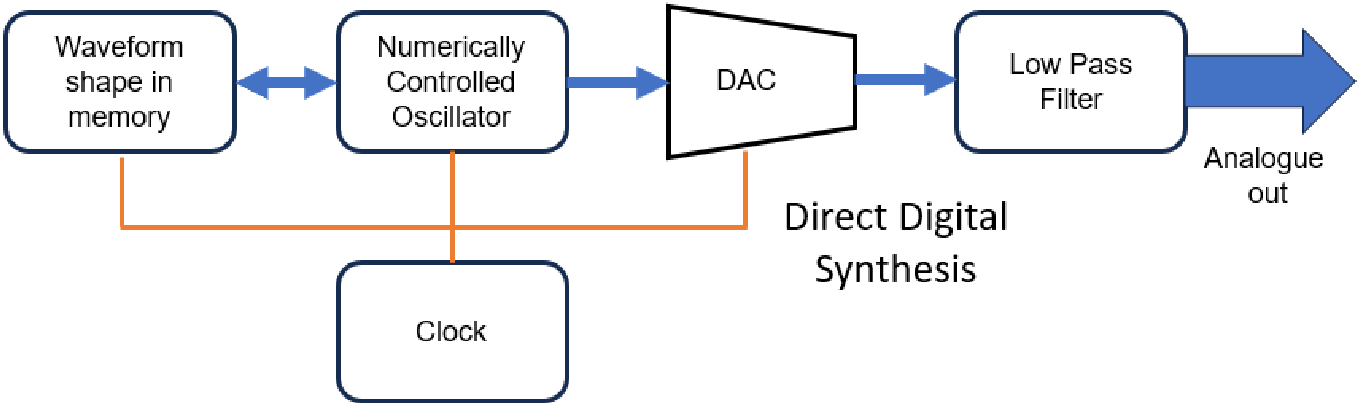
FIR filters

FIR filters are generally chosen for applications where memory and computational performance are available. They have a widely deployed in audio and biomedical signal enhancement applications. Their all-zero structure ensures they never become unstable for any type of input signal, which gives them a distinct advantage over the IIR.

Due to the various advantages, FIR filters are preferred over IIR filters. FIR filters are mainly used in bandpass and band stop filtering applications. While low pass and anti-aliasing filtering applications require IIR digital filters

Direct Digital Synthesis (DDS)

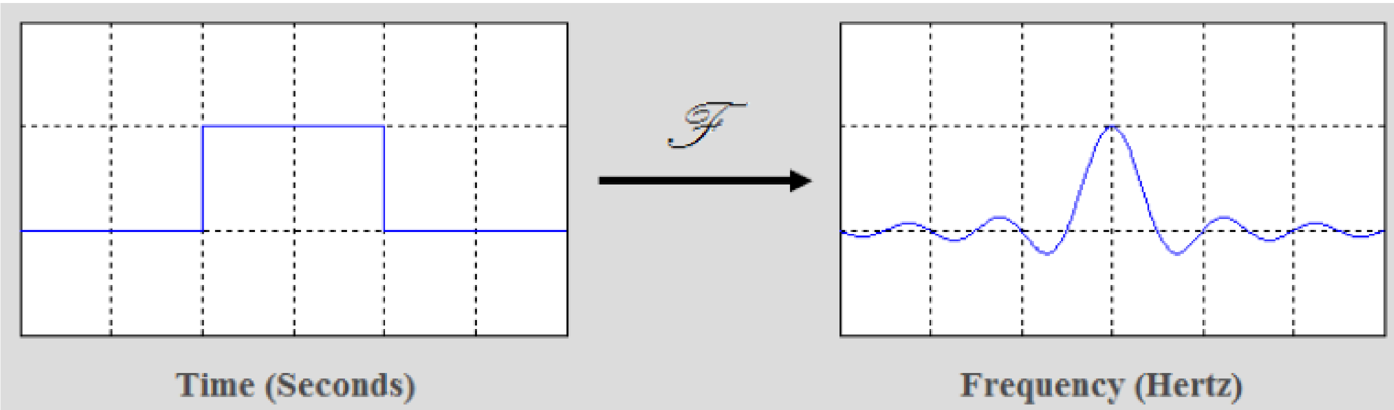
DDS is a method used in frequency synthesizers for creating waveforms from a single, fixed-frequency reference clock. DDS is a common technique for generating signals as the desired waveform is stored digitally in memory as a series of amplitude points.



The clock provides a stable time base for the system and determines the accuracy of the DDS. The NCO is controlled by the digital word contained in memory. The digital waveform is converted to an analogue waveform by the DAC. The LPF rejects unwanted signals.

Fourier Transformation

The Fourier transform (FT) converts a function into a form that describes the frequencies present in the original function. The Fourier transform can be compared to decomposing the sound of a musical chord into terms of the intensity of its constituent pitches.



The Fourier Transform breaks a waveform into an alternate representation, characterized by the sine and cosine functions of varying frequencies. The Fourier Transform shows that any waveform can be re-written as the sum of sinusoidal waves.
If you wish to study the maths of FT, go HERE.

Discrete Fourier Transform (DFT)

The discrete Fourier transform (DFT) algorithm transforms samples of signals from the time domain into the frequency domain.

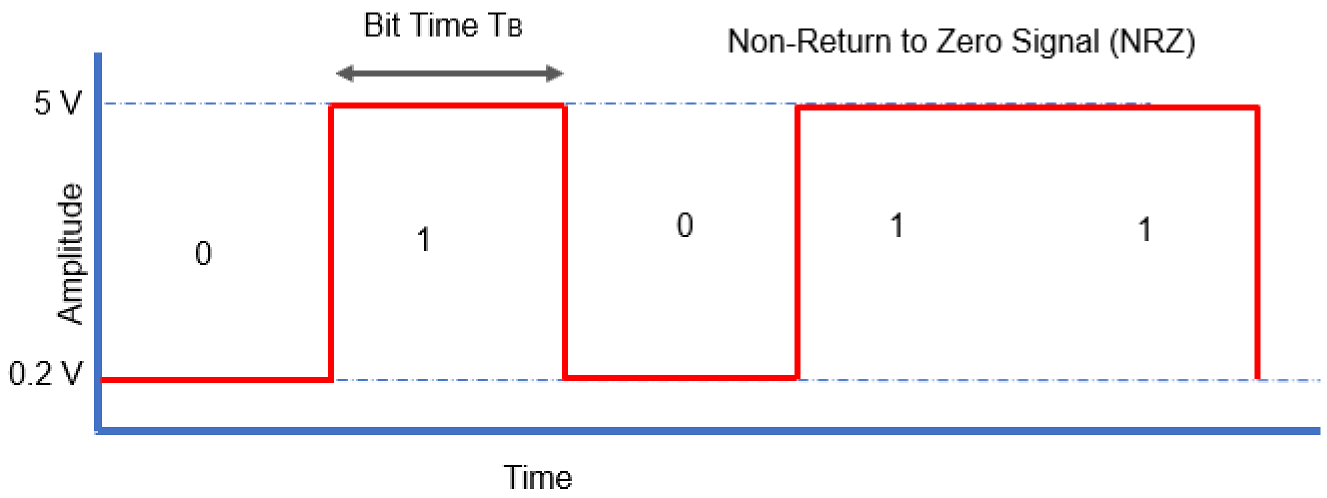
Fast Fourier Transform (FFT)

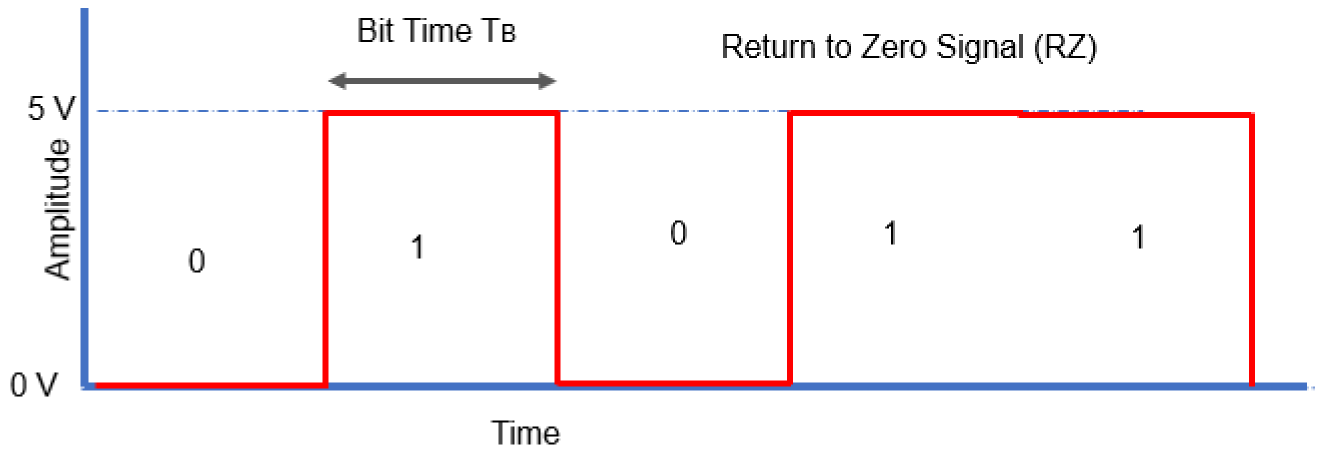
FFT is any algorithm for calculating the DFT.

Digital Modulations Methods

Most data communications over networks are serial-data transmissions where data bits (Binary Digits) are transmitted one at a time. The data shifts between two voltage levels, such as +5 V for a binary 1 and +0.2 V for a binary 0. This form of signal never goes to zero and is referred to as Non-Return to Zero signal (NRZ). A Return-to-Zero signal (RZ) could have +5 V for a binary 1 and 0 V for a binary 0.

Bits and bytes are both units of data where a byte is equivalent to eight bits. An octet is a collection of eight bits and is used when it is necessary to specify that there are only eight bits in the collection.





Bit Rate

The speed of the data is expressed in bits per second (bits/s or bps). The data rate R is calculated from the duration of the bit or bit time (TB). The Bit Rate is also called channel capacity (C).

$$Bit\ Rate = \frac{1}{T_B}$$

Example: TB = 5 μS, the bit rate would be 200,00 bits per second or 200 kBPS.

Payload and Overhead

Not all bits in a data stream are actual information being transferred. The data stream includes bits representing source address, destination address, error detection and correction codes. This data is referred to as the “Overhead”. The actual data being transferred is referred to as the “Payload” The overhead may be substantial—up to 20% to 50% depending on the total payload bits sent over the channel.

Communication Protocols

A communication protocol is a set of rules that enables two or more entities to exchange information. A protocol defines the rules, syntax, semantics, and synchronization of communication and possible error recovery methods.

A common internet protocol, known as Transmission Control Protocol (TCP), is a framework for communications between computers.

Baud Rate

Baud rate is the measure of the number of changes to the signal (per second) that propagate through a transmission medium. The baud rate may be higher or lower than the bit rate, which is the number of bits per second that the user can push through the transmission system. There may be more than two symbols per transmission interval, whereby each symbol represents multiple bits. With more than two symbols, data is transmitted using modulation techniques.

Example: A modem has a specified bit rate of 4,800 bps. Each bit represents two bits of data when the signal changes from one state to another.

Bit Rate = 4800 bps
Baud rate = 4,800 bps / 2 characters per bit
Baud rate = 2,400 baud

Bandwidth

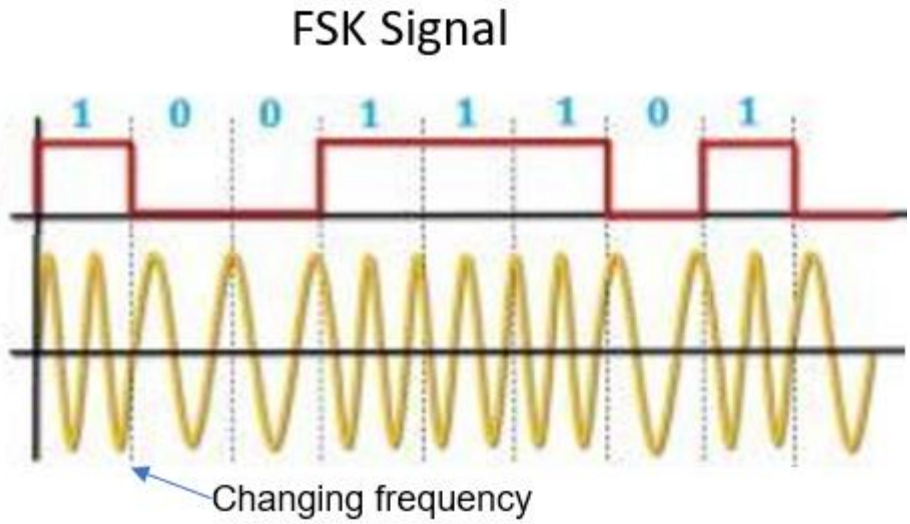
The bandwidth of an analogue transmission channel is usually defined as the difference between the highest and lowest frequencies that the channel can support measured in hertz. In a digital system, the bandwidth is defined by the bit rate and is expressed in bits per second. Data transfers today support bit rates in the tens, hundreds, and even thousands of megabits per second.

Modulation

Digital modulation is the process of encoding information into the amplitude, phase or frequency of the transmitted signal. The goal of digital modulation is to include as much data into the least amount of spectrum possible. This is called spectral efficiency and quantifies how quickly data can be transmitted in an assigned bandwidth. Now different modulation methods can include very high bit rates. Common digital modulation techniques used by amateur operators are shown below.

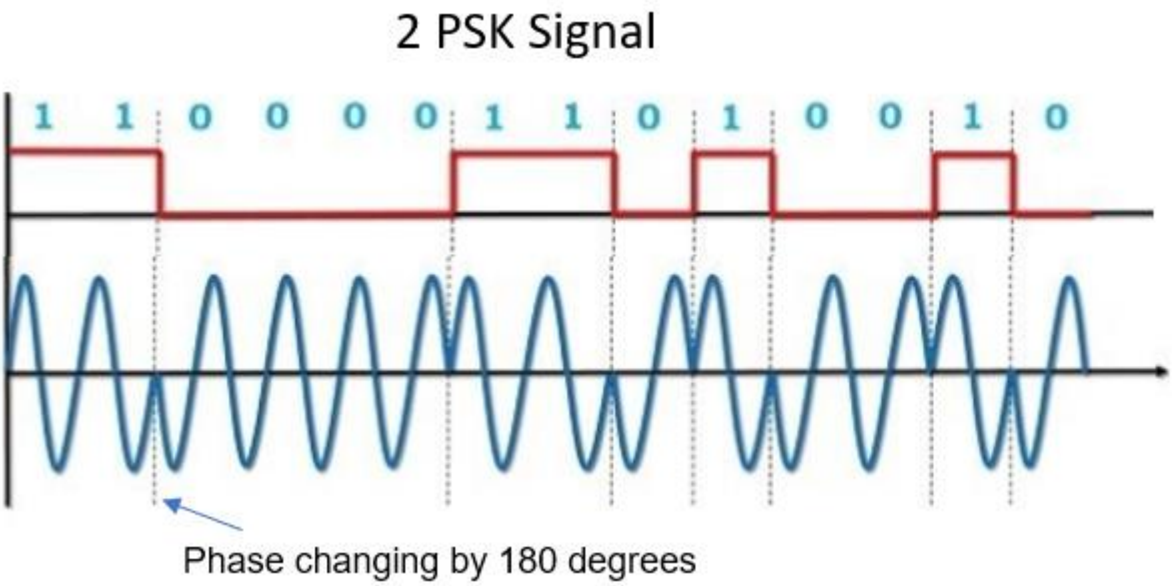
Frequency Shift Keying (FSK)

FSK is a means of transmitting digital signals using analogue signals. A logic 0 is represented by a wave at a specific frequency and logic 1 is represented by a wave at a different frequency. A device such as a modem converts the binary data to FSK for transmission over telephone lines, cables, optical fibre or wireless. The modem also reconverts incoming FSK signals to digital low and high states.



2-Phase Shift Keying (2-PSK)

2 PSK, also called Binary Phase Shift Keying (BPSK) or Phase Reversal Keying (PRK), is the simplest form of PSK. PSK uses two phases such as 0° and 180°. PSK handles the highest noise level or distortion before the demodulator reaches an incorrect decision making PSK the most robust of all the PSKs. The downside is that PSK is unsuitable for high data-rate applications.

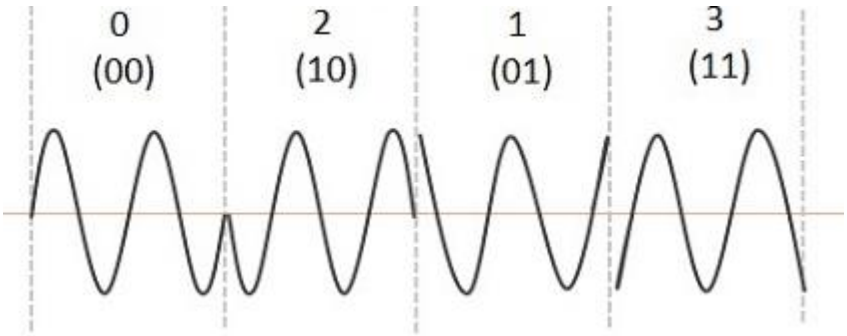


4 Phase – Frequency Shift Keying (4- PSK)

In binary, the combination of 1 and 0 can only be rearranged four ways.

Count	Binary Combination
0	00
1	10
2	01
3	11

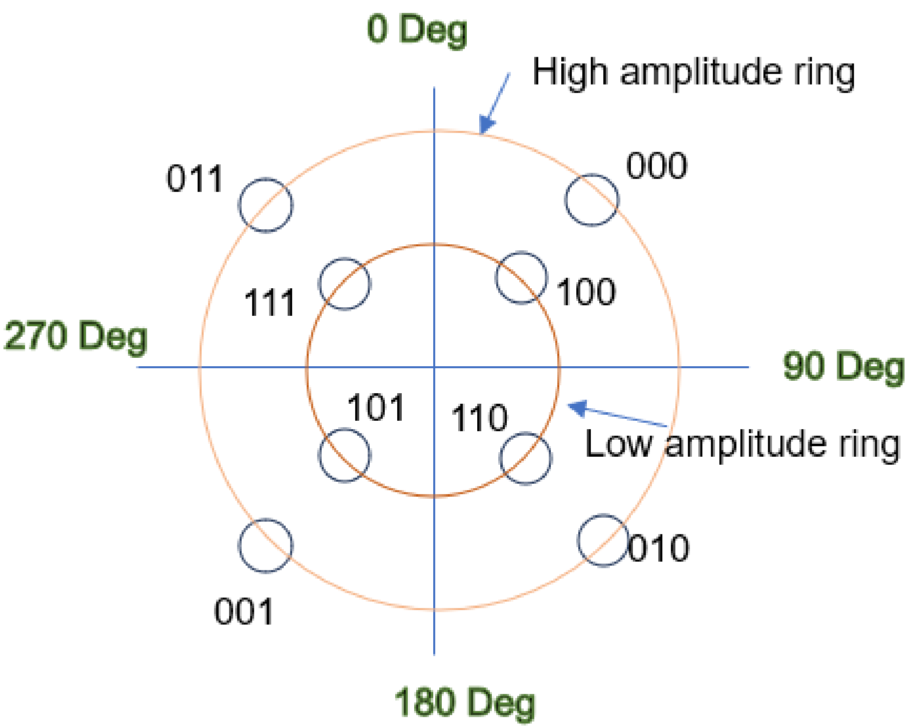
4-PSk or Quadrature Phase Shift Keying (QPSK) enables one frequency phase (0, 90, 180, or 270 degrees) to represent two bits. This enables the signal to carry twice as much information as ordinary PSK using the same bandwidth.



Quadrature Amplitude Modulation (QAM)

QAM a combination of amplitude changes and phase variations to transmit groups of data bits. The table below is compiled as an example only.

Bits	Amplitude	Phase Degrees
000	High	45
100	Low	45
010	High	135
110	Low	135
001	High	225
101	Low	225
011	High	315
111	Low	315



Error Correction

Error correction is the process of detecting errors in transmitted data and reconstructing the original error-free data. Error correction ensures that corrected and error-free messages are obtained at the receiver.

Where a system detects a bad message, the protocol would generate an automatic repeat request (ARQ).

ARQ is an error control (error correction) method that uses error-detection codes and positive and negative acknowledgments.

Error-correcting code (ECC) or forward error correction (FEC) is a method that involves adding parity data bits to the message. These parity bits will be read by the receiver to determine whether an error occurred during transmission. The receiver checks and corrects errors when they occur. It does not ask the transmitter to resend the frame or message.

A hybrid method that combines both ARQ and FEC functionality is also used for error correction. In this case, the receiver asks for retransmission only if the parity data bits are not enough for successful error detection and correction.

AMTOR and Packet radio are two examples of amateur communications using error correcting.

Amateur Used Protocols

Listed below are a few of the protocols used by amateurs to modulate and transmit digital data.

APCO P25 - APCO P25 is a digital standard developed for public safety and land mobile applications.

D-STAR - Digital Smart Technologies for Amateur Radio is a digital voice and data protocol specification for amateur radio. The system uses minimum-shift keying in its packet-based standard.

DMR - DMR stands for Digital Mobile Radio and is an international standard that has been defined for two-way radios. The DMR standard allows equipment developed by different manufacturers to operate together on the same network.

System Fusion - System Fusion is Yaesu's implementation of Digital Amateur Radio DMR, utilizing C4FM 4-level FSK Technology to transmit digital voice and data over the Amateur radio bands.

M17 - M17 uses Codec 2, a low bitrate voice codec developed by David Rowe VK5DGR et al. Codec 2 was designed to be used for amateur radio and other high compression voice applications.

Amateur teleprinting over radio (AMTOR) - AMTOR is rarely used today, as other protocols such as PSK31 are becoming favoured by amateur operators for real-time text communications.

PSK31 - PSK31 is distinguished from other digital modes in that it is specifically tuned to have a data rate close to typing speed, and has an extremely narrow bandwidth, allowing many conversations in the same bandwidth as a single voice channel.

Multi Tone 63 (MT63) - MT63 is a digital radio modulation mode for transmission in high-noise situations. MT63 is designed for keyboard-to-keyboard conversation modes, on HF amateur radio bands.

Multiple frequency-shift keying (MFSK) - MFSK is a variation of frequency-shift keying (FSK) that uses more than two frequencies. M is usually between 4 and 64

Packet radio (AX25) - Packet radio is the application of packet switching techniques to digital radio communications. Packet radio uses a packet switching protocol as opposed to circuit switching or message switching protocols to transmit digital data via a radio communication link.

Amateur Packet Radio Network (AMPRNet) - AMPRNet or Network 44 is used in amateur radio for packet radio and digital communications between computer networks managed by amateur radio operators.