Jules' Workshop

Standard

Unit 19
DIGITAL

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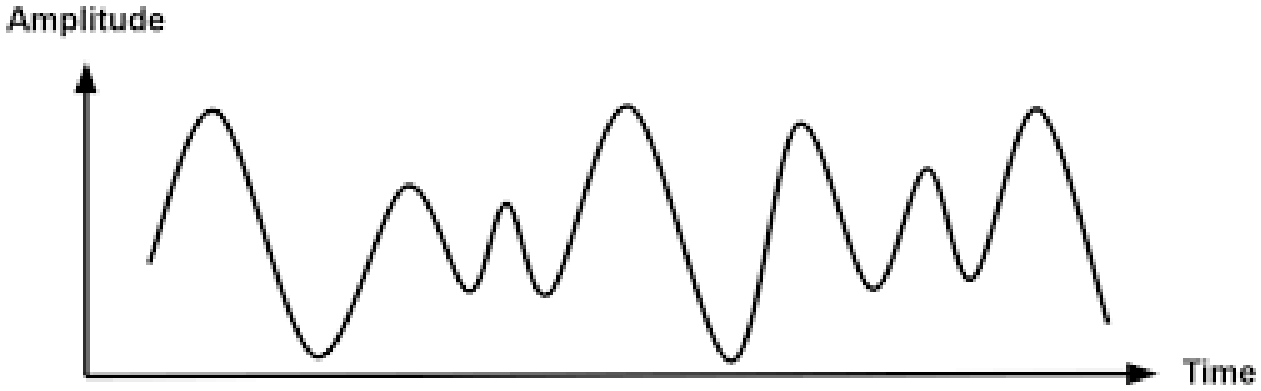
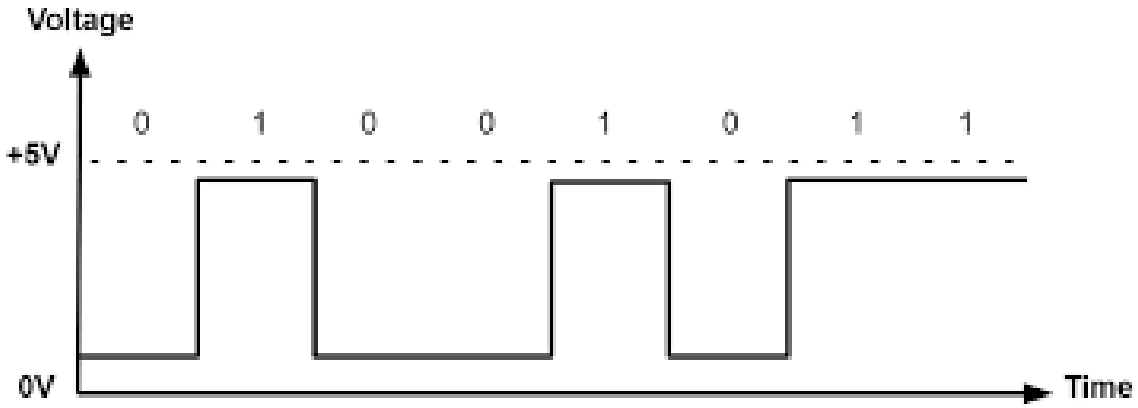
Digital Signals

An analogue signal is a voltage that continuously and infinitely varies in accordance with some time parameter. For example, radio waves, television waves, or sound waves are all examples of analogue signals.



A digital signal represents data as discrete values.



Analogue systems use continuous wave signals to represent data, while digital systems use discrete, binary values (0s and 1s).

Analogue is suited for natural, high-detail signals (e.g., sound), whereas digital is more robust, less prone to noise, and easier for computing, making it more accurate for storage and transmission.

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Computer Numbering

Binary

Our numbering system is based on the number 10. But computers are not that smart. They can only detect if a device is on or off. These two states limit their counting to binary numbering only.

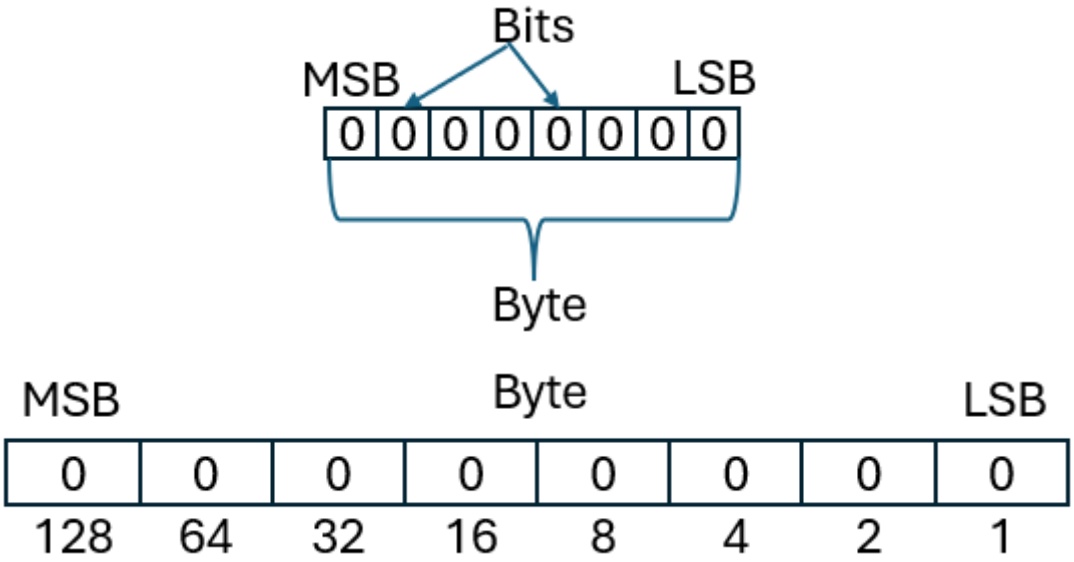
As binary numbering can only show two states, let's show them as a 1 or a 0 and call each of these a "bit".

One bit does not do much, so join a few bits together. Four bits makes a "nibble", and eight bits makes a "byte." The word byte was coined in 1956 by IBM engineer Werner Buchholz who spelt the group as a byte instead of bite, so it would not be confused with bit.

When computing power first exploded on the public scene in the 1970s, a popular processor was the Zilog Z80. This popular processor is an 8-bit microprocessor that became a cornerstone of early computing,

Eight bits, in those days, was considered practical. But how big was it? Let us find out!

Join eight bits together, to make a byte.



Identifying the bits, the LSB is position 1 and the MSB is position 8.

Each bit can only have two states, 1 or 0. Yes, zero is counted as a number. As computers can only count in twos, so if we double the number for every bit, we can represent a much larger number.

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

	MSB							LSB
Number	**P7**	**P6**	**P5**	**P4**	**P4**	**P3**	**P2**	**P1**
Value	128	64	32	16	8	4	2	1
1	0	0	0	0	0	0	0	**1**
2	0	0	0	0	0	0	**1**	0
3	0	0	0	0	0	0	**1**	**1**
4	0	0	0	0	0	**1**	0	0
5	0	0	0	0	0	**1**	0	**1**
6	0	0	0	0	0	**1**	**1**	0
7	0	0	0	0	0	**1**	**1**	**1**
8	0	0	0	0	**1**	0	0	0
9	0	0	0	0	**1**	0	0	**1**
10	0	0	0	0	**1**	0	**1**	0

Examples:

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

Practice making up some binary numbers.

Hexadecimal System (For info)

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

Our number is based on 10, 11 to 15 are represented by letters.

Decimal	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
Hex	0	1	2	3	4	5	6	7	8	9	A	B	C	D	E	F

Example: When selecting a colour in your computer, you notice letters in the number sequence.

The colour in the rectangle is identified as #3FC9AF. This is a hex number for the colour.

Octet (For info)

A digital octet is a unit of information consisting of exactly eight bits. This is used primarily in networking and computing to avoid the ambiguity of the term "byte" (which can sometimes vary in size). An octet is defined by international standards like IEC 60027-2.

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Letters, Numbers and Characters

Many computing concepts were developed and applied during the early days of processor development.

ASCII

One important standard was the ASCII (American Standard Code for Information Interchange) character set. ASCII was designed as a 7-bit code in the early 1960s for teletype machines and systems. ASCII was first used commercially on model 33 and 35 Teleprinters in 1963.

The ASCII set has a total of 128 unique controls and characters represented. The first 32 controls are non-printable. The remaining 96 characters include number, upper / lower case alphabet and common punctuation.

A computer byte can represent all the English alphabet, numbers and punctuation needed at that time.

$b_7 b_6 b_5$ → / Bits					0 0 0	0 0 1	0 1 0	0 1 1	1 0 0	1 0 1	1 1 0	1 1 1
b_4 ↓	b_3 ↓	b_2 ↓	b_1 ↓	Row ↓ / Column →	0	1	2	3	4	5	6	7
0	0	0	0	0	NUL	DLE	SP	0	@	P	`	p
0	0	0	1	1	SOH	DC1	!	1	A	Q	a	q
0	0	1	0	2	STX	DC2	"	2	B	R	b	r
0	0	1	1	3	ETX	DC3	#	3	C	S	c	s
0	1	0	0	4	EOT	DC4	$	4	D	T	d	t
0	1	0	1	5	ENQ	NAK	%	5	E	U	e	u
0	1	1	0	6	ACK	SYN	&	6	F	V	f	v
0	1	1	1	7	BEL	ETB	'	7	G	W	g	w
1	0	0	0	8	BS	CAN	(	8	H	X	h	x
1	0	0	1	9	HT	EM	)	9	I	Y	i	y
1	0	1	0	10	LF	SUB	*	:	J	Z	j	z
1	0	1	1	11	VT	ESC	+	;	K	[	k	{
1	1	0	0	12	FF	FS	,	<	L	\	l	¦
1	1	0	1	13	CR	GS	—	=	M	]	m	}
1	1	1	0	14	SO	RS	.	>	N	^	n	~
1	1	1	1	15	SI	US	/	?	O	_	o	DEL

Unicode

In 1991, Unicode was officially established. Unicode is a massive, evolving standard designed to encode all characters from all writing systems. Unicode is not a fixed-bit system but an encoding standard, commonly implemented as 8-bit (UTF-8), 16-bit (UTF-16), or 32-bit (UTF-32).

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Analogue to Digital Conversion (ADC)
and
Digital to Analogue Conversion (DAC)

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

ADC is essential when you need to digitize, measure, or control a real-world, analogue signal with a computer. The ADC samples the signal at set intervals and quantizes it into a digital representation. Conversely, a DAC is used when you need to send a signal to a device that expects analogue input, such as speakers, actuators, or a screen.

ADC

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

DAC

The digital to analogue conversion is referred to as DAC (Digital to Analogue Conversion)

CODEC

A codec (short for coder-decoder) is hardware or software that compresses raw digital data/signals for efficient storage and transmission and decompresses them for playback or editing.

Codecs are essential for streaming media, video conferencing, and digital audio, enabling high-quality content via lower bandwidth.

This a combination of an ADC and a DAC.

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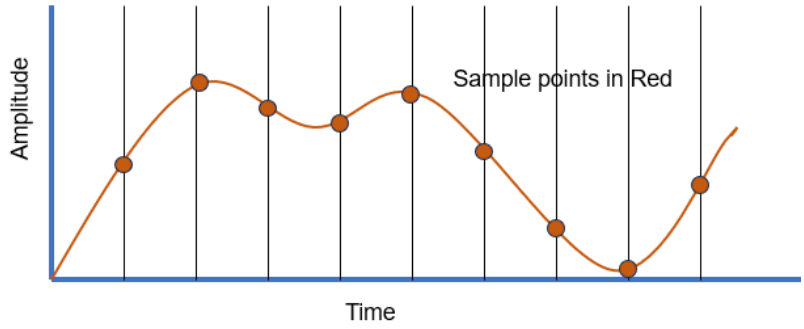
Digital Signal Processing (DSP)

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 samples 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.

Nyquist Theorem

The Nyquist-Shannon sampling theorem states that to accurately reconstruct an analogue signal from its digital samples, the sampling rate must be greater than twice the highest frequency of the sampled signal.
Sampling at or above this rate prevents signal distortion known as aliasing.

$$\text{Sampling rate } (f_s) = > 2\ f_{max}$$

Example: What is the digital sampling rate if the maximum analogue signal is 20k Hertz?

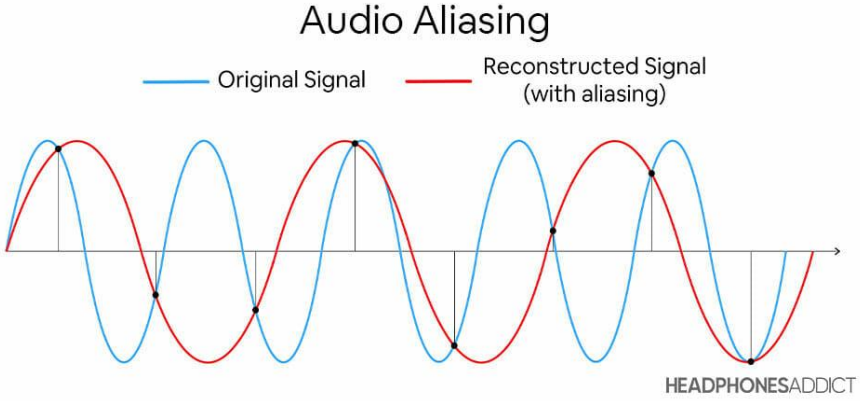
$$\text{Sampling rate } (f_s) = > 2 \times 20{,}000 \text{ Hertz}$$
$$f_s = > 40{,}000 \text{ samples per second}$$

Aliasing

Aliasing is a phenomenon in signal processing and computing where high-frequency signals or data, when sampled too slowly, appear as false, lower-frequency signals (aliases).

In the figure, the blue signal is the original and red is the constructed signal. The red signal is a slightly different frequency than the original signal.

The original signal was not sampled fast enough as required by the Nyquist theorem.



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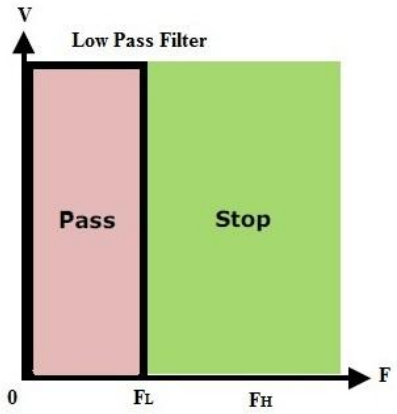
Anti-Aliasing Filter (AAF)

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 X.

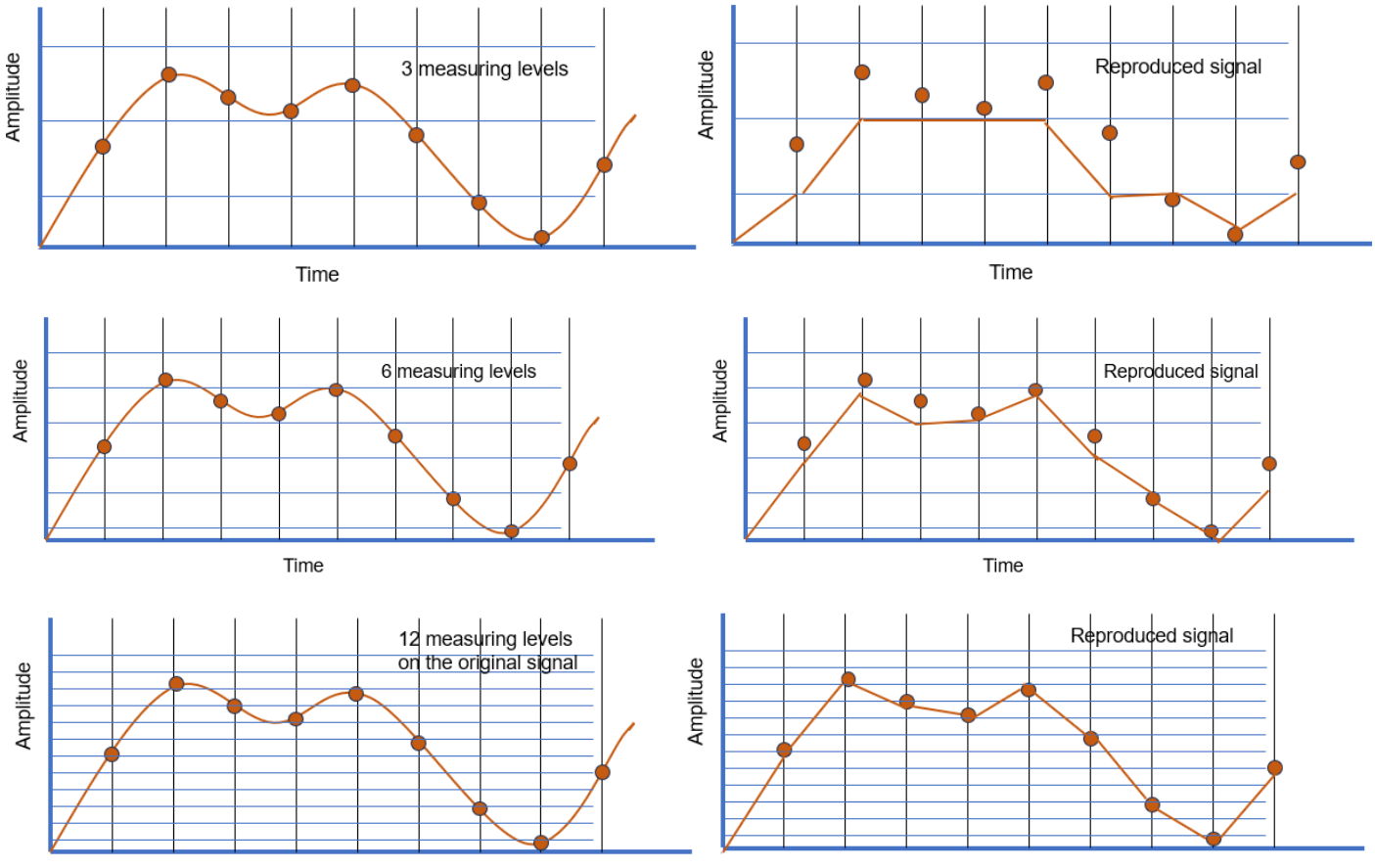
The AAF is a low pass filter which keeps the signal being sampled to a limit of X. Removes the potential for signal aliasing.

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.



Quantisation

The level of sampling is important 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.



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Digital Filters

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

A digital filter's impulse response is the output a filter produces when given a single, instantaneous impulse.

This response characterizes linear time-invariant (LTI) systems, defining how the filter acts on signals over time, often called FIR (Finite) or IIR (Infinite) depending on its duration.

Infinite Impulse Response (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.

Finite Impulse Response (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 used in bandpass, and band stop filtering applications. While low pass and anti-aliasing filtering applications require IIR digital filters

Preference

Finite Impulse Response (FIR) filters are chosen over Infinite Impulse Response (IIR) filters when linear phase response (zero phase distortion), guaranteed stability, or precise frequency response control is required.

IIR filters are often more computationally efficient, FIR filters are favoured when signal integrity outweighs the need for low latency and high computational efficiency.

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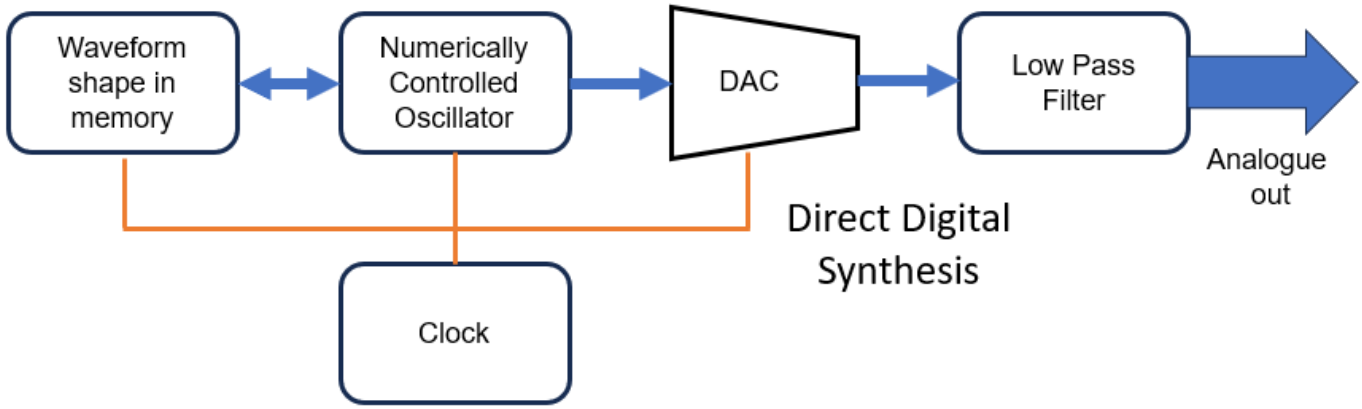
Direct Digital Synthesis (DDS)

Direct Digital Synthesis (DDS) is a method for generating analogue waveforms (typically sine waves) by producing a time-varying digital signal and converting it to analogue using a DAC.

DDS creates waveforms from a single, fixed-frequency reference clock. The clock provides a stable time base for the system and determines the accuracy of the DDS. The Numerically Controlled Oscillator (NCO) is controlled by the digital word contained in memory. The speed of the reference clock determines the maximum output frequency.

The digital waveform is converted to an analogue waveform by the DAC.

The Low Pass Filter (LPF) rejects unwanted signals.



DDS is used in signal generators & test equipment as they enable rapid frequency sweeping to check for resonance or calibrate sensors.

In communication systems, DDS is used as frequency synthesizers for local oscillators in receivers and transmitters to generate stable carrier frequencies.

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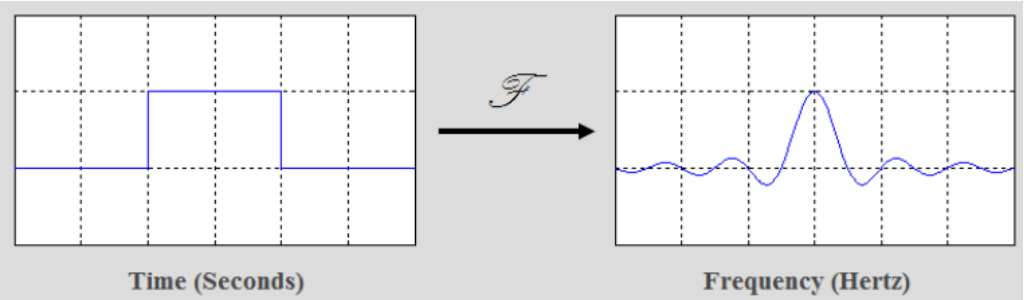
Fourier Transformation

The Fourier Transform (FT) breaks down a complex signal—like sound or images—into its fundamental, simple sine wave frequencies.

The FT converts data from the time domain, signal changing over time, to the frequency domain, energy across different frequencies. This mathematical tool identifies the amplitude and phase of each component frequency, allowing for complex data processing.

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 expressed as the sum of sinusoidal waves.

The formula to perform the transformations is shown below. Just be aware of the complexity involved in the transformation. Computers today are able to make the transformation instantly and graphically. Previously each point had to be calculated.

Fourier transform

$$\widehat{f}(\xi) = \int_{-\infty}^{\infty} f(x)\, e^{-i2\pi\xi x}\, dx, \quad \forall \xi \in \mathbb{R}.$$

Do not need to know the formula.

Discrete Fourier Transform (DFT)

The Discrete Fourier Transform (DFT) is a mathematical algorithm that converts a finite sequence of equally spaced samples of a signal (time or spatial domain) into a representation of its constituent frequencies (frequency domain).

DFT is the fundamental tool for digital signal processing, enabling spectral analysis, filtering, and compression by calculating complex coefficients that represent the amplitude and phase of specific frequency components.

Fast Fourier Transform (FFT)

The Fast Fourier Transform (FFT) is an efficient algorithm that computes the Discrete Fourier Transform (DFT) of a sequence, converting signals between time/space and frequency domains.

Difference

FFT has the advantage due to speed and memory efficiency. The DFT can process sequences of any size efficiently but is slower than FFT.

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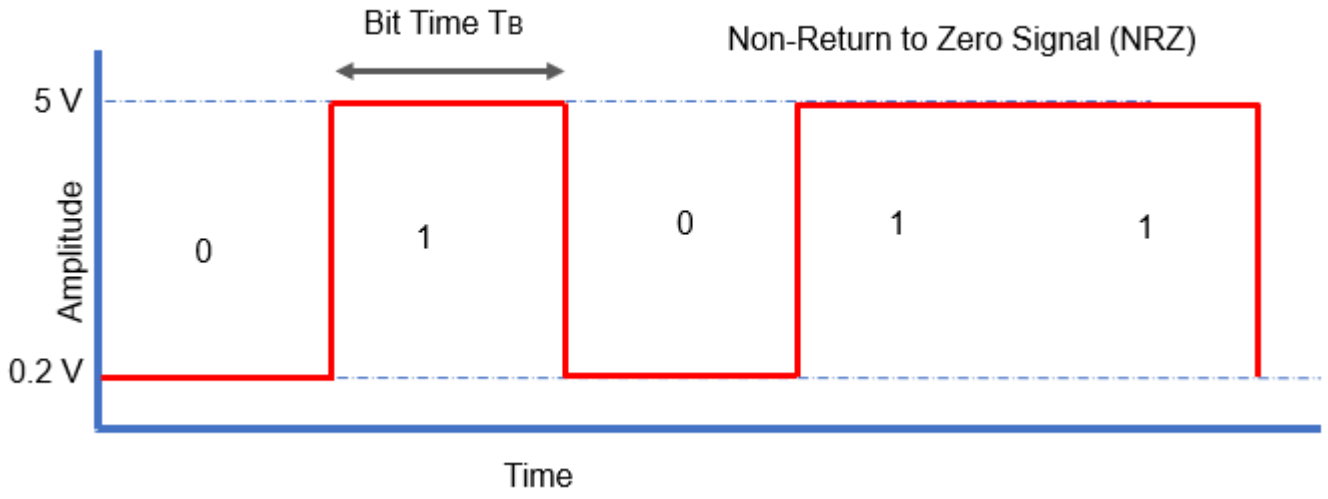
Digital Coding Techniques

Most data communications over networks are serial-data transmissions where data bits (Binary Digits) are transmitted one at a time.

A binary digit is either a 1 or a 0. The difference between 0 and 1 is the voltage level.

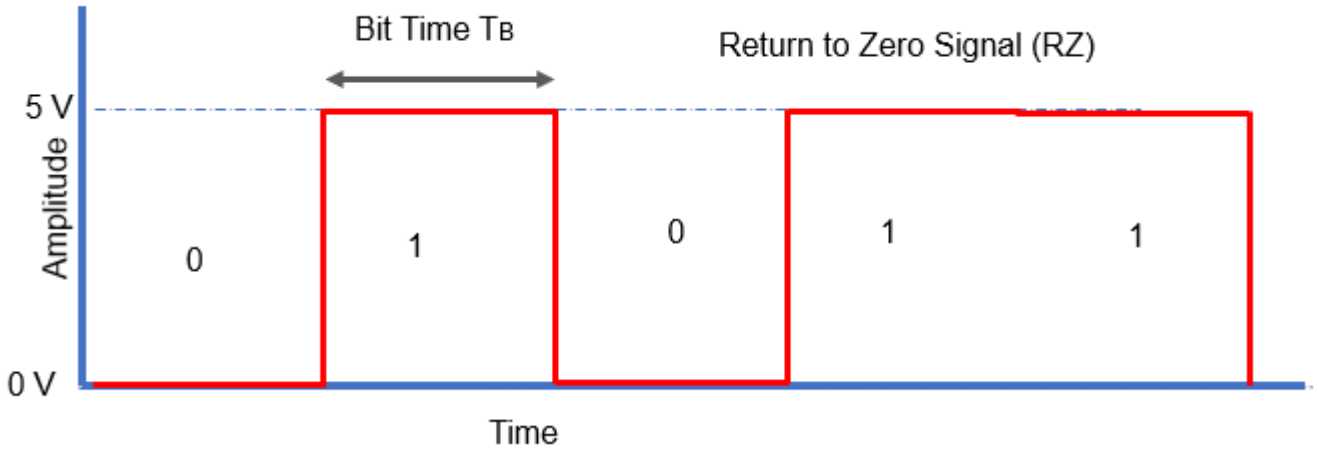
'NRZ

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).



RZ

A Return-to-Zero signal (RZ) could have +5 V for a binary 1 and 0 V for a binary 0.



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 (T_B). The Bit Rate is also called channel capacity (C).

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

Example: TB = 5 µS, the bit rate would be 200,000 bits per second or 200 kBPS.

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Bits and Bauds

Bit rate is the number of bits (0s and 1s) transmitted per second, i.e. bits per second (bit/s).

Baud rate is the number of signal units (symbols or characters) transmitted per second.

Baud Rate vs. Bit Rate

1:1 or Simple Coding.

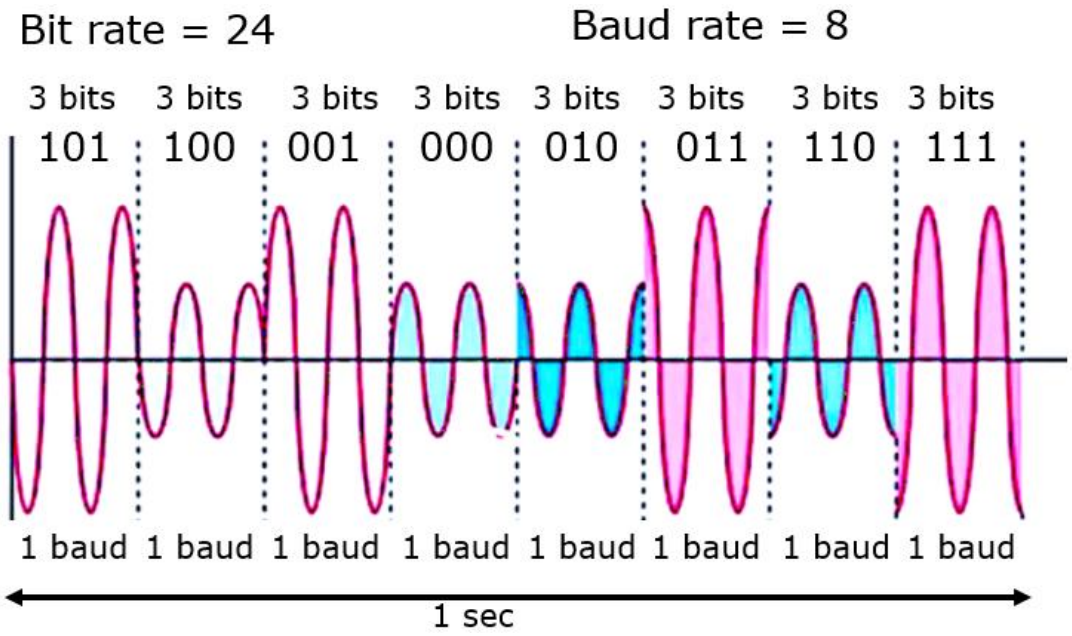
If 1 symbol = 1 bit. e.g. 9600 baud = 9600 bps. This simple coding would only send a one or a zero.

1:N or Complex Modulation.

If 1 symbol = 4 bits e.g. QAM 1000 baud requires 4000 bps. In this case, 4 bits would represent a more complex character. So, 1000 complex characters are sent at a rate of 4000 bits per second.

In the figure, 24 bits are sent every second. Bit Rate = 24

3 bits represent one character. Baud Rate = 8



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Digital Signalling Terms

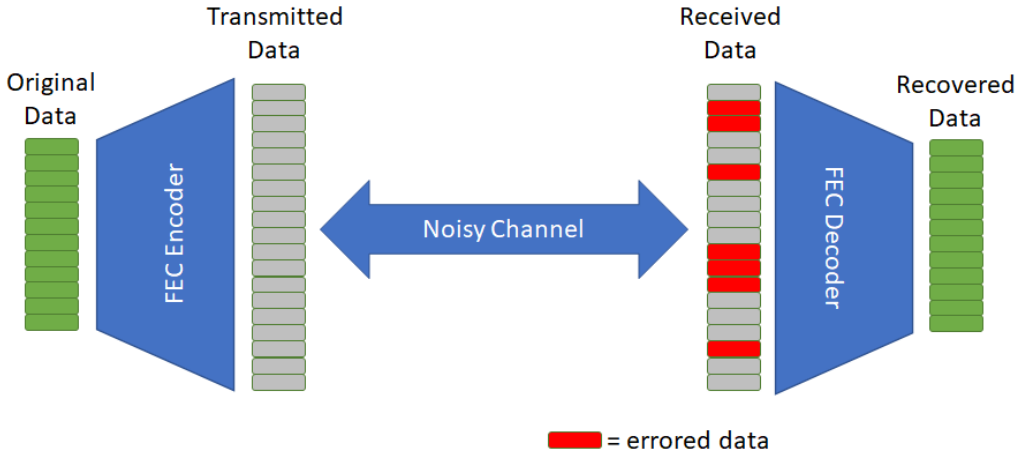
Cyclic Redundancy Check (CRC)

Cyclic Redundancy Check (CRC) is an error-detecting code used in network protocols to ensure data integrity. Used in protocols like TCP.

When a receiver detects a CRC mismatch, the packet is discarded and a request for retransmission is sent. This method ensures reliable transmission but can reduce throughput and increase latency.

Forward Error Correction (FEC)

FEC is a digital signal processing technique used to control errors in data transmission over noisy or unreliable communication channels. FEC adds error-correcting codes to the original data stream. The receiver can detect and correct a limited number of errors without needing to request retransmission.



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 and this data is referred to as the “Overhead”. The overhead may be substantial—up to 20% to 50% depending on the total payload bits sent over the channel.

The actual data being transferred is referred to as the “Payload”.

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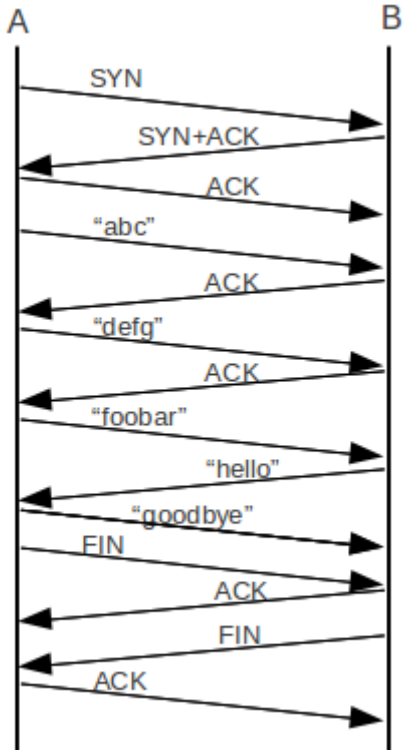
Communication Protocols

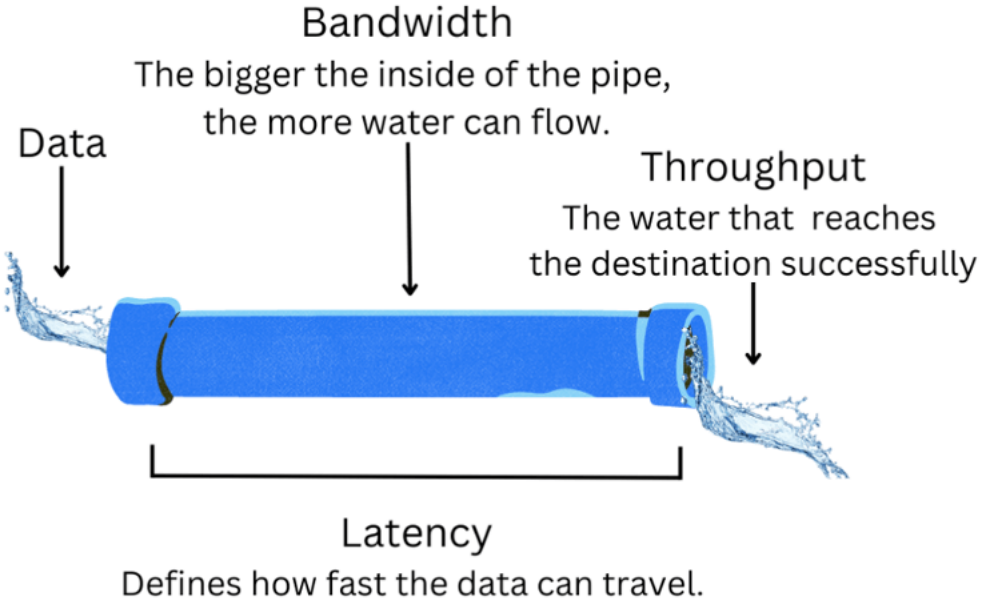
A communication protocol is a set of rules that enables two or more entities to exchange information. A protocol defines the rules, syntax, semantics, and synchronization of communication and possible error recovery methods.

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



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.



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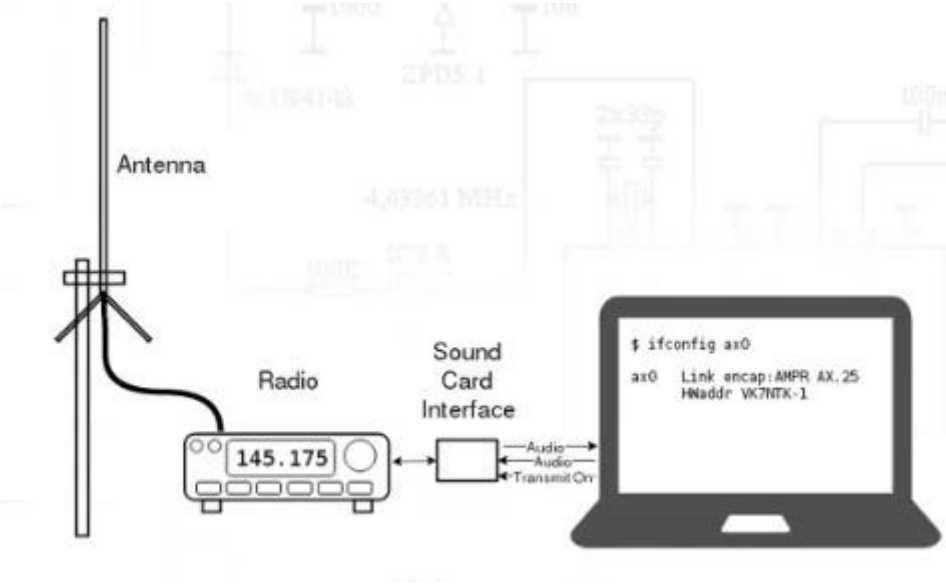
Packet Radio

Packet radio is a digital amateur radio mode, developed in the 1970s, that transmits data in small, error-corrected blocks (packets) rather than continuous streams.

Packet radio uses AX.25 protocols to transfer text messages, email, and GPS location data (APRS) across VHF/UHF/HF frequencies.

While traditionally requiring a Terminal Node Controller (TNC) to interface with a radio, many modern users use software modems like Dire Wolf.



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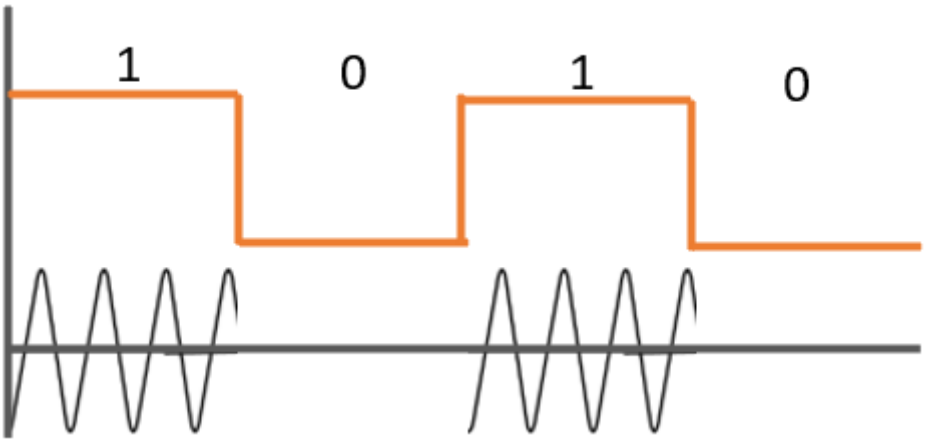
Digital Data Encoding

When modulating an analogue signal with digital data, there are there three parameters that can be varied. amplitude, frequency or phase.

Amplitude Shift Keying (ASK)

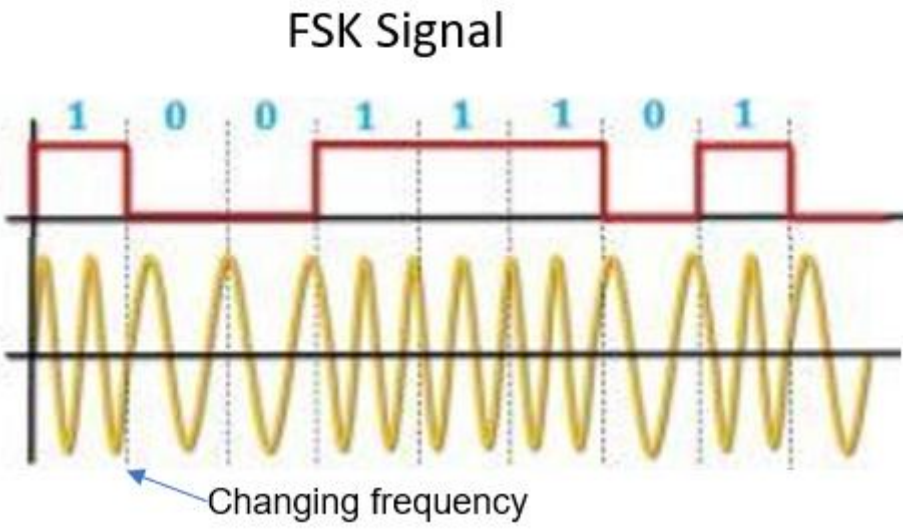
ASK represents binary data (0s and 1s) by varying the amplitude of a carrier wave.

Used in low-power applications like RFID, ASK often uses On-Off Keying (OOK), where a "1" is a high-amplitude signal and a "0" is no signal.



Frequency Shift Keying (FSK) AKA 2FSK

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.



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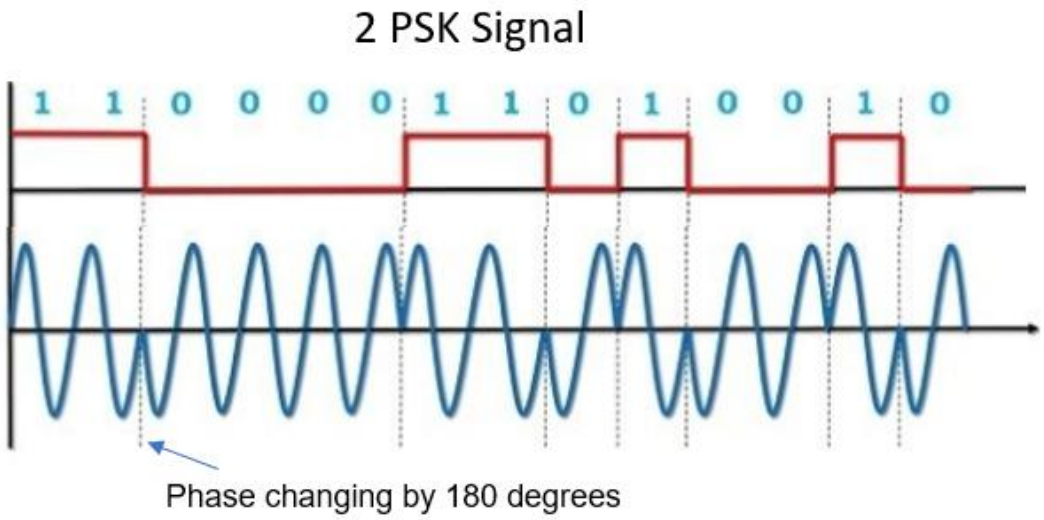
2-Phase Shift Keying (2-PSK)

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.

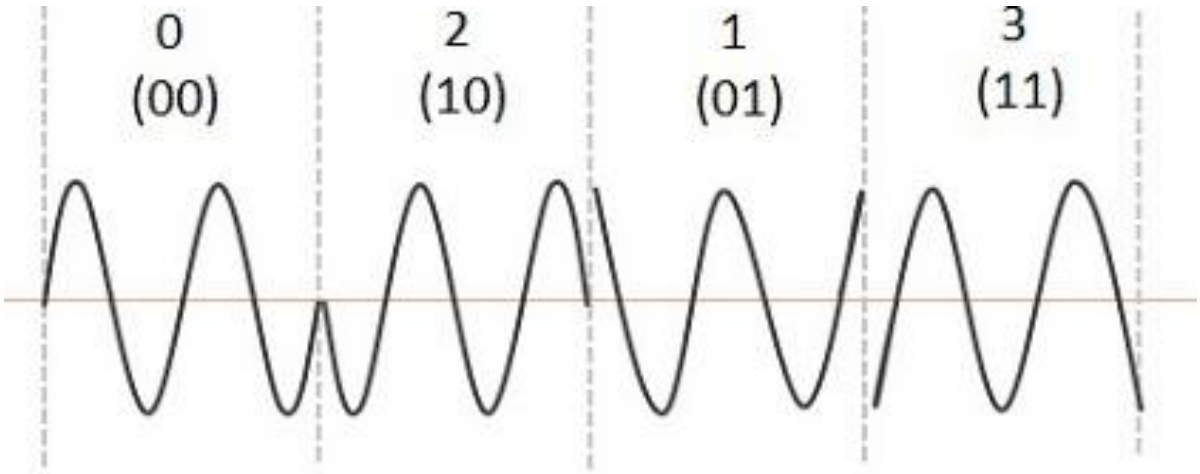


4 Phase Shift Keying (4- PSK)

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.



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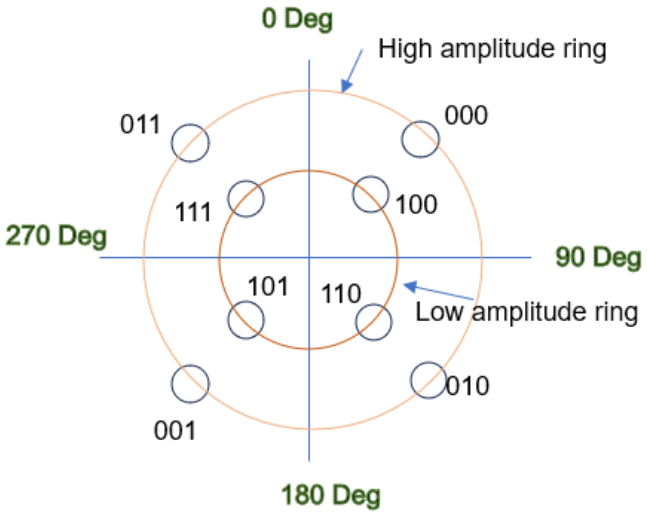
Quadrature Amplitude Modulation (QAM)

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

Example: To send the binary characters 101, the amplitude is low and the signal would be at 225 degrees.



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Amateur Radio Digital 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. Allows different agencies (police, fire, EMS) to communicate seamlessly, even with equipment from different vendors.

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 ("Phase Shift Keying, 31 Baud") 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

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.

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