LAB ASSIGNMENT No. 1

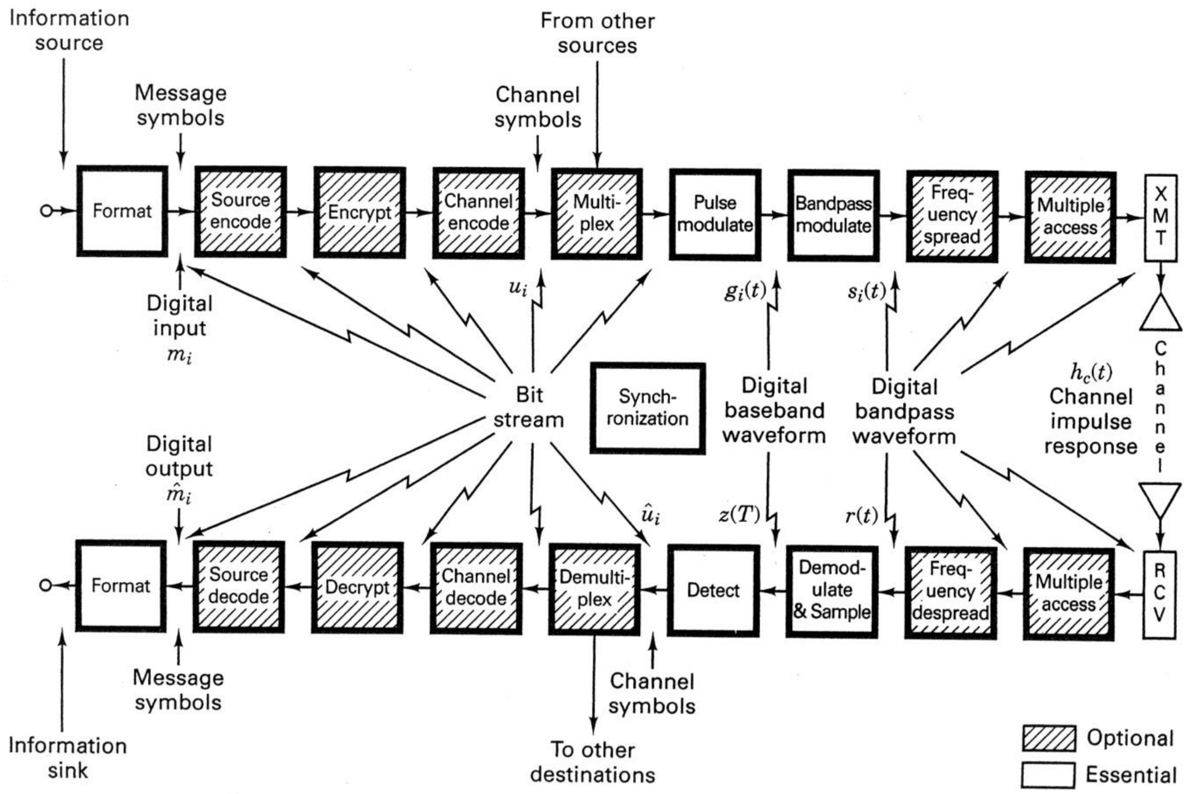
To Implement "Formatting" Of Digital Signal For Optical Fiber Communication

Digital communications is the physical transfer of data a digital bit stream over a point-to-point or point-to-multipoint communication channel. Examples of such channels are copper wires, optical fibers, wireless communication channels, storage media and computer buses. The data are represented as an electromagnetic signal, such as an electrical voltage, radio wave, microwave, or infrared signal.

While analog transmission is the transfer of a continuously varying analog signal, digital communications is the transfer of discrete messages. The messages are either represented by a sequence of pulses by means of a line code baseband transmission, or by a limited set of continuously varying wave forms pass band transmission, using a digital modulation method. The pass band modulation and corresponding demodulation also known as detection is carried out by modem equipment. According to the most common definition of digital signal, both baseband and pass band signals representing bit-streams are considered as digital transmission, while an alternative definition only considers the baseband signal as digital, and pass band transmission of digital data as a form of digital-to-analog conversion.

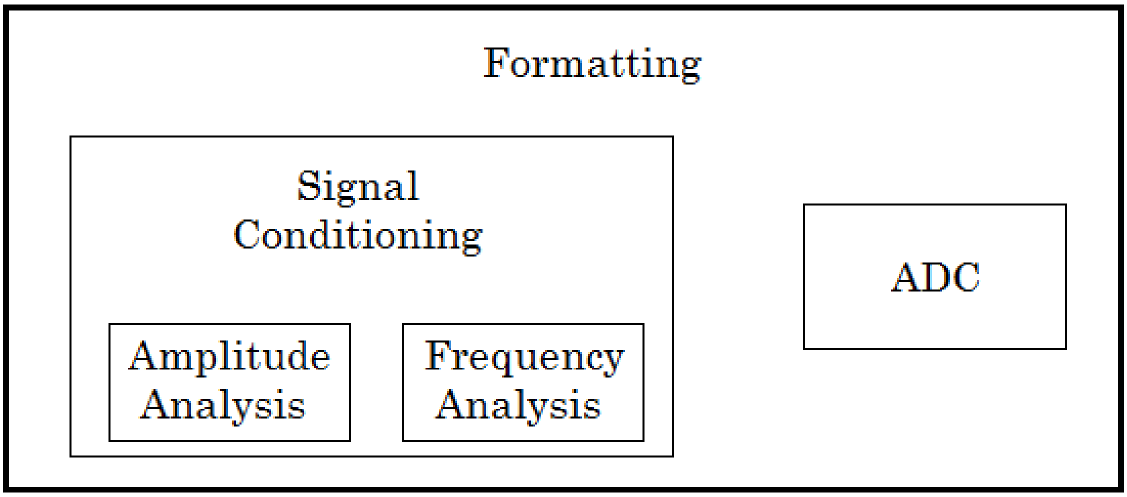
Data transmitted may be digital messages originating from a data source, for example a computer or a keyboard. It may also be an analog signal such as a phone call or a video signal, digitized into a bit-stream for example using pulse-code modulation (PCM) or more advanced source coding schemes.

Block Diagram of Digital Communication:



Formatting:

Digital Signal formatting is the process of transforming information from one format into another. This is often used in many digital devices and for communication processes. A digital system is a data technology that uses discrete values. By contrast, non-digital systems use a continuous range of values to represent information. Although digital representations are discrete, the information represented can be either discrete, such as numbers, letters or icons, or continuous, such as sounds, images, and other measurements of continuous systems.



Signal Conditioning:

In electronics, signal conditioning means manipulating an analog signal in such a way that it meets the requirements of the next stage for further processing. Most common use is in analog-to-digital converters.

It is common to have a sensing stage which consists of a sensor, a signal conditioning stage where usually amplification of the signal is done and a processing stage normally carried out by an ADC and a micro-controller.

In digital electronics, digital computers have taken a major role in near every aspect of life in our modern world. Digital electronics is at the heart of computers, but there are lots of direct applications of digital electronics in our world. All these digital electronics need data to be presented to them in a digital format i.e. the data have to be digitally conditioned. This is called digital conditioning. Since computers are electronics devices, all the information they work with has to be digitally formatted. Therefore, if they are used to control a variable such as temperature, then the temperature has to be represented digitally. That's why we need digital signal conditioning to condition process-control signal to be an approximated digital format.

Signal conditioning can include amplification, filtering, converting, range matching, isolation and any other processes required to make sensor output suitable for processing after conditioning.

Filtering:

Filtering is the most common signal conditioning function, as usually not all the signal frequency spectrum contains valid data. The common example are 60 Hz AC power lines, present in most environments, which will produce noise if amplified.

Amplifying:

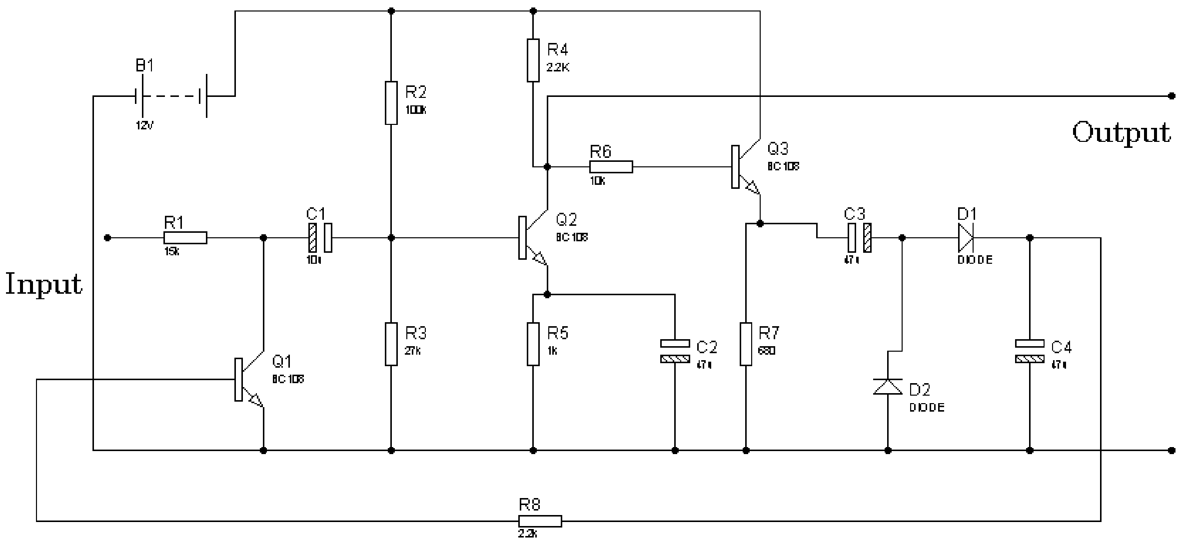
Signal amplification performs two important functions: increases the resolution of the input signal, and increases its signal-to-noise ratio. For example, the output of an electronic temperature sensor, which is probably in the millivolts range is probably too low for an Analog-to-digital converter (ADC) to process directly. In this case it is necessary to bring the voltage level up to that required by the ADC.

Auto Gain Control:

The actual signal amplifier stage is Q2, which operates in common emitter mode, the output signal being taken from its collector. A portions of the output signal is fed through emitter follower Q3 to a peak rectifier comprising D1/D2 and C4. The voltage on C4 is used to control the base current of Q1, which forms part of the input attenuator. At low signal levels the voltage on C4 is small and Q1 draws little current. As the input signal level increases the voltage on C4 rises and Q1 turns on mode, thus attenuating the input signal. The net result is that as the input signal increases it is subject to a greater and greater degree of attenuation and the output signal therefore remains fairly constant for a wide range of input levels.

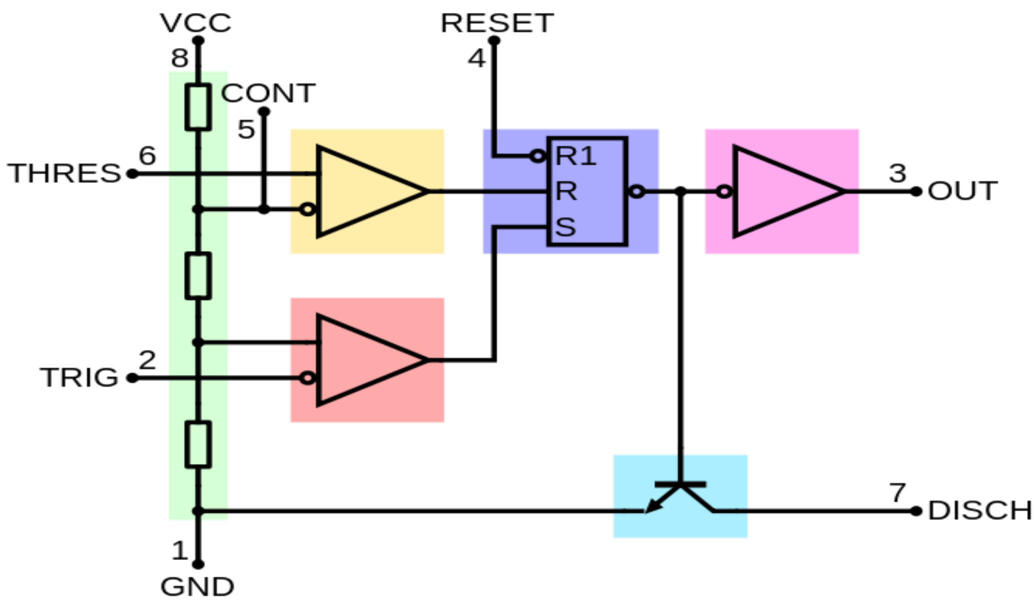
This preamplifier incorporates automatic gain control, which keeps the output level fairly constant over a wide range of input levels.

Automatic Gain Controller



Oscillator:

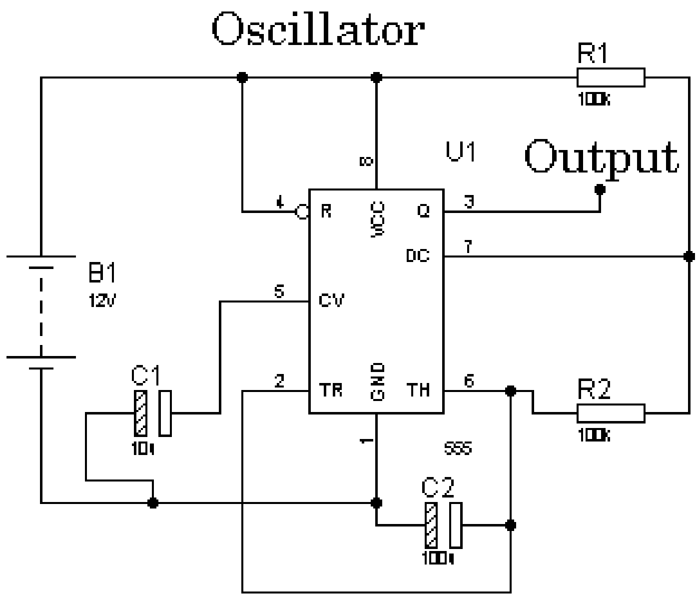
The 555 timer IC is an integrated circuit chip used in a variety of timer, pulse generation, and oscillator applications. The 555 can be used to provide time delays, as an oscillator, and as a flip-flop element. Derivatives provide up to four timing circuits in one package.



Astable (free-running) mode:

The 555 can operate as an oscillator. Uses include LED and lamp flashers, pulse generation, logic clocks, tone generation, security alarms, pulse position modulation and so on. The 555 can be used as a simple ADC, converting an analog value to a pulse length. e.g. selecting a thermistor as timing resistor allows the use of the 555 in a temperature sensor. The period of the output pulse is determined by the temperature. The use of a microprocessor based circuit can then convert the pulse period to temperature, linearize it and even provide calibration means.

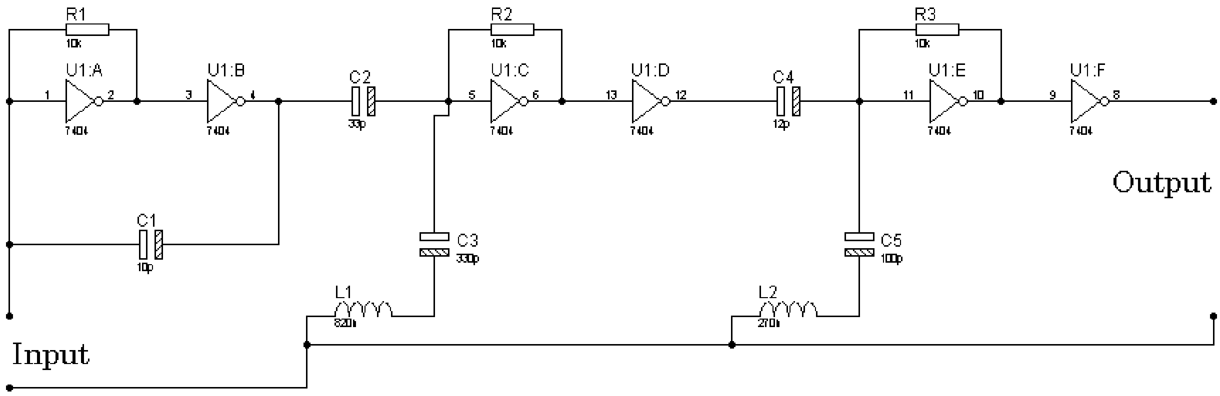
In astable mode, the 555 timer puts out a continuous stream of rectangular pulses having a specified frequency. Resistor R_1 is connected between V_{CC} and the discharge pin (pin 7) and another resistor (R_2) is connected between the discharge pin (pin 7), and the trigger (pin 2) and threshold (pin 6) pins that share a common node. Hence the capacitor is charged through R_1 and R_2, and discharged only through R_2, since pin 7 has low impedance to ground during output low intervals of the cycle, therefore discharging the capacitor.



Frequency Multiplier:

Whenever higher frequencies are required a frequency multiplier is placed after the oscillator. The resulting output signal is then a whole multiple of the frequency. Other frequency multipliers often use transistors, which produce harmonics due to their non-linearity. These are subsequently filtered from the signal. One way of doing this is to put a parallel L-C filter in the collector arm. This filter could then be tuned to three times the input frequency. This circuit contains only a single IC and a handful of passive components, and has a complete oscillator and two frequency triplers. The output is therefore a signal with a frequency that is 9 times as much as that of the crystal. Two gates from IC1, which contains six high-speed CMOS inverters, are used as an oscillator. This works at the fundamental frequency of the crystal and has a square wave at its output. A square wave can be considered as the sum of a fundamental sine wave plus an infinite number of odd multiples of that wave. The second stage has been tuned to the first odd multiple (3 x).

Frequency Multiplier

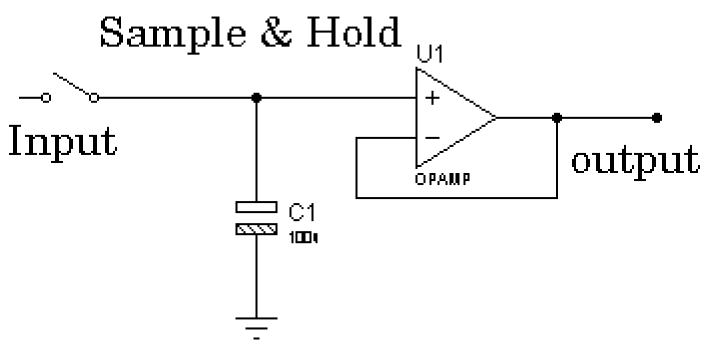


The filter used here is a band-rejection series LC type. Worse still, the rejection frequency is equal to the fundamental crystal frequency. The fundamental frequency is therefore attenuated, which is good. But the third harmonic is boosted by the smaller capacitor of 33 pF in combination with the inductor. Together they form the required band-pass filter. The same applies to the 12 pF capacitor in the next stage.

Through the careful selection of components, this filter is therefore capable of rejecting the fundamental and boosting the third harmonic.

Sample and Hold Circuit:

If we want to measure a signal, we cannot ignore the fact that it changes with time. There are several approaches to dealing with this. We could measure the signal very fast, do it many times, and after we're done figure out what the right time scale would have been. We could average the signal. Or we can snatch the value of the signal, hold that value steady, and digitize the single, sampled value at our leisure. A sample and hold circuit can be employed with digitizers to pluck a single value from an analog source, keep that value stable for at least the time required for digitization, and can then be set to grab a value at a later time. Consider the following circuit:



If the switch is closed, the capacitor is charged to the potential V_{in}. When the switch opens, the capacitor holds its charge, so the follower op amp has an output potential of what potential was on the capacitor at the moment the switch opened. If the switch is typically closed, but opens momentarily when we want to make a measurement, this is a track and hold circuit. If we typically leave the switch open, but momentarily close it when we want to grab a measurement, it is a sample and hold circuit.

Analog-To-Digital Converter:

An analog-to-digital converter is a device that converts a continuous physical quantity usually voltage to a digital number that represents the quantity's amplitude.

The conversion involves quantization of the input, so it necessarily introduces a small amount of error. Instead of doing a single conversion, an ADC often performs the conversions periodically. The result is a sequence of digital values that have been converted from a continuous-time and continuous-amplitude analog signal to a discrete-time and discrete-amplitude digital signal.

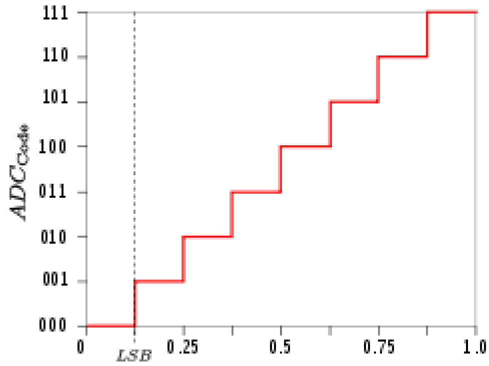
An ADC is defined by its bandwidth the range of frequencies it can measure and its signal to noise ratio how accurately it can measure a signal relative to the noise it introduces. The actual bandwidth of an ADC is characterized primarily by its sampling rate, and to a lesser extent by how it handles errors such as aliasing. The dynamic range of an ADC is influenced by many factors, including the resolution the number of output levels it can quantize a signal to, linearity and accuracy how well the quantization levels match the true analog signal and jitter small timing errors that introduce additional noise.

ADCs are chosen to match the bandwidth and required signal to noise ratio of the signal to be quantized. If an ADC operates at a sampling rate greater than twice the bandwidth of the signal, then perfect reconstruction is possible given an ideal ADC and neglecting quantization error.

The presence of quantization error limits the dynamic range of even an ideal ADC, however, if the dynamic range of the ADC exceeds that of the input signal, its effects may be neglected resulting in an essentially perfect digital representation of the input signal.

Resolution:

The resolution of the converter indicates the number of discrete values it can produce over the range of analog values. The resolution determines the magnitude of the quantization error and therefore determines the maximum possible average signal to noise ratio for an ideal ADC without the use of oversampling. The values are usually stored electronically in binary form, so the resolution is usually expressed in bits.



In consequence, the number of discrete values available, or "levels", is assumed to be a power of two. For example, an ADC with a resolution of 8 bits can encode an analog input to one in 256 different levels, since $2^8 = 256$. The values can represent the ranges from 0 to 255 i.e. unsigned integer or from −128 to 127 i.e. signed integer, depending on the application.

Resolution can also be defined electrically, and expressed in volts. The minimum change in voltage required to guarantee a change in the output code level is called the least significant bit (LSB) voltage. The resolution Q of the ADC is equal to the LSB voltage.

Sampling rate:

The analog signal is continuous in time and it is necessary to convert this to a flow of digital values. It is therefore required to define the rate at which new digital values are sampled from the analog signal. The rate of new values is called the sampling rate or sampling frequency of the converter.

A continuously varying band limited signal can be sampled that is, the signal values at intervals of time T, the sampling time, are measured and stored and then the original signal can be exactly reproduced from the discrete-time values. The accuracy is limited by quantization error. However, this faithful reproduction is only possible if the sampling rate is higher than twice the highest frequency of the signal. This is essentially what is embodied in the Shannon-Nyquist sampling theorem.

Since a practical ADC cannot make an instantaneous conversion, the input value must necessarily be held constant during the time that the converter performs a conversion called the conversion time. An input circuit called a sample and hold performs this task—in most cases by using a capacitor to store the analog voltage at the input, and using an electronic switch or gate to disconnect the capacitor from the input. Many ADC integrated circuits include the sample and hold subsystem internally.

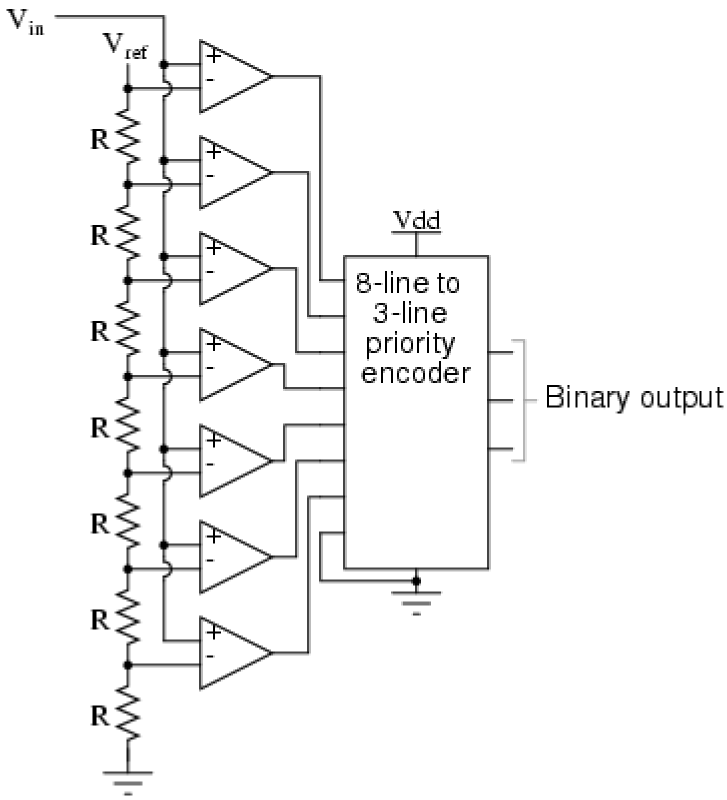
Aliasing:

An ADC works by sampling the value of the input at discrete intervals in time. Provided that the input is sampled above the Nyquist rate, defined as twice the highest frequency of interest, then all frequencies in the signal can be reconstructed. If frequencies above half the Nyquist rate are sampled, they are incorrectly detected as lower frequencies, a process referred to as aliasing.

Aliasing occurs because instantaneously sampling a function at two or fewer times per cycle results in missed cycles, and therefore the appearance of an incorrectly lower frequency. For example, a 2 kHz sine wave being sampled at 1.5 kHz would be reconstructed as a 500 Hz sine wave.

Flash ADC:

Flash ADC has a bank of comparators sampling the input signal in parallel, each firing for their decoded voltage range. The comparator bank feeds a logic circuit that generates a code for each voltage range. Direct conversion is very fast, capable of gigahertz sampling rates, but usually has only 8 bits of resolution or fewer, since the number of comparators needed, $2^N - 1$, doubles with each additional bit, requiring a large, expensive circuit.



V_{ref} is a stable reference voltage provided by a precision voltage regulator as part of the converter circuit, not shown in the schematic. As the analog input voltage exceeds the reference voltage at each comparator, the comparator outputs will sequentially saturate to a high state. The priority encoder generates a binary number based on the highest-order active input, ignoring all other active inputs.

Conclusions:

Formatting of signal involves signal conditioning and conversion of analog signal into digital signal. In signal conditioning amplitude analysis and frequency analysis is performed. Amplitude of input signal coming from any sensor usually not suitable for next stage, sometimes amplitude of source signal is very low hence we need to amplify this signal, similarly sometimes source signal is very high, to overcome these over gain and under gain problems automatic gain controller is designed to maintain the amplitude of the signal. Frequency analysis is performed for two reasons, to find the fundamental frequency of input circuit and to design oscillator circuit required for sampling in ADC. Using fast Fourier analysis we find the fundamental frequency. Sampling of source signal is done at frequency at least double than the source signal. For better and more accurate digital signal sampling is done at frequency 10 times greater than the source signal. For this reason variable oscillators and frequency multipliers are used.

References:

http://www.ukessays.com/essays/engineering/digital-signal-formatting.php#ixzz30O4MASbZ

http://en.wikipedia.org/wiki/Digital_signal_conditioning

http://en.wikipedia.org/wiki/Signal_conditioning

LAB ASSIGNMENT No. 2

To Implement "PCM" Of Digital Signal

Non-Return to Zero:

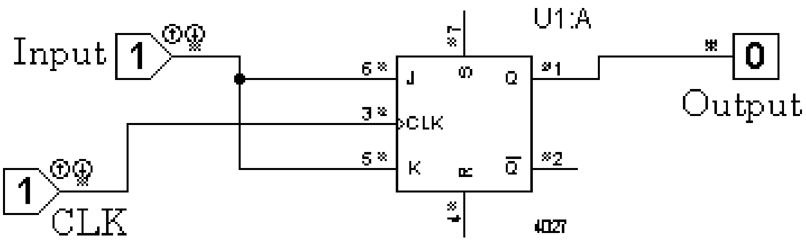
A non-return-to-zero (NRZ) line code is a binary code in which 1s are represented by one significant condition usually a positive voltage and 0s are represented by some other significant condition usually a negative voltage, with no other neutral or rest condition. The pulses have more energy than a return-to-zero (RZ) code. Unlike RZ, NRZ does not have a rest state.

Non Return to Zero Mark (NRZ-M) is a method of mapping a binary signal to a physical signal for transmission over some transmission media. The two level signal has a transition at a clock boundary if the bit being transmitted is a logical 1, and does not have a transition if the bit being transmitted is a logical 0.

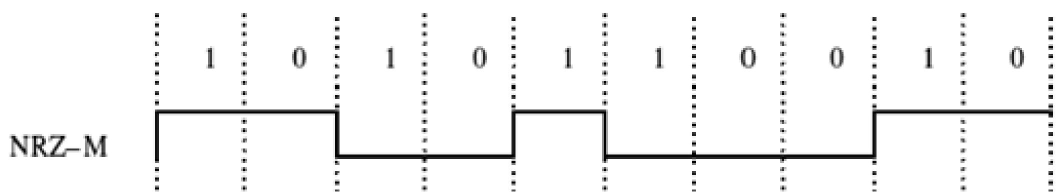
"One" is represented by a transition of the physical level.

"Zero" has no transition.

Circuit Diagram:



Output Wave:



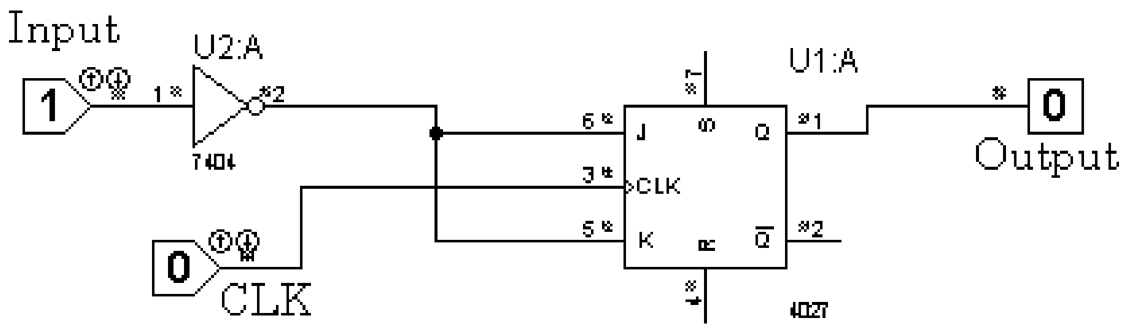
Non Return to Zero Space:

A binary encoding scheme in which a signal parameter, such as electric current or voltage, undergoes a change in a significant condition or level every time that a "zero" occurs, but when a "one" occurs, it remains the same, i.e., no transition occurs.

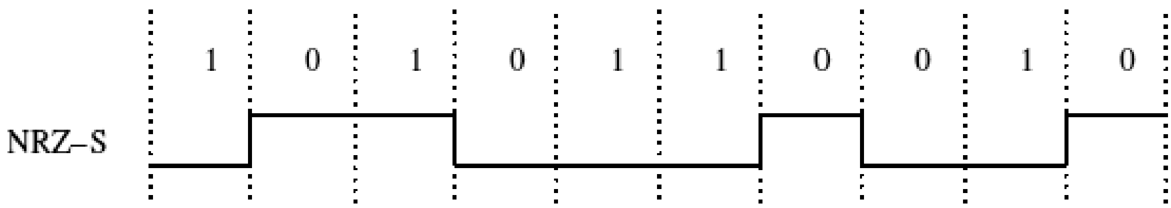
"One" is represented by no change in physical level.

"Zero" is represented by a change in physical level.

Circuit Diagram:



Output Wave:

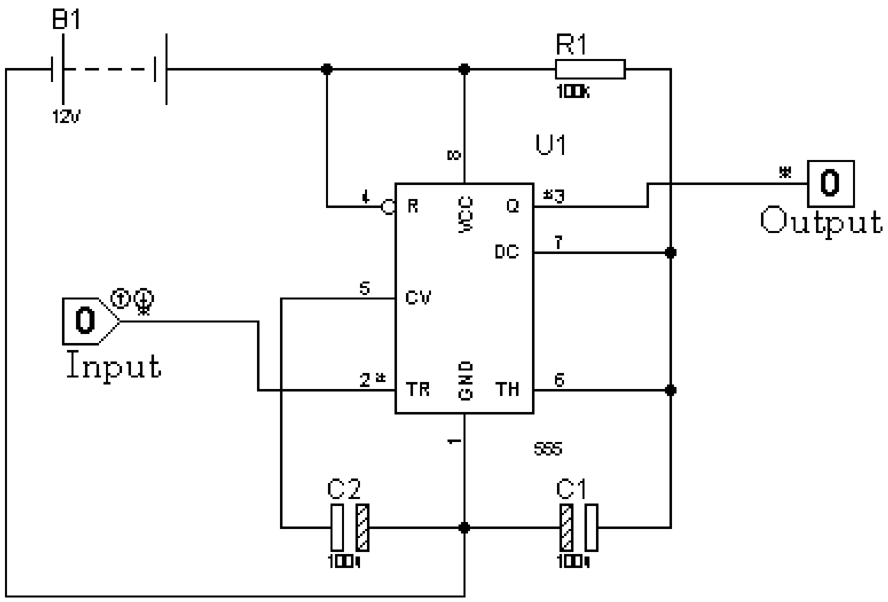


Unipolar Return-to-Zero:

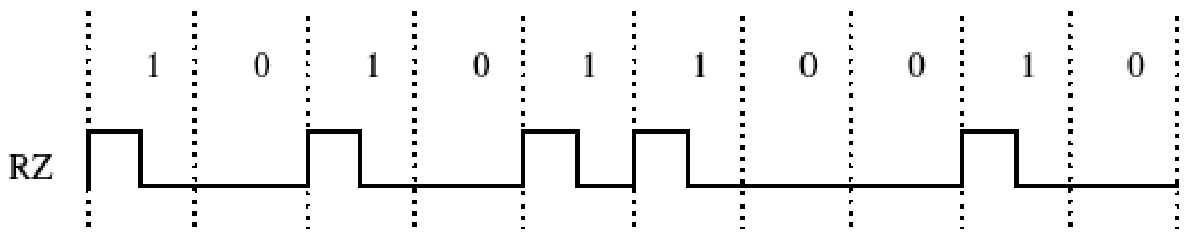
Unipolar encoding is a line code. A positive voltage represents a binary 1, and zero volts indicates a binary 0. It is the simplest line code, directly encoding the bit stream, and is analogous to on-off keying in modulation.

Its drawbacks are that it is not self-clocking and it has a significant DC component, which can be halved by using return-to-zero, where the signal returns to zero in the middle of the bit period. With a 50% duty cycle each rectangular pulse is only at a positive voltage for half of the bit period. This is ideal if one symbol is sent much more often than the other and power considerations are necessary, and also makes the signal self-clocking.

Circuit Diagram:



Output Wave:

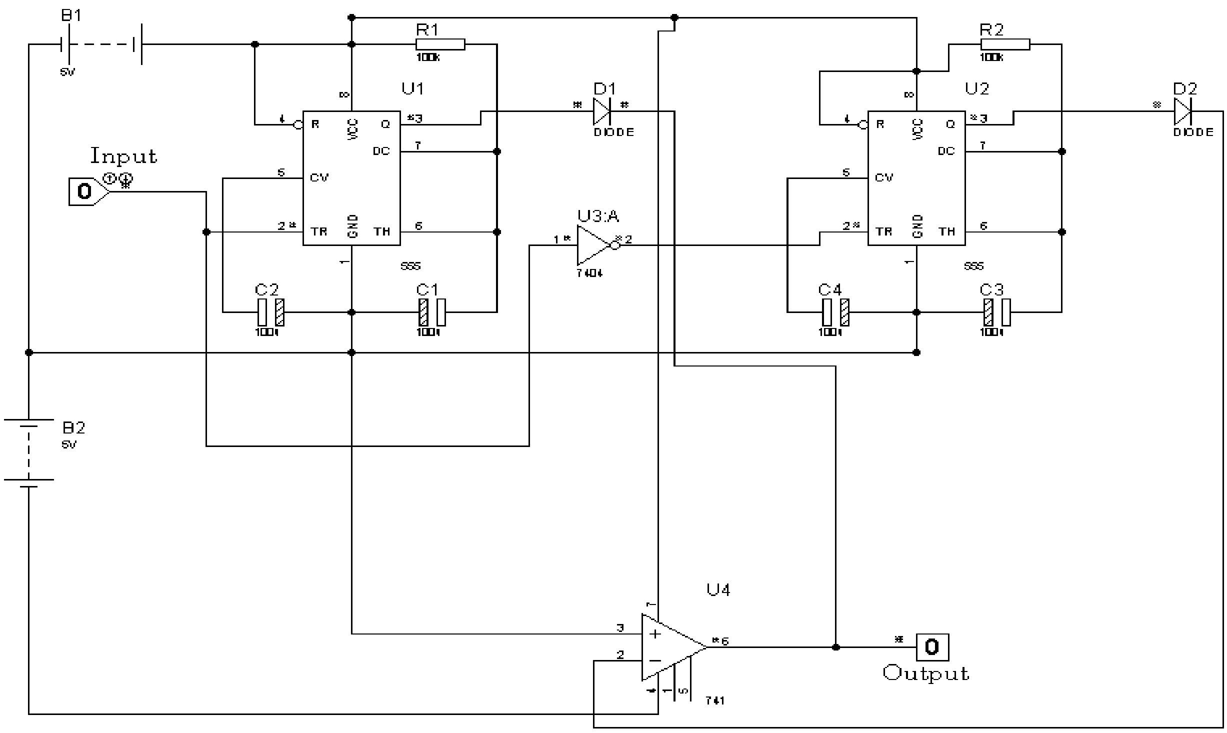


Bipolar Return-to-zero

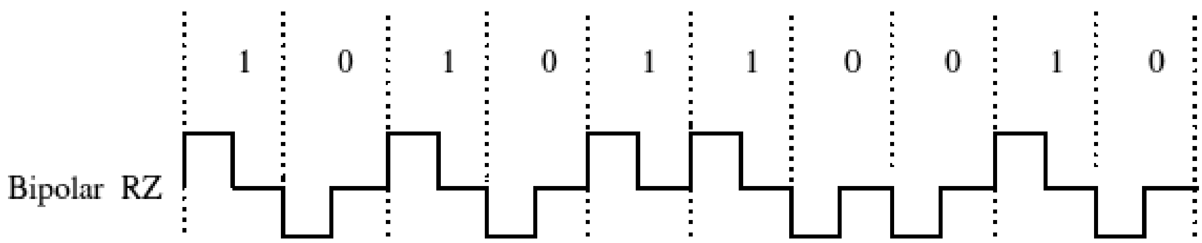
Bipolar Return-to-zero (RZ) describes a line code used in telecommunications signals in which the signal drops to zero between each pulse. This takes place even if a number of consecutive 0s or 1s occur in the signal.

That "zero" condition is typically halfway between the significant condition representing a 1 bit and the other significant condition representing a 0 bit.

Circuit Diagram:



Output Wave:



Bi-ø-L:

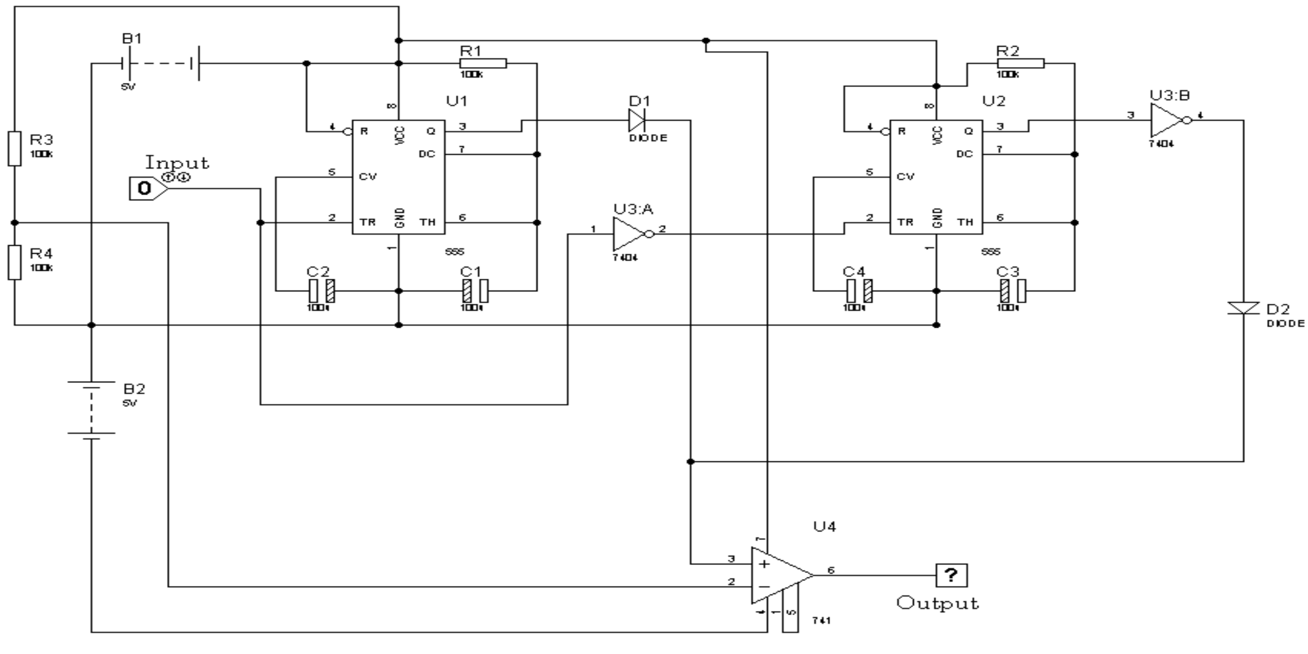
One form of encoding asynchronous binary data is BiPhase encoding. Given that a message consists of '0's and '1's, and that two different types of signal can be transmitted 'L' and 'H', bi-phase encoding consists of transmitting either 'LH' or 'HL' for every bit. The name comes from the fact that every bit period looks like one cycle of a digitized sine wave, either with phase 0 (HL) or phase π (LH). The level always changes in the middle of the pulse.

1: level changes from high to low. 0: level changes from low to high.

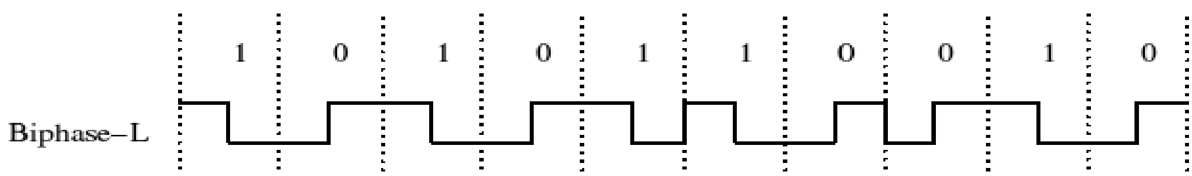
Level change occurs at the beginning of every bit period.

"One" is represented by a "One" level with transition to the "Zero" level "Zero" is represented by a "Zero" level with transition to the "One" level.

Circuit Diagram:



Output Wave:



Bi-Phase-S:

This is the same as Bi-Phase-L but with the logic levels reversed.

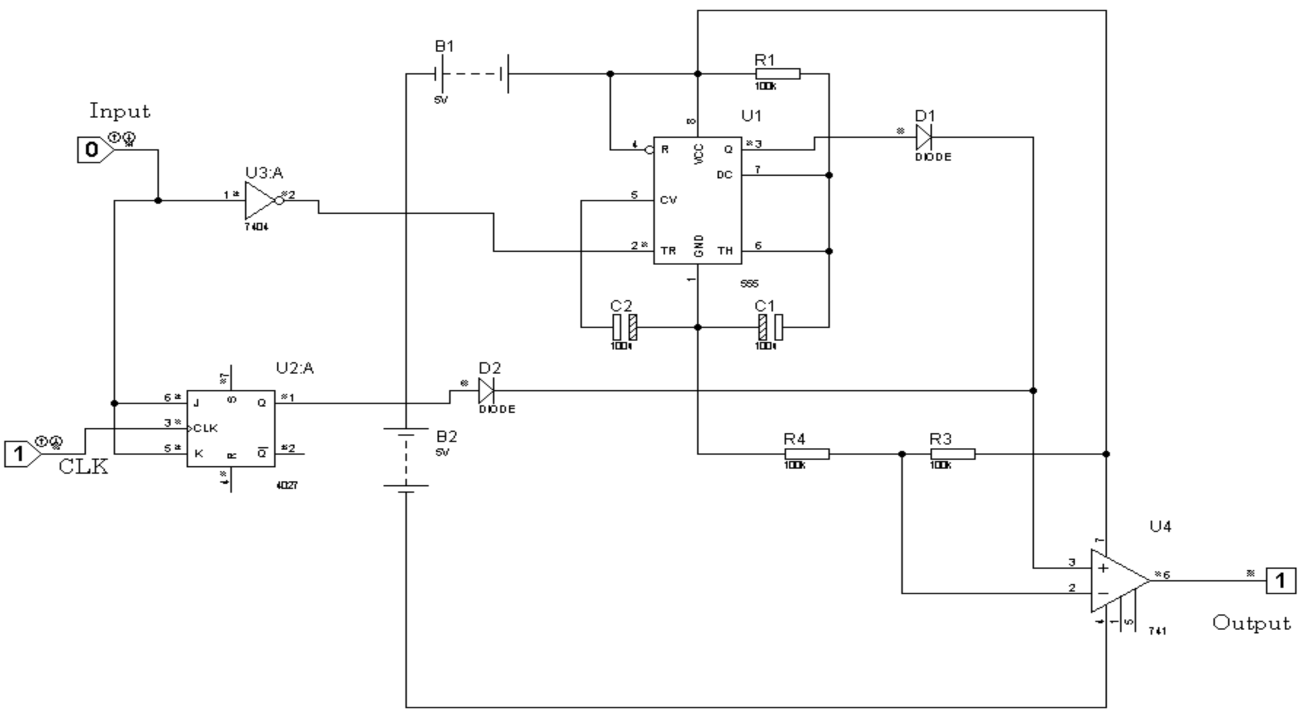
1: no level change in middle of pulse. 0: level changes in the middle of the pulse.

Level change occurs at the beginning of every bit period.

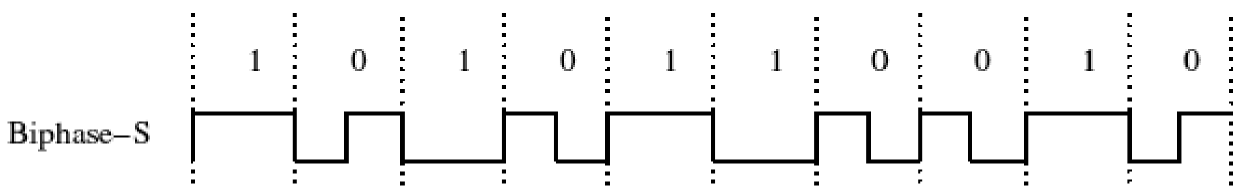
"One" is represented by no midbit level change.

"Zero" is represented by a midbit level change.

Circuit Diagram:



Output Wave:



References:

http://everything2.com/title/Bi-Phase+encoding

http://en.wikipedia.org/wiki/Differential_Manchester_encoding

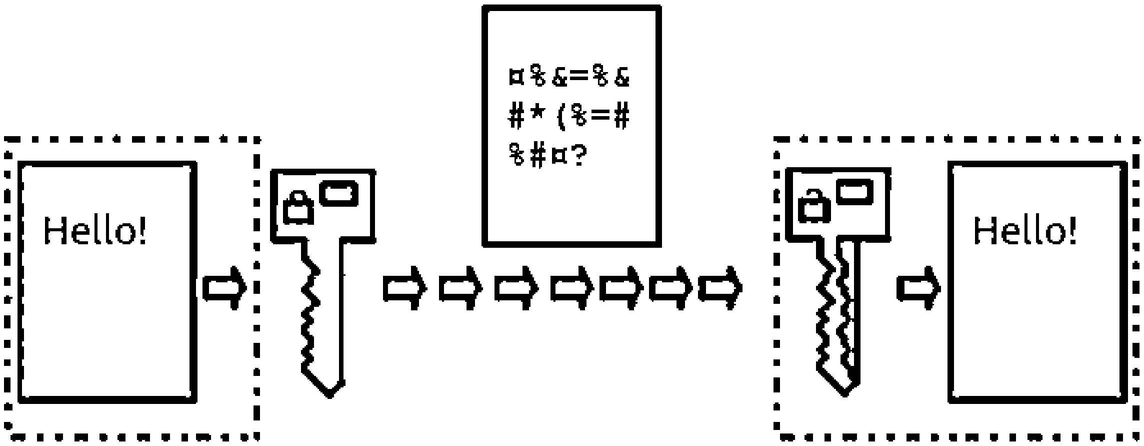
http://blog.oureducation.in/pulse-code-modulation-concepts/

http://ckp.made-it.com/encodingschemes.html

LAB ASSIGNMENT No. 3

To Perform Encryption of Data Signal

Encryption is the process of encoding messages or information in such a way that only authorized parties can read it. Encryption does not of itself prevent interception, but denies the message content to the interceptor. In an encryption scheme, the message or information, referred to as plaintext, is encrypted using an encryption algorithm, generating cipher text that can only be read if decrypted. For technical reasons, an encryption scheme usually uses a pseudo-random encryption key generated by an algorithm. It is in principle possible to decrypt the message without possessing the key, but, for a well-designed encryption scheme, large computational resources and skill are required. An authorised recipient can easily decrypt the message with the key, provided by the originator to recipients but not to unauthorised interceptors.

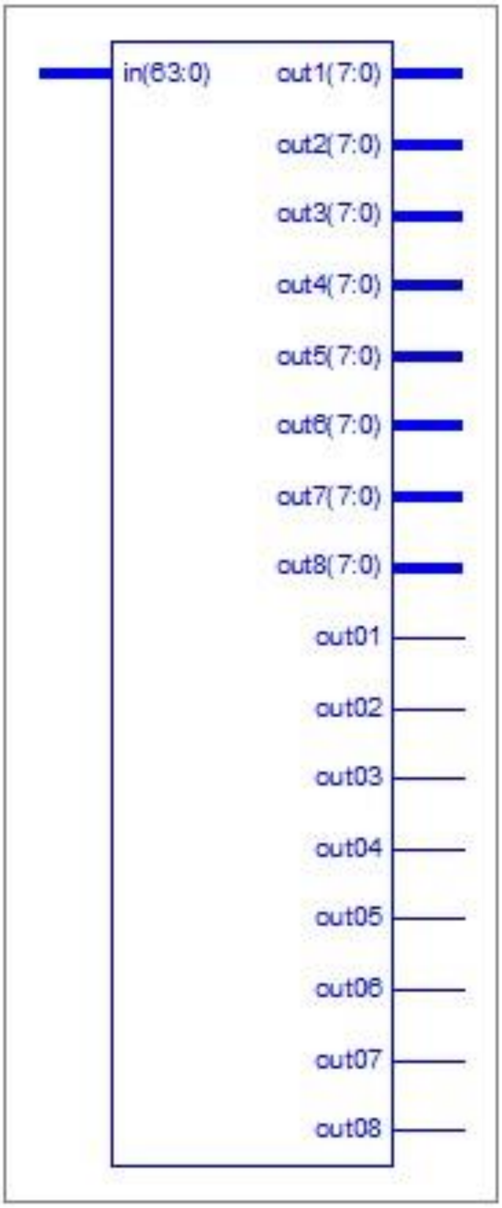


Chunks:

In this part, Larger message is distributed into small parts, e.g. message data is 16 bit longer 0001001000110100 it will be divided into 4 parts of 4 bits. i.e. 0001 - 0010 - 0011 - 0100

```
modulechunks(out1,out2,out3,out4,out5,out6,out7,out8,out01,out02,out03,ou
t04,out05,out06,out07,out08,in);
output [7:0] out1,out2,out3,out4,out5,out6,out7,out8;
output out01,out02,out03,out04,out05,out06,out07,out08;
input [63:0] in;
reg  [7:0] out1,out2,out3,out4,out5,out6,out7,out8;
reg  out01,out02,out03,out04,out05,out06,out07,out08;
always @(in)
begin
out1 = in[63:56];
out2 = in[55:48];
out3 = in[47:40];
out4 = in[39:32];
out5 = in[31:24];
out6 = in[23:16];
out7 = in[15:8];
out8 = in[7:0];

out01 = 1;
out02 = 2;
out03 = 3;
out04 = 4;
```



```
out05 = 5;
out06 = 6;
out07 = 7;
out08 = 8;
end
endmodule
```

Transformation:

In this section every part of message is encrypted. 1st part is encrypted as its all bits are inverted instead of 1st bit, 2nd part is encrypted as its all bits are inverted instead of 2nd bit, and similarly every part is encrypted. e.g. encrypted code of above example is

Data	0	0	0	1		0	0	1	0		0	0	1	1		0	1	0	0
Encryption	0	1	1	0		1	0	0	1		1	1	1	0		1	0	1	0

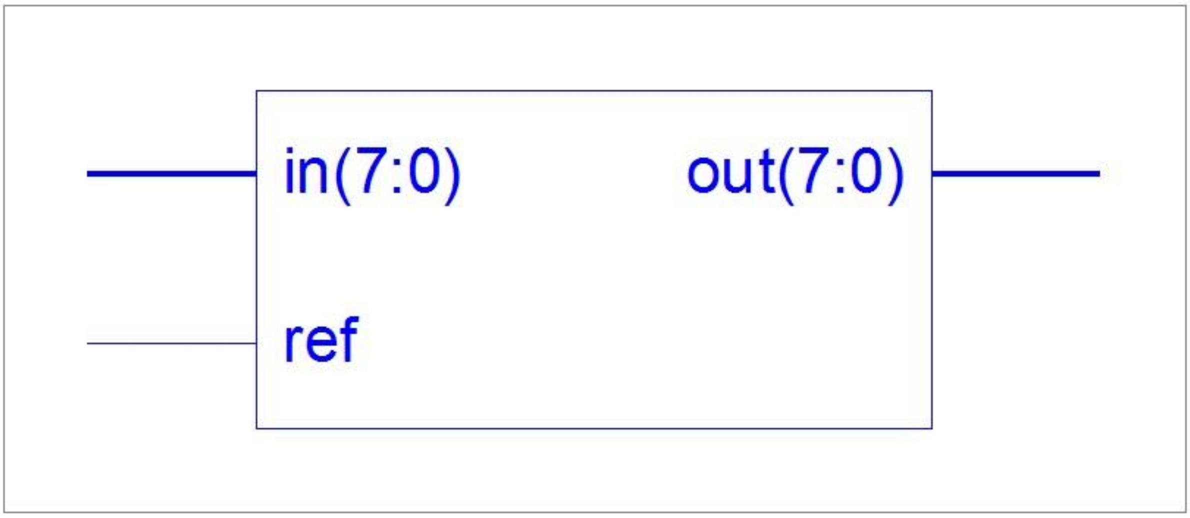
```
module Transformer(out,in,ref);
output [7:0]out;
input ref;
input [7:0]in;
reg [7:0]out;
always @(in,ref)
begin
```

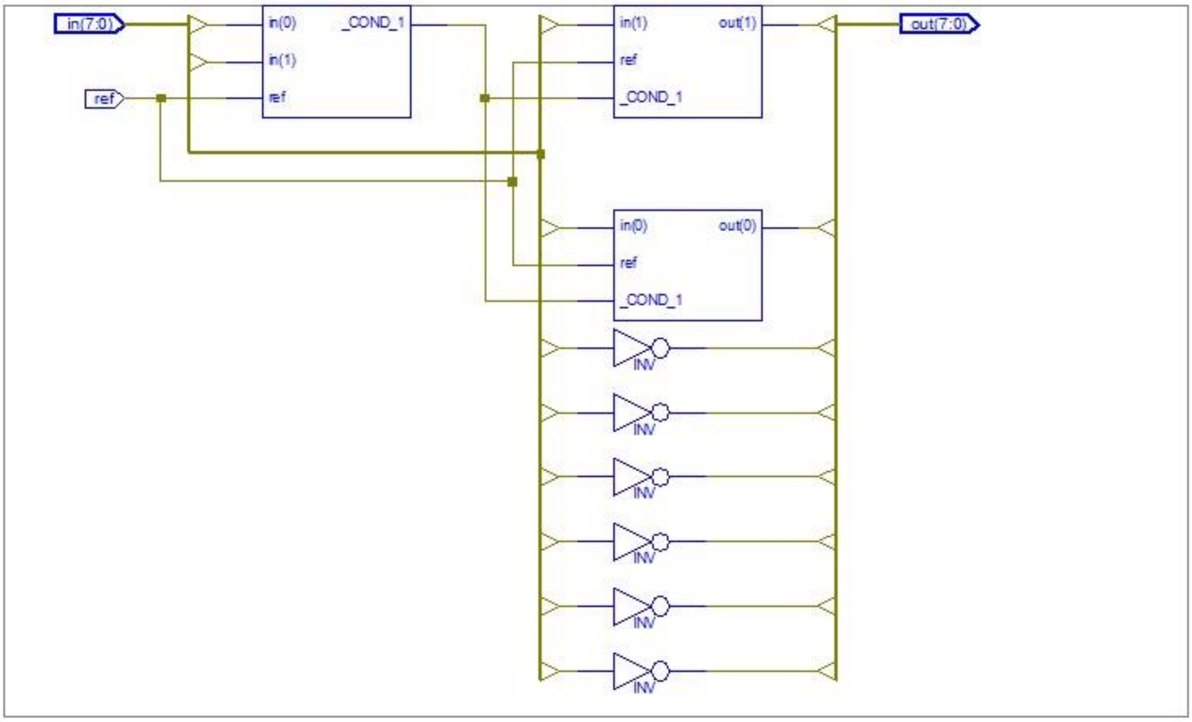
```
out = ~in;

out[ref] = ~in[ref];

end

endmodule
```



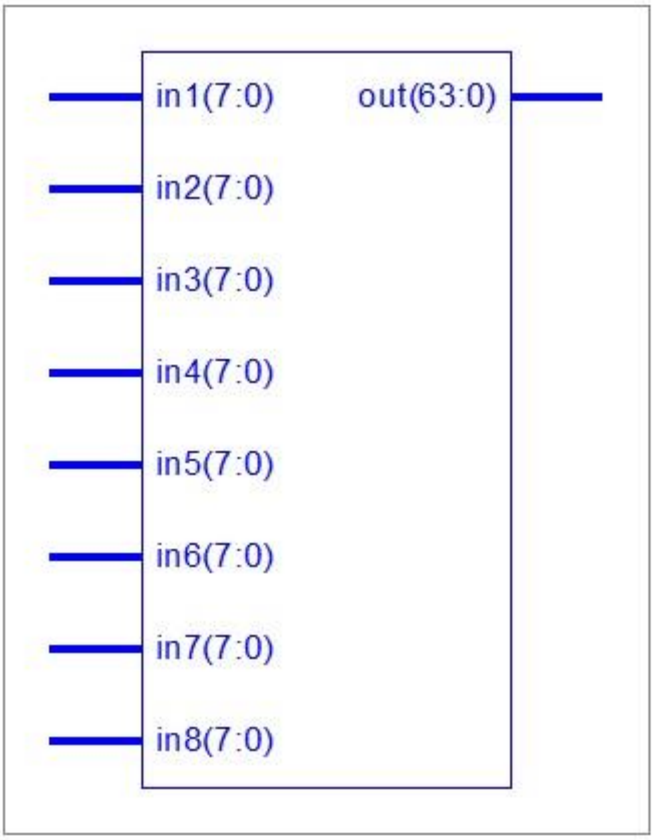


Scrambler:

In this section, every encrypted part is replaced by other part. If we take above example.

Code	0	1	1	0	x	1	0	0	1	-	1	1	1	0	x	1	0	1	0
Replacement	1	0	0	1		0	1	1	0		1	0	1	0		1	1	1	0

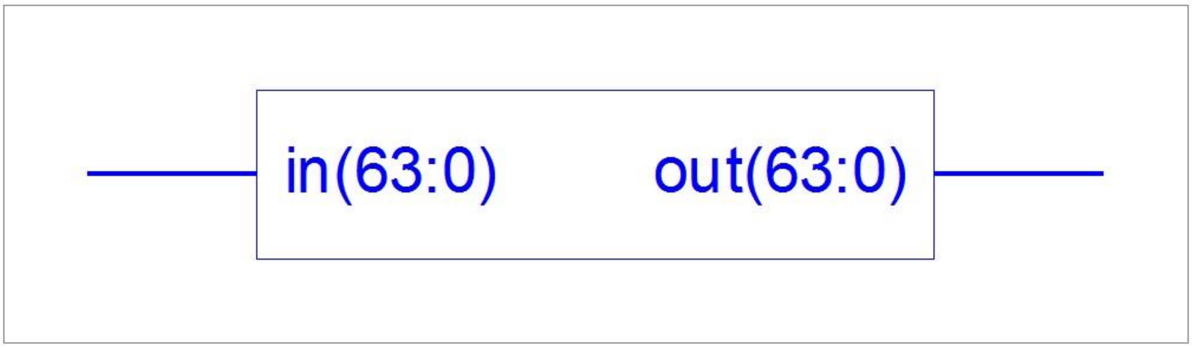
```
module scrambler(out,in1,in2,in3,in4,in5,in6,in7,in8);
output [63:0]out;
input [7:0]in1,in2,in3,in4,in5,in6,in7,in8;
reg [63:0]out;
always @(in1,in2,in3,in4,in5,in6,in7,in8)
begin
out = {in2,in1,in4,in3,in6,in5,in8,in7};
end
endmodule
```

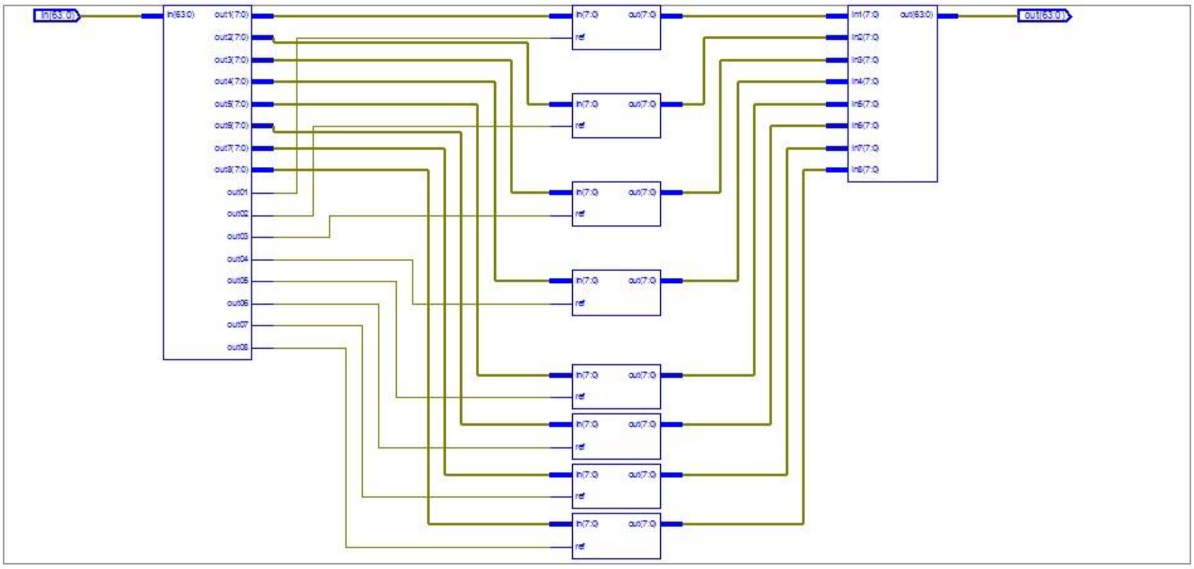


```
module encrypt(out,in);
output [63:0]out;
input [63:0]in;
wire [7:0] w1,w2,w3,w4,w5,w6,w7,w8;
wire [7:0] wi1,wi2,wi3,wi4,wi5,wi6,wi7,wi8;
wire w01,w02,w03,w04,w05,w06,w07,w08;

always @(in);

chunks
m(w1,w2,w3,w4,w5,w6,w7,w8,w01,w02,w03,w04,w05,w06,w07,w08,in);
Transformer m1(wi1,w1,w01);
Transformer m2(wi2,w2,w02);
Transformer m3(wi3,w3,w03);
Transformer m4(wi4,w4,w04);
Transformer m5(wi5,w5,w05);
Transformer m6(wi6,w6,w06);
Transformer m7(wi7,w7,w07);
Transformer m8(wi8,w8,w08);
scrambler m9(out,wi1,wi2,wi3,wi4,wi5,wi6,wi7,wi8);
endmodule
```


in(63:0)
out(63:0)



LAB ASSIGNMENT No. 4

To Perform Digital Modulation Technique FSK

In electronics and telecommunications, modulation is the process of varying one or more properties of a periodic waveform, called the carrier signal i.e high frequency signal, with a modulating signal that typically contains information to be transmitted.

A modulator is a device that performs modulation. A demodulator sometimes detector or de-mod is a device that performs demodulation, the inverse of modulation. A modem from modulator demodulator can perform both operations.

The aim of digital modulation is to transfer a digital bit stream over an analog band pass channel, for example over the public switched telephone network where a band pass filter limits the frequency range to 300–3400 Hz, or over a limited radio frequency band.

In digital modulation, an analog carrier signal is modulated by a discrete signal. Digital modulation methods can be considered as digital-to-analog conversion, and the corresponding demodulation or detection as analog-to-digital conversion.

Fundamental digital modulation methods:

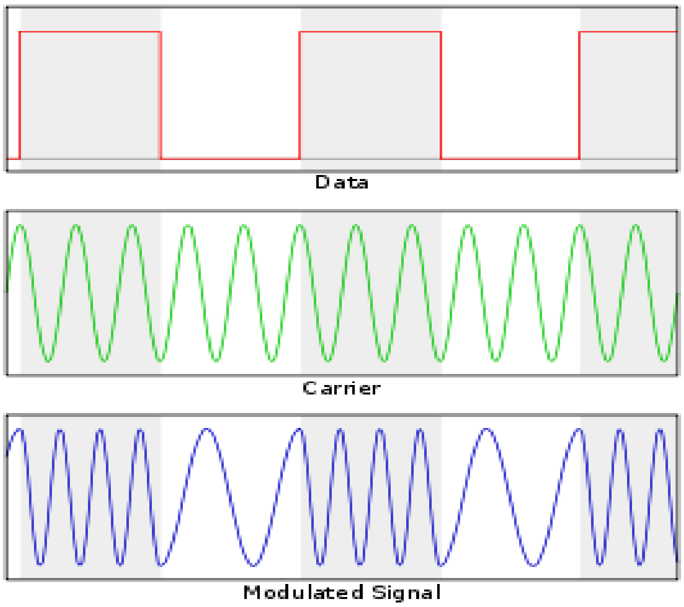
The most fundamental digital modulation techniques are based on keying:

- **PSK** (phase-shift keying): a finite number of phases are used.
- **FSK** (frequency-shift keying): a finite number of frequencies are used.
- **ASK** (amplitude-shift keying): a finite number of amplitudes are used.
- **QAM** (quadrature amplitude modulation): a finite number of at least two phases and at least two amplitudes are used.

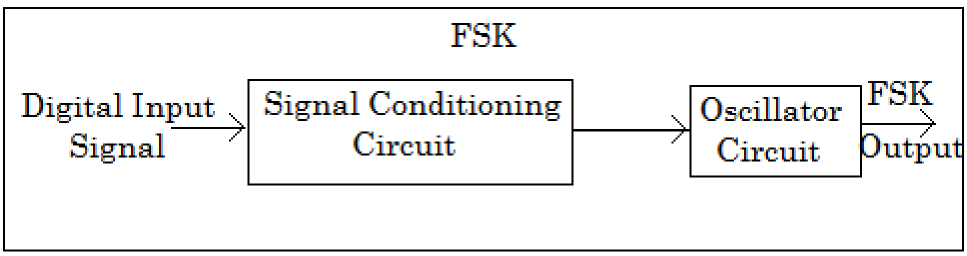
Frequency-shift keying:

Frequency-shift keying (FSK) is a frequency modulation scheme in which digital information is transmitted through discrete frequency changes of a carrier wave. The simplest FSK is binary FSK (BFSK).

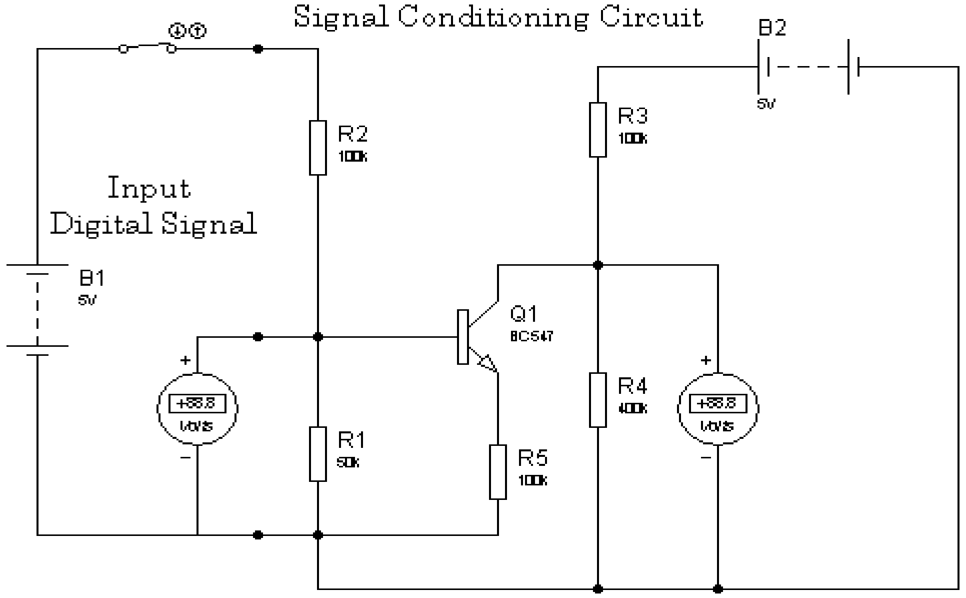
BFSK uses a pair of discrete frequencies to transmit binary 0s and 1s information. With this scheme, the "1" is called the mark frequency and the "0" is called the space frequency.

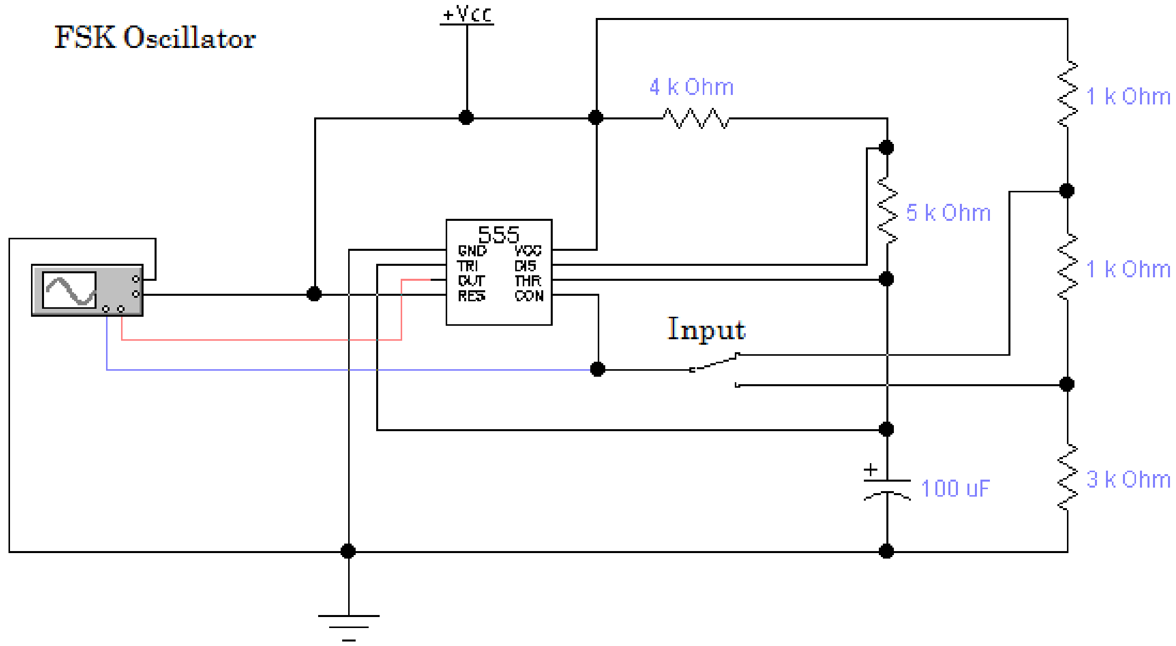


Block Diagram of FSK:

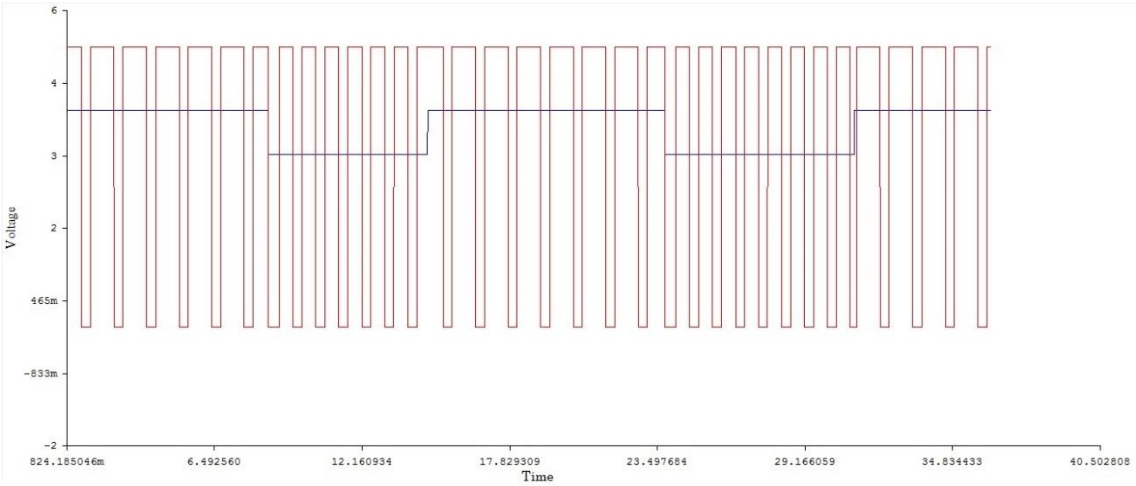


FSK Circuit Diagram:





FSK Wave Form:



LAB ASSIGNMENT No. 5

To Perform Digital Modulation Technique ASK

In electronics and telecommunications, modulation is the process of varying one or more properties of a periodic waveform, called the carrier signal i.e high frequency signal, with a modulating signal that typically contains information to be transmitted.

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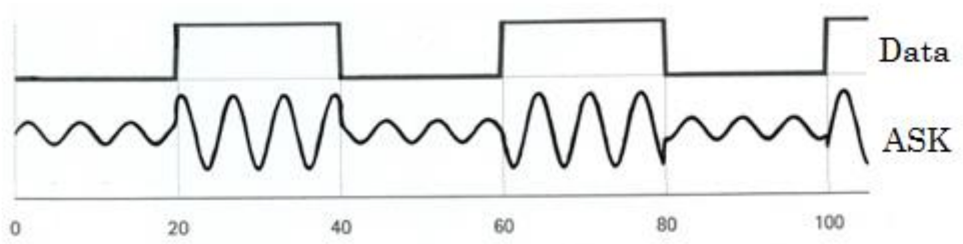
- **FSK** (frequency-shift keying): a finite number of frequencies are used.
- **ASK** (amplitude-shift keying): a finite number of amplitudes are used.
- **QAM** (quadrature amplitude modulation): a finite number of at least two phases and at least two amplitudes are used.

Amplitude-shift keying:

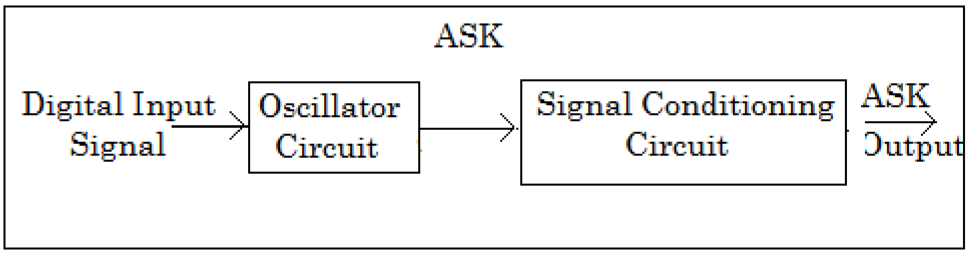
Amplitude-shift keying (ASK) is a form of amplitude modulation that represents digital data as variations in the amplitude of a carrier wave. In an ASK system, the binary symbol 1 is represented by transmitting a fixed amplitude carrier wave and fixed frequency for a bit duration of T seconds. If the signal value is 1 then the carrier signal will be transmitted, otherwise, a signal value of 0 will not be transmitted.

Any digital modulation scheme uses a finite number of distinct signals to represent digital data. ASK uses a finite number of amplitudes, each assigned a unique pattern of binary digits. Usually, each amplitude encodes an equal number of bits. Each pattern of bits forms the symbol that is represented by the particular amplitude.

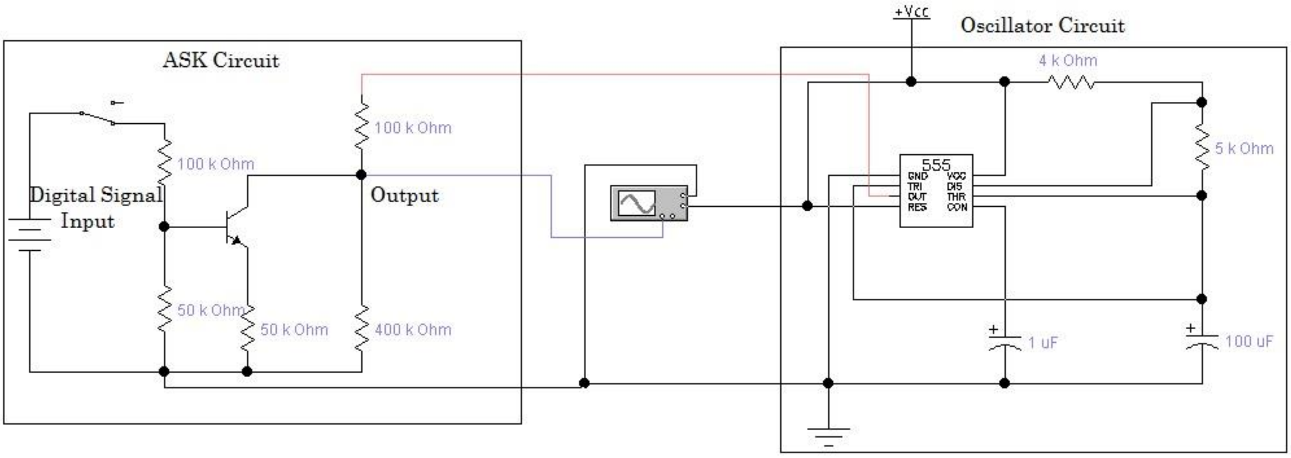
The demodulator, which is designed specifically for the symbol-set used by the modulator, determines the amplitude of the received signal and maps it back to the symbol it represents, thus recovering the original data. Frequency and phase of the carrier are kept constant.



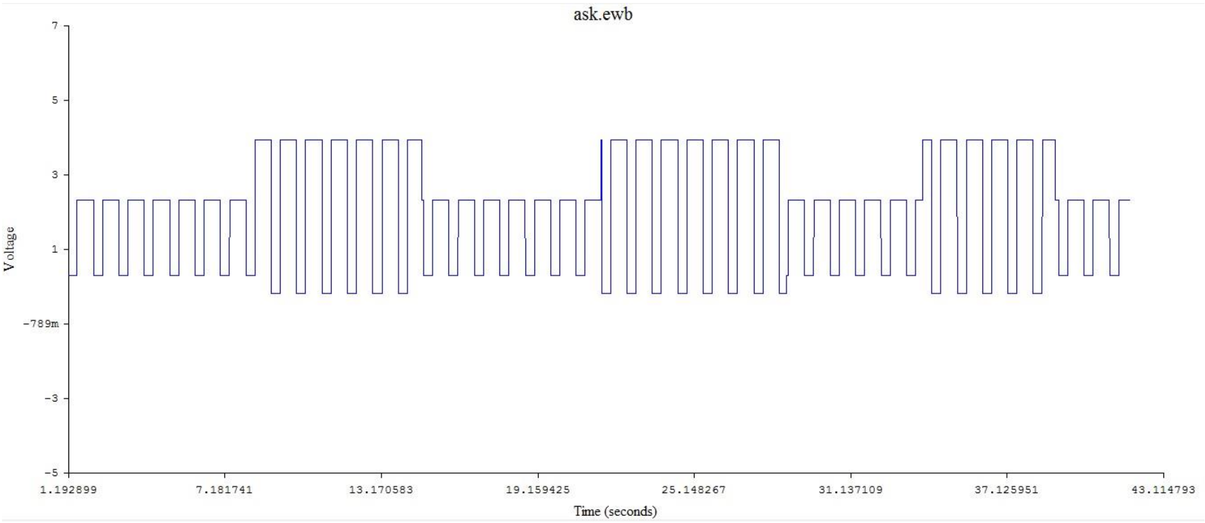
Block Diagram of ASK:



ASK Circuit Diagram:



ASK Wave from:



LAB ASSIGNMENT No. 6

To Perform Digital Modulation Technique PSK

In electronics and telecommunications, modulation is the process of varying one or more properties of a periodic waveform, called the carrier signal i.e high frequency signal, with a modulating signal that typically contains information to be transmitted.

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Fundamental digital modulation methods:

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- **FSK** (frequency-shift keying): a finite number of frequencies are used.
- **ASK** (amplitude-shift keying): a finite number of amplitudes are used.
- **QAM** (quadrature amplitude modulation): a finite number of at least two phases and at least two amplitudes are used.

Phase Shift Keying:

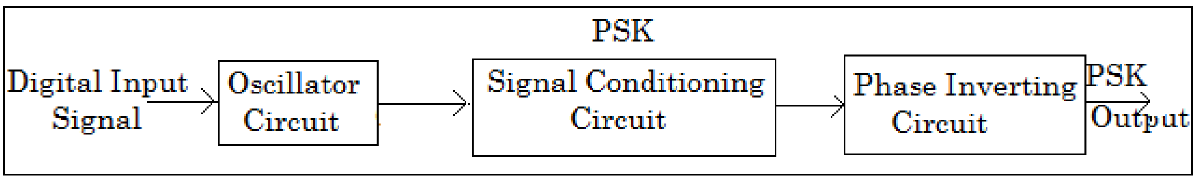
Phase-shift keying (PSK) is a digital modulation scheme that conveys data by changing, or modulating, the phase of a reference signal (the carrier wave).

Any digital modulation scheme uses a finite number of distinct signals to represent digital data. PSK uses a finite number of phases, each assigned a unique pattern of binary digits. Usually, each phase encodes an equal number of bits. Each pattern of bits forms the symbol that is represented by the particular phase. The demodulator, which is designed specifically for the symbol-set used by the modulator, determines the phase of the received signal and maps it back to the symbol it represents, thus recovering the original data. This requires the receiver to be able to compare the phase of the received signal to a reference signal.

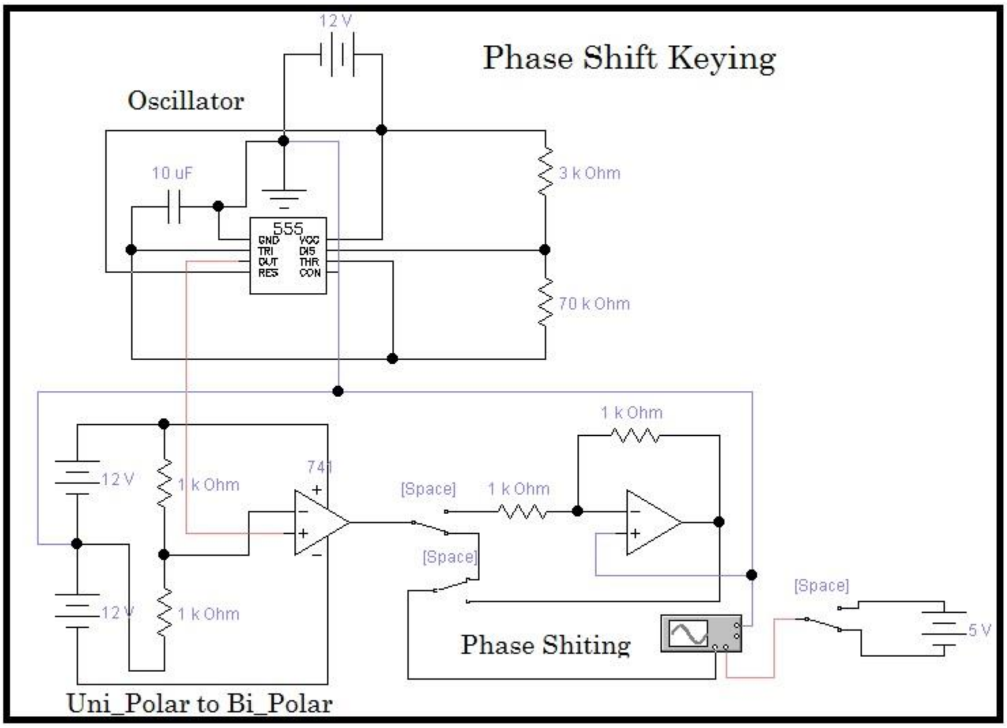
In the case of PSK, the phase is changed to represent the data signal. There are two fundamental ways of utilizing the phase of a signal in this way:

- By viewing the phase itself as conveying the information, in which case the demodulator must have a reference signal to compare the received signal's phase against; or
- By viewing the change in the phase as conveying information differential schemes, some of which do not need a reference carrier

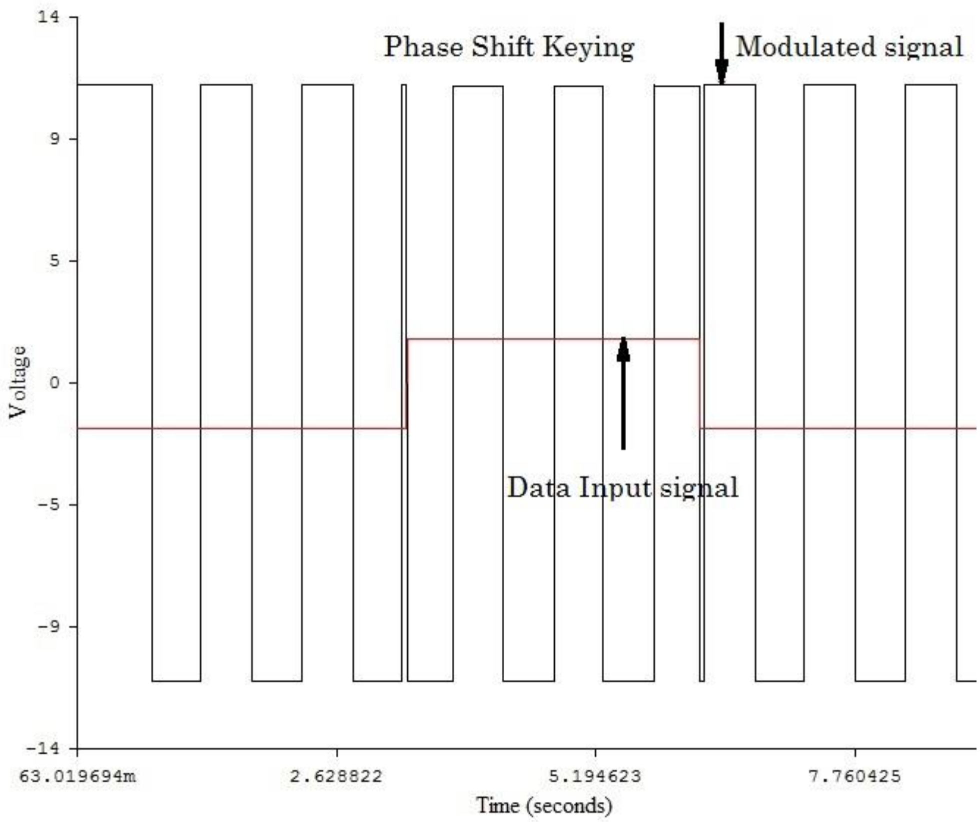
Block Diagram:



Circuit Diagram:



PSK Waveform:



LAB ASSIGNMENT No. 7

To Perform Multiplexing of Two Signals

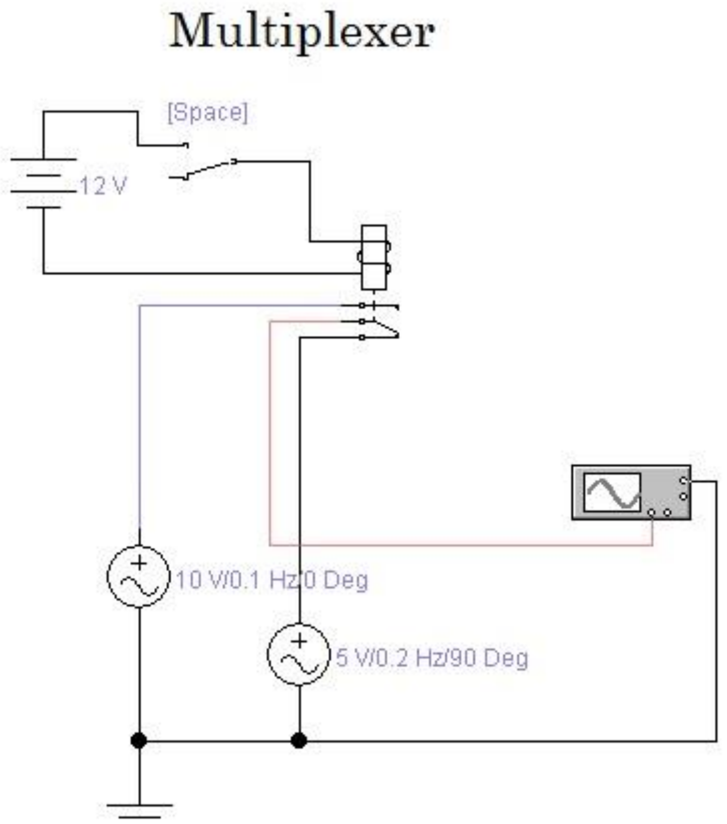
In telecommunications and computer networks, multiplexing sometimes contracted to muxing is a method by which multiple analog message signals or digital data streams are combined into one signal over a shared medium. The aim is to share an expensive resource. For example, in telecommunications, several telephone calls may be carried using one wire. Multiplexing originated in telegraphy in the 1870s, and is now widely applied in communications.

The multiplexed signal is transmitted over a communication channel, which may be a physical transmission medium (e.g. a cable). The multiplexing divides the capacity of the high-level communication channel into several low-level logical channels, one for each message signal or data stream to be transferred. A reverse process, known as de-multiplexing, can extract the original channels on the receiver side.

A device that performs the multiplexing is called a multiplexer (MUX), and a device that performs the reverse process is called a de-multiplexer (DEMUX or DMX).

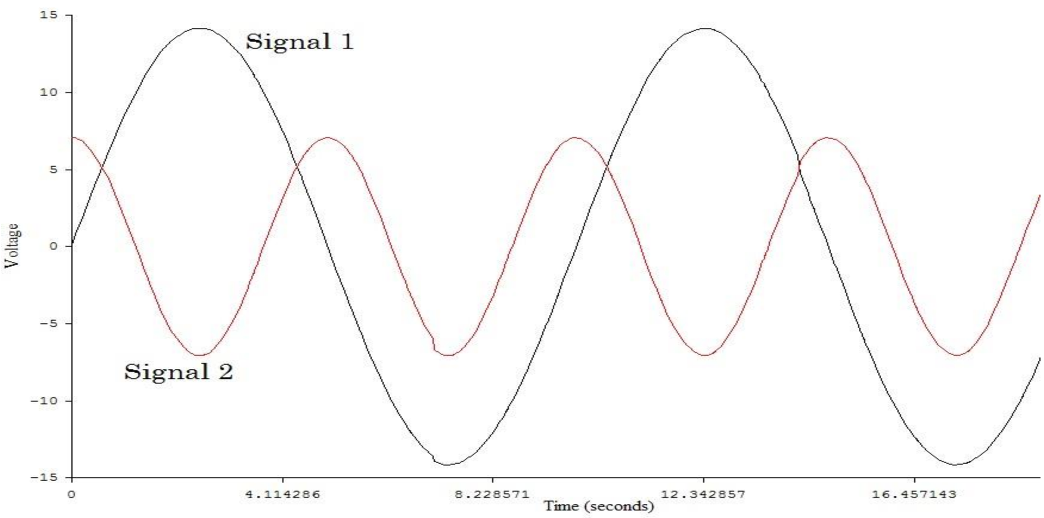
Inverse multiplexing (IMUX) has the opposite aim as multiplexing, namely to break one data stream into several streams, transfer them simultaneously over several communication channels, and recreate the original data stream.

Circuit Diagram:

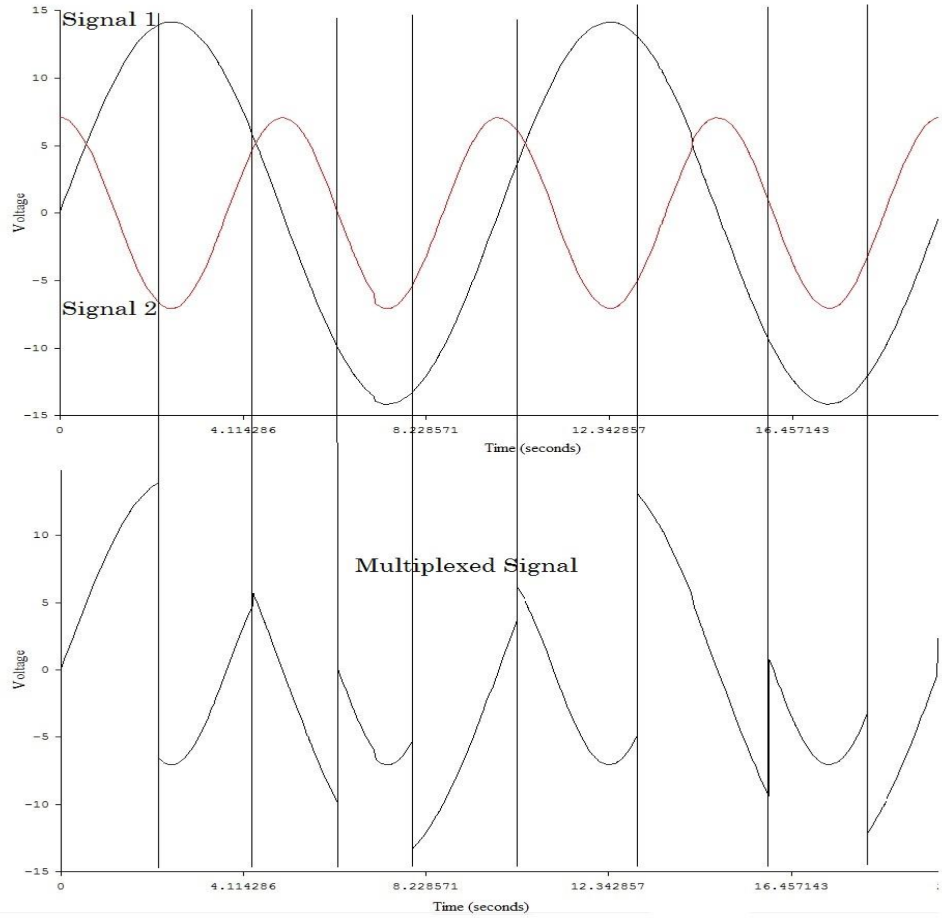


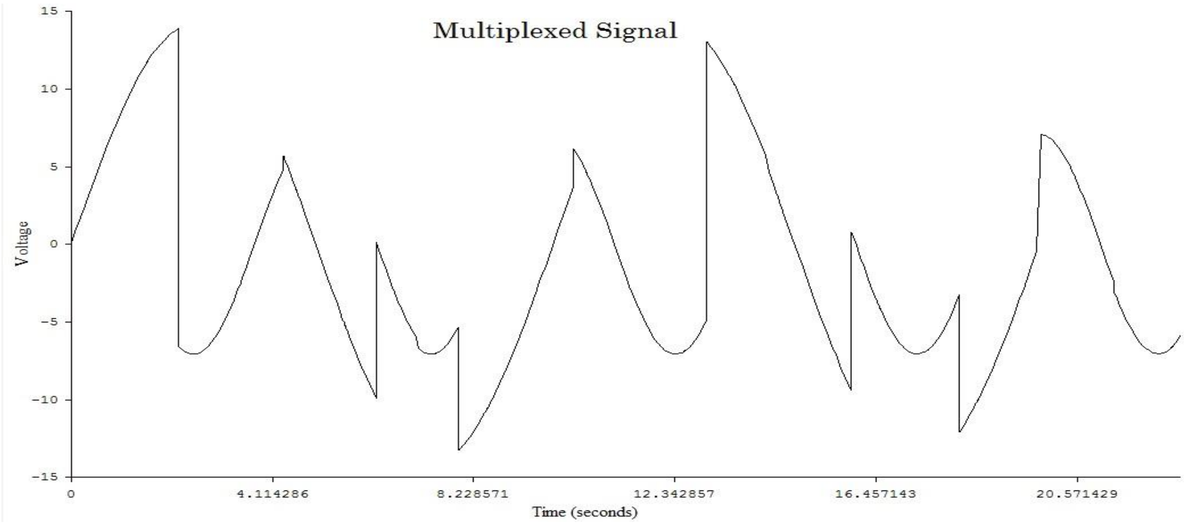
Waveforms:

Two Input Signals:



Output Signal:





LAB ASSIGNMENT No. 8

To Perform Digital Modulation Technique QPSK

In electronics and telecommunications, modulation is the process of varying one or more properties of a periodic waveform, called the carrier signal i.e high frequency signal, with a modulating signal that typically contains information to be transmitted.

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Fundamental digital modulation methods:

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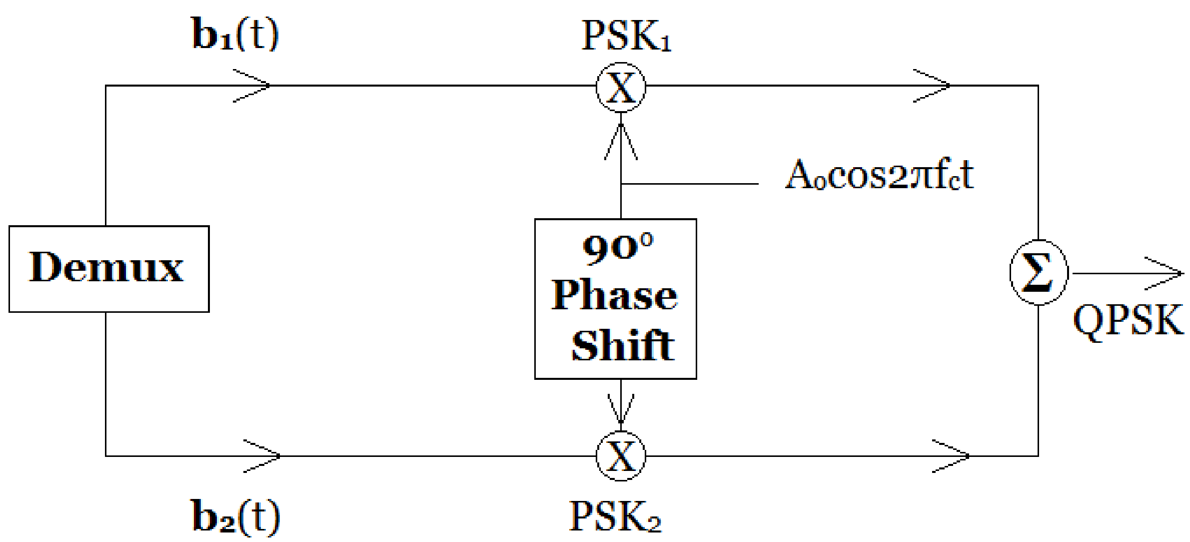
- **ASK** (amplitude-shift keying): a finite number of amplitudes are used.
- **QAM** (quadrature amplitude modulation): a finite number of at least two phases and at least two amplitudes are used.

Quadrature Phase Shift Keying:

Sometimes this is known as quadri-phase PSK, 4-PSK, or 4-QAM. Although the root concepts of QPSK and 4-QAM are different, the resulting modulated radio waves are exactly the same. QPSK uses four points on the constellation diagram, equi-spaced around a circle. With four phases, QPSK can encode two bits per symbol.

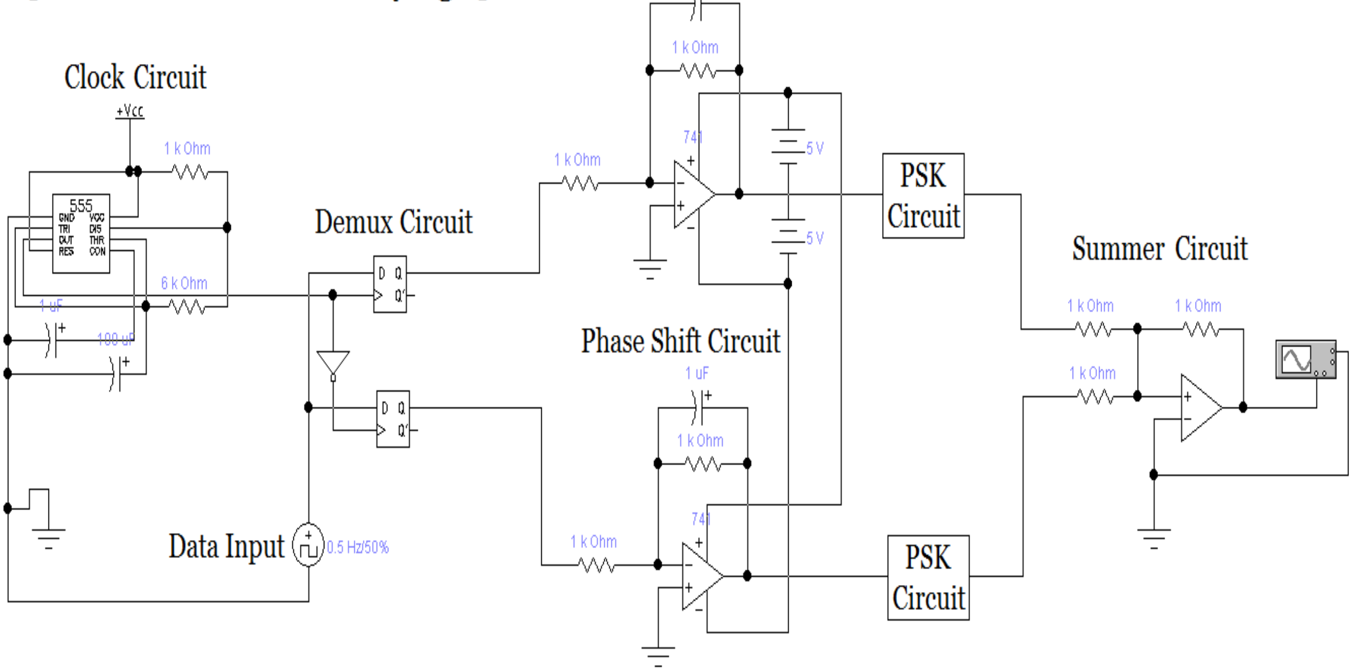
The mathematical analysis shows that QPSK can be used either to double the data rate compared with a BPSK system while maintaining the same bandwidth of the signal, or to maintain the data-rate of BPSK but halving the bandwidth needed. In this latter case, the BER of QPSK is exactly the same as the BER of BPSK and deciding differently is a common confusion when considering or describing QPSK. The transmitted carrier can undergo numbers of phase changes.

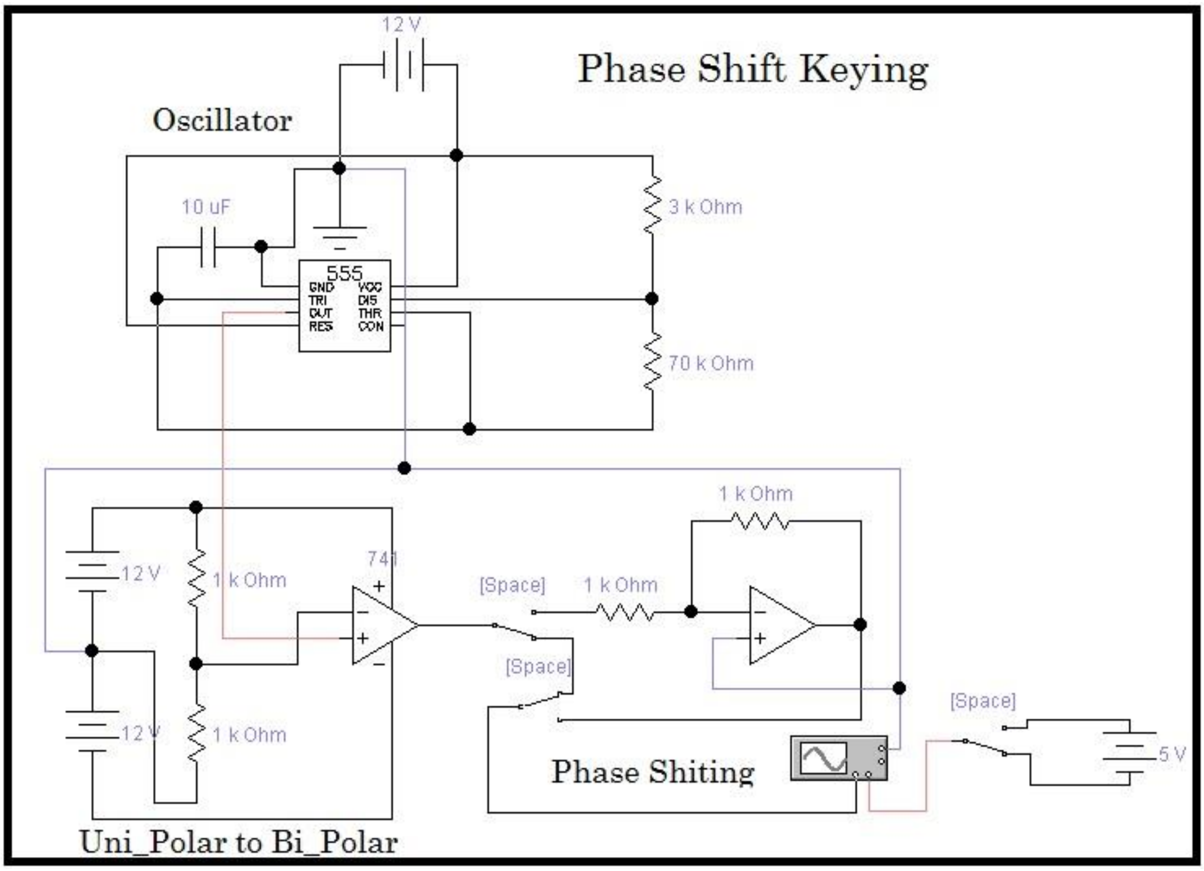
Block Diagram:



Circuit Diagram:

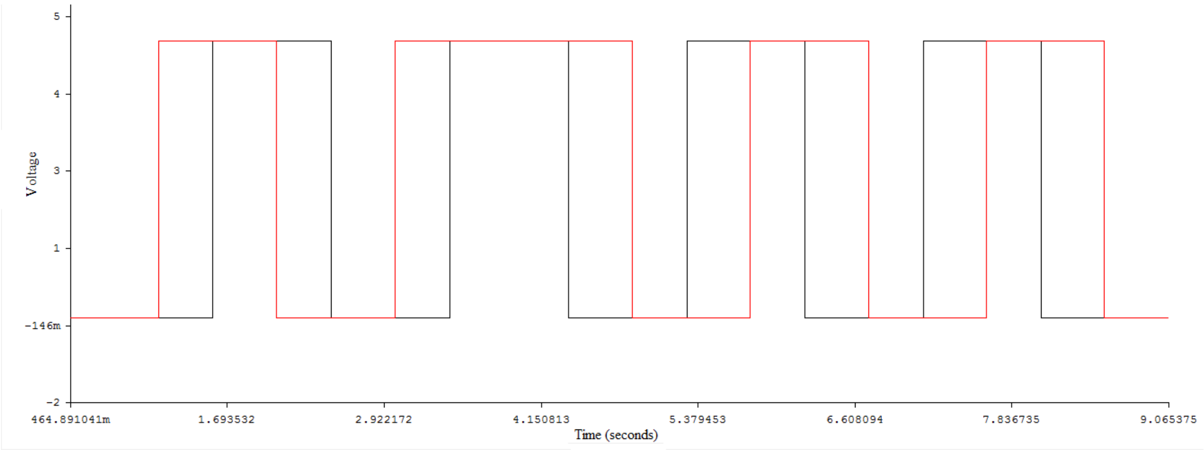
Quadrature Phase Shift Keying(QPSK)



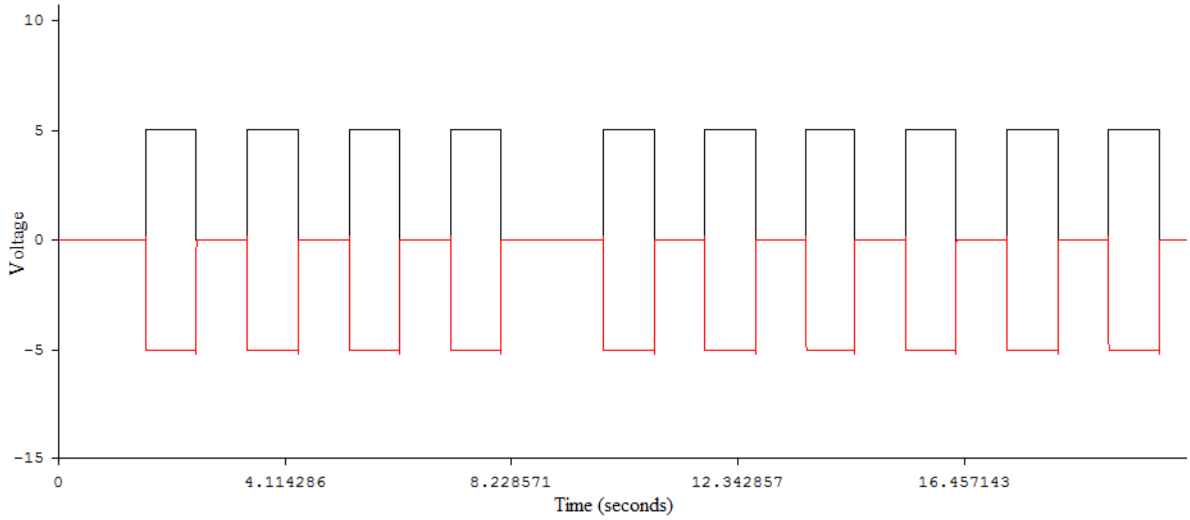


Wave forms:

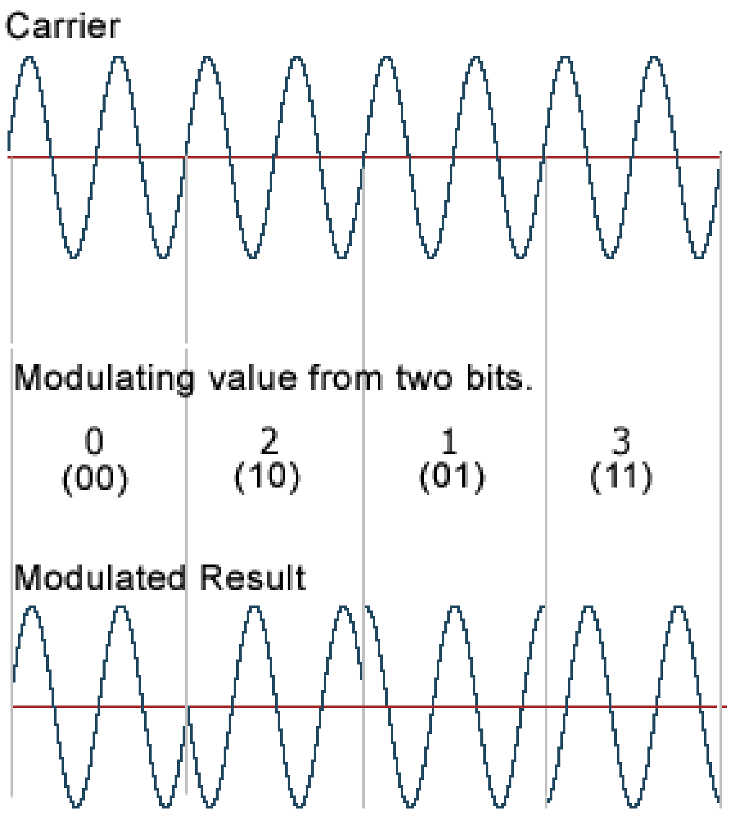
Output of Demux:



Output of phase shift:



QPSK Wave:


Carrier
Modulating value from two bits.
0
(00)
2
(10)
1
(01)
3
(11)
Modulated Result

LAB ASSIGNMENT No. 9

To Perform Digital Modulation Pulse Position Modulation (PPM)

In electronics and telecommunications, modulation is the process of varying one or more properties of a periodic waveform, called the carrier signal i.e high frequency signal, with a modulating signal that typically contains information to be transmitted.

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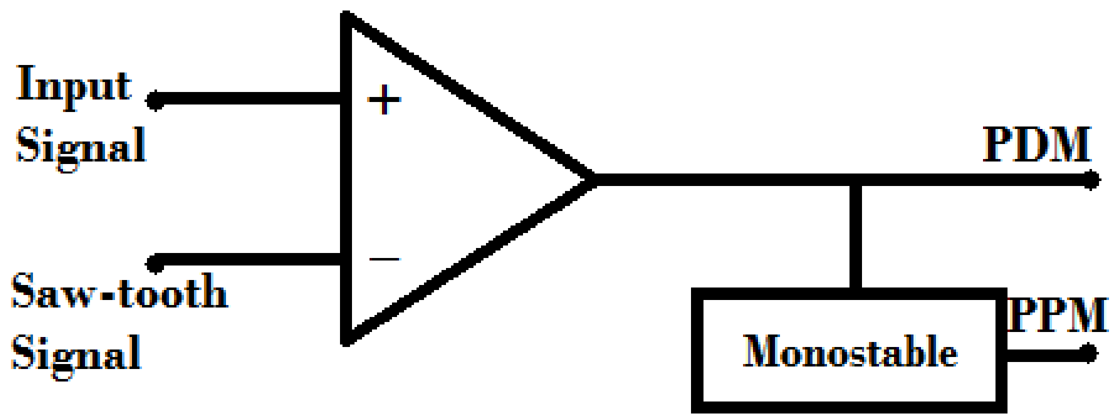
- **QAM** (quadrature amplitude modulation): a finite number of at least two phases and at least two amplitudes are used.

Pulse Position Modulation:

Pulse-position modulation (PPM) is a form of signal modulation in which M message bits are encoded by transmitting a single pulse in one of 2^M possible time-shifts. This is repeated every T seconds, such that the transmitted bit rate is M/T bits per second. It is primarily useful for optical communications systems, where there tends to be little or no multipath interference.

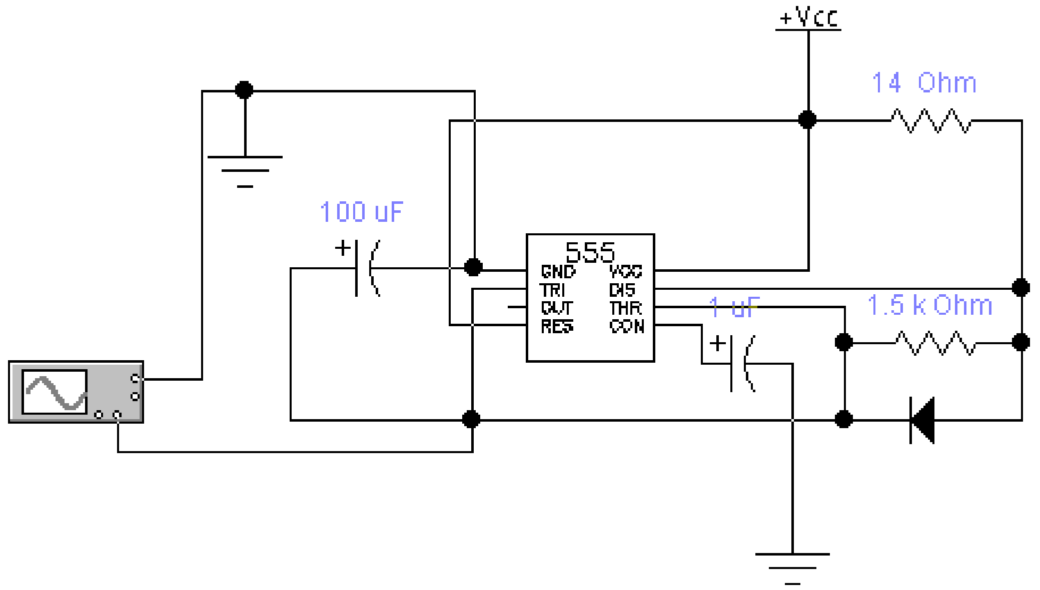
One of the key difficulties of implementing this technique is that the receiver must be properly synchronized to align the local clock with the beginning of each symbol. Therefore, it is often implemented differentially as differential pulse-position modulation, whereby each pulse position is encoded relative to the previous, such that the receiver must only measure the difference in the arrival time of successive pulses. It is possible to limit the propagation of errors to adjacent symbols, so that an error in measuring the differential delay of one pulse will affect only two symbols, instead of affecting all successive measurements.

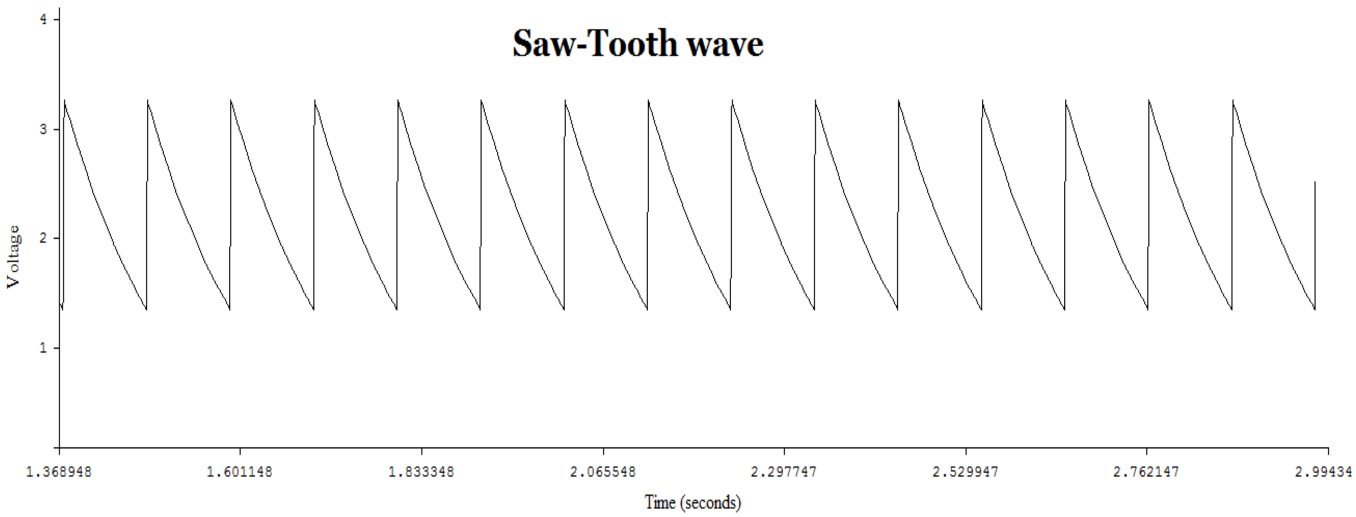
Block Diagram:



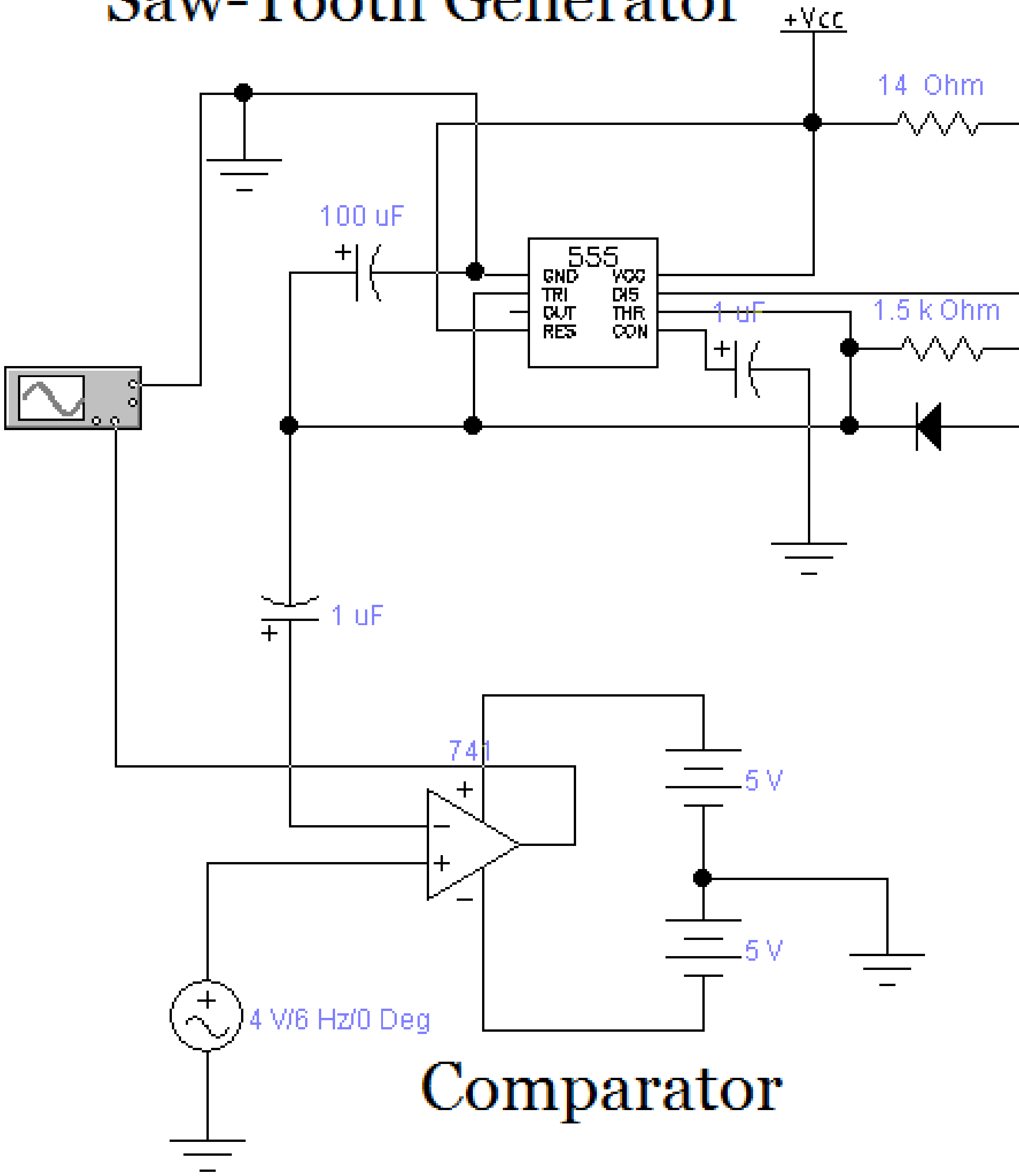
Circuit Diagram:

Saw-tooth Wave Circuit



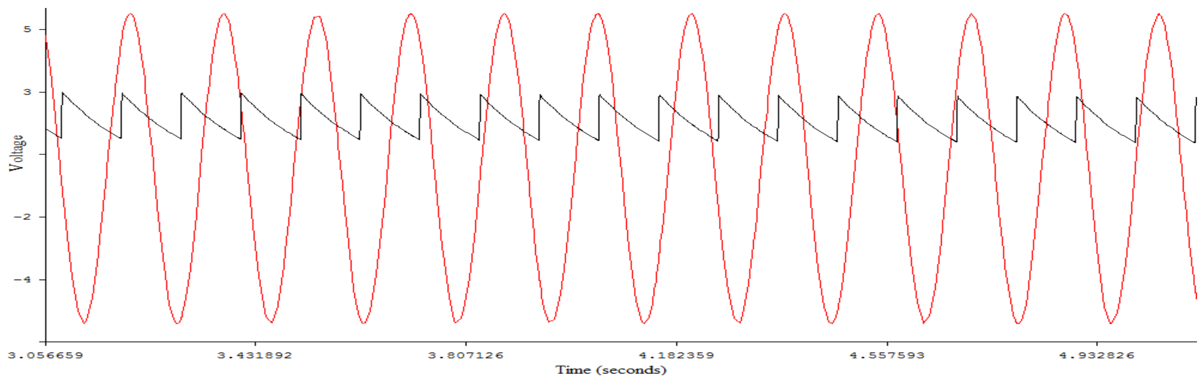


Saw-Tooth Generator

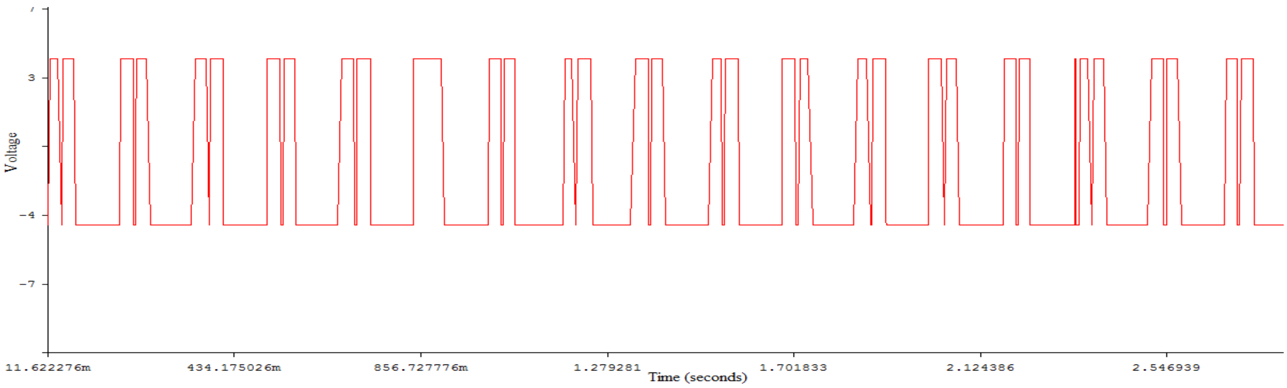


Comparator

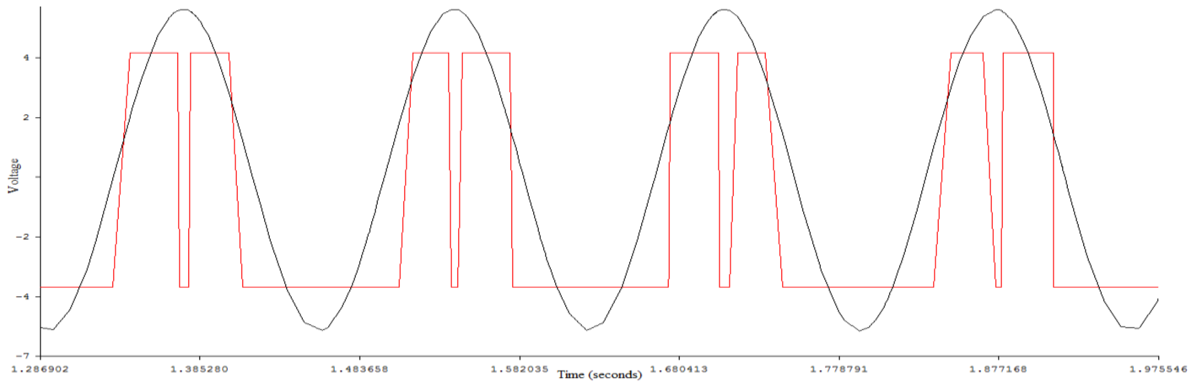
Saw-tooth signal Vs Data input:



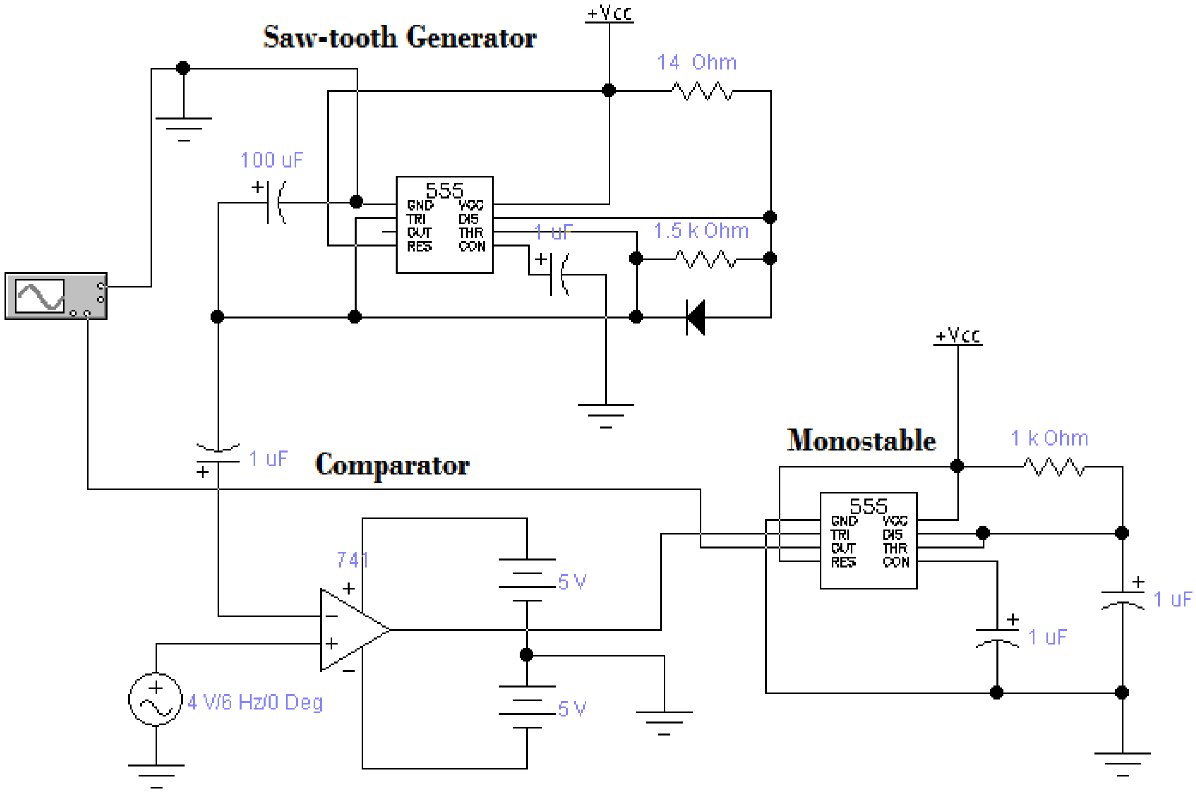
PDM(Output of Comparator) Wave:



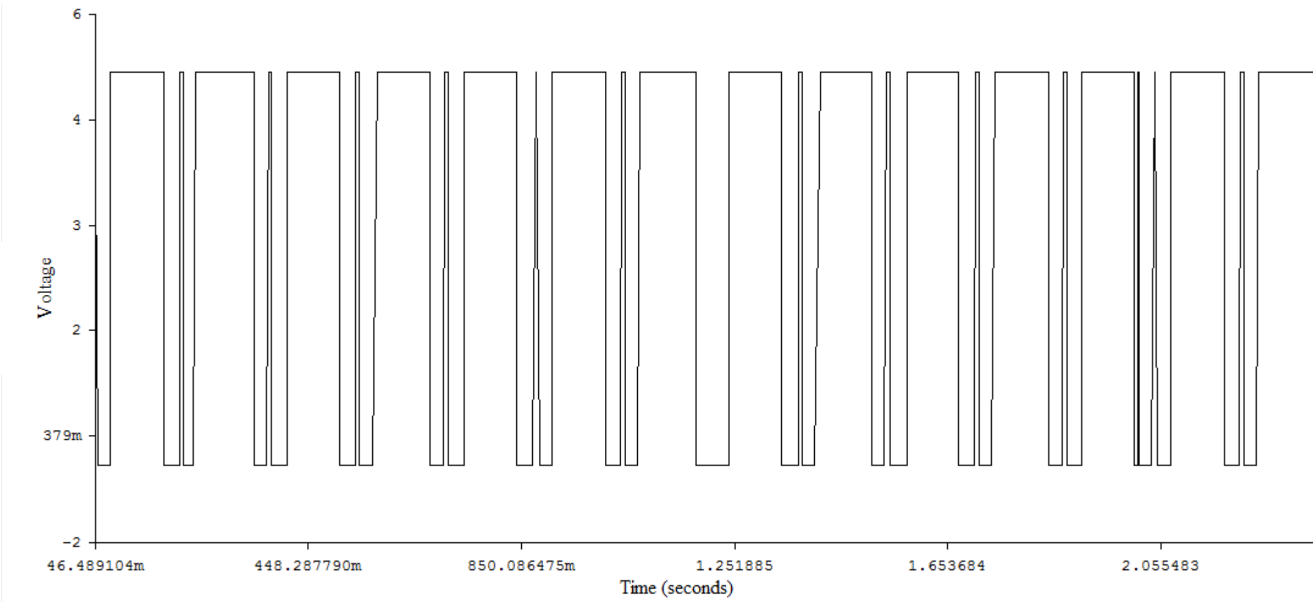
Data Input VS PDM Wave:



Pulse Position Modulation



PPM Wave:



LAB ASSIGNMENT No. 10

Design A Digital Communication System

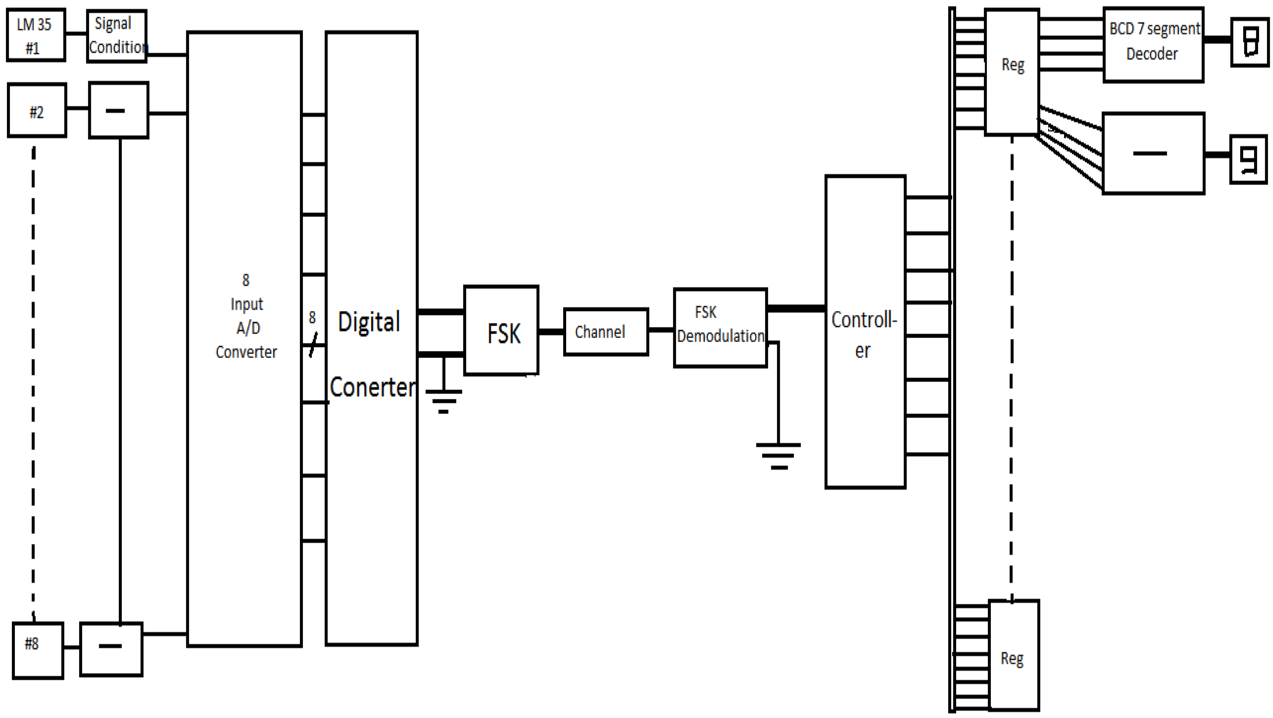
For 8 Channel Temperature Sensor

Temperature is a very critical and widely measured variable for engineers. Many processes must have either a monitored or controlled temperature. Temperature monitoring and controlling of furnace as furnace brazing application which are used in automotive industry is very crucial for increasing lifetime of the machine. A continuous real time monitoring and controlling system has been developed with the use of data acquisition system. A real time PC based logging systems can be used for the measurement, monitoring and controlling of different temperature parameters of furnace. Data storage and recording is a very common measurement application. In its most basic form, data storage is the measurement and recording of different temperature parameters over a period of time.

The device temperature parameter of different zones of furnace has been constantly monitored and hence it can be further controlled by using data acquisition and control system. The constant monitoring of such different furnace zone temperature can produce data-base for scheduling of the machine servicing, troubleshooting and also for future references. Due to use of microcontroller the computation task has been handled most effectively. Hence the proposed system has been widely used in automotive engineering, instrumentation and power quality monitoring and control applications. Data acquisition system for monitoring such temperature of brazing furnace provides advantages of design simplicity, portability and less cost.

Analog and digital signals are used to transmit information, usually through electric signals. In both these technologies, the information, such as any audio or video, is transformed into electric signals. The difference between analog and digital technologies is that in analog technology, information is translated into electric pulses of varying amplitude. In digital technology, translation of information is into binary format (zero or one) where each bit is representative of two distinct amplitudes.

Circuit Diagrm:



LM 335:

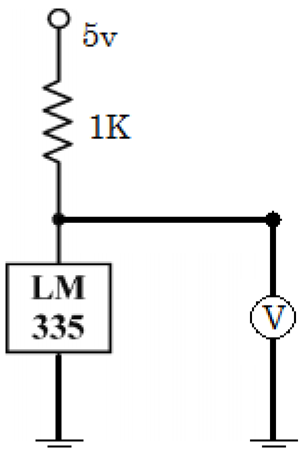
The transducers RTD, thermistor and thermocouples have some significant limitations, e.g. thermocouples have a low output signal which varies none linearly with temperature. Also, they need some form of reference compensation. RTD's are more linear than thermocouple but the change in their resistance is very small even for large change in input temperatures i.e. they have low sensitivity. Thermistor has high sensitivity but they exhibit highly non-linear resistance temperature characteristics.

For each of these transducers, electronic compensation circuits have to be used in order to overcome their shortcomings. Also additional circuitry may be needed to increase their voltage or current output. Usually this additional electronics circuitry takes the form of monolithic integrated circuits. Thus it requires combining temperature sensing element with signal conditioning electronics to produce single monolithic IC package. The one used in ST2302 is LM 335.

This is an IC containing 16 transistors 9 resistance and 2 capacitors contained in a transistor type package. It provides an output of 10mV/° K, measurements of output voltage therefore indicate the temperature directly in degrees Kelvin e.g. at a temperature of 20°C (293 °K) the output voltage will be 2.93V.

An LM335 unit is mounted on the type K thermocouple panel, external to the heated closure and fitted in a heat sink together with another type K thermocouple. Its output is available at the REF socket. The output form this can be used as an indication of the ambient temperature outside the heated enclosure.

Circuit Diagram:



Signal Conditioning:

In electronics, signal conditioning means manipulating an analog signal in such a way that it meets the requirements of the next stage for further processing. Most common use is in analog-to-digital converters.

It is common to have a sensing stage which consists of a sensor, a signal conditioning stage where usually amplification of the signal is done and a processing stage normally carried out by an ADC and a micro-controller.

In digital electronics, digital computers have taken a major role in near every aspect of life in our modern world. Digital electronics is at the heart of computers, but there are lots of direct applications of digital electronics in our world. All these digital electronics need data to be presented to them in a digital format i.e. the data have to be digitally conditioned. This is called digital conditioning. Since computers are electronics devices, all the information they work with has to be digitally formatted. Therefore, if they are used to control a variable such as temperature, then the temperature has to be represented digitally. That's why we need digital signal conditioning to condition process-control signal to be an approximated digital format.

Signal conditioning can include amplification, filtering, converting, range matching, isolation and any other processes required to make sensor output suitable for processing after conditioning.

Analog-To-Digital Converter:

An analog-to-digital converter is a device that converts a continuous physical quantity usually voltage to a digital number that represents the quantity's amplitude.

The conversion involves quantization of the input, so it necessarily introduces a small amount of error. Instead of doing a single conversion, an ADC often performs the conversions periodically.

The result is a sequence of digital values that have been converted from a continuous-time and continuous-amplitude analog signal to a discrete-time and discrete-amplitude digital signal.

An ADC is defined by its bandwidth the range of frequencies it can measure and its signal to noise ratio how accurately it can measure a signal relative to the noise it introduces. The actual bandwidth of an ADC is characterized primarily by its sampling rate, and to a lesser extent by how it handles errors such as aliasing. The dynamic range of an ADC is influenced by many factors, including the resolution the number of output levels it can quantize a signal to, linearity and accuracy how well the quantization levels match the true analog signal and jitter small timing errors that introduce additional noise.

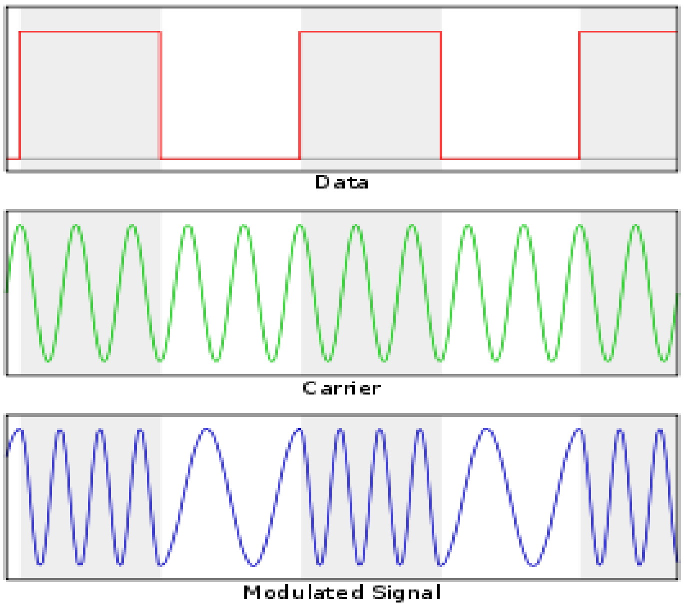
ADCs are chosen to match the bandwidth and required signal to noise ratio of the signal to be quantized. If an ADC operates at a sampling rate greater than twice the bandwidth of the signal, then perfect reconstruction is possible given an ideal ADC and neglecting quantization error.

The presence of quantization error limits the dynamic range of even an ideal ADC, however, if the dynamic range of the ADC exceeds that of the input signal, its effects may be neglected resulting in an essentially perfect digital representation of the input signal.

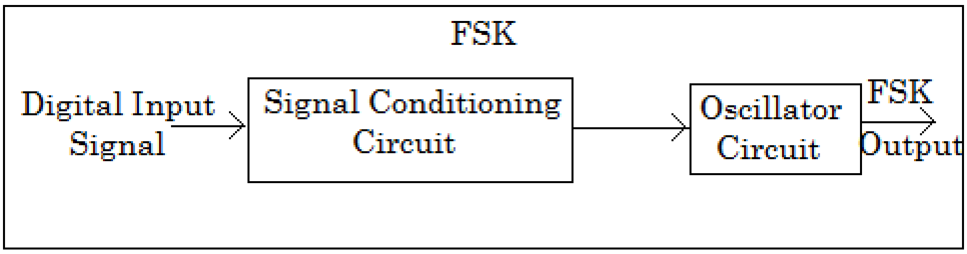
Frequency-shift keying:

Frequency-shift keying (FSK) is a frequency modulation scheme in which digital information is transmitted through discrete frequency changes of a carrier wave. The simplest FSK is binary FSK (BFSK).

BFSK uses a pair of discrete frequencies to transmit binary 0s and 1s information. With this scheme, the "1" is called the mark frequency and the "0" is called the space frequency.



Block Diagram of FSK:



Register:

The Shift Register is a type of sequential logic circuit that is used for the storage or transfer of data in the form of binary numbers. This sequential device loads the data present on its inputs and then moves or "shifts" it to its output once every clock cycle, hence the name "shift register".

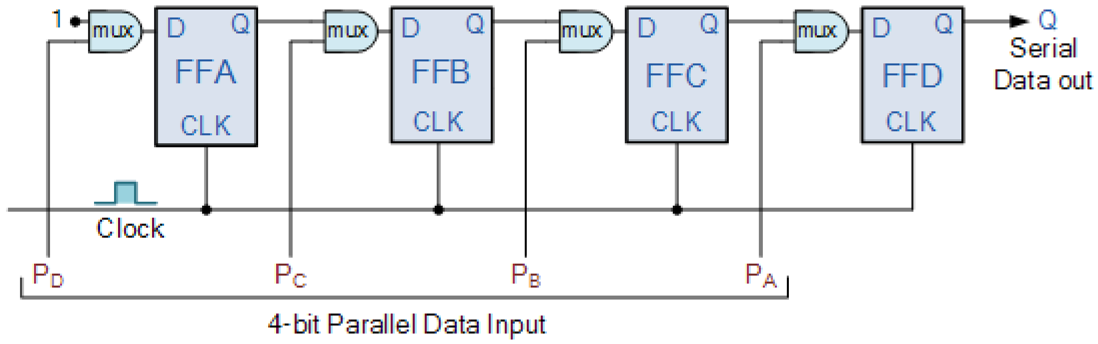
A shift register basically consists of several single bit "D-Type Data Latches", one for each data bit, either a logic "0" or a "1", connected together in a serial or daisy-chain arrangement so that the output from one data latch becomes the input of the next latch and so on.

The data bits may be fed in or out of a shift register serially, that is one after the other from either the left or the right direction, or all together at the same time in parallel.

The number of individual data latches required to make up a single Shift Register device is usually determined by the number of bits to be stored with the most common being 8-bits (one byte) wide constructed from eight individual data latches.

The Shift Register is used for data storage or data movement and are used in calculators or computers to store data such as two binary numbers before they are added together, or to convert the data from either a serial to parallel or parallel to serial format. The individual data latches that make up a single shift register are all driven by a common clock (Clk) signal making them synchronous devices. Shift register IC's are generally provided with a clear or reset connection so that they can be "SET" or "RESET" as required. Generally, shift registers operate in one of four different modes with the basic movement of data through a shift register.

Two registers are used at input of 8 to 1 mux. These registers contain any random sequence of bits to inform the mux that the data from all ADCs is being transferred to the mux and the cycle starts again.



7 Segment BCD decoder:

Typically 7-segment displays consist of seven individual coloured LED's (called the segments), within one single display package. In order to produce the required numbers or HEX characters from 0 to 9 and A to F respectively, on the display the correct combination of LED segments need to be illuminated and BCD to 7-segment Display Decoders such as the 74LS47 do just that.

A standard 7-segment LED display generally has 8 input connections, one for each LED segment and one that acts as a common terminal or connection for all the internal display segments. Some single displays have also have an additional input pin to display a decimal point in their lower right or left hand corner.

Obviously, the use of so many connections and power consumption is impractical for some electronic or microprocessor based circuits and so in order to reduce the number of signal lines required to drive just one single display, display decoders such as the BCD to 7-Segment Display Decoder and Driver IC's are used instead.

So in order to display the number 3 for example, segments a, b, c, d and g would need to be illuminated. If we wanted to display a different number or letter then a different set of segments would need to be illuminated.