Experiment No. 1

CO Addressed: CO2

AIM :-

To modulate and demodulate BPSK & its spectral analysis (DSO)

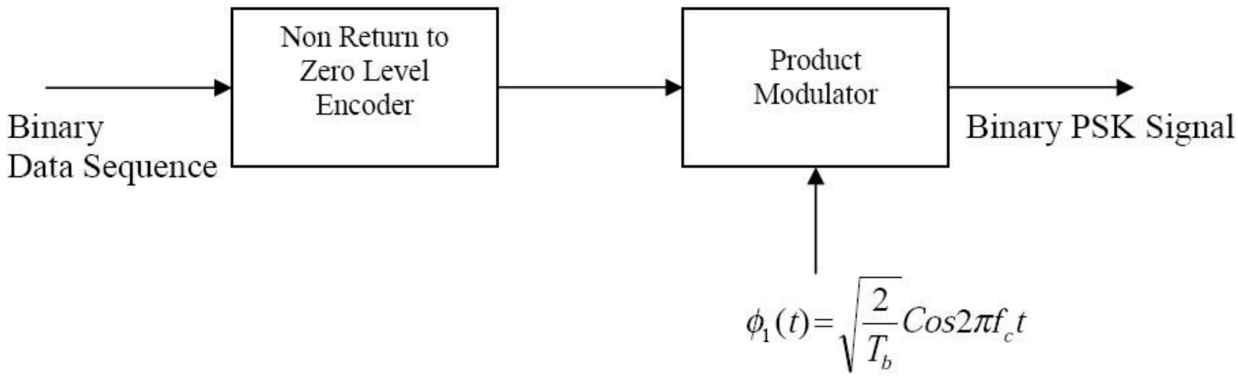
EQUIPMENT: - BPSK modem, DSO, connecting wires, probes.

THEORY:-

In binary PSK the transmitted signal is a sinusoidal of fixed amplitude. It has one fixed phase when the data is at one level & when the data is at the other level the phase is differ by 180. In this kit we have derived a sinusoid from a crystal clock using IC4017's & frequency divider & 2nd order butter worth filter & for phase shifted wave form we have used OP-AMP Inverter.

Also two different bit patterns are also derived from a basis clock. For transmitter section we have used IC 4052 [anolog switch].A sinusoid & phase shifted sinusoid are two I/P's to Anolog switch & bit patttern is connected to control I/P of 4052.O/P of 4052 is a required BPSK output. Also for synchronization purpose a start signal (power on mono) is send. This signal is ORed with bit pattern & then given to transmitter. At the receiver we have to derive carrier frequency from BPSK waveform. So this waveform is given to squaring the circuit (IC1496 multiplier). At the O/P of 1496 we have 2 fc frequencies. By using Band pass filter with center frequency '2fc & frequency divider (division factor 2). We get carrier frequency. For detection of data from BPSK signal, phase comparator section of PLL (IC565) is used. One I/P to PLL is BPSK signal & Other I/P is recovered carrier. O/P of PLL is then given to the filter. O/P of filter is either as digital data or inverted one. Then with the help of start signal we corrected the data. We know that at the 'Power on' we have send '1' as digital data. O/P of filter is latched at this instant. If it is ‘1’ then we get correct data. If this is zero then O/P is inverted & therefore we get correct data.

Coherent Binary PSK:



Fig(a) Block diagram of BPSK transmitter

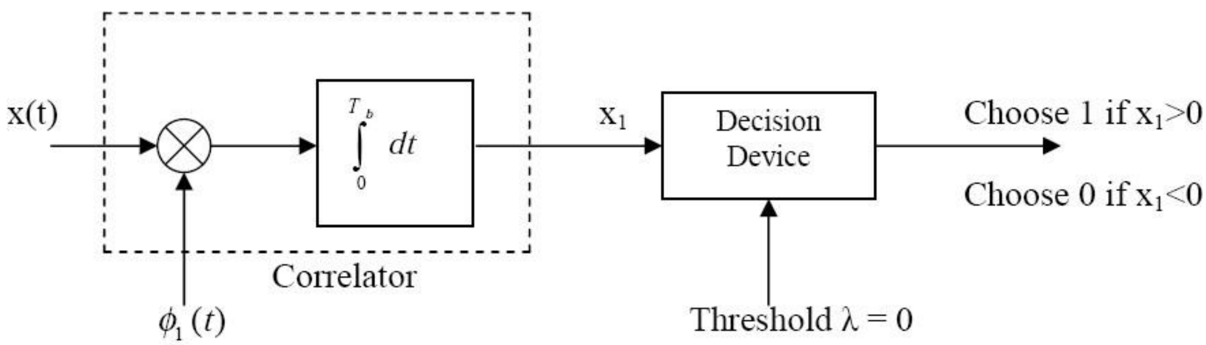


Fig (b) Coherent binary PSK receiver

In a Coherent binary PSK system the pair of signals $S_1(t)$ and $S_2(t)$ are used to represent binary symbol '1' and '0' respectively.

$$S_1(t)=\sqrt{\frac{2E_b}{T_b}}\,Cos2\pi f_c t \qquad \text{---------} \qquad \text{for Symbol '1'}$$

$$S_2(t)=\sqrt{\frac{2E_b}{T_b}}\,Cos(2\pi f_c t+\pi) = -\sqrt{\frac{2E_b}{T_b}}\,Cos2\pi f_c t \quad \text{------- for Symbol '0'}$$

Where E_b= Average energy transmitted per bit $E_b = \frac{E_{b0}+E_{b1}}{2}$

In the case of PSK, there is only one basic function of Unit energy which is given by

$$\phi_1(t) = \sqrt{\frac{2}{T_b}}\, Cos2\pi f_c t \qquad 0 \le t \le T_b$$

Therefore the transmitted signals are given by

$$S_1(t) = \sqrt{E_b}\,\phi_1(t) \qquad 0 \le t \le T_b \qquad for \quad Symbol\, 1$$

$$S_2(t) = -\sqrt{E_b}\,\phi_1(t) \qquad 0 \le t \le T_b \qquad for \quad Symbol\, 0$$

A Coherent BPSK is characterized by having a signal space that is one dimensional (N=1) with two message points (M=2)

$$S_{11} = \int_0^{T_b} S_1(t)\phi_1(t)\,dt = +\sqrt{E_b}$$

$$S_{21} = \int_0^{T_b} S_2(t)\phi_1(t)\,dt = -\sqrt{E_b}$$

The message point corresponding to $S_1(t)$ is located at $S_{11} = +\sqrt{E_b}$ and $S_2(t)$ is located at $S_{21} = -\sqrt{E_b}$.

To generate a binary PSK signal we have to represent the input binary sequence in polar form with symbol '1' and '0' represented by constant amplitude levels of $+\sqrt{E_b}\ \&-\sqrt{E_b}$ respectively. This signal transmission encoding is performed by a NRZ level encoder. The resulting binary wave [in polar form] and a sinusoidal carrier $\phi_1(t)$ [whose frequency $f_c = \frac{n_c}{T_b}$] are applied to a product modulator. The desired BPSK wave is obtained at the modulator output.

To detect the original binary sequence of 1's and 0's we apply the noisy PSK signal x(t) to a Correlator, which is also supplied with a locally generated coherent reference signal $\phi_1(t)$ as shown in fig (b). The correlator output x_1 is compared with a threshold of zero volt.

If $x_1 > 0$, the receiver decides in favour of symbol 1.

If $x_1 < 0$, the receiver decides in favour of symbol 0.

PROCEDURE:-

1] O/P of pattern Gen is connected to I/P of ‘OR’ Gate.
2] O/P of OR gate is connected to I/P of transmitter i.e. (i/p of MULT. Block).
3] O/P of ‘MULT’ Block i.e BPSK O/P is connected to I/P of 1496 Sq.cct.
4] O/P of 1496 Sq.cct is connected to I/P of Band Pass Filter.
5] O/P of Band Pass fitter is connected to I/P of %2 N/W.
6] O/P of %2 N/W is connected to I/P of phase comparator.
7] BPSK O/P is connected to I/P 2 of phase comparator.
8] Observed signal at different test points together with I/P bit pattern.
9] Observed filter O/P & COMP. Block O/P. The O/P of COMP. Block is required detected O/P.

OBSERVATION TABLE:

Sr. No.	Name of Signal	Amplitude	Frequency

CONCLUSION:-

Staff Signature

Experiment No. 2

CO Addressed: CO2

AIM:

To study QPSK modulation and demodulation & its spectral analysis (DSO).

EQUIPMENT: QPSK modem, DSO, connecting wires, probes.

THEORY:

To transmit digital data on analog lines or even into space, modulation of analog signal is required. Simplest way is BPSK, where one phase of carrier is transmitted for '1' and inverted carrier is transmitted for digital '0'. Here if bit rate is 'Tb' then bandwidth required is '2Tb'. To reduce this bandwidth requirement QPSK can be used. For QPSK bandwidth is 'Tb' i.e half that of BPSK.

QPSK techniques come under carrier modulation type. Here i/p to the transmitter is digital data, in between processing is in analog form & finally o/p of receiver is again digital data same as fed to transmitter.

'Q' in 'QPSK' means quadrature i.e 4 four phases of carrier are transmitted depending upon bit pattern. Incoming bit pattern is divided into ODD & EVEN bit patterns. Odd pattern is multiplied by sine wave, & even wave is multiplied by cos wave. Sine & cos waves are 90 degree phase shifted. Now resulting two PSKs are added & we get vector addition o/p i.e if both bit is '1' we get 45 degree phase shifted carrier. If odd bit is 1 and even bit is 0 we get 135 degree phase carrier. If odd bit is 0 & even also 0 we get 225 degree. If odd bit is 0 & even bit is 1 we get 315 degree.

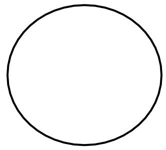
In QPSK, two consecutive bits are stored & for resulting four combinations (4) different phases of carrier are transmitted. By using 'D' flip-flop type arrangement incoming bit pattern is divided into two bit patterns viz. odd & even pattern, for obtaining this, basic clock whose frequency is 'fb' is divided by two, resulting odd & even clock frequencies are 'fb/2' & they are complementary. Each bit is stored for '2Tb' & even bit pattern will have bit no 2, 4, 6, etc stored for '2Tb'.

Here active edges of, odd & even clocks are separated by time 'Tb'. So out of two bits only one bit is changing after each 'Tb' period but every bit is there for 2Tb time; so in this offset QPSK system every time phase changes by 90 degree only.

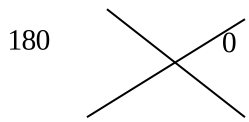
At receiver carrier is recovered from QPSK signal itself. This is synchronous reception. To recover carrier QPSK signal is raised to power four using analog multiplier. Then resulting signal is passed through bandpass filter whose center frequency is adjusted to 4 times carrier frequency. Then o/p of bandpass filter is divided by 4 to get carrier frequency. In this kit IC 1496 is used as analog multiplier.

Then QPSK signal is multiplied by 'SINE & COS' carrier waves. As a result we get odd & even patterns after filtering & integrating multiplier outputs. Now by combining these two patterns we get original bit patterns. This is done by using switch.

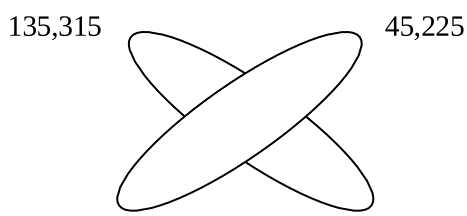
To observe QPSK, we have given two bit patterns so that on analog CRO we can observe the wave forms. Here carrier phase changes every time 'tb' depending upon odd & even bit combination. It is difficult to observe this on analog CRO. Details of these phase changes are shown in diagram attached. To observe QPSK we can use lissageous patterns. If we connect SINE wave to one channel & COS wave to other channel & press 'XY' button of CRO we get circle on screen.



Now if we connect SINE & its associated PSK signal to two channels & press 'XY' mode buttons we get two crossed lines.



If 'SINE' & 'QPSK' signals are connected to two channels, on 'XY' mode we get two crossed ellipses. This is because for 45, 135, 225, 315 degree we get ellipse a lissageous fig.



Also at transmitter observe that 'SINE', 'COS' wave amplitudes are lesser than resulting 'QPSK' wave because of vector addition.

We are doing this complex processing to save on bandwidth requirement of the system. This can be observed on CRO also. Observe bit pattern on CRO along with odd or even bit pattern, you will come to know that odd or even bit pattern frequency is lesser than original bit pattern frequency.

PROCEDURE:

1) Observe 'CLK' O/P, measure its frequency. This is nothing but 'Fb'.
2) Observe two patterns of pattern gen. & connect first pattern of I/P of O & E Data generator.
3) Observe 'O Data' & 'E Data' along with I/P pattern on DSO.
4) Connect 'O' Data to 'O' data pt. of 1495 Multiplier.
5) Connect 'E' Data to 'E' data pt. of 1495 Multiplier.
6) Observe 'SINE' & 'COS' waves & measure their frequencies. Also observe on 'XY' mode of DSO.
7) Observe pt. A with 'O Data', this is PSK of 'O Data'.
8) Observe pt. A with SINE wave on DSO. ('XY' mode)
9) Observe pt. B with 'E Data', this is PSK of 'E Data'.
10) Observe pt. B with COS wave on DSO. ('XY' mode)

11) Observe QPSK O/P with bit pattern & then with SINE wave. Press 'XY' mode & observe two ellipses.
12) Connect final O/P with the original I/P bit pattern.
13) Observe pt. 'C' & 'D'. These are odd & even bit patterns received at receiver.

OBSERVATION TABLE:

Sr. No.	Name of Signal	Amplitude	Frequency

CONCLUSION:

Staff Signature

Experiment No. 3

CO Addressed: CO3

AIM :-

To modulate and demodulate Frequency-hopping spread spectrum (FHSS).

EQUIPMENT: - FHSS Modem , DSO, connecting wires, probes.

THEORY:-

Frequency Hopping Spread Spectrum (FHSS) is a modulation technique and a method of transmitting radio signals in which the carrier frequency of a communication signal hops rapidly and randomly between predefined frequencies within a certain frequency band. FHSS is commonly used in wireless communication systems, such as Wi-Fi, Bluetooth, and some military and industrial applications.

1. Basic Principle: FHSS is designed to reduce interference and improve the security and reliability of wireless communication. It achieves this by spreading the signal over a wide frequency band and constantly changing the frequency at which the signal is transmitted. This makes it more resistant to narrowband interference and jamming.

2. Frequency Hopping: In FHSS, the transmitter and receiver both follow a predefined hopping sequence. At each time interval, typically a small fraction of a second, the transmitter switches to a different frequency within the specified band. The receiver follows the same hopping pattern to synchronize with the transmitter.

3. Pseudorandom Sequences: The hopping sequence is often generated using pseudorandom sequences, which are predictable by authorized receivers but appear random to unauthorized parties. This adds a level of security to the communication.

4. Interference Mitigation: FHSS is effective at mitigating narrowband interference because the signal quickly moves away from the interfering frequency. This helps maintain a more reliable and stable connection in noisy environments.

5. Robustness: FHSS is known for its robustness against various types of interference, including jamming and multipath fading. By hopping over a wide range of frequencies, the communication system can maintain connectivity even in challenging conditions.

6. Coexistence: FHSS is often used in systems where multiple devices or networks need to coexist in the same frequency band. Since FHSS hops between different frequencies, it reduces the chances of collisions and interference among multiple devices operating in the same area.

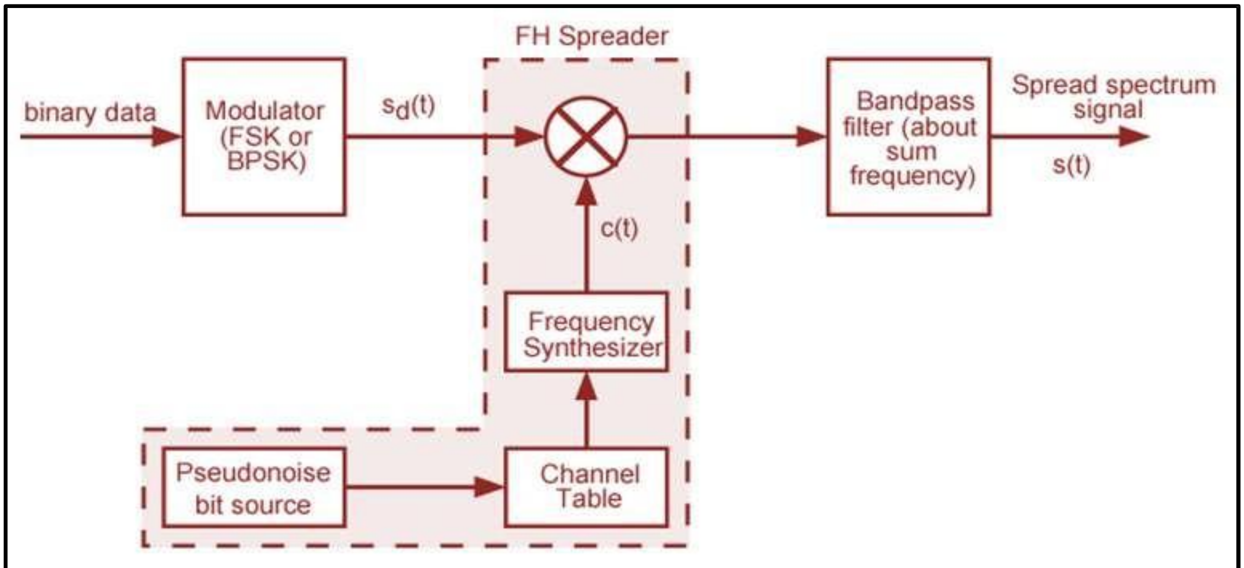
7. Applications: FHSS has been used in various applications, including:

- Wireless LANs (Local Area Networks): FHSS was one of the original modulation techniques used in early Wi-Fi standards (e.g., IEEE 802.11-1997).
- Bluetooth: Some early versions of Bluetooth used FHSS for communication.
- Military Communications: FHSS is employed in military radio systems to enhance security and reliability.
- Industrial Control Systems: FHSS is used in some industrial wireless communication systems for control and monitoring.
- Spread Spectrum Technologies: FHSS is part of a broader family of spread spectrum techniques, which also includes Direct Sequence Spread Spectrum (DSSS).

8. Drawbacks: FHSS requires precise synchronization between the transmitter and receiver. If synchronization is lost, it can lead to communication failures. Additionally, FHSS can be less efficient in terms of data transmission compared to other modulation techniques like DSSS.

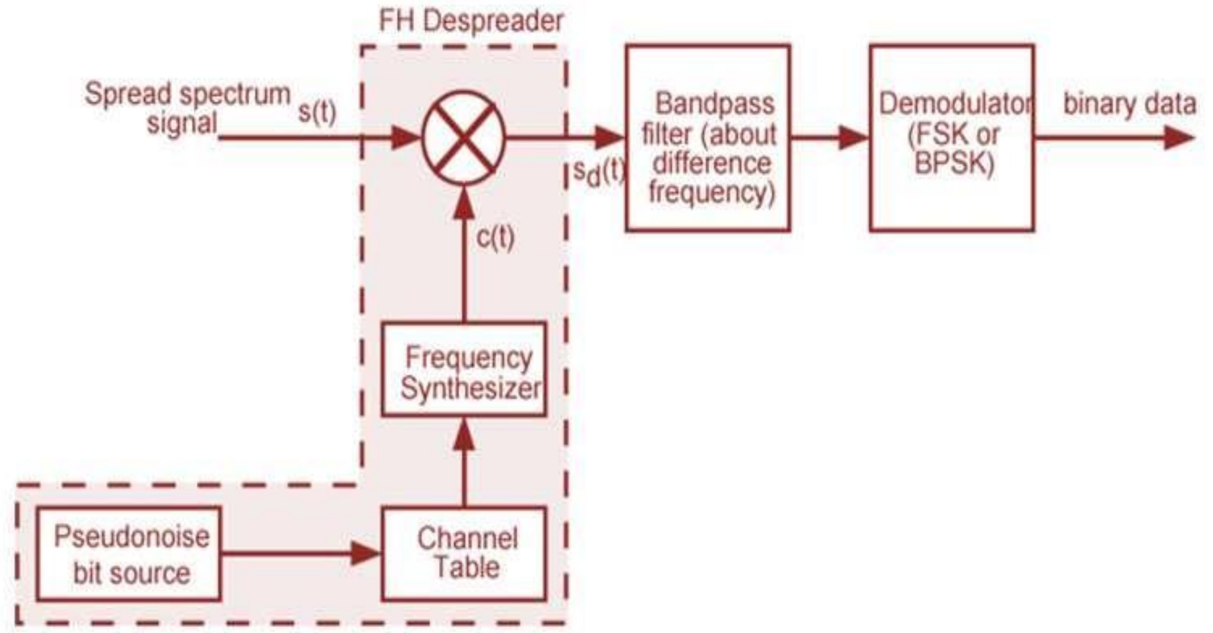
In summary, Frequency Hopping Spread Spectrum is a modulation technique that provides robustness and security in wireless communication systems by rapidly and randomly changing the carrier frequency of the transmitted signal. It has found applications in various domains due to its interference resistance and coexistence capabilities.

- **FHSS Transmitter :**

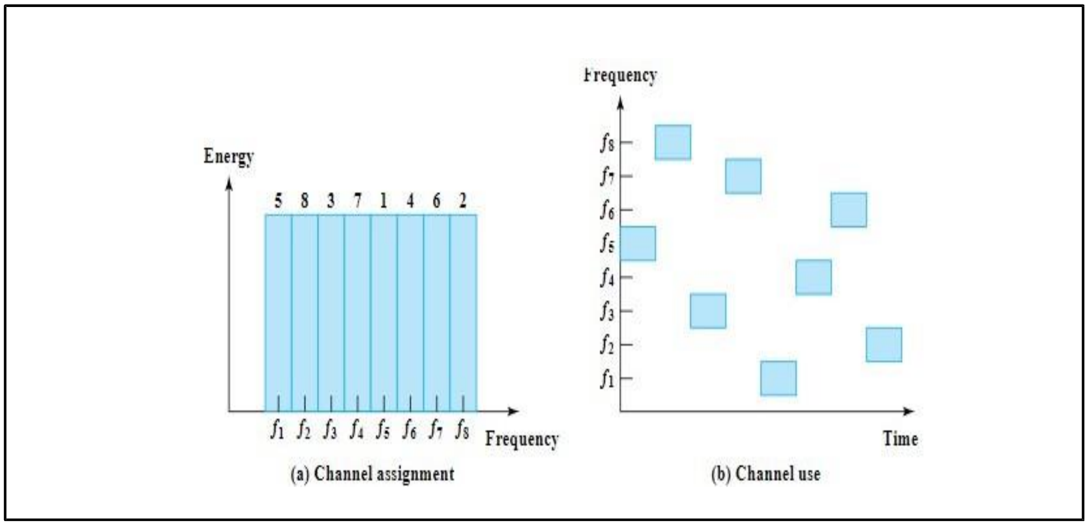


FHSS Transmitter

- **FHSS Receiver :**



FHSS Receiver



Frequency Spectrum

➢ Working :

In our kit we have provided two carrier frequencies fc1 & fc2. In practical systems no of frequencies are very large. But as it is a trainer kit for understanding purpose we have used only two frequencies.

The carrier freq. is determined by PN sequence. We have used 4-Bit PN sequence. It repeats after 2^4-1 = 15 clock cycles. If o/p of PN seq. is at logic '0' then fc1 is selected & if o/p of PN seq. is at logie '1' then fc2 is selected. This fel or fc2 is then generates FSK signal depending on logie level of i/p DATA pattern. We have provided 2 Bit patterns. FSK generator is designed using XR2206 & demodulator using XR 2211.

- **PROCEDURE**:-

1) Switch on the power supply.
2) Observe P1, P2 PN seq, o/p on CRO,

A] Now we first get four different frequency in FHSS signal

I) Keep pot of FHSS modulator near to minimum position (POT increases in clockwise direction). So keep pot to left most position Marking is given on panel.
II) Connect PN i/p of Modulator block to PN i/p of demodulator block. Connect PN i/p to GND.
III) Connect Pattern i/p to 5 V. Measure freq. of FHSS modulator o/p. (fcl-dF)
IV) Connect Pattern i/p to GND. Measure freq. of FHSS modulator o/p. (fcl+dF)
V) Connect PN ip to 5V.
VI) Connect Pattern ip to 5 V. Measure freq. of FHSS modulator o/p. (fc2-dF)
VII) Connect Pattern i/p to GND. Measure freq, of FHSS modulator o/p. (fc2+dF)

B]

I) First connect PN I/P of FHSS Modulator & FHSS Demodulator to GND i.e. logie '0'
II) Connect either P1 or P2 to Pattern i/p of FHSS Modulator.
III) observe o/p of FHSS Modulator. It is FSK at Fel.
IV) Observe exaggerated FSK by varying pot (Frequency Shift Pot) & then keep it at minimum position
V) Connect o/p of FHSS Modulator to FHSS Demodulator.
VI) Observe o/p of FHSS Demodulator & adjust POT 1 to get proper detected o/p [Do not vary POT 1 here onwards.]

C]

I) Now connect PN I/P of FHSS Modulator & FHSS Demodulator to '5V' logic '1' [All other connections unchanged.]
II) observe o/p of FHSS Modulator. It is FSK at Fc2
III) Observe o/p of FHSS Demodulator & adjust POT 2 to get proper detected o/p.

D]

I) Now connect PN I/P of FHSS Modulator to PN sequence o/p & PN I/P FHSS Demodulator Kept open
II) observe o/p of FHSS Modulator. We will get FSK at fcl or fc2 varying randomly.
III) Observe o/p of FHSS Demodulator. We will not get proper detected o/p as we have not connected PN seq. to FHSS Demodulator.
IV) Now Connect PN sequence to PN i/p of FHSS demodulator
V) Observe Detected o/p.

OBSERVATION TABLE:

Sr. No.	Name of Signal	Amplitude	Frequency

<u>CONCLUSION</u>:-

Staff Signature

Experiment No. 04

CO Addressed: CO3

AIM:-

To study the theory of direct sequence spread spectrum (DS-SS) communication.

EQUIPMENTS: - DS-SS PSK kit, DSO, patch cords, probes

THEORY: -

Spread spectrum techniques are used in military applications because of their high security & their susceptibility to interface from other parties.

Multiple users share the same bandwidth without significantly interfacing with each other. The spreading waveform is controlled by PN sequence which is binary random sequence .This PN is then multiplied with original baseband signal, which has a lower frequency, which yields a spread waveform that has a noise property. In receiver, the opposite happens, when pass band signal is first demodulated, and then dispreads using the same PN waveform. An imp factor is synchronization between two generated sequences.

Pseudo-noise (PN):

PN is the key factor in DS-SS systems. A pseudo noise sequence is a binary sequence with an autocorrelation that resembles, over a period. The autocorrelation that resembles over a period, the autocorrelation of random binary sequence (RAP 96) generated by a shift register and a combinational logic circuit as its feedback. The logic circuit determines the PN word. In this design i used the so called max length PN sequence. Its sequence of period 2m.1 generated by a linear feedback shift register, which has feedback logic of only modulo-2 adders (XOR gates). Some properties of maximum length sequence are as follows:

Properties OF PN Sequence:-

PN **sequences** have many of the properties possessed by a truly random binary sequence. A random binary sequence is a sequence in which the presence of binary symbol 1 or 0 is equally probable. The properties of PN sequences are as follows:

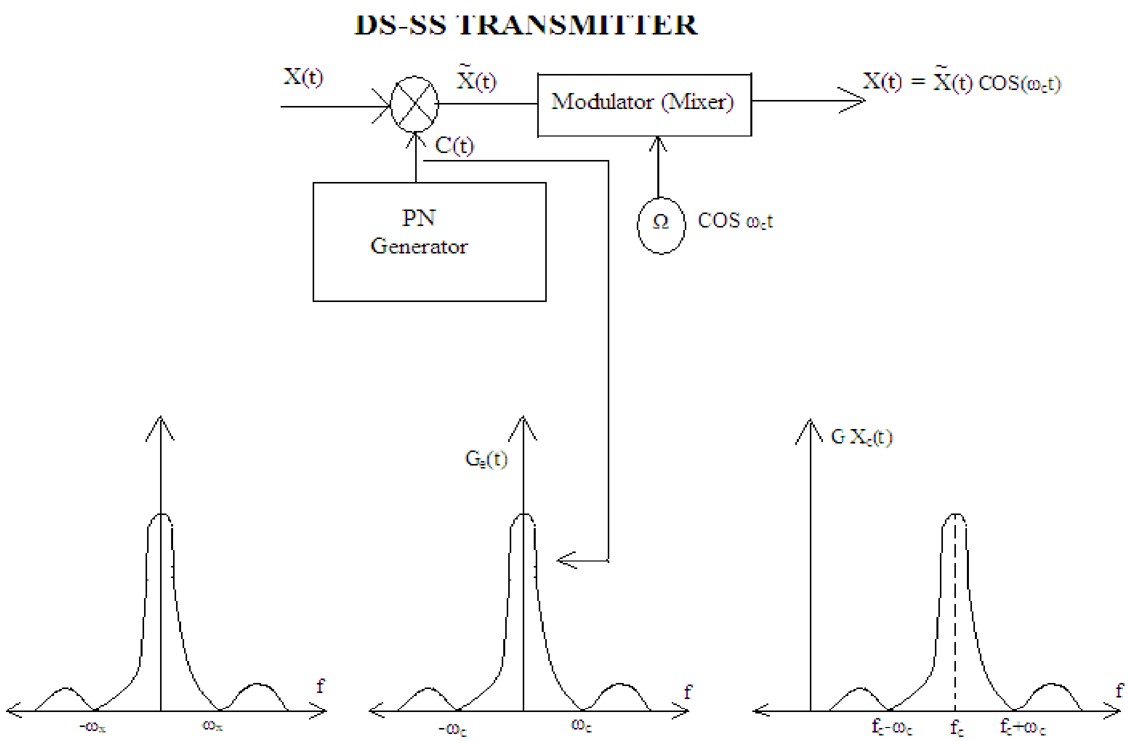
1. In each period of PN sequence, the no. of 1's is always one more than the no. of 0's. This property is called as ***balance property*.**
2. Among the runs of 1's and 0's in each period of PN sequence , one half the runs of each kind are of length one, one-fourth are of length two, one –eighth are of length three, and so on as long as these fractions represent meaningful numbers of runs. This property is called as ***run property.***
3. **T**he autocorrelation function of a PN sequence is periodic and binary-valued. This property is called the ***correlation property.***

Direct sequence spread spectrum:

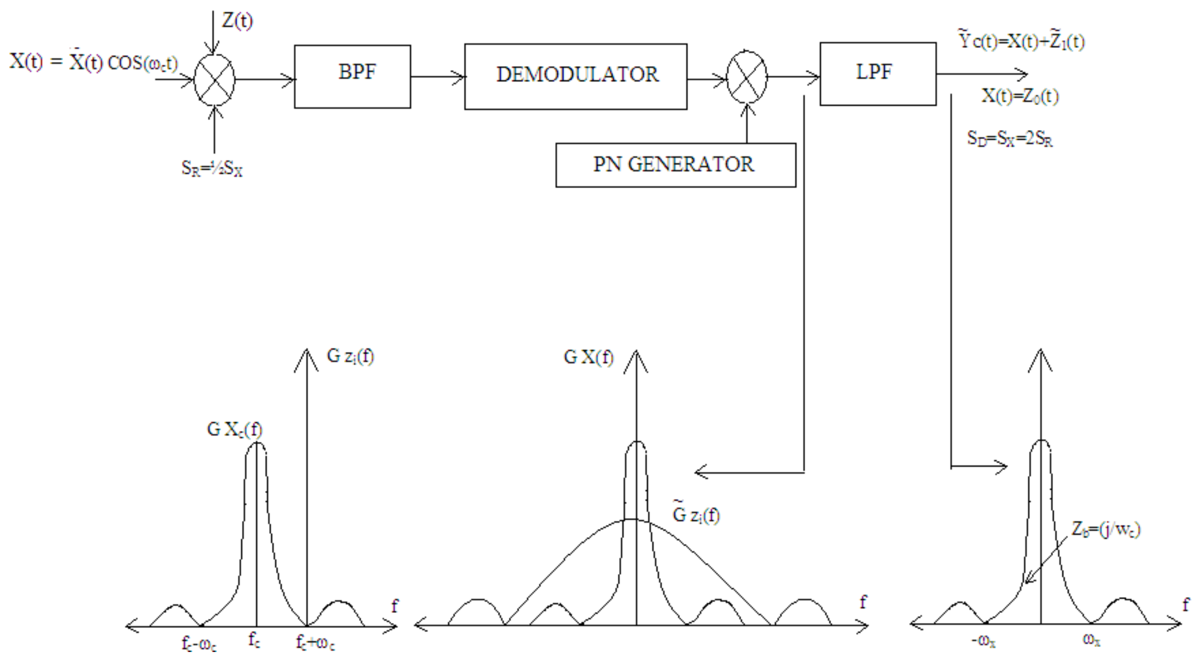
The baseband waveform is multiplied by PN sequence. The PN sequence is produced using a PN generator, frequency of which is higher than data signal. This generator consists of shift register & the logic circuit that determines the PN signal. After spreading the signal true signal is modulated & transmitted. The most widely modulation scheme is **BPSK**. The equation that represents the DS-SS signal & is shown in equation & the block diagram is shown:

$$S_{ss} = \sqrt{(2E_s/T_s)} \times m(t)p(t) \cos(2\pi f_c t + \varphi)$$

Where, m(t) is data sequence p(t) is pn spreading sequence f_c is the carrier frequency & is the carrier phase angle of t=0. Each symbol is m(t) represents a data of symbol & has duration of



DS-SS RECEIVER



T_b. Each symbol is m(t) represents a data of symbol & has a duration of Tc. The transmission of data symbols and chips coincide such that the ratio Ts to Tc is an integer. Here we notice that the higher frequency of the spreading signal p(t). The resulting spread signal is then modulated using BPSK scheme. The carrier frequency fc should have a frequency at least 5 times the chip frequency p(t).In the demodulator section, we simply reverse the process. We demodulate the BPSK signal first, Low Pass Filter the signal, & then Dispread the filtered signal, to obtain the original massage. The process is described by following equations:

$$M(t)=Sss(t)*cos(2piefct+Q)$$

In demodulation design, we simply reverse the process. The signal is filtered to obtain the original message.

PROCEDURE: -

1) Connected CRO ch1 at clock (W_k) socket & observed it.
2) Connected CRO ch2 at the bit clock (B_k) socket & observed it.
3) Connected CRO ch1 at RF carrier socket & observed the waveform.
4) Connected CRO ch1 NRZ data socket & observed it.
5) Pushed on start switch & observed PN signal.
6) Observed spreading signal.
7) Observed DS-SS signal with reference the NRZ data & spreading code signal.
8) Observed recovered spreading code at o/p of DS-SS modulator.
9) Observed the filter spreading code at o/p of LPF. Keep noise level at minimum.
10) Observed recovered pure NRZ data at o/p of comparator.

OBSERVATION TABLE:

Sr. No.	Name of Signal	Amplitude	Frequency

<u>CONCLUSION</u>: -

Staff Signature

Experiment No. 05

CO Addressed: CO3

AIM:- Simulation study of Performance of M-ary PSK.

EQUIPMENTS: - MATLAB software

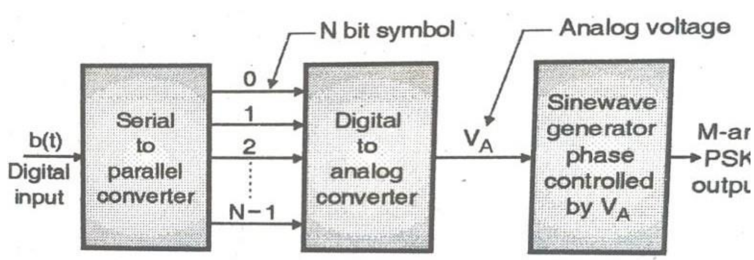
THEORY: -

M-ary Phase Shift Keying or MPSK is a modulation where data bits select one of M phase shifted versions of the carrier to transmit the data . Thus, the M possible waveforms all have the same amplitude and frequency but different phases. The signal constellations consists of M equally spaced points on a circle
Generation : Group 'N' successive bits together to form N bits symbol.Each symbol will extend over a period of N T_b where T_b is the duration of one bit. Due to grouping we have 2^N=M symbols.These M symbols are represented by sinusoidal signals of duration T_s=NT_b having phase shift of 2 π/M radians.
Thus the M-ary PSK waveform can be mathematically represented as

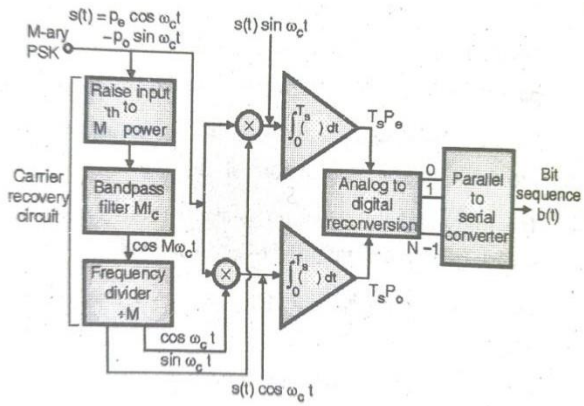
$$V_{M\text{-ary}} = \sqrt{2Ps}\cos(w_c t+\phi_m) \qquad \text{where } m=0,1,2,\ldots(M-1)\ ; \qquad \phi_m=(2m+1)\frac{\pi}{M}$$

M-ary PSK Transmitter :

The bit stream b(t) is applied to serial to parallel converter. This can store N bits of symbol. Output of serial to parallel converter remains unchanged for NTb duration. N bit output is then applied to D/A converter. This N bits are converted into an analog signal V. D/A converter can have 2^N=M number of distinct values , corresponding to the M symbols. Finally this analog voltage is applied to a sinusoidal signal generator ,which produces constant amplitude sinusoidal output voltage , the phase ϕm of which has one to one correspondence to N bit symbols. The phase will change once per symbol time Ts=NTb. Thus M-ary PSK is generated



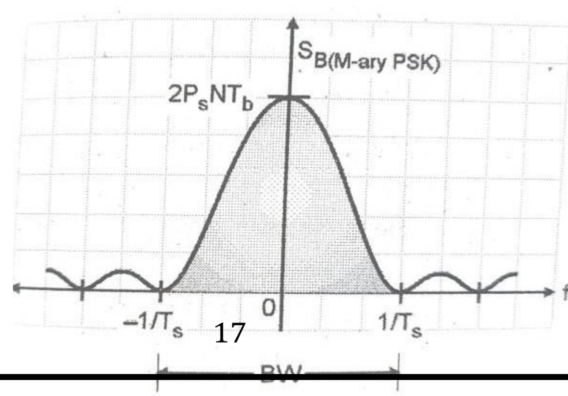
M-ary PSK Reciever :

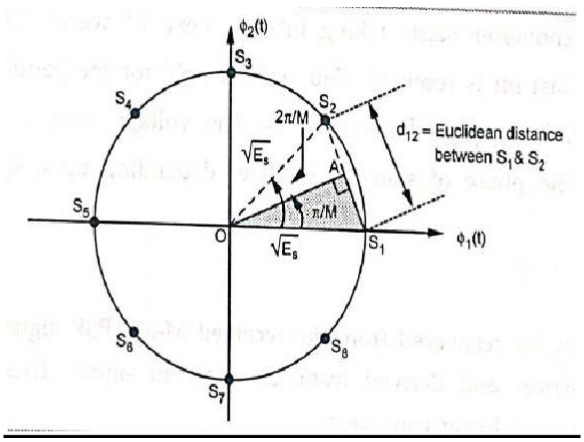


The M-ary receiver works on principle of synchronous demodulation as in BPSK and QPSK. The carrier recovery system will require a device which can raise received signal to M^{th} power. This signal is then applied to band pass filter. At filter output we get sinusoidal signal at frequency Mfc i.e M^{th} harmonic of fc. The frequency is then divided by M to obtain carrier at frequency f_c. Two carriers produced at filter output are $\cos w_c t$ and $\sin w_c t$ which are applied to two multipliers. The other input to multiplier is connected to received M-ary PSK signal. Output is then given to integrators. Since M-ary PSK is non-offset type system, integrators will extend integration over same time interval. The outputs of integrators are proportional to $T_s p_e$ and $T_s p_o$ respectively and they change symbol rate . output is connected to A to D converter which yields N bit transmitted signal. These signal is converted into b(t) using parallel to serial converter. Now the operating systems with N=4 and M=16 are common.

$$\text{BandWidth} = \frac{2fb}{N} = \frac{2fb}{4} = \frac{fb}{2}$$

$$S_{B(\text{M-ary PSK})}(f) = 2P_sNT_b \left(\frac{\sin(\pi f N T_b)}{\pi f N T_b}\right)^2$$





CONCLUSION: -

Staff Signature

Experiment No. 06

CO Addressed: CO3

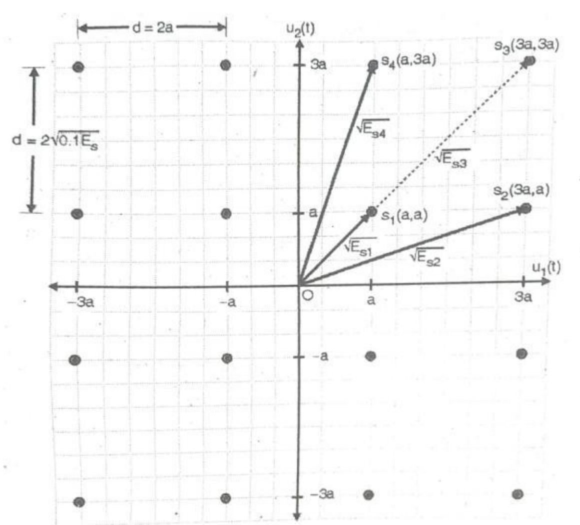
AIM:- To study the theory of QASK (Quadrature Amplitude Shift Keying).

EQUIPMENTS: - DSO, patch cords, probes

THEORY: -

Symbols transmitted using BPSK,QPSK or M-ary PSK are of same amplitude. Also noise immunity will improve if the signal vector differ not only in phase but also in amplitude. Such a system is called amplitude and phase shift keying system. In this system the direct modulation of carriers in quadrature is involved, therefore this system is called as quadrature amplitude phase shift keying

Let us assume that using QASK we want to transmit a symbol consisting of 4 bits. That means N=4 and there are 16 different possible symbols. Hence the QASK system should be able to generate 16 different distinguishable signals . A possible geometric representation of 16 signals is as shown



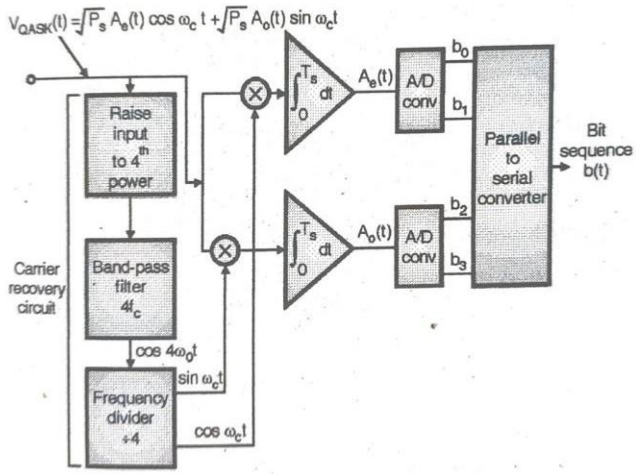
QASK Transmitter :

The bit stream b(t) is applied to a serial to parallel converter operating on a clock which has period of Ts sec. the four signals are b_{k+3} , b_{k+2}, b_{k+1}, b_k. out of these four bits first two are applied to first D/A converter and other two to second D/A converter.output of first converter is

$A_e(t)$ used to modulate carrier $\sqrt{Ps}\cos wct$ and second converter gives $A_o(t)$ used to modulate $\sqrt{Ps}.\sin wct$ with help of balanced modulators. balanced modulators outputs are added together to get QASK signal which is expressed as

$$V_{QASK} = A_e(t)\sqrt{Ps}\cos(wct) + A_o(t)\sqrt{Ps}\sin(wct)$$

QASK Reciever :



Like QPSK this is also synchronous demodulation which requires locally generated set of quadrature carriers(90 phase shift). The input QASK signal is first raised to fourth poer and using band pass filter with centre frequency 4fc alongside with frequency divider(÷4). i.e

$$V_{QASK}^{4}(t) = [Ps]^2\,[A_e(t)\cos(wct) + A_o(t)\sin(wct)]^4$$

Then passed through BPF with freq. 4fc so we neglect all other terms expect which have freq 4fc

$$V_{QASK}^{4}(t) = \frac{Ps}{8}\,[A_e^{\,4}(t) + A_o^{\,4}(t) - 6A_e^{\,2}(t)\;A_e^{\,2}(t)]\cos 4wct \;+$$

$$\frac{Ps}{2}\,[A_e(t) + A_o(t)[\,A_e^{\,2}(t) - A_o^{\,2}(t)]\sin 4wct$$

The two balanced modulators are used with two integrators to recover the signals $A_e(t)$ and $A_o(t)$. Both integrate over period Ts. Finally the original bits are obtained from $A_e(t)$ and $A_o(t)$

By using A/D converters .outputs of converters are applied to serial to parallel converter to obtain sequence b(t)

$$BW = f_s - (-f_s) = 2\,f_s = \frac{2}{Ts} = \frac{2}{NTb}$$

CONCLUSION: -

Staff Signature

EXPERIMENT NO: 7

Aim: Simulation study of OFDM transmitter and receiver.

CO Addressed: CO2,CO3

Theory: The set of 4G wireless standards is based on the revolutionary new technology of Orthogonal Frequency Division Multiplexing (OFDM). Orthogonal Frequency Division Multiplexing forms the basis of fourth generation wireless communication systems. OFDM is used in 4G wireless cellular standards such as Long Term Evolution (LTE) and WiMAX (Worldwide Interoperability for Microwave Access). OFDM is a keyword broadband wireless technology which supports data rates in excess of 100 Mbps. OFDM is a combination of modulation and multiplexing. In this technique, the given resource (bandwidth) is shared among individual modulated data sources. The multiple access technology based on OFDM is termed Orthogonal Frequency Division for Multiple Access (OFDMA).

Necessity of OFDM:

OFDM is very effective for communication over channels with frequency selective fading (different frequency components of the signal experience different fading). It is very difficult to handle frequency selective fading in the receiver, in which case, the design of the receiver is hugely complex. Instead of trying to mitigate frequency selective fading as a whole (which occurs when a huge bandwidth is allocated for the data transmission over a frequency selective fading channel), OFDM mitigates the problem by converting the entire frequency selective fading channel into small flat fading channels (as seen by the individual subcarriers). Flat fading is easier to combat (when compared to frequency selective fading) by employing simple error correction and equalization schemes.

An OFDM system is defined by IFFT/FFT length – N, the underlying modulation technique (BPSK/QPSK/QAM), supported data rate, etc. The FFT/IFFT length N defines the number of total subcarriers present in the OFDM system.

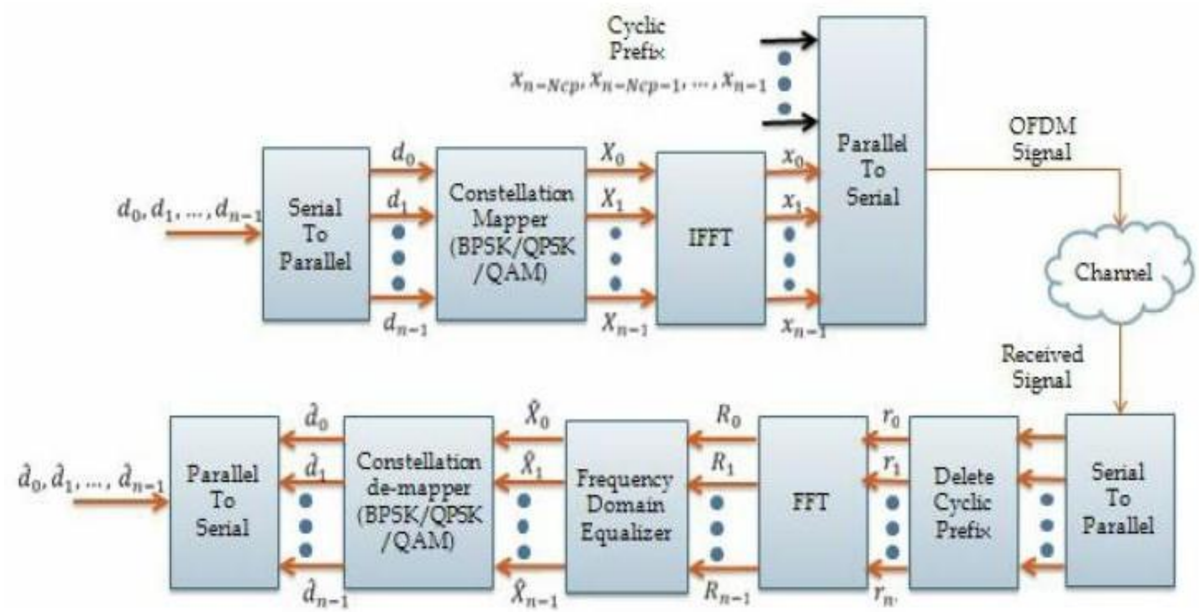


Fig. 1 Model for simulating the OFDM system

In the figure, symbols represented by small case letters are assumed to be in time domain, whereas the symbols represented by uppercase letters are assumed to be in frequency domain. Consider data bits D = {d_0,d_1,d_2,…). Select number of subcarriers required to send the given data. As a generic case, assume N subcarriers. The data (D) is first converted from serial stream to parallel stream depending on the number of sub-carriers (N). Decide digital modulation techniques such as BPSK/ QAM. Apply IFFT algorithm on generated symbols and add cyclic prefix bits . Finally, the resultant output from the N parallel arms are summed up together to produce the OFDM signal. The channel in this case is modeled as a simple AWGN channel. Since the channel is considered to be an AWGN channel, there is no need for the frequency domain equalizer in the OFDM receiver as shown in Fig. 1. Reverse the process at the receiver side. Compare the transmitted and received bits to compute bit error rate.

<u>CONCLUSION</u>: -

Staff Signature

EXPERIMENT NO: 8

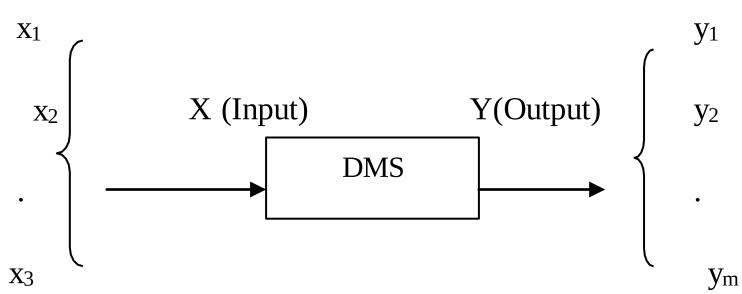
TITLE: Simulation study of various Entropies and mutual information in a communication system

CO Addressed: CO4
THEORY:

Discrete Communication Channel (DCC)

Channel Representation:

A communication channel is the path or medium through which the symbols flow to the receiver. A discrete channel (DMC) is a statistical model with an input X and an output Y. During each unit of the time (signaling interval), the channel accepts an input symbol from X, and in response it generates an output symbol from Y. The channel is "**discrete**" when the alphabets of X and Y are both finite. It is "**Memory-less**" when the current output depends on only the current input and not on any of the previous inputs.



A diagram of a DMC with m inputs and n outputs. The input X consists of input symbols x_1, x2, x3,.x_m. The a priori probabilities of these source symbols $P(x_i)$ are assumed to be known. The output Y consists of output symbols y_1, y2, ...y_n.

Channel Matrix:

Channel Transition Probability

Each possible input-to-output path is indicated along with a conditional probability $P(y_j/x_i)$, where $P(y_j/x_i)$ is the conditional probability of obtaining output y_j given that the input is x_i, and is called channel transition probability.

A channel is completely specified by the complete set of transition probabilities.

$$[P(Y/X)] = \begin{pmatrix} P(y_1/x_1) & P(y_2/x_1) & \ldots\ldots & P(y_n/x_1) \\ P(y_1/x_2) & P(y_2/x_2) & \ldots\ldots & P(y_n/x_2) \\ . & . & . & \\ . & . & . & \\ P(y_1/x_m) & P(y_2/x_m) & \ldots\ldots\ldots & P(y_n/x_n) \end{pmatrix}_{m*n}$$

Matrix of channel transition probabilities i.e. [P(Y/X)] is called the channel matrix.

Important property of [P(Y/X)] matrix: Since each input to the channel results in some output, each row of the channel matrix must sum to unity, that is

$$\sum_{j=1}^{n} p\left(\frac{yj}{xi}\right) = 1 \; for\ all\ i$$

Mutual Information

Mutual information is the net amount of information that passes through the discrete communication channel. The mutual information I(X;Y) of a channel is defined by

$$I(X; Y) = H(X) – H(X/Y) \text{ bits/ Symbol}$$

We know that in DCC, the channel output Y is a noisy version of the channel output. Where H(X) represents the uncertainty about channel input before the channel output is observed and

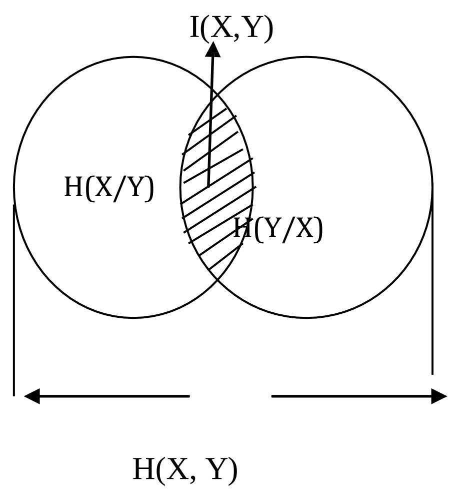
H(X/Y) represents the uncertainty about the channel input after the channel output is observed. The difference H(X)-H(X/Y) must represent our uncertainty about channel input that is resolved by observing the channel output. This important quantity is called mutual information and is represented by I(X;Y).

Properties of mutual information:

(a) Mutual information is always symmetric, i.e. $I(X;Y) = I(Y;X)$

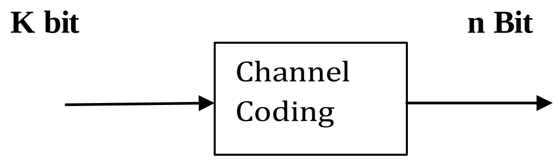
(b) Mutual information is always nonnegative , $I(X;Y) \geq 0$

(c) Mutual information of a channel is related to the joint entropy of the channel input and channel output by equation $I(X;Y) = H(X) + H(Y) – H(X,Y)$



Channel Coding: Channel coding consist of mapping the incoming data sequence into a channel input sequence and inverse mapping the channel output sequence into an output data sequence in such a way that the overall effect of channel noise on the system is minimized. The first mapping operation is performed in the transmitter by a channel encoder, whereas the inverse mapping operation is performed in the receiver by a channel decoder.

The channel encoder and channel decoder are both under the designer’s control and should be designed to optimize reliability of the communication system. The approach taken is to introduce redundancy in the channel encoder so as to reconstruct the original source sequence as accurately as possible. We may view channel coding as the dual of source coding in that former introduces controlled redundancy to improve reliability, whereas the latter reduces to improve efficiency.

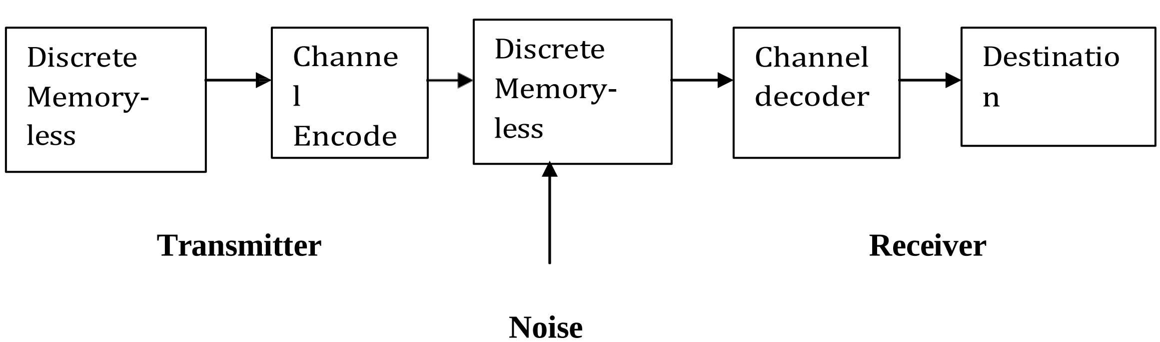


The number of redundant bits added by the encoder to each transmitted block is n-k bits. The ratio k/n is called the code rate and is denoted by r. where r is code rate and is less than unity.

The accurate reconstruction of the original source sequence at the destination requires that the average probability of symbol error be arbitrarily low. This raises the following important question: does there exist a channel coding scheme such that the probability that a message bit will be in error is less than any positive number £(i.e., as small as we want it), and yet the channel coding scheme is efficient in that the code rate need not be small. The answer to this fundamental question is an emphatic "Yes". Indeed, the answer to the question is provided by Shannon's second theorem in terms of the channel capacity Cs.

Consider DMS, has the source alphabet x and entropy H(x) bits per source symbol. If the source emits symbols once every T_s seconds. Hence, the average information rate of the source is H(x)/ Seconds. The decoder delivers decoded symbols to the destination from the source alphabet x and at the same source rate of one symbol every T_s Seconds.

The discrete memory less channel has a channel capacity equal to Cs bits per use of the channel. We assume that the channel is capable of being used once every T_c Seconds. Hence, the channel capacity per unit time is Cs/ T_c bits per second, which represents the maximum rate of information transfer over the channel.



Shannon's Second Theorem (Channel Coding Theorem):

The channel coding theorem for a discrete memory-less channel is stated in two parts as follows.

(i) Let a discrete memory-less source with an alphabet x have entropy H(x) and produce symbols once ever T_s seconds. Let a discrete memory-less channel have capacity Cs and be used once every T_c Seconds. Then, if

$$H(x)/ T_s \leq Cs/ T_c$$

There exists a coding scheme for which the source output can be transmitted over the channel and be reconstructed with an arbitrarily small probability of error. The parameter of Cs/ T_c is called the critical rate. When equation is satisfied with the equality sign the system is said to be signaling at the critical rate.

(ii) Conversely, if

$$H(x)/ T_s > Cs/ T_c$$

It is not possible to transmit information over the channel and reconstruct it with an arbitrarily small probability of error.

The channel coding theorem is the single most important result of information theory. The theorem specifies the channel capacity Cs as a fundamental limit on the rate at which the transmission of reliable error –free messages can take place over a discrete memory-less channel.

Limitation of Channel Coding Theorem:

(a) The channel coding theorem does not show how to construct a good code. Rather, the theorem should be viewed as an existence proof in the sense that it tells us that if the condition is satisfied, then good code do exist.

(b) The theorem does not have a precise result for the probability of symbol error after decoding the channel output. Rather, it tell us that the probability of symbol error tends to zero as the length of the code increases, again provided that the condition is satisfied.

Important Formulae:

1) $P(x_i)=\{p(x_1), p(x_2), p(x_3)\ldots\ldots p(x_i)\ldots\ldots p(x_m)\}$
2) $P(y_j)=\{p(y_1), p(y_2), p(y_3)\ldots\ldots p(y_j)\ldots\ldots p(y_n)\}$
3) $P(x_i, y_j)= P(x_i/y_j).p(y_j)$

   $= P(y_j/x_i).p(x_i)$
4) $H(X) = \sum_{i=1}^{m} P(Xi)\log_2(1/P(Xi))$
5) $H(Y) = \sum_{j=1}^{n} P(Yj)\log_2(1/P(Yj))$
6) $H(X,Y) = \sum_{i=1}^{m}\sum_{j=1}^{n} P(Xi, Yj)\log_2(1/P(Xi, Yj))$
7) $H\left(\frac{X}{Y}\right) = \sum_{1}^{m}\sum_{1}^{n} P(xi, yj)\log(1/(P(xi/yj))$
8) $H\left(\frac{y}{x}\right) = \sum_{1}^{m}\sum_{1}^{n} P(xi, yj)\log(1/(P(yj/xi))$
9) $H(X,Y) = H\left(\frac{X}{Y}\right) + H(Y)$
10) $H(X,Y) = H\left(\frac{Y}{X}\right) + H(X)$
11) $I(X,Y) = H(X) - H\left(\frac{X}{Y}\right)$ **or** $I(X,Y) = H(Y) - H(Y/X)$
12) $I(X,Y) = H(X) - H\left(\frac{X}{Y}\right)$ **or** $I(X,Y) = H(Y) - H(Y/X)$
13) $I(X,Y) = \sum_{i=1}^{m}\sum_{j=1}^{n} P(Xi, Yj)\log_2(P(Xi/Yj)/P(xi))$
14) $I(y,x) = \sum_{i=1}^{m}\sum_{j=1}^{n} P(Xi, Yj)\log_2(P(yj/xi)/P(yi))$
15) $P(x_i) = \sum_{1}^{m} P(xi, yj)$
16) $P(y_j) = \sum_{1}^{n} P(xi, yj)$

Algorithm

1. Read out the channel matrix from the user.

   Read out m & n from the user. Where,

   m- no. of inputs to DMC. (No of rows of channel Matrix)

   n-no. of outputs of DMC.(No. of columns of channel matrix)

2. Read out elements of the channel matrix (transition probabilities).
3. Identify the type of matrix.
   - If it is joint probability matrix, then addition of all elements should be equal to 1.
   - If it is not joint probability matrix then there is a possibility of conditional probability P(Y/X) either P(X/Y).
   - If it is a conditional P(Y/X) then individual row wise addition must be 1.
   - If individual column wise addition is 1, then it is P(X/Y) matrix. Finally print the matrix.
4. If the matrix is joint probability matrix go to next step. But if matrix is conditional then convert it into joint probability matrix.
   - If matrix is P(Y/X) then take input probability distribution from user $P(X_0)$, $P(X_1)$, $P(X_2)$,…….$P(X_{m-1})$.

$$\sum_{i=0}^{m-1} (xi) = 1$$

   & after that you multiply $P(X_0)$ to the first row of P(Y/X) & second row of P(Y/X) to the $P(X_1)$. Similarly multiply $(m-1)^{th}$ row of P(Y/X) to the $P(X_{m-1})$.
   Resultant matrix is joint probability matrix.
   - If matrix is P(X/Y) then take $P(Y_0)$, $P(Y_1)$, $P(Y_2)$……..$P(Y_{n-1})$ from user. Multiply first column with $P(Y_0)$, second column with $P(Y_1)$, & last column with $P(Y_{n-1})$ &resultant matrix is P(X, Y).
5. If the matrix is joint probability matrix, the row wise addition of joint probability matrix is $P(X_0)$, $P(X_1)$, .$P(X_{m-1})$. &column wise addition of joint probability matrix is $P(Y_0)$, $P(Y_1)$,…..$P(Y_{n-1})$.
6. By using P(X) value calculates H(X).

$$H(X) = \sum_{i=0}^{m-1} P(Xi) \log_2(1/P(Xi))$$

   By using P(Y) values calculate H(Y).

$$H(Y) = \sum_{j=0}^{n-1} P(Yj) \log_2(1/P(Yj))$$

After calculating H(X) & H(Y) which is a transmission (input)

& reception (output) entropies, calculate joint entropy.

$$H(X,Y) = \sum_{i=0}^{m-1} \sum_{j=0}^{n-1} P(Xi, Yj) \log_2(1/P(Xi, Yj))$$

7. Calculation of conditional entropies:

- $H(X,Y) = H\left(\frac{X}{F}\right) + H(Y)$
- $H\left(\frac{X}{F}\right) = H(X,Y) - H(Y)$
- $H(X,Y) = H\left(\frac{F}{X}\right) + H(X)$
- $H\left(\frac{F}{X}\right) = H(X,Y) - H(X)$

8. Calculation for mutual information:

   $(X,Y) = H(X) - H\left(\frac{X}{F}\right)$ or $I(Y,X) = H(Y) - H(Y/X)$

9. Channel capacity:

$$Cs = \max_{\{P(Xi)\}} I(X,Y)$$

Maximization of mutual information is called Channel capacity C_S where maximization is possible on probability distribution P(X)={$P(X_0)$,$P(X_1)$,…..... $P(X_{m-1})$}

&

$$\sum_{i=0}^{m-1} (Xi) = 1$$

10. Read out the channel rate from the user. $r_c=1/T_C$ no. of symbols/sec. Where r_c is no. of symbol transmitted (passes) by the Communication channel

$$C = r_c * C_s = C_s/T_c$$

Conclusion:-

EXPERIMENT NO: 9

TITLE: Write a program for generation and evaluation of variable length source coding using Huffman coding.

CO Addressed: CO5

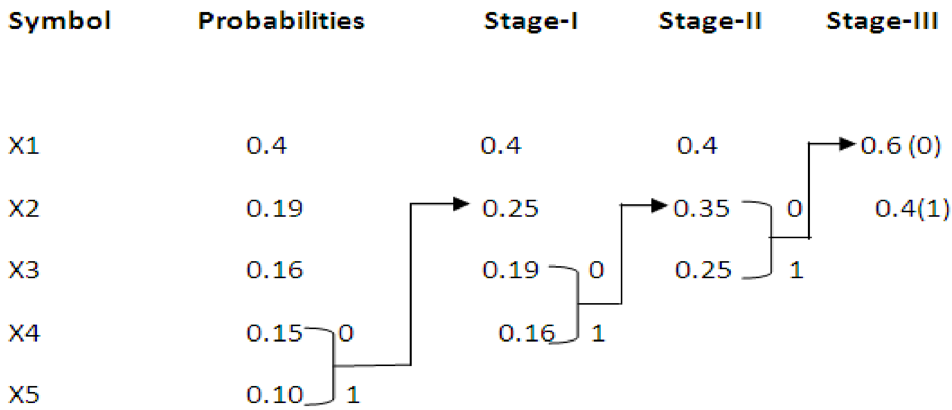
THEORY:

Huffman Coding:

Huffman code is an important class of Huffman code. The basic idea behind Huffman coding is to assign to each symbol of an alphabet of bits roughly equal in length to the amount of information conveyed by the symbol in question. The end result is a source code whose average codeword length approaches the fundamental limit set by the entropy of a discrete memory-less source , namely, H(X).The essence of the algorithm used to synthesize the Huffman code is to replace the prescribed set of source statistics of a discrete memory-less source with a simpler one. This reduction process is continued in a step-by-step manner until we are left with a final set of only two source statistics (symbols) , for which (0,1) is an optimal code. Starting from this trivial code, we then work backward and thereby construct the Huffman code for the given source.

Numerical

1) A DMS has 5 symbols x1 x2 x3 x4 x5 with probabilities 0.4,0.19,0.16,0.15,0.10.Calculate the code efficiency



Symbol	Probabilities	Source code	Source code Length (l_k)
X1	0.4	1	1
X2	0.19	000	3
X3	0.16	001	3
X4	0.15	010	3
X5	0.10	011	3

$$H(X) = 2.3125$$

$$\bar{L} = 2.3125$$

$$\% \eta = H(X)/\bar{L} \times 100$$

$$\eta = 100\%$$

- Perform the practical to evaluate variable length source coding using Huffman coding using C Programming or Matlab.

Huffman Encoding Algorithm

1. The source symbols are listed in order of decreasing probability. The two source symbols of lowest probability are assigned a 0 and a 1. This part of the step is referred to as a splitting stage.
2. These two source symbols are regarded as being combined into a new source symbol with probability equal to the sum of the two original probabilities. (The list of source symbols, and therefore source statistics, is thereby reduced in size by one). The probability of the new symbol is placed in the list in accordance with its value.
3. The procedure is repeated until we are left with a final list of source statistics (Symbols) of only two for which a 0 and a 1 are assigned.

   The code for each (original) source symbol is found by working backward and tracing the sequence of 0s and 1s assigned to that symbol as well as its successors.

Conclusion:

Staff Signature:

EXPERIMENT NO: 10

AIM: Write a Program for coding & decoding of Linear block codes

CO Addressed: CO5

THEORY:

Error Correcting Codes

- The basic idea behind error correcting codes is to add some redundancy in the form of extra symbol to a message prior to its transmission through a noisy channel. This redundancy is added in a controlled manner.
- The encoded message when transmitted might be corrupted by noise in channel.
- At the receiver, the original message can be recovered from the corrupted one if the numbers of errors are within the limit of which the code has been designed.

Objective of a good error control coding scheme

1. Error correcting capability in terms of the number of errors that it can rectify.
2. Fast & efficient encoding of the message.
3. Fast and efficient decoding of the received message.
4. Maximum transfer of information bits per unit time (i.e. fewer overheads in terms of redundancy).

The first objective is the primary one. In order to increase the error correcting capability of a coding scheme one must introduce more redundancies. However, increased redundancy leads to a slower rate of transfer of the actual information. Thus the objective (1) & (4) are not totally compatible. Also, as the coding strategies become more complicated for correcting larger number of errors, the objectives (2) & (3) also become difficult to achieve.

Basic Definitions:

- **Word**: A Word is a sequence of symbols.
- **Code**: A Code is a set of vectors called code words.

- **Hamming Weight**: The Hamming Weight of a codeword is equal to the number of nonzero element in the codeword.
  - The Hamming Weight of a codeword c is denoted by w(c).
- **Hamming Distance**: The hamming distance between two code words is the number of places the code words differ. The Hamming distance between two codeword c1 & c2 is denoted by $d(c_1,c_2)$

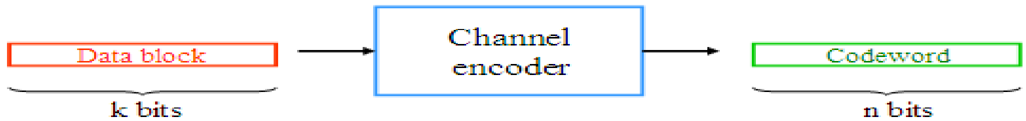
  i.e. $d(c_1,c_2) = w(c_1-c_2)$
- **Block Code**: A Block code consists of a set of fixed length code words.
- **Block Length**: The fixed length of these code words is called the Block Length and is typically denoted by n.
- **Code Rate**: The code rate of an (n, k) code is defined as the ratio (k/n) ,and is denotes the fraction of the codeword that consist of information symbol.
- **Minimum Distance**: The minimum distance of a code is the minimum hamming distance between any two code words.
- **Minimum Weight**: The minimum weight of a code is the smallest weight of any non-zero codeword.
- **For a linear code the minimum distance is equal to the minimum weight of the code. i.e.** $d_{min} = d^* = w_{min} = w^*$

Linear Code

- A linear code has the following properties.
  - The sum of two codeword belonging to the code is also a codeword belonging to the code.
  - The all-zero word is always a codeword.
  - The minimum hamming distance between two codeword of a linear code is equal to the minimum weight of the code. i.e. $d^* = w^*$.

Note that if the sum of two codeword is another codeword, the difference of two codeword will also yield a valid codeword.

Linear Block Code:



$n\text{-}k$ Redundant bits

$R_c = \frac{k}{n}$ Code rate

Generator Matrix:

- The Generator matrix provides a concise and efficient way of representing a linear block code. The generator matrix converts (encodes) a vector of length k to a vector of length n. The generator matrix will be k x n matrix with rank k.
  $G = [I_{k \times k} \quad P_{k \times n-k}]_{k \times n}$
  Where I is identity matrix size is k x k, P is parity matrix and size is k x n-k , Size of G matrix is k x n

Systematic block code (n,k)

For a systematic code, the first (or last) k elements in the codeword are information bits.

Encoding in (n,k) block code

- C = mG or dG
  - Where c is called the codeword and d or m is called the information word.
  - Where C is 1 x n vector, d is 1 x k vector and G is generator matrix.
  - C= $(c_1, c2, \ldots\ldots c_n) = (m_1, m2,..m_k \; p_1, p_2, \ldots p_{n-k})$. o

- Note: Mod 2 arithmetic: Multiplication similar to AND operation, addition (subtraction) similar to EX-OR operation.

Linear Block Coding Example:

Consider (7,4) LBC with the generator matrix

G = 1 0 0 0 1 1 0

0 1 0 0 0 1 1

0 0 1 0 1 1 1

0 0 0 1 1 0 1

Find out all possible code vectors ,error detection & correction capability

Message Word (Length k bits)	Code Vector C = mG (Length n bits)	Weight of code vectors (Wc)	Remarks
0000	0000000	0	Zero Code Vector
0001	0001101	3	All remaining code vectors are non zero Code vector. d_{min} = Minimum weight of non zero code vector = 3 (All these code vector are transmitted by transmitter it is also called valid code vector and it is equal to 2^k , where k is message length)
0010	0010111	4	
0011	0011010	3	
0100	0100011	3	
0101	0101110	4	
0110	0110100	3	
0111	0111001	4	
1000	1000110	3	
1001	1001011	4	
1010	1010001	3	
1011	1011100	4	
1100	1100101	4	
1101	1101000	3	
1110	1110010	4	
1111	1111111	7	

Minimum Distance: d_{min} = Minimum weight of non zero code vectors =3

Error Detection Capabilities of code $t_d \leq d_{min} - 1 = 2$

Error Correction Capabilities of code $t_c \leq (d_{min} - 1)/2 = 1$

Algorithm of Linear Block Coding:

a. Read out the message length (k) and code vector length(n)
b. Read out the parity matrix (P).
c. Construct G matrix.
d. Find out all possible messages.
e. Find out all possible code vector generated by encoder (valid code vector) C= d x G
f. Find out weight of individual code vector (Wc)
g. Calculate d_{min} , Error detection capability and Error correction capability of the code.

Linear block decoding (minimum distance consideration)

Decoding is the process of detecting and correcting errors, when message in the form of code words are transmitted over noisy channel. How many errors can be detected and corrected????The answer of above question is it will depend on the design of the code. The number of errors, code can detect or correct is called error detecting and error correcting capability of the code

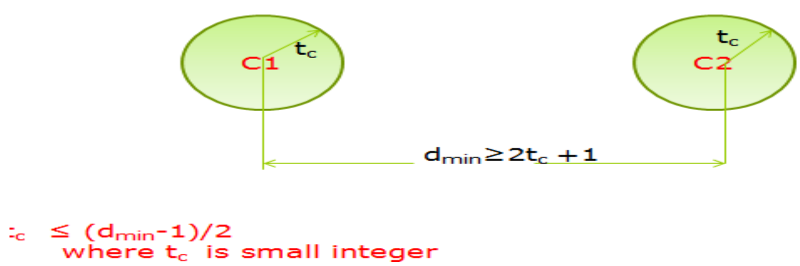
Error Detection

As long as one code word is not transformed into another valid codeword, we can detect whether there was an error in transmission or not. One valid code word is transformed into another valid code word after distance dmin. Thus the number of errors can be detected depends on dmin.$t_d \leq (d_{min}-1)$ where t_d is error detection capability of the code.

Error Correction:

In two cases an error correction is not possible.

- If the received code vector is near to another valid code vector (which is not transmitted) then there is wrong decoding.
- If received codeword might be at same distance from two or more valid code vectors, it is then not possible to correct the code.



Consider a code c with minimum distance $d_{min} \geq 2t_c + 1$. The spheres of radius t_c centered at the codeword's { c_1 , c_2 , c_3, ………C_m} of c will then be disjoint.

- **Decoding Process→** Any received vector can be represented as a point in this space . If this point lies within a sphere, then by nearest neighbor decoding it will be decoded as the center of the sphere.

If t_c or fewer error occur the received word will definitely lie within the sphere of the codeword that was transmitted, and will be correctly decoded.

If however, larger than t_c errors occur, it will escape the sphere, thus resulting in incorrect decoding.

Important Note: Linear Block Decoding

- If the multiplication of the received word (at the receiver) with the transpose H yields a non-zero vector, it implies that error has occurred.
  - This methodology, however will fail if the errors in the transmitted codeword exceed the number of errors for which the coding scheme is designed.

Numerical Example:

Consider the previous example

Prepare error decoding table (look up table)

If the received code vector is 1111110, find out corrected code vector.

Error Decoding Table:

Bit in Error	**Single Bit Error Pattern**	**Syndrome Vector**
	E1 E2 E3 E4 E5 E6 E7	**$S = EH^T$**
1^{st} Bit in Error	1 0 0 0 0 0 0	1 1 0
2^{nd} Bit in Error	0 1 0 0 0 0 0	0 1 1
3^{rd} Bit in Error	0 0 1 0 0 0 0	1 1 1
4^{th} Bit in Error	0 0 0 1 0 0 0	1 0 1
5^{th} Bit in Error	0 0 0 0 1 0 0	1 0 0
6^{th} Bit in Error	0 0 0 0 0 1 0	0 1 0
7^{th} Bit in Error	0 0 0 0 0 0 1	0 0 1

If the received code vector R= 1111110, then syndrome of received code vector is RH^T = 001.

If we compare with error decoding table or H^T matrix, it will matches with last row (or 7^{th} row). It indicates that last bit (or 7^{th} bit) in error. So

Corrected code vector $= R + E = 1111110 + 0000001 = 1111111$

- Perform the practical of coding & decoding of Linear block codes using C Programming or Matlab.

Algorithm of Linear Block Decoding:

a. Read out the message length (k) and code vector length(n)
b. Read out the parity matrix (P).
c. Construct H^T (Transpose of Parity Check Matrix).
d. Read out the received messages.
e. Find out syndrome vector $S= R\ H^T$
f. Compare syndrome vector(S) with H^T Matrix, if it matches with i^{th} row of the H^T matrix, then i^{th} bit in error.
g. Find out corrected code vector $C = R + E$

Conclusion:

Staff Signature: