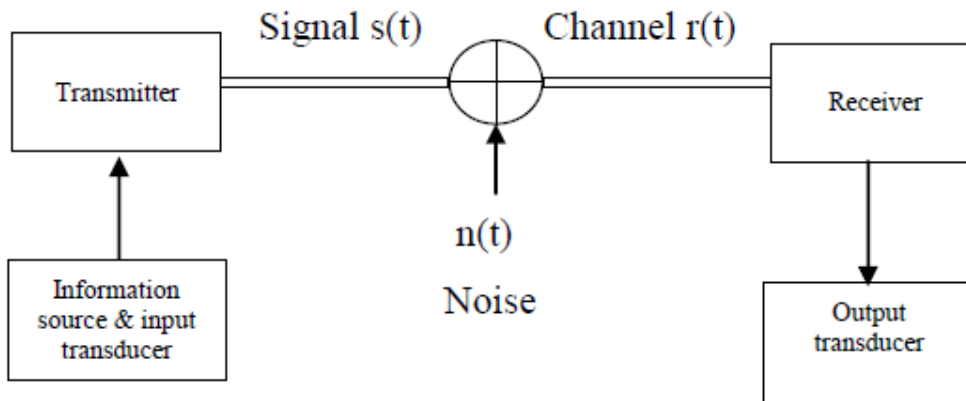


<b>UNIT-01</b>
<b>UNIT-01/LECTURE-01</b>
<b>DATA COMMUNICATIONS</b>
<p><b>DATA COMMUNICATIONS</b> Transmission of signals in a reliable and efficient matter.</p> <p><b>COMMUNICATION MODEL</b></p> <ul style="list-style-type: none"> <li>The purpose of a communications system is to exchange data between two entities.</li> </ul> <p><b>Source:</b></p> <ul style="list-style-type: none"> <li>Entity that generates data. For example a person who speaks into the phone, or a computer sending data to the modem.</li> </ul> <p><b>Transmitter:</b></p> <ul style="list-style-type: none"> <li>A device to transform/encode the signal generated by the source. The transformed signal is actually sent over the transmission system. For example a modem transforms digital data to analog signal that can be handled by the telephone network.</li> </ul> <p><b>Transmission System (Channel):</b></p> <ul style="list-style-type: none"> <li>Medium that allows the transfer of a signal from one point to another. For example a telephone network for a computer/modem.</li> </ul> <p><b>Receiver:</b></p> <ul style="list-style-type: none"> <li>A device to decode the received signal for handling by destination device. For example a modem converts the received analog data back to digital for the use by the computer.</li> </ul> <p><b>Destination:</b></p> <ul style="list-style-type: none"> <li>Entity that finally uses the data. For example Computer on other end of a receiving modem.</li> </ul> <p><b>Data Communications:</b></p> <ul style="list-style-type: none"> <li>Data communications is the transfer of information that is in digital form, before it enters the communication system.</li> </ul> <p><b>Basic Elements of a Communication System</b></p> <ul style="list-style-type: none"> <li>Signal <math>s(t)</math> Channel <math>r(t)</math> Output transducer Information source &amp; input transducer Transmitter Receiver <math>n(t)</math> Noise</li> </ul>



### Basic Elements of a Communication System

#### Information:

- Information generated by the source may be in the form of voice, a picture or a plain text. An essential feature of any source that generates information is that its output is described in probabilistic terms; that is, the output is not deterministic. A transducer is usually required to convert the output of a source in an electrical signal that is suitable for transmission.

#### Transmitter:

- A transmitter converts the electrical signal into a form that is suitable for transmission through the physical channel or transmission medium. In general, a transmitter performs the matching of the message signal to the channel by a process called modulation.
- The choice of the type of modulation is based on several factors, such as:
  - the amount of bandwidth allocated,
  - the type of noise and interference that the signal encounters in transmission over the channel,
  - and the electronic devices that are available for signal amplification prior to transmission.

#### Channel:

- The communication channel is the physical medium that connects the transmitter to the receiver. The physical channel may be a pair of wires that carry the electrical signals, or an optical fibre that carries the information on a modulated light beam or free space at which the information-bearing signal are electromagnetic waves.

#### Receiver:

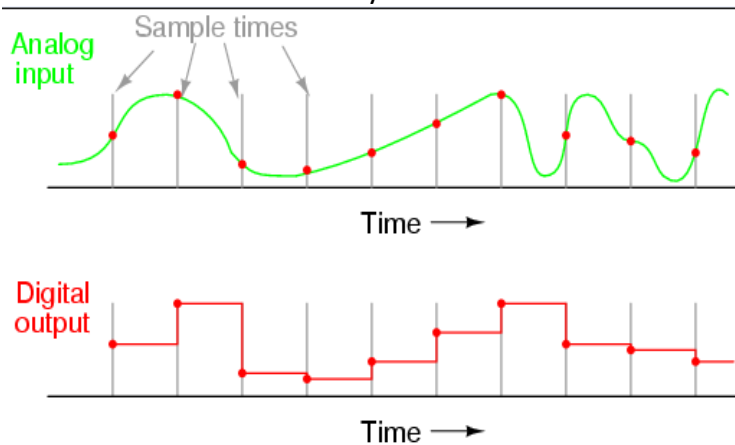
- The function of a receiver is to recover the message signal contained in the received signal. The main operations performed by a receiver are demodulation, filtering and decoding.

## UNIT-01/LECTURE-02

### Analog And Digital Signals

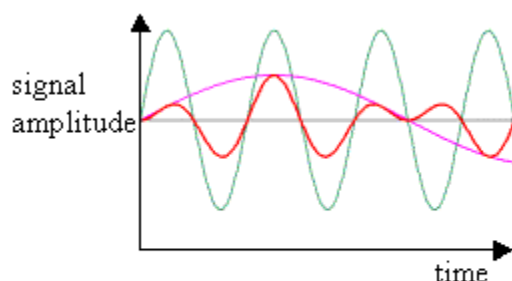
#### Analog And Digital Signals:

The terms analog and digital correspond to continuous and discrete, respectively. These two terms are frequently used in data communications. Analog data takes on continuous values on some interval. The most familiar example of analog data is audio signal. Frequency components of speech may be found between 20 Hz and 20 kHz. The basic speech energy is concentrated between 300-3400 Hz. The frequencies up to 4000 Hz add very little to the intelligibility of human ear. Another common example of analog data is video. The outputs of many sensors, such as temperature and pressure sensors, are also examples of analog data. Digital data takes on discrete values. Digital transmission is the transfer of information through a medium in digital form. A digital signal can be transmitted only for a limited distance. Data communications is the transfer of information that is in digital form, before it enters the communication system.



#### Time Domain:

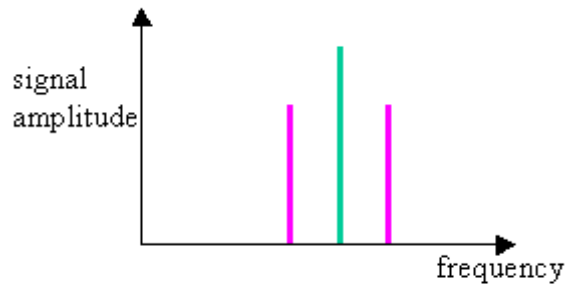
Time domain is the analysis of mathematical functions, physical signals or time series with respect to time. In the time domain, the signal or function's value is known for all real numbers, for the case of continuous time, or at various separate instants in the case of discrete time.



**Time domain**

### Frequency Domain :

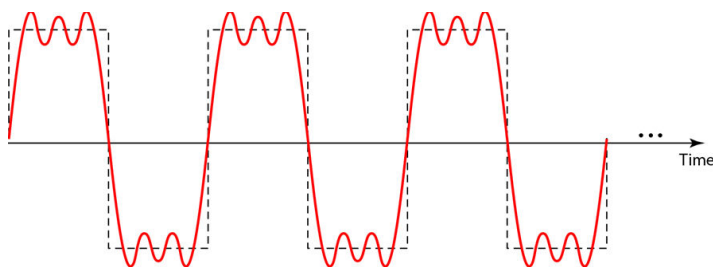
The frequency domain refers to the analysis of mathematical functions or signals with respect to frequency. In other words a frequency-domain graph shows how much of the signal lies within each given frequency band over a range of frequencies. A frequency-domain representation can also include information on the phase shift that must be applied to each sinusoid in order to be able to recombine the frequency components to recover the original time signal.



**Frequency domain**

### Composite Signal:

Composite signal is an indication of an exact reproduction of an image or picture on a receiving end. The signal usually consists of intensity information and both horizontal and vertical retrace signals.



**Composite Signal**

### Bit Rate:

The bit rate is the number of bit intervals per second. This means that the bit rate is the number of bits sent in one second, usually expressed in bits per second (bps).

### Bit length:

Bit length is the number of binary digits, called bits, necessary to represent an integer

**Bandwidth:**

Width of the spectrum or band of frequencies containing energy of the signal.

In networking, we use the term bandwidth in two contexts.

- The first, bandwidth in hertz, refers to the range of frequencies in a composite signal or the range of frequencies that a channel can pass.
- The second, bandwidth in bits per second, refers to the speed of bit transmission in a channel or link. Often referred to as Capacity.
- The bandwidth of a subscriber line is 4 kHz for voice or data. The bandwidth of this line for data transmission
- can be up to 56,000 bps using a sophisticated modem to change the digital signal to analog.
- If the telephone company improves the quality of the line and increases the bandwidth to 8 kHz, we can send 112,000 bps by using the same technology.

**Baseband transmission : (RGPV Dec 21012)**

A type of digital data transmission in which each medium (wire) carries only one signal, or channel, at a time. 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. This source coding and decoding is carried out by codec equipment. The passband 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 passband signals representing bit-streams are considered as digital transmission, while an alternative definition only considers the baseband signal as digital, and passband transmission of digital data as a form of digital-to-analog conversion.

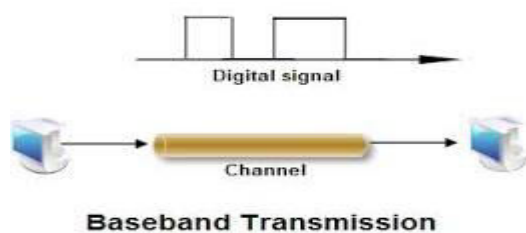


Diagram (a)

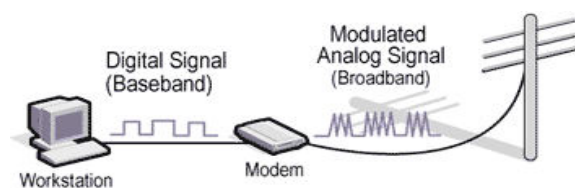


diagram (b)

S.NO	RGPV QUESTIONS	Year	Marks
Q.1	What do you understand by baseband transmission.	Dec.2012	4

## UNIT-01/LECTURE-03

### Broadband Transmission

#### Broadband Transmission : (RGPV dec 2013)

- Broadband transmission enables a single wire to carry multiple signals at the same time. When we enter data into the computer via keyboard, each keyed element is encoded by the electronics within the keyboard into an equivalent binary coded pattern, using one of the standard coding schemes that are used for the interchange of information. To represent all characters of the keyboard, a unique pattern of 7 or 8 bits in size is used. The use of 7 bits means that 128 different elements can be represented, while 8 bits can represent 256 elements. A similar procedure is followed at the receiver that decodes every received binary pattern into the corresponding character.

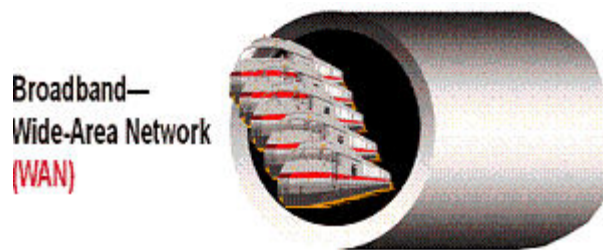


Diagram : Broadband Transmission

#### Attenuation: (RGPV dec 2012)

- Reduction of signal strength during transmission. Attenuation is the opposite of *amplification*, and is normal when a signal is sent from one point to another. If the signal attenuates too much, it becomes unintelligible, which is why most networks require repeaters at regular intervals. Attenuation is measured in *decibels*.

#### Distortion: (RGPV dec 2012)

- Distortion (or warping) is the alteration of the original shape (or other characteristic) of something, such as an object, image, sound or waveform. Distortion is usually unwanted, and so engineers strive to eliminate distortion, or minimize it. In some situations, however, distortion may be desirable. The important signal processing operation of heterodyning is based on nonlinear mixing of signals to cause intermodulation.

#### Noise :

- Noise is unwanted electrical or electromagnetic energy that degrades the quality of signals and data. Noise occurs in digital and analog systems, and can affect files and communications of all types, including text, programs, images, audio, and telemetry.
- In a hard-wired circuit such as a telephone-line-based Internet hookup, *external noise*

is picked up from appliances in the vicinity, from electrical transformers, from the atmosphere, and even from outer space. Normally this noise is of little or no consequence. However, during severe thunderstorms, or in locations where many electrical appliances are in use, external noise can affect communications. In an Internet hookup it slows down the data transfer rate, because the system must adjust its speed to match conditions on the line. In a voice telephone conversation, noise rarely sounds like anything other than a faint hissing or rushing

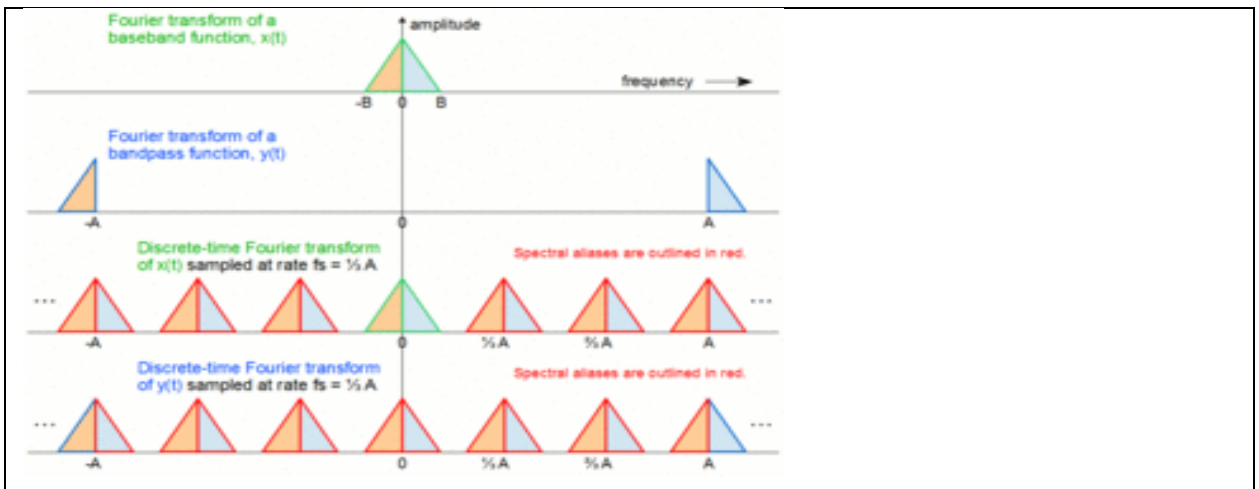
### **Nyquist bit rate :**

In signal processing, the Nyquist rate, named after Harry Nyquist, is twice the bandwidth of a bandlimited function or a bandlimited channel. This term means two different things under two different circumstances:

1. as a lower bound for the sample rate for alias-free signal sampling
  1. as an upper bound for the symbol rate across a bandwidth-limited baseband channel such as a telegraph line or passband channel such as a limited radio frequency band or a frequency division multiplex channel.
- The Nyquist bit rate formula is used to calculate the theoretical maximum bit rate.
  - $\text{Bit rate} = 2 * \text{bandwidth} * \log_2 L$
  - Where bandwidth is the bandwidth of the channel, 'L' is the signal level used to represent data.

### **Nyquist rate relative to sampling (RGPV dec 2012)**

- When a continuous function,  $x(t)$ , is sampled at a constant rate,  $f_s$  (samples/second), there is always an unlimited number of other continuous functions that fit the same set of samples. But only one of them is bandlimited to  $\frac{1}{2} f_s$  (hertz), which means that its Fourier transform,  $X(f)$ , is 0 for all  $|f| \geq \frac{1}{2} f_s$  (see Sampling theorem). The mathematical algorithms that are typically used to recreate a continuous function from samples create arbitrarily good approximations to this theoretical, but infinitely long, function. It follows that if the original function,  $x(t)$ , is bandlimited to  $\frac{1}{2} f_s$ , which is called the Nyquist criterion, then it is the one unique function the interpolation algorithms are approximating. In terms of a function's own bandwidth (B), as depicted above, the Nyquist criterion is often stated as  $f_s > 2B$ . And  $2B$  is called the Nyquist rate for functions with bandwidth B. When the Nyquist criterion is not met ( $B > \frac{1}{2} f_s$ ), a condition called aliasing occurs, which results in some inevitable differences between  $x(t)$  and a reconstructed function that has less bandwidth. In most cases, the differences are viewed as distortion.



- Fig 2: The top 2 graphs depict Fourier transforms of 2 different functions that produce the same results when sampled at a particular rate. The baseband function is sampled faster than its Nyquist rate, and the bandpass function is undersampled, effectively converting it to baseband (the high frequency shadows can be removed by a linear filter). The lower graphs indicate how identical spectral results are created by the aliases of the sampling process.

### Intentional aliasing

- It depicts a type of function called baseband or lowpass, because its positive-frequency range of significant energy is  $[0, B)$ . When instead, the frequency range is  $(A, A+B)$ , for some  $A > B$ , it is called bandpass, and a common desire (for various reasons) is to convert it to baseband. One way to do that is frequency-mixing (heterodyne) the bandpass function down to the frequency range  $(0, B)$ . One of the possible reasons is to reduce the Nyquist rate for more efficient storage. And it turns out that one can directly achieve the same result by sampling the bandpass function at a sub-Nyquist sample-rate that is the smallest integer-sub-multiple of frequency  $A$  that meets the baseband Nyquist criterion:  $f_s > 2B$ . For a more general discussion, see bandpass sampling.

### Nyquist rate relative to signaling

#### Shannon Capacity : (RGPV dec 2012)

To determine the theoretical highest data rate for a noisy channel, we can use

Shannon capacity.

$$\text{Capacity} = \text{bandwidth} * \log_2(1 + \text{SNR})$$



When SNR is very high, we can use the following formula for calculate highest data

Rate then

$$\text{Capacity} = \text{bandwidth} * \text{SNR}^{\text{db}/3}$$

SNR : signal to noise ratio

S.NO	RGPV QUESTIONS	Year	Marks
Q1	<b>What do you understand by broadband transmission</b>	Dec 2013	4
Q2	<b>Explain the nyquist criteria.</b>	Dec 2012	4
Q3	<b>Discuss about Shannon capacity of channel.</b>	Dec 2012	4
Q4	<b>Differentiate between aatenuation and distortion.</b>	Dec 2012	6

## UNIT-01/LECTURE-04

### Throughput: (RGPV dec 2012/2011)

1. In communication networks, such as Ethernet or packet radio, throughput or network throughput is the rate of successful message delivery over a communication channel. This data may be delivered over a physical or logical link, or pass through a certain network node. The throughput is usually measured in bits per second (bit/s or bps), and sometimes in data packets per second or data packets per time slot.
2. The system throughput or aggregate throughput is the sum of the data rates that are delivered to all terminals in a network. Throughput is essentially synonymous to digital bandwidth consumption; it can be analyzed mathematically by means of queueing theory, where the load in packets per time unit is denoted arrival rate  $\lambda$ , and the throughput in packets per time unit is denoted departure rate  $\mu$ .

### Question : 1

A network with bandwidth of 10 Mbps can pass only an average of 12,000 frames per minute with each frame carrying an average of 10,000 bits. What is the throughput of this network?

### Solution:

We can calculate the throughput as

$$\text{Throughput} = \frac{12,000 \times 10,000}{60} = 2 \text{ Mbps}$$

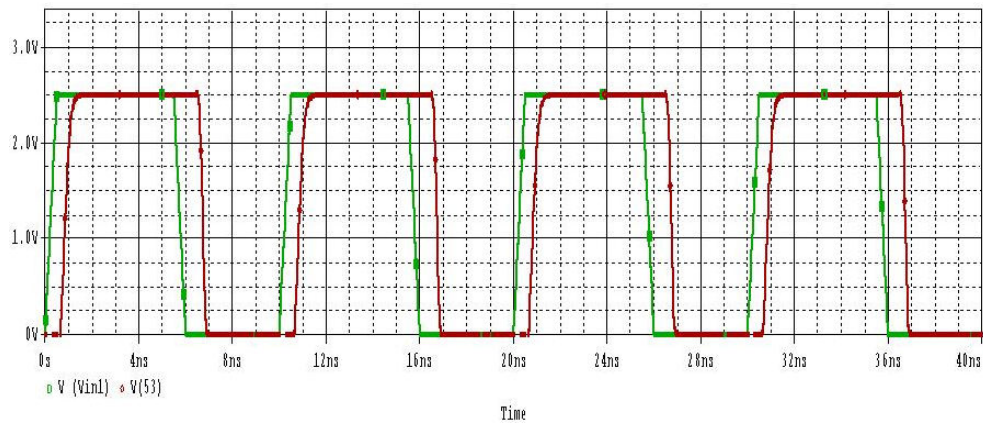
The throughput is almost one-fifth of the bandwidth in this case.

### DELEY:

### Propagation & Transmission delay

- Propagation speed - speed at which a bit travels through the medium from source to destination.
- Transmission speed - the speed at which all the bits in a message arrive at the destination. (difference in arrival time of first and last bit)
- Propagation Delay = Distance/Propagation speed
- Transmission Delay = Message size/bandwidth bps
- Latency = Propagation delay + Transmission delay + Queueing time + Processing time

Diagram :



### Propagation & Transmission delay

**Question : 2**

What is the propagation time if the distance between the two points is 12,000 km? Assume the propagation speed to be  $2.4 \times 10^8$  m/s in cable.

**Solution:**

We can calculate the propagation time as

$$\text{Propagation time} = \frac{12,000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}$$

The example shows that a bit can go over the Atlantic Ocean in only 50 ms if there is a direct cable between the source and the destination.

**Question : 3**

What are the propagation time and the transmission time for a 2.5-kbyte message (an e-mail) if the bandwidth of the network is 1 Gbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at  $2.4 \times 10^8$  m/s.

**Solution:**

We can calculate the propagation and transmission time as:

$$\text{Propagation time} = \frac{12,000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}$$

$$\text{Transmission time} = \frac{2500 \times 8}{10^9} = 0.020 \text{ ms}$$

Note that in this case, because the message is short and the bandwidth is high, the dominant factor is the propagation time, not the transmission time. The transmission time can be ignored.

**Question : 4**

What are the propagation time and the transmission time for a 5-Mbyte message (an image) if the bandwidth of the network is 1 Mbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at  $2.4 \times 10^8$  m/s.

**Solution:**

We can calculate the propagation and transmission times as:

$$\text{Propagation time} = \frac{12,000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}$$

$$\text{Transmission time} = \frac{5,000,000 \times 8}{10^6} = 40 \text{ s}$$

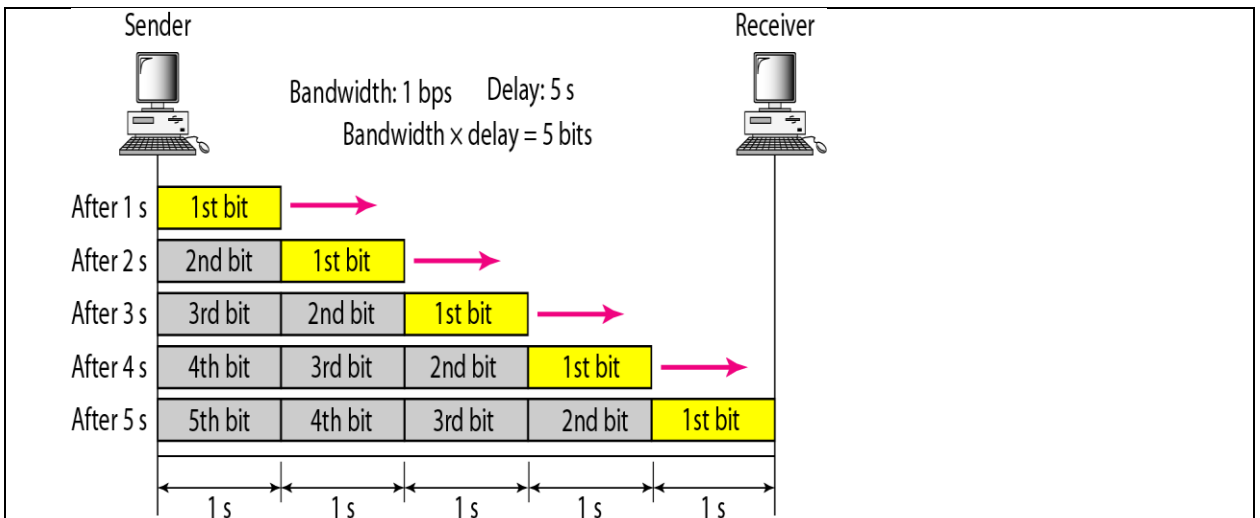
Note that in this case, because the message is very long and the bandwidth is not very high, the dominant factor is the transmission time, not the propagation time. The propagation time can be ignored.

S.NO	RGPV QUESTIONS	Year	Marks
Q.1	Discuss the throughput and delay	DEC2012	4
		Dec 2011	4

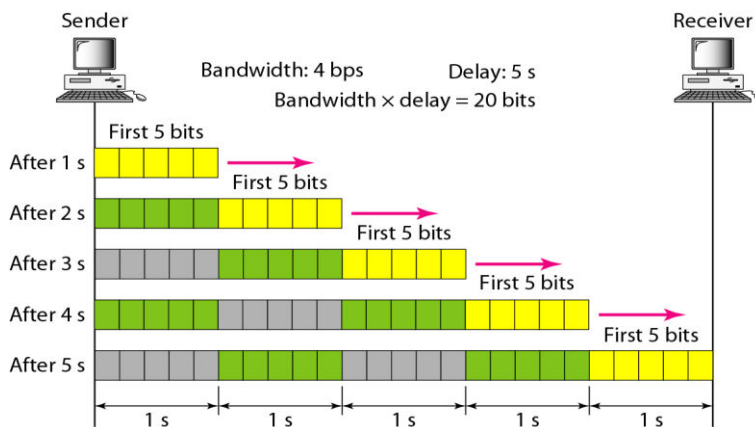
**UNIT-01/LECTURE-05****Bandwidth Delay Product : (RGPV dec 2012/2011)**

1. In data communications, bandwidth-delay product refers to the product of a data link's capacity and its end-to-end delay. The result, an amount of data measured in bits, is equivalent to the maximum amount of data on the network circuit at any given time, i.e., data that has been transmitted but not yet acknowledged. Sometimes it is calculated as the data link's capacity multiplied by its round trip time.
2. Ultra-high speed LANs may fall into this category, where protocol tuning is critical for achieving peak throughput, on account of their extremely high bandwidth, even though their delay is not great.
3. An important example of a system where the bandwidth-delay product is large is that of GEO satellite connections, where end-to-end delivery time is very high and link throughput may also be high. The high end-to-end delivery time makes life difficult for stop-and-wait protocols and applications that assume rapid end-to-end response.
4. A high bandwidth-delay product is an important problem case in the design of protocols such as TCP in respect of performance tuning, because the protocol can only achieve optimum throughput if a sender sends a sufficiently large quantity of data before being required to stop and wait until a confirming message is received from the receiver, acknowledging successful receipt of that data. If the quantity of data sent is insufficient compared with the bandwidth-delay product, then the link is not being kept busy and the protocol is operating below peak efficiency for the link. Protocols that hope to succeed in this respect need carefully designed self-monitoring, self-tuning algorithms. The TCP window scale option may be used to solve this problem caused by insufficient window size, which is limited to 65535 bytes without scaling.

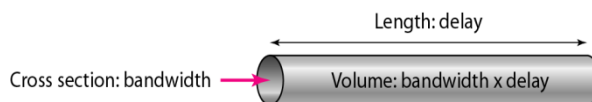
**Filling the link with bits for case 1**



### Filling the link with bits in case 2



- The bandwidth-delay product defines the number of bits that can fill the link.
- Concept of bandwidth-delay product



### QUESTION:

**Q :** What is channel capacity of a noisy channel having BW = 4 khz. Given  $s/n = 0$

**Sol :** channel capacity for a noisy channel is given by

$$C = BW \times \log_2 \left( 1 + \frac{S}{N} \right)$$

$$= (4 \times 10^3) \times \log_2(1 + 0)$$

$$= 0$$

Hence channel capacity is zero. It will not transmit data.

S.NO	RGPV QUESTIONS	Year	Marks
Q.1	Discuss the bandwidth delay product ?	DEC2012	4
		Dec 2013	4

### UNIT-01/LECTURE-06

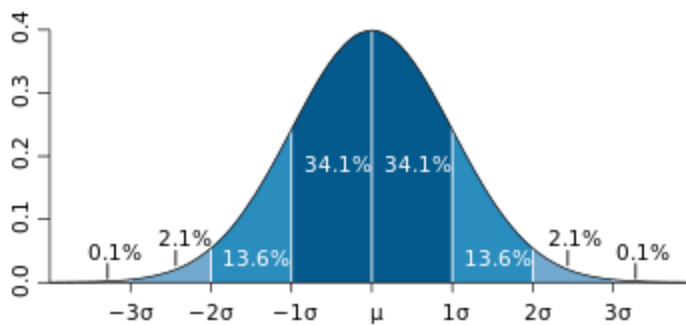
#### Jitter: (RGPV dec 2012/2013)

1. Jitter is the undesired deviation from true periodicity of an assumed periodic signal in electronics and telecommunications, often in relation to a reference clock source. Jitter may be observed in characteristics such as the frequency of successive pulses, the signal amplitude, or phase of periodic signals. Jitter is a significant, and usually undesired, factor in the design of almost all communications links. In clock recovery applications it is called timing jitter.
2. Jitter can be quantified in the same terms as all time-varying signals, e.g., RMS, or peak-to-peak displacement. Also like other time-varying signals, jitter can be expressed in terms of spectral density.
3. Jitter period is the interval between two times of maximum effect (or minimum effect) of a signal characteristic that varies regularly with time. Jitter frequency, the more commonly quoted figure, is its inverse classifies jitter frequencies below 10 Hz as wander and frequencies at or above 10 Hz as jitter.
4. Jitter may be caused by electromagnetic interference and crosstalk with carriers of other signals. Jitter can cause a display monitor to flicker, affect the performance of processors in personal computers, introduce clicks or other undesired effects in audio signals, and loss of transmitted data between network devices. The amount of tolerable jitter depends on the affected application.

#### Jitter metrics

For clock jitter, there are three commonly used metrics: absolute jitter, period jitter, and cycle to cycle jitter. Absolute jitter is the absolute difference in the position of a clock's edge from where it would ideally be. Period jitter is the difference between any one clock period and the ideal/average clock period. Accordingly, it can be thought of as the discrete-time derivative of absolute jitter. Period jitter tends to be important in synchronous circuitry like digital state machines where the error-free operation of the circuitry is limited by the

shortest possible clock period, and the performance of the circuitry is limited by the average clock period. Hence, synchronous circuitry benefits from minimizing period jitter, so that the shortest clock period approaches the average clock period. Cycle-to-cycle jitter is the difference in length/duration of any two adjacent clock periods. Accordingly, it can be thought of as the discrete-time derivative of period jitter. It can be important for some types of clock generation circuitry used in microprocessors and RAM interfaces. Since they have different generation mechanisms, different circuit effects, and different measurement methodology, it is useful to quantify them separately. In telecommunications, the unit used for the above types of jitter is usually the Unit Interval which quantifies the jitter in terms of a fraction of the ideal period of a bit. This unit is useful because it scales with clock frequency and thus allows relatively slow interconnects such as T1 to be compared to higher-speed internet backbone links such as OC-192. Absolute units such as picoseconds are more common in microprocessor applications. Units of degrees and radians are also used.



In the normal distribution one standard deviation from the mean (dark blue) accounts for about 68% of the set, while two standard deviations from the mean (medium and dark blue) account for about 95% and three standard deviations (light, medium, and dark blue) account for about 99.7%.

If jitter has a Gaussian distribution, it is usually quantified using the standard deviation of this distribution. Often, jitter distribution is significantly non-Gaussian. This can occur if the jitter is caused by external sources such as power supply noise. In these cases, peak-to-peak measurements are more useful. Many efforts have been made to meaningfully quantify distributions that are neither Gaussian nor have meaningful peaks (which is the case in all real jitter). All have shortcomings but most tend to be good enough for the purposes of engineering work. Note that typically, the reference point for jitter is defined such that the mean jitter is 0. In networking, in particular IP networks such as the Internet, jitter can refer to the variation (statistical dispersion) in the delay of the packets.

## Types

### Random jitter

Random Jitter, also called Gaussian jitter, is unpredictable electronic timing noise. Random jitter typically follows a Gaussian distribution or Normal distribution. It is believed to follow



this pattern because most noise or jitter in an electrical circuit is caused by thermal noise, which has a Gaussian distribution. Another reason for random jitter to have a distribution like this is due to the central limit theorem. The central limit theorem states that composite effect of many uncorrelated noise sources, regardless of the distributions, approaches a Gaussian distribution. One of the main differences between random and deterministic jitter is that deterministic jitter is bounded and random jitter is unbounded.

### **Deterministic jitter**

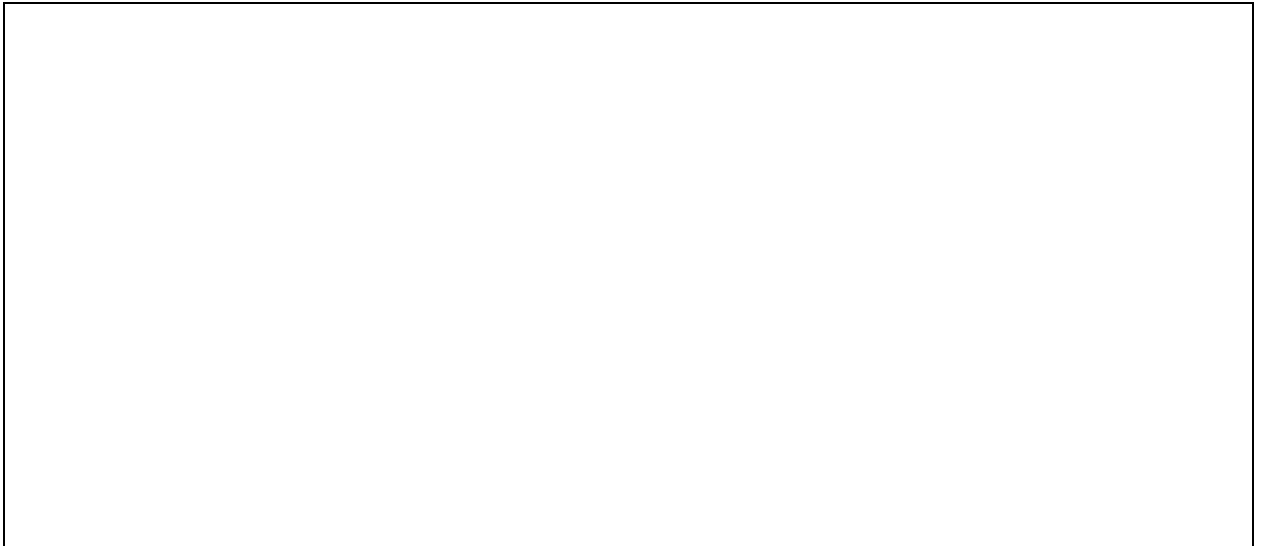
Deterministic jitter is a type of clock timing jitter or data signal jitter that is predictable and reproducible. The peak-to-peak value of this jitter is bounded, and the bounds can easily be observed and predicted. Deterministic jitter can either be correlated to the data stream (data-dependent jitter) or uncorrelated to the data stream (bounded uncorrelated jitter). Examples of data-dependent jitter are duty-cycle dependent jitter (also known as duty-cycle distortion) and intersymbol interference. Deterministic jitter has a known non-Gaussian probability distribution.

**Q : A system sends a signal that can assume 8 different voltage levels. It sends 400 of these signals per second. What is baud rate ?**

**Sol :** Baud rate is defined as the number of signals transmitted per second. Since 400 symbols are transmitted per second.

Baud rate = 400 symbols / sec

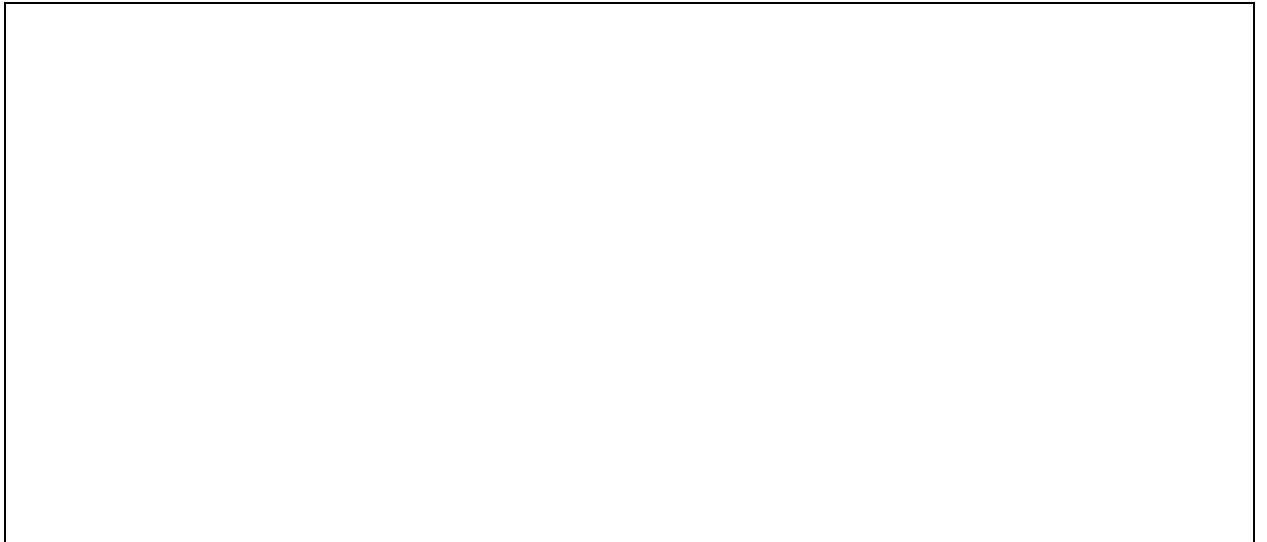
<b>S.NO</b>	<b>RGPV QUESTIONS</b>	<b>Year</b>	<b>Marks</b>
Q.1	Explain the jitter and types of jitter	Dec 2012 Dec 2013	7



**UNIT-01/LECTURE-07**



**UNIT-01/LECTURE-08**



**UNIT-01/LECTURE-09**



### UNIT-02/LECTURE-10

An FM wave consists of infinite number of side bands so that the bandwidth is theoretically infinite. But, in practice, the FM wave is effectively limited to a finite number of side band frequencies compatible with a small amount of distortion.

#### Carson's Rule:-

An empirical formula for the bandwidth of a single tone wideband FM is given by carson's rule. According to this rule, the Fm bandwidth is given by

$$BW = 2(\Delta\omega + w_m)$$

$$BW = 2 \Delta\omega \left(1 + \frac{w_m}{\Delta\omega}\right)$$

Since  $\frac{\Delta\omega}{w_m} = m_f$

So  $\frac{w_m}{\Delta\omega} = \frac{1}{m_f}$

$$BW = 2 \Delta\omega \left(1 + \frac{1}{m_f}\right) \text{ radians}$$

Or

$$BW = 2 \Delta f \left(1 + \frac{1}{m_f}\right) \text{ Hz}$$

- (i) When  $\Delta\omega \ll w_m$  (narrowband FM) i.e.  $m_f \ll 1$   
Then  
 $BW = 2 w_m$   
Which is equivalent to AM

- (ii) When  $\Delta\omega \gg w_m$  (wideband FM) i.e.  $m_f \gg 1$

Then

$$BW = 2 \Delta\omega$$

**DEVIATION RATIO:-**

$$D = \frac{\text{Peak frequency deviation corresponding to the maximum possible amplitude of } f(t)}{\text{the maximum frequency component present in the modulating signal } f(t)}$$

$$D = \frac{\Delta\omega}{\omega_m}$$

**Example:-**

A carrier  $A\cos\omega_c t$  is modulated by a signal  $f(t) = 2\cos 10^4 2\pi t + 5\cos 10^3 2\pi t + 3\cos 10^4 4\pi t$ . Find the bandwidth of the fm signal by using Carson's rule. Assume  $K_f = 15 \times 10^3$  Hz per volt. Also find modulation index  $m_f$ .

**Solution:-**

The maximum frequency component in  $f(t)$  is 20 kHz.

The second term in  $f(t)$  has the maximum amplitude i.e.  $E_m = 5$  volts

Therefore the frequency deviation  $\Delta f$  is given by,

And deviation ratio is given by

$$m_f = \frac{\Delta f}{\omega} = \frac{\Delta f}{f_m}$$

$$m_f = \frac{75 \times 10^3}{20 \times 10^3}$$

$$m_f = \frac{7.5}{2}$$

And the bandwidth is given as

$$BW = 2 \Delta f \left( 1 + \frac{1}{m_f} \right)$$

$$B = 2 \left( \frac{2}{7.5} + 1 \right) 75$$

$$B = 190 \text{ kHz}$$

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**SOME REMARKS ABOUT PHASE MODULATION:-**

The total phase angle of the PM signal is given by

$$\Psi_i(t) = \omega_c t + K_p f(t)$$

For a single tone modulating signal

$$f(t) = E_m \cos \omega_m t$$

Hence

$$\Psi_i(t) = \omega_c t + K_p E_m \cos \omega_m t$$

the maximum departure in the phase is  $Kp E_m$ . This is known as phase deviation denoted by  $\theta_d$ . thus  $\theta_d = Kp E_m$  and the expression for the PM wave is

$$\Phi_{PM} = A \cos \Psi_i(t) = A \cos[\omega_c t + \theta_d \cos \omega_m t]$$

The instantaneous frequency corresponding to  $\Psi_i$  is given by

$$\omega_i = \frac{d\Psi}{dt} = \omega_c - Kp E_m \omega_m \sin \omega_m t$$

The maximum departure in the frequency from  $\omega_c$  is  $Kp E_m \omega_m$ . Therefore, a frequency deviation produced in PM is given as

$$(\Delta\omega)_{PM} = Kp E_m \omega_m$$

Which is independent on the modulating frequency  $\omega_m$ .

The frequency deviation in FM is,

$$(\Delta\omega)_{FM} = Kf E_m.$$

Therefore, for an equivalent bandwidth in PM and FM

$$Kf E_m = Kp E_m \omega_m$$

$$Kf = Kp \omega_m$$

#### **BANDWIDTH OF PM:-**

The PM bandwidth is given by carson's rule

$$(BW)_{PM} \cong 2(\Delta\omega) = 2Kp E_m \omega_m$$

The phase modulation index  $m_p$  is the same as the deviation  $\theta_d$  and is given by

$$m_p = Kp E_m = \theta_d$$

#### **Example:-**

A modulating signal  $5 \cos 2\pi 15 \times 10^3 t$ , angle modulates carrier  $A \cos \omega_c t$ :

(i) Find the modulation index and the bandwidth for fm and pm system.

(ii) Determine the change in the bandwidth and the modulation index for both fm and pm, if fm is reduced to 5 kHz.

#### **Solution:-**

Given  $A_m = 5$ ,  $f_m = 15$  kHz

1)

#### **FM system**

Frequency deviation

$$\Delta f = kf A_m$$

Assume  $kf = kp = 15$  kHz/volt

So

$$\Delta f = 15 \times 10^3 \times 5$$

$$\Delta f = 75 \text{ kHz}$$

Therefore, modulation index

$$mf = \frac{\Delta f}{f_m}$$

$$mf = \frac{75}{15}$$

$$mf = 5$$

$$\text{Bandwidth} = 2(mf + 1)f_m$$

$$\text{Bandwidth} = 2(5 + 1)15$$

$$\text{Bandwidth} = 2 \times 6 \times 15$$

$$\text{Bandwidth} = 180 \text{ kHz}$$

### PM system

Frequency deviation

$$\Delta f = k_p A_m f_m$$

$$\Delta f = 15 \times 10^3 \times 5 \times 15 \times 10^3$$

$$\Delta f = 1125 \text{ MHz}$$

modulation index

$$m_p = k_p A_m$$

$$m_p = 15 \times 10^3 \times 5$$

$$m_p = 75000$$

Corresponding band width

$$\text{Bandwidth} = 2 \Delta f \text{ since } \Delta f \text{ is so much greater than } f_m$$

$$\text{Bandwidth} = 2 \times 1125 \text{ MHz}$$

$$\text{Bandwidth} = 2250 \text{ MHz}$$

The band width is quite large as compared to FM. This is because the Modulation index in PM is quite large

(i) Now, if  $f_m = 5 \text{ kHz}$

**For FM:** The deviation  $\Delta f$  is independent of  $f_m$ , and remains 75 kHz.

$$mf = \frac{75}{5}$$

$$mf = 15$$

$$\text{Bandwidth} = 2(mf + 1)f_m$$

$$\text{Bandwidth} = 2(15 + 1)5$$

$$\text{Bandwidth} = 2 \times 16 \times 5$$

$$\text{Bandwidth} = 160 \text{ kHz}$$

Thus in FM the modulation index changes considerably with a change in the modulating frequency, but the bandwidth changes only slightly.

**For PM:**

**Deviation**  $\Delta f$  is dependent on  $f_m$  and is given by

$$\Delta f = k_p A_m f_m$$

$$\Delta f = 15 \times 10^3 \times 5 \times 5 \times 10^3$$



$$\Delta f = 375 \text{ MHz}$$

Hence

$$\text{Bandwidth} = 2 \Delta f \quad \text{since } \Delta f \text{ is so much greater than } f_m \quad \text{Bandwidth} = 2 \times 375 \text{ MHz}$$

$$\text{Bandwidth} = 750 \text{ MHz}$$

The modulation index is independent of  $f_m$ , so

$$m_p = k_p A_m$$

$$m_p = 15 \times 10^3 \times 5$$

$$m_p = 75000$$

$$m_p = 75 \text{ kHz}$$

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