

## ***Question1***

### ***Introduction***

Analogue Bandwidth is the difference between the highest and lowest frequencies present in the signal. The units of bandwidth in analogue are hertz.

Digital Signal is a way of sending voice, video, or data that reconstructs the signals using binary codes (1s and 0s) for transmission through wire, fiber optic...

Digital Bandwidth is the Difference between the Highest and Lowest how analogue speech signal of telephone call is converted to digital signal.

*The main steps involved in the conversion process,*

### **Filtering**

The first step to convert the signal from analog to digital is to filter out the higher frequency component of the signal. This make things easier downstream to convert this signal. Most of the energy of spoken language is somewhere between 200 or 300 hertz and about 2700 or 2800 hertz. Roughly 3000–hertz bandwidth for standard speech and standard voice communication is established. Therefore, they do not have to have precise filters (it is very expensive). A bandwidth of 4000 hertz is made from an equipment point if view. This band–limiting filter is used to prevent aliasing (antialiasing).

### **Sampling**

The second step to convert an analog voice signal to a digital voice signal is to sample the Filtered input signal at a constant sampling frequency. It is accomplished by using a process called pulse amplitude modulation (PAM). This step uses the original analog signal to modulate the amplitude of a pulse train that has a constant amplitude and frequency. The pulse train moves at a constant frequency, called the sampling frequency. The analog voice signal can be sampled at a million times per second or at two to three times per second. How is the sampling frequency determined? A scientist by the name of Harry Nyquist discovered that the original analog signal can be reconstructed if enough samples are taken. He determined that if the sampling frequency is at least twice the highest frequency of the original input analog voice signal,

this signal can be reconstructed by a low-pass filter at the destination.

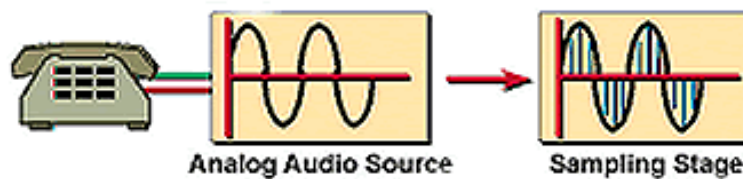
The Nyquist criterion is stated like this:

$$F_s > 2(BW)$$

$F_s$  = Sampling frequency

$BW$  = Bandwidth of original analog voice signal

Figure 1: Analog Sampling



## Digitize Voice

After you filter and sample (using PAM) an input analog voice signal, the next step is to digitize these samples in preparation for transmission over a Telephony network. The process of digitizing analog voice signals is called PCM. The only difference between PAM and PCM is that PCM takes the process one step further. PCM decodes each analog sample using binary code words. PCM has an analog-to-digital converter on the source side and a digital to analog converter on the destination side. PCM uses a technique called quantization to encode these samples.

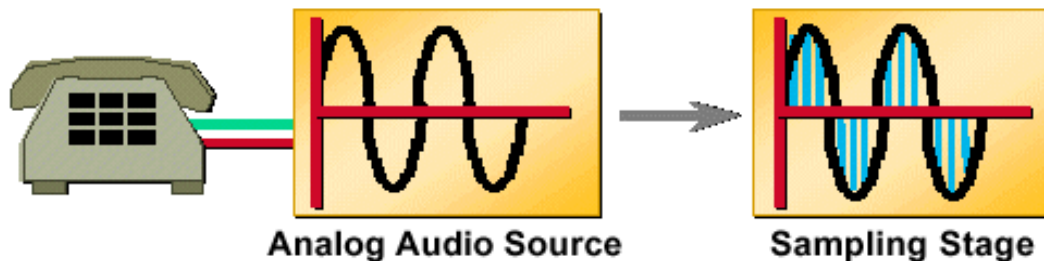
*Why the required bit-rate is 64 kbit/s.*

- ❖ *The predominance of 64 kbit/s channels in many multiplexing schemes is a direct result of the fact that this is the bit-rate used for single telephone channels.*
- ❖ *The minimum theoretical rate is of the order of tens or at most hundreds of bits per second, and that the 64 kbit/s PCM coding must be highly redundant.*

## Quantization and Coding

# Pulse Code Modulation—Nyquist Theorem

Voice Bandwidth =  
200 Hz to 3400 Hz



## Codec Technique

1 = Sample  
8 bits per sample  
8 kHz (8,000 Samples/Sec)

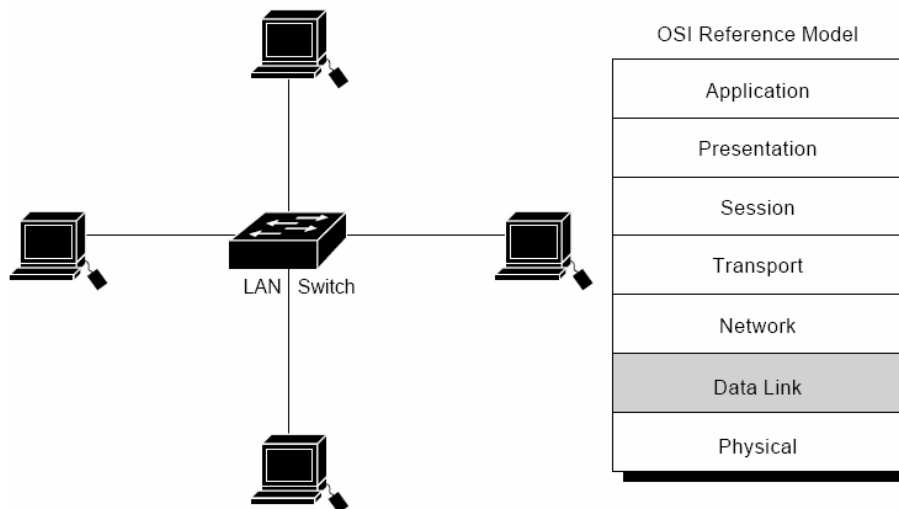
As the input signal samples enter the quantization phase, they are assigned to a quantization interval. All quantization intervals are equally spaced (uniform quantization) throughout the dynamic range of the input analog signal. Each quantization interval is assigned a discrete value in the form of a binary code word. The standard word size used is eight bits. If an input analog signal is sampled 8000 times per second and each sample is given a code word that is eight bits long, then the maximum transmission bit rate for Telephony systems using PCM is 64,000 bits per second. Figure illustrates how bit rate is derived for a PCM system.

## Question2

LAN is a high-speed, fault-tolerant data network that covers a relatively small geographic area. It typically connects workstations, personal computers, printers, and other devices. LANs offer computer users many advantages, including shared access to devices and applications, file exchange between connected users, and communication between users via electronic mail and other applications. LAN *switch* is a device that provides much higher port density at a lower cost than traditional bridges. For this reason, LAN switches can accommodate network designs featuring fewer users per segment, thereby increasing the average available bandwidth per user. This chapter provides a summary of general

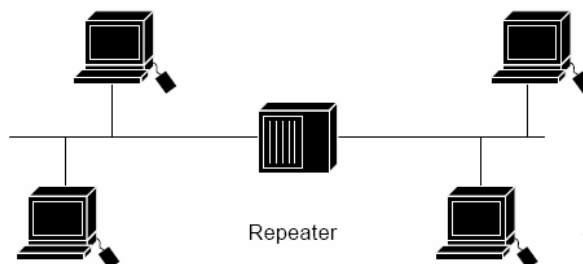
LAN switch operation and maps LAN switching to the OSI reference model.

**A LAN switch is a data link layer device.**



repeater is a physical layer device used to interconnect the media segments of an extended network. A repeater essentially enables a series of cable segments to be treated as a single cable. Repeaters receive signals from one network segment and amplify, retiming, and retransmit those signals to another network segment. These actions prevent signal deterioration caused by long cable lengths and large numbers of connected devices. Repeaters are incapable of performing complex filtering and other traffic processing. In addition, all electrical signals, including electrical disturbances and other errors, are repeated and amplified. The total number of repeaters and network segments that can be connected is limited due to timing and other issues. Figure below shows a repeater connecting two network segments.

**A repeater connects two network segments.**



Bridges, It connects two or more local area networks (LANs) together. The data use the bridge to travel to and from different areas of the network. The device is similar to a router, but it does not analyze the data being forwarded. Because of this, bridges are typically fast at transferring data, but not as versatile as a router. For example, a bridge cannot be used

as a firewall like most routers can. A bridge can transfer data between different protocols (i.e. a Token Ring and Ethernet network) and operates at the "data link layer" or level 2 of the OSI (Open Systems Interconnection) networking reference model.

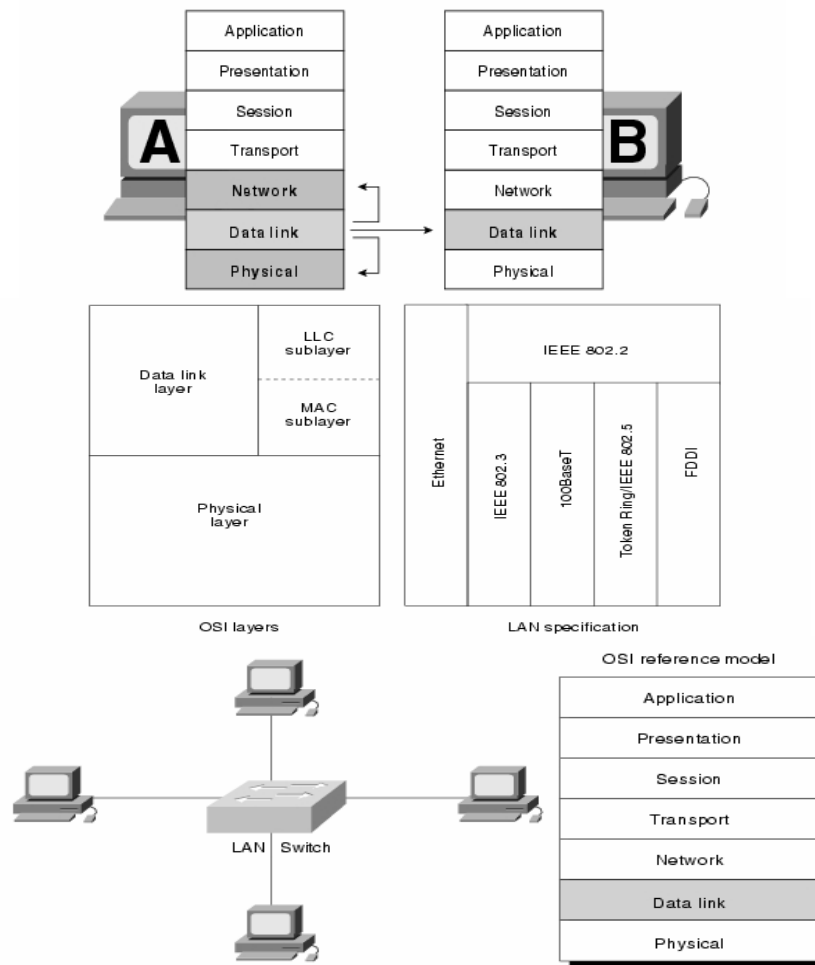
Router is a hardware device that routes data (hence the name) from a local area network (LAN) to another network connection. A router acts like a coin sorting machine, allowing only authorized machines to connect to other computer systems. Most routers also keep log files about the local network activity.

### Bit Rate

A signal received in a wavelength division multiplexing system is delayed and an exclusive OR logical operation is performed on the received signal and the delayed received signal. Then, the direct current voltage of a selection signal which is a result of the operation is measured to identify the bit-rate of the received signal. When a mixture of various signals having different bit-rates are used over a network in the wavelength division multiplexing system, a receiving terminal can automatically recognize information on the bit-rate of a received optical signal and extract a reference clock signal from the received signal, thereby reproducing the received optical signal without distortion, using the clock signal.

Bit rate through a to f, its move from router to switch. The bandwidth of bit rate between Routers to switch is back to the media speed and so on the contrary. A protocol is used between the routers in a Multi Protocol Label Switching (MPLS) network to assign labels to IP network and exchange label information with other routers. The most commonly used protocol currently used today is Label Distribution Protocol (LDP Port number 646), which is TCP-based and runs on the MPLS Label Switch Router (LSR).

LAN protocols function at the lowest two layers of the OSI reference model between the physical layer and the data link layer. Figures below illustrate how several popular LAN protocols map to the OSI reference model.



### Question3

Satellite Communications Systems (SCS) was founded in 1979 to make it easier for individuals and organizations to access and utilize sophisticated technologies in order to improve the way they communicate. Satellite Communication Systems, Inc. (SCSI) is a rapidly growing facilities-based international telecommunications carrier. SCSI provides telecommunications services

A communications satellite (sometimes abbreviated to comsat) is an artificial satellite stationed in space for the purposes of telecommunications. Modern communications satellites use a variety of orbits including geostationary orbits, Molniya orbits, other elliptical orbits and low (polar and non-polar) Earth orbits.

*Propagation delay* is the length of time starting from when the input to a logic gate becomes stable and valid, to the time that the output of that logic gate is stable and valid.

In a wide variety of broadband applications, there is a need to distribute information to a potentially large number of receiver sites that are widely dispersed from each other. Communication satellites are a natural technology option and are extremely well suited for carrying such services because of the inherent broadcast capability of the satellite channel. Despite the potential of satellite multicast, there exists little support for multicast services over satellite networks. Although several multicast protocols have been proposed for use over the Internet, they are not optimized for satellite networks. One of the key multicast components that is affected when satellite networks are involved in the communication is the transport layer.

*The maximum propagation delay is:*

Time for the reception of the data:

$$v = d/t$$

$$t = d/v = 20000/3.06108 = 6533 \text{ s.}$$

Time for the retransmission of the received data, taking into account the circumference of the circle to be covered by the signal:

$$t = (d + 2[r])/v = (20000 + 2[30000])/3.06108 = 68.08 * 10^3 \text{ s}$$

So the propagation delay is:

$$(6.533 * 10^3 + 68.08 * 10^3) = 74.613 * 10^3 \text{ s} = 149.226 \text{ ms.}$$

For a geostationary system,  $d = 36 \text{ Mm}$

The propagation delay is 23582.102 ms.

*Reference*

- *T305 DIGITAL COMMUNICATION Block1 part1 and 2*
- *Tutorial 1,2,3,4 and 13*
- *T529 ICT CD-ROM*
- [www.tpub.cim](http://www.tpub.cim)
- <http://en.wikipedia.org/wiki/pulse-code>

