CS542 HW2

1. Summarize all access networks briefly and list their main characteristics/technologies.

Access Network – It is defined as the network which connects subscriber to their immediate service provider

There are 3 types of access networks – Home network, Enterprise network and Mobile network.

Home Network-

Five types of broadband residential access are Digital Subscriber Line, cable internet access, FTTH, Dial-Up and Satellite.

Typically, DSL is used by resident. Internet service is provided by local telephone company. DSL modem is used for exchanging data with digital subscriber line access multiplexer (DSLAM) via existing telephone line.

Analog signals from all houses are transmitted back to digital format at DSLAM. Telephone call and internet connection can share the DSL link at same time. Splitter at customer side splits the data and telephone signal and forwards data signals to DSL modem. Also, splitter at DSLAM separates phone and data line and forwards data to internet.

As per the ITU2003, standard DSL transmission rate is 24 Mbps downstream and 2.5 Mbps upstream.

These rates may not be achieved because of DSL provider may limit residential rate when it offered at tiered level or maximum rate may be limited by distance between home and telephone company central office (CO), gauge of twisted pair line and electrical interference. Typical distance should be 5 to 10 miles between CO and home.

Cable internet access make use of cable company's existing cable line infrastructure for data exchange.

In this network, coaxial cable is used from cable head end to individual house and head end is connected to immediate internet provider (neighborhood junction) by fiber optics cable. Each junction can support 500 to 5000 homes. This is often referred as hybrid fiber coax (HFC) because both fiber and coaxial cables are used.

It works with cable model. It is a device that connects home PC through ethernet port. Cable modem termination system (CMTS) is present at each cable head end and it serves same as DSLAM of DSL network. Here also downstream and upstream channels are present. The DOCSIS 2.0 standard defines downstream rates up to 42.8 Mbps and upstream rates of up to 30.7 Mbps.

Cable internet access is shared broadcast medium. Every packet from head end flows downstream to every link to every home and each packet from home travels upstream to head end. Hence each time customer cannot access data at actual downstream rate of head end.

The new technology called fiber to the home (FTTH) provides even more faster rate than DSL and cable network. It provides optical fiber path from Co to directly home. There are two optical distribution network FTTH: Active Optical Network (AON) and Passive Optical Network (PON). AON and PON are used to split into individual customer specific fiber. FTTH can provide internet access in rate of gigabits per second range with average downstream speed of 20 Mbps in 2011.

Satellite link and dial-up access can be used where, above mentioned technology is not present. Speed of satellite link is more than 1 Mbps. StarBand and HugesNet are satellite access providers.
Dial-up model uses phone line to connect to modem of ISP. Maximum speed of dial-up link is 56 Kbps.

Enterprise network-

1. Ethernet

For corporate and education institutions, ethernet is the most commonly used local area network (LAN) technology. twisted pair copper cables are used to connect end users to ethernet. Ethernet switch or network of switches are connected to internet connections. Generally, ethernet switch speed is 100Mbps and 1Gbps to 10Gbps speed for the server.

2. WiFi

Wireless local area network (LAN) is based on IEEE 802.11 technology. Wireless user transmits or receives packets to or from access points. Wireless user must be within 10 meters from access points. Access point are connected the corporate network is connected to wired internet via access points. It is also used to create home network.

Wide-Area Wireless Access

1 **3G (Third Generation)**
Packet switched wide area wireless used by telecommunication companies. Speed for internet is more than 1Mbps.

2 **LTE (Long Term Evolution)**
It uses 3G technology as base and has speed of 10Mbps. Downstream speed is in multiple of 10Mbps in commercial networks.

2. Explain why dial up modem Internet connection and a phone call cannot co-exist at the same time.

Dial-up connects users to the Internet via an existing phone line. To use dial-up Internet, the user must dial and call a modem at the Internet service provider’s office. Then, data is transferred between the ISP’s modem and the user’s modem. But, the signal must be analog to transfer of data to occur over the phone line. It can only be translated to digital data at the user’s modem. Dial-up Internet uses a user’s phone line while he is online. Also, there is no any splitter present to split the data and voice to travel the data to customer PC. Therefore, the user can’t be online and make or receive phone calls at the same time.[1]

3. Research why theoretical maximum throughput of a dialup modem cannot surpass 56 Kbps.

A 56Kbps dialup modem can download data at up to 56 kilobits per second. That is a theoretical maximum of the hardware; for various reasons, this speed can never actually be achieved.

To get voice data onto this digital network, it must first be digitized. So now instead of just going through a lowpass filter, it goes through a lowpass filter, then is sampled, and then is run into an analog-to-digital converter. the lowpass filter cuts off all frequencies above 4 kHz. This means that the filtered signal can be sampled at 8 kHz (which is to say, it is sampled 8000 times per second) with no fidelity loss.

To quantize our signal so that we have a purely digital signal, an 8-bit analog-to-digital converter (ADC) is used.

A DS-1 transmits data in frames. Each frame contains one complete sample from each DS-0, plus a framing bit. Since there are 24 DS-0s multiplexed to make each DS-1, and a single sample from a DS-0 is 8 bits of data. 24 * 8 makes 192 bits. Toss in the framing bit and you have 193 bits.

So, we have our 193-bit frame, and we want to pack some control data into it. Every 6 frames, the least significant bit of each voice channel is "stolen". The data bit is thrown out, and control data is sent instead. Because it is the *least* significant bit that is lost, this very slight alteration to the voice signal is not detectable by human ears.

That system worked fine until people wanted to transmit more than just voice on these networks. If you're sending voice, losing the least significant bit of every sixth byte is no big deal. If you are sending data, it is a very big deal. Neither modem has any way to know which bytes will be affected. Maybe this sample, maybe the next one. This is a problem.

The solution is simple enough: assume you are always going to lose that least significant bit. Assume that you can only reliably send or receive 7 bits with each sample. When the receiver gets its data, it simply throws away the least significant bit of every byte. That yields a maximum data rate of 7 bit/sample * 8000 samples/s, or **56 kbit/s**. [2]

4. Describe

A. What is a digitized voice

Digitized voice is in the form of one of two digits, either 0 or 1. Analog signals are converted into digital form i.e. in binary form (0 or 1).

B. Why a digitized voice is needed in PSTN

Analog speech signals of telephone contain frequencies that extend up to a maximum of 4KHz. Instead of dealing with continuous analog signals, it is possible to digitize a speech as a sequence of binary digits, so that it appears to be a data. This has several advantages. It allows integration of voice and data services, switching and transmission of binary data becomes more reliable and convenient within the network. Hence digitized voice is needed in PSTN. [3]

C. How a digitized voice can be produced

Sound is converted into a single binary code in digitization. The main process in digitization is to process between the capturing device and the player device so that the final result represents the original source with the most possible quality of sound. Speed in which this form of information can be transmitted with no quality degradation compared with analog information is the main advantage of digitization.

Digital information exists as one of two digits, either 0 or 1. These are known as bits and the sequences of 0s and 1s that contains information are called bytes. Analog signals are continuously variable, both in the number of possible values of the signal *at* a given time, as well as in the number of points in the signal *in* a given period of time.

Digitization occurs in two parts:

Discretization

The reading of an analog signal *A*, and, at regular time intervals, sampling the value of the signal at the point. Each such reading is called a sample and may be considered to have infinite precision at this stage.

Quantization

Samples are rounded to a fixed set of numbers such as integers, a process known as quantization. These processes can occur at the same time. [4]

D. What is the bandwidth of a digitized voice in a PSTN

The bandwidth of digitized voice is 64Kbps in a PSTN.

E. Why the bandwidth is produced (numerical analysis is required)

Maximum frequency of digitization standard in PSTN is 4KHz. Hence, by Nyquist sampling theorem, sampling rate = 2 * 4KHz = 800(samples/sec). There are 8 bits per sample.

So, Bit rate = number of bits per sample * number of sample per second.

Bit rate = 8 bits/sample * 8000 samples/sec = 64 kbps

Hence bandwidth in PSTN is 64 kbps. [5]

5. List US/NAR standard multiplexing level and their bandwidth in a PSTN

Trunk lines between exchanges carried multiple voice channels simultaneously using frequency division multiplexing (FDM). This entailed the use of expensive modulators, demodulators and filters for each voice channel, and commercial pressures created a need to develop more cost-effective exchange equipment. The digitization of an analogue voice channel into a 64kbps digital channel, designated as digital signaling level zero or simply *D*S0

The T1 carrier originally carried 24 digital voice channels, each with a bit rate of 64 kbps. Each TDM frame on the T1 line carried 24 bytes of voice data (8 bits per voice channel) plus a single framing bit to facilitate synchronization and de-multiplexing by the receiver. Since each byte of data represents a 125μs voice sample, the required frame rate was 8,000 frames per second. The total bit rate for a T1 line is therefore 8,000 x ((24 x 8) + 1) = 1,544,000 bits per second (1.544 Mbps) and is designated as *digital signaling level 1* or *DS1*. [6]

PDH Levels and Bit Rates			
Carrier level	North America	Europe	Japan
Voice/data channel	64 kbps (DS0)	64 kbps (E0)	64 kbps
First level	1.544 Mbps (DS1) 24 channels	2.048 Mbps (E1) 32 channels	1.544 Mbps 24 channels

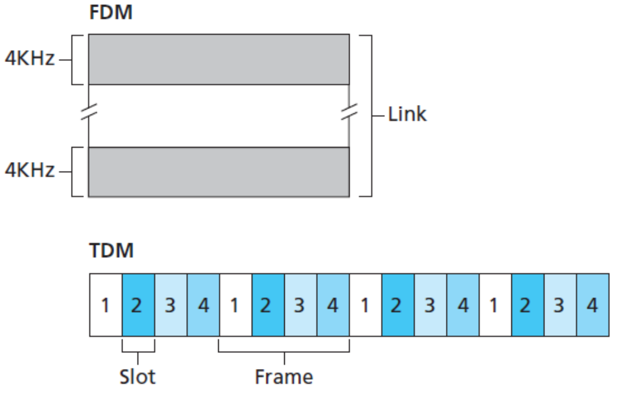
6. Describe

A. What is FDM

Suppose the channel supports N nodes and that the transmission rate of the channel is R bps. Frequency Division Multiplexing (FDM) divides R bps channel into different frequencies with N nodes with the rate of R/N and assign it to N nodes. Hence FDM produces N small channel of rate R/N from single large R bps channel. FDM avoids collisions of frame. It divides total bandwidth among all nodes. In FDM, node is limited to bandwidth of R/N bps even if it is the only node with the packet to send.

B. What is TDM

TDM divides time into **time frames** and further divides each time frame into *N* **time slots.** Each time slot is then assigned to one of the *N* nodes. Whenever a node has a packet to send, it transmits the packet's bits during its assigned time slot in the revolving TDM frame. TDM allows one node to send packet for fixed amount of time and then allow another node to send packet for the same amount of time and so on. This pattern would repeat after every node is done with packet sending. TDM eliminates collision. Each node gets dedicated rate of R/N bps to send frame during each frame type. It also has drawbacks. Every node gets only R/N bps even when it the only node with packet to send. Node has to wait every time for its time frame to send packet even if it is the only node with frame to send.



[8]

C. What are differences of above 6.A and 6.B

Difference between FDM and TDM

- FDM divides the channel into two or more frequency ranges that do not overlap, while TDM divides and allocates certain time periods to each channel in an alternating manner.
- For TDM, each signal uses all of the bandwidth some of the time, while for FDM, each signal uses a small portion of the bandwidth all of the time.
- As TDM allocates time periods, only one channel can transmit at a given time, and some data would often be delayed. channels in FDM can transmit at any time, their latencies would be much lower compared to TDM.
- In tandem, FDM channels are divided and further each FDM channel is occupied by TDM Channels. [7]

D. Which multiplexing is used by xDSL

Digital-Subscriber-Line-Access-Multiplexer, is a network distribution device that combines individual subscriber lines into uplink. xDSL uses digital encoding to provide more bandwidth over existing twisted-pair telephone lines. It uses **frequency division multiplexing** for data transfer over the network. The term ‘x’ denotes whether DSL is synchronous or asynchronous. xDSL ranges from 6.1Mbps to 155Mbps download speed, and from 600Kbps to 15Mbps for upload speed. [9]

E. Which multiplexing is used by Cable Modem

Cable internet requires a modem, called cable modem. It is an external device that connects home PC through ethernet port. At the cable head end, cable modem termination system (CMTS) is used to turn the analog signals from home PC into digital format. Both upstream and downstream channels used for data transmission are shared among many subscribers, using a **time-division multiplexing technique**. Within the subscriber's home or office, a splitter is employed to direct ordinary television signals to a television and the data channel to a cable modem, which can serve one or a network of PCs.

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