Overview of Networks

- The note is mainly based on Chapters 1-2 of High Performance of Computer Networks and Data and Computer Communications, William Starlings, 7th Edition.

- **Computer/communication networks:**

  Telephone networks, computer networks, cable TV networks, and wireless networks.

  Telephone networks: Alexander Graham Bell invented a pair of telephones in 1876. Early networks connect telephones by manually operated switches. The networks use analog signals and *circuit switching* to connect a pair of telephones. In modern telephone networks, electronic switches and digital signals are used, the call control and voice transfer are separated, and integrated services can be provided. The key innovations in telephone networks include circuit switching, digitization, separation of call control from voice transfer, and service integration.

  Computer networks: Networks are used for data communication. Computer networks can be classified into broadcast networks and point-to-point (switched) networks. A broadcast network has a single communication link which is shared by all terminals attached to it. Most local area networks (LANs) are broadcast networks. A switched network consists of terminals and switches connected by point-to-point communication links. A switched network can be expressed by a graph with the nodes for switches/terminals and the edges for links. Computer networks can be further classified based on the following:

  - Point-to-point/multi-access connection
  - Local area networks (LAN)
  - Wide area networks (WAN)
  - Internet/Intranet
  - Parallel/distributed computing systems

  Cable TV networks: Networks were originally used for the *community antenna television* (CATV). Now, the networks deliver television signals to end users by coaxial cable. CATV uses frequency-division multiplexing to transmit dozens of TV channels.

  Wireless networks: Radio/TV transmission/broadcast are examples of communications of using wireless networks. Other examples include cellular phone networks, wireless LANs, ad hoc wireless networks, sensor networks, and satellite networks.

- **Network elements**

  Softwares: Protocols.

  Protocols are rules for communication and specify the details of message formats, how a source sends a message, how a destination responses when a message arrives, how to deal with errors, etc. Protocols allow us to discuss data communication independent of any particular network hardware. Protocols to communication is similar as algorithms to computation. Protocols for computer/communication networks are very complicated. Layered structure is an efficient approach to design and implement such protocols. Models for layered structures include TCP/IP model and OSI model.

Hardwares: links, switches, and terminals.

- **Network links**

  A link transfers data from one node (source) to another node (destination). A link can be a shared medium (e.g., a bus in the Ethernet) or a point-to-point link. A point-to-point link can be unidirectional (expressed by a directed edge) and bidirectional (expressed by an undirected edge or a pair of directed edges, one in each direction).

- **Transmission on link**

  A transmission on a point-to-point link can be simplex, half duplex, and full duplex.

  In simplex transmission, data can be transmitted in only one direction on the link. A unidirectional link can support the simplex transmission in the direction denoted by the directed edge.

  In half duplex transmission, data can be transmitted in both directions, but only in one direction at a time. A bidirectional link expressed by an undirected edge can support the half duplex transmission.

  In full duplex, data can be transmitted in both directions simultaneously. A bidirectional link expressed by a pair of directed edges can support the full duplex transmission.

  Important physical links include optical fibers, copper cables, and microwave or radio ”wireless” links.

- **Network nodes**

  Networks include switches and end systems. A switch is a device which transfers data from its input links to its output links. An end system is a user of the network.

- **Networks and graphs**

  A network is an interconnection of network elements (nodes and links). A network can be modeled as a graph G with a vertex of G for a network node and an edge of G for a network link. In the model, a simplex link, a half-duplex link, and a full-duplex link are expressed by a directed edge, an undirected edge, and a pair of directed edges (one in each direction), respectively.

- **More on telephone networks**

  Early days networks use analog transmission, circuit-switching and manually operated switches.

  Modern networks use analog/digital transmission, circuit-switching, electronic switches, common channel signaling (CCS) which separates the call control from voice transfer, digital signal hierarchy, and digital carrier system: trunk capacity divided into a hierarchy of logical channels like, (T-1,DS-1),(T-2,DS-2),..,(OC-1,STS-1),(OC-3,STS-3)..

Integrated services digital network (ISDN) provides the services for voice transfer and digital communications.

Internet telephone uses digital transmission, voice transferred into digital streams, streams transmitted in packets on a virtual-circuit in the Internet.

- **More on computer networks**

  Point-to-point transmission

  RC-232-C standard is for serial port of computer devices. Each character is encoded into seven bits and an additional parity bit is used for error detection. Successive characters are separated by time intervals in the transmission.

  Synchronous data link control (SDLC) improves the data rate of RC-232-C by grouping many characters into a packet to eliminate the time intervals between characters. A packet is a sequence of bits with a special bit pattern called *header* at the top and a special pattern called *tailer* at the end of the sequence.

  Point-to-point link networks

  Two computers can exchange data on a link using by either RC-232-C or SDLC. Communications among multiple computers are more complicated. Store-and-forward packet-switching was proposed for communications among multiple computers connected by point-to-point links. As an example, assume that Computer A is directly connected to Computer C by a link, C is directly connected to Computer B by a link, and A and B are not connected directly. When Computer A sends a packet to Computer B, A puts the source address $addr_A$ and destination address $addr_B$ into the header of the packet and sends the packet to computer C. Computer C, on receiving the packet, stores the packet and checks the destination address. If $addr_B = addr_C$ then C gets the packet, otherwise, C forwards the packet to B. What C performs at the above is called *packet-switched routing*. There are several different approaches for C to choose another computer for forwarding the packet. We will study those approaches later.

  Multiple access networks

  Another approach for communications among multiple computers is to connect all computers to a shared communication medium. When Computer A sends a packet to Computer B, A broadcasts the packet on the shared medium with destination address B. All the computers in the network can hear the broadcast but only the computer with address B picks up the packet. Examples include Ethernet, Token Ring, etc.

  Local area networks (LAN)

  Ethernet, Token Ring, etc

  Wide area networks (WAN)

  Backbones of Internet, ARPANET, NSFNET

Metropolitan area networks (MAN)

Internet

Networks for parallel/distributed computing systems

Networks for multicomputers, networks for multiprocessors, and networks of workstations (NOW's).

- **More on wireless networks**

  Radio/TV broadcast, wireless communication

  Early wireless communication uses a central transmitter that broadcasts over an entire area. On a specific frequency, only one transmission is allowed at a time. Simultaneous transmissions can be realized by frequency-division multiplexing. The wireless network with one central transmitter has limited capacity.

  Cellular phone networks

  The idea is to partition a large area into cells, each cell has a central transmitter, the signal power is limited within the cell, and frequencies are reused in different cells. Both analog and digital transmissions are used in cellular phone networks.

  Wireless LAN

  Packet-switched wireless network started from Alohanet, a network used to interconnect the computers on Hawaii islands. Now, packet-switched wireless transmission technologies have been widely used in local area networks.

  Ad hoc wireless networks

  The networks are called mobile ad hoc networks (MANETs). A MANET uses packet-switched wireless transmission, does not have any fixed infrastructure, but has a set of nodes, every node in the network can move arbitrarily and works as both an end system and a router (transmitter).

  Sensor networks

  A sensor network consists of a set of sensors distributed in an area. Senors collect information and transmit data via wireless communication. Each sensor acts both an end system and a router. Usually, a sensor has a fixed location.

  Satellite networks

  In a satellite network, earth stations communicate with each other via satellite based antennas. The wireless link from an earth station to a satellite is called an uplink and the wireless link from a satellite to an earth station is called a downlink.

- **Network services**

  Networks provide services for users to exchange information via transmitting messages in bit streams or analog signals. Applications on networks include World Wide Web, audio/video streams, networked games, client/server, and many others.

- **Network service type**

  To support network applications, network provide connection-oriented service and connectionless service for transmitting messages. In connection-oriented service, the messages are delivered in the correct order from the source to the destination. There are three phases in the connection oriented service: connection establishment, data transfer, and connection termination. In connectionless service, packets are transmitted independently. The network does not guarantee the order of the packets. The receiver rearranges the received packets into a correct order.

- **Traffic characterization**

  Information is exchanged by transmitting data in bit streams or analog signals. For transmitting data streams, the traffic can be classified into three types: constant bit rate (CBR), variable bit rate (VBR), and messages. Information in some applications, like voice or video signals, is converted into a bit stream with a constant data rate (CBR) or variable data rate (VBR). While some other applications exchange messages each of which is a bit string.

- **Network performances**

  Major measures for the network performances include the end-to-end delay (latency), bit rate (throughput), and the error rate. The end-to-end delay is the time elapsed between the bit transmitted at the source and received at the destination. The throughput is the total number of bits per time unit that can be delivered over the network (or a link, or passing through a node). The error rate is measured by *bit error rate* of *BER* which is the fraction of the corrupted bits in the entire bit stream.

- **Quality of service (QoS)**

  Many applications require guaranteed delay, throughput, and error rate. It is a challenge to provide such QoS in the Internet.

- **Basic network mechanisms**

  Switching, multiplexing, error control, flow control, congestion control, resource allocation.

- **Switching**

  Two important switching techniques: circuit switching and packet switching.

- **Circuit switching**

  Used in telephone networks and all-optical networks. In circuit switching, a dedicated path is established from source to destination.

  Three phases in circuit switching: (1) circuit establishment, (2) data transfer, and (3) circuit disconnect.

  Connection establishment: A dedicated path is set-up to connect the source and destination. The path can be set-up by control signals on an auxiliary network. Another approach for the set-up is routing prob messages. A routing prob message contains the destination address and other control information. The source sends the prob message into the network and the message travels to the destination and reserves the links it passes through to form a path between the source and destination.

Message Transmission: Messages are transferred on the path in the FIFO order.

Connection Termination: The reserved path can be released by the source, destination, or the last part of the message.

Circuit-switching can be realized by either space-division switching or time-division switching. In space-division switching, paths are separated by physical spaces. In time-division switching, paths are separated by time slots. A device called *time slot exchanger* is needed at switches for the time-division switching.

The latency (transmission delay) of circuit-switching depends on the time delay of connection set-up and the delay of data transmission which is usually small and constant. So, circuit switching provides very high quality of services but is resource consuming because an entire path is reserved for one connection. It is appropriate for infrequent and large message transmissions (constant bit rate traffic).

- **Store-and-forward**

  In circuit-switching, if one link is not available in a path, then the path can not be used. This may result in the waste of resources. To improve this, the store and forward routing is proposed. In this approach, the source node attaches a header to the message to be sent that contains the destination address and sends the message to an output link if it is available. An intermediate node, on receiving a packet, stores the message at its buffer, checks the destination address in the header of the message, selects a next node on the route to the destination, and forwards the message to the next node if the link to the next node is available.

- **Message-switching**

  In message-switching, the entire message is stored and forwarded. Variable sizes of messages make the approach less efficient.

- **Packet Switching**

  Packet switching is an improved version of message switching. In packet-switching approach, a message is partitioned into packets. Each packet contains a header and a data area. A packet is transmitted from a node (source/switch) to a next node connected by a link. A node, on receiving a packet, stores the entire packet at its buffer, checks the destination address in the header of the packet, selects a next node on the route to the destination, and forwards the packet to the next node if the link to the next node is available. The routing decision can be made in two ways: Datagram and virtual circuit.

  In datagram, each packet is treated independently. Different packets from the same data stream can be routed in different links.

In virtual circuit, a route is established before the data transmission (the route is not a dedicate one). There are three phases in virtual circuit routing: path establishment, data transfer, path disconnection.

Performance issues for the packet-switched networks include the throughput of the network, the latency, and the buffer size at the switches. The latency can be very large if the network is congested. The queueing model is used to analyze the performances of packet-switched networks.

- **Multiplexing**

  Multiplexing refers to the technologies that allow multiple channels sharing a common medium. *Frequency division multiplexing* (FDM), *time division multiplexing* (TDM), *statistical TDM*, and *code division multiplexing* (CDM) are commonly used multiplexing technologies.

- **Frequency-division multiplexing (FDM)**

  In FDM, the frequency band of the transmission medium is partitioned into sub-bands, each sub-band is centralized at a specific frequency and supports a communication channel. Multiple channels share the medium using distinct sub-bands. Examples of FDM include broadcast radio and TV. In optical networks, wavelength is commonly used for frequency and FDM is called wavelength-division multiplexing (WDM).

- **Time-division multiplexing (TDM)**

  In TDM, a time unit is partitioned into time slots which are assigned to channels. Multiple channels share the transmission medium using distinct slots. A simple TDM is to assign one slot in every k slots to each of the k channels. This approach, however, may wast some slots if some channels do not have any data to transfer. Statistical TDM is proposed to solve this problem. In this approach, each channel has an FIFO input queue and the multiplexer reads packet in each queue in turn. If the multiplexer finds the queue of a channel is empty queue then it does not assign the slot to this channel and tries the next channel. This approach improves the network performance especially for busty traffics. Hierarchies of TDM structures have been developed for high speed data communication. In North America, the TDM carrier standards include DS-1,...,DS-4 for telephone networks and STS1,...,STS3072 for SONET (Synchronous Optical Network).

- **TDM vs time-division switching**

  The key difference between time-division multiplexing and time-division switching is that the former does not change the order of time slots assigned to the data streams while the later can change the order arbitrarily. A device called time slot interchanger (TSI) is used to realize the switching.

- **Code division multiplexing**

  CDM is a multiplexing technique using orthogonal codes to allow multiple data streams to share a common frequency channel. Here is an example: Assume that a frequency channel is shared by two users A and B. A and B are assigned codes $C_A = (a_1, ..., a_6) = (1, -1, -1, 1, -1, 1)$ and $C_B = (b_1, ..., b_6) = (1, 1, -1, -1, 1, 1)$, respectively. To transmit a sequence of binary numbers, A sends C_A for each binary 1 and $-C_A$ for each binary 0. Similarly, B sends C_B for 1 and $-C_B$ for 0. The receiver, who wants to get the message from A, performs $\sum_{i=1}^{6} d_i \times a_i$ for each received $(d_1, ..., d_6)$. It is easy to check that $\sum_{i=1}^{6} d_i \times a_i$ is 6 for 1 and -6 for 0 transmitted from A but $\sum_{i=1}^{6} d_i \times a_i = 0$ for the codes from B. Similarly, $\sum_{i=1}^{6} d_i \times b_i = 0$ for the codes from A. The codes assigned to A and B are called *chip code* and the data rate for transmitting the codes are called *chip data rate* which is larger than the data rate for the data of A or B only. The codes assigned to A and B have the property that $\sum_{i=1}^{6} a_i \times b_i = 0$. Such codes are called *orthogonal*.

- **Error Control**

  Error control in data communication includes error detection and error correction. Errors in data transmission include *damaged message* and *lost message*. For the damaged message, the receiver receives the message but some bits of the message are changed during the transmission.

- **Error detection**

  A simple method for detecting errors in the received message is the *parity check* approach. In this approach, a parity bit is appended to the end of a block of data such that the total number of 1s in the block is even (even parity) or odd (odd parity). On receiving the block of data, the parity is checked. If the pre-defined parity is changed, then there is an error in the transmission, otherwise no error is considered. The approach is simple but can only detect errors caused by odd number of bit changes. It is not reliable for large block of data.

  The cyclic redundancy check (CRC) is a more powerful error detection scheme. The CRC works as follows:

  - Choose an $(r + 1)$ bits number G
  - Make a binary string of $P2^r$ for message P
  - Divide $P2^r$ by G to get remainder R
  - Send $M = P2^r + R$
  - Receiver checks if M is a multiple of G, i.e., if M divided by G is zero. If so, no error, otherwise error.

  The CRC approach can not detect the error which changes the value of M into another multiple of G. However, this can happen with extremely low probability for an appropriately chosen G.

  Another error-detection approach for large block of data is the checksum code (CKS) which works as follows:

  - Partition the message into segments of n bits
  - Find sum S of segments by bit-wise addition

  - Send $-S$ as checksum code with the message
  - Receiver calculate S, if S plus checksum code (bit-wise sum) is 0, no error, otherwise error detected

- **Error correction**

  A simple approach for error correction is to retransmit the message if an error is detected during the transmission. If a receiver detects an error in a received message, it sends the source a negative acknowledgment (ACK) and retransmission request. A receiver can send the source a positive ACK on receiving a message without finding any error. By the positive ACK and a timer, the source can detect a lost message. The above retransmission is known as *automatic repeat request* (ARQ).

  Using error correction code is another approach for error correction. In this approach, redundant bits are added to the transmitted data in such a way that the receiver can finds which bit has an error. For this purpose, the number of redundant bits is much larger (usually proportional to the length of the data) than that used in CRC or CKS.

- **Flow Control**

  Flow control is a process to manage the data rate in a transmission. The basic technique for flow control is stop and wait. Sliding window is an improved version of stop and wait.

- **Stop and wait**

  The source sends packet P_i only when a positive ACK for P_{i-1} has been received. This is a reliable approach and combined with a timer it can be used for error control. However, this approach has a poor performance on data rate.

- **Sliding window**

  This approach allows the source sending more packets before a specific ACK is received. In this approach, the source has a list (called window) on the packets that can be sent out before an ACK is received. Similarly, the destination has a window on the packets it can accept. On the source, once a packet is sent out, the packet is deleted from the window and once an ACK is received, new packets are added to the window for sending. On the destination, once a received packet is processed the packet is deleted from the window and once an ACK is sent out and the window is expanded to accept more packets. If the packets are arranged in the order to be sent/received, the windows slide on the sequence of packets. The sliding window flow control has improved performance over the stop and wait approach.

- **Resource Allocation**

  Different applications have different requirements on the quality of service, which in turn require different resources, of a network. A network can provide a specific QoS for an application only if it dedicates the required resources to the application. Resource allocation includes bandwidth allocation, buffer allocation, and admission control. In circuit-switched network, the bandwidth is assigned to an application by a dedicated path for the application. In packet-switched network, if applications are on a connectionless service, the network does not have information to distinguish packets from different applications and thus can not allocate resources for a particular application in a well defined way. However, some rough allocation schemes are possible. For example, give the packets from different applications from different priorities to be routed in the network. For the virtual circuit-switched networks, the switches in the network can treat the packets in different connections based on their virtual circuit identifiers. It is important that applications reserve the resources they required from the network. The resource reservation protocol (RSVP) is a protocol for this purpose used in the Internet.

- **Layered structure of protocols**

  Functions of network are organized into layers. Service of a layer n is realized by protocols at layer n using services of layer $n-1$. The layered structure makes the design of protocols easier.

  TCP/IP Model

  The TCP/IP model is a result from the research funded by the US Defense Advanced Research Projects Agency (DARPA) for ARPANET (DARPA has also been called ARPA, Advanced Research Projects Agency). Now it is used by the Internet. There are five layers in the model:

  Application layer, deals with user applications

  Transport layer, deals with end to end reliable delivery of data. Transmission Control Protocol (TCP) is used in this layer.

  Internet layer, deals with data transmission across multiple networks. Internet Protocol (IP) is used in this layer.

  Network access layer, deals with data exchange between the end device and network.

  Physical layer, deals with the access to physical medium.

  OSI Model

  Open System Interconnection (OSI) model is developed by International Organization for Standardization (ISO). The model has seven layers.

  Application layer: It enables the user, human or software, to access the network. It provides user interfaces and support for services such as email, etc.

  Presentation layer: It ensures interoperability among communicating devices. It provides the necessary translation of different control codes, character sets, etc. to allow two devices to understand the transmission in the same way.

  Session layer: It establishes, maintains, and terminates connections (sessions) between cooperating applications.

  Transport layer: It provides end-to-end delivery of entire message, error recovery, and flow-control.

Network layer: It deals with end-to-end delivery of a data packet. It handles switching and routing.

Data link layer: It provides the reliable transfer of data across the physical link.

Physical layer: It deals with access to physical medium.

- **Protocol Data Unit (PDU)**

  A Protocol Data Unit (PDU) is data packet which contains the data from the next higher layer and a layer head which contains the control information. For example, a PDU from the transport layer has the transport layer head which contains the information on source/destination service access points (called ports in TCP), sequence number of the packet, error-detection code, and so on.