# Module 1 Introduction to computer networks

# Introduction

In this module, we will look at different components of a computer network, popularly known as layers, and discuss their functions. There are two basic models used to describe computer networks, OSI and TCP/IP. The TCP/IP model is the prevalent model and will be the center of discussion throughout this course.

# Why study computer networks

We need to generate and disseminate information. Following are some of the daily objects of information exchange of user community.

- 1. Music files
- 2. Emails
- 3. Phone Calls
- 4. Blogs
- 5. Remote Database Servers
- 6. Facebook Posts
- 7. Skype Calls, audio, and Video

Consider the services that are listed above or a few other similar services that you may think of yourself. You will find one common thread that binds all of them together - the network; either a local network or the Internet. In all above cases, the resources are being shared and the network is the media for sharing.

For people like us, who regularly use services like above, the major need is the networking infrastructure. An interesting question that we would like to get an answer of during these course is, how this network infrastructure work, and how we put it to use for providing services like above.

## Layers

One of the most common strategies to solve a problem is "Divide and Conquer". Computer networks follow this strategy. The entire networking problem is solved in pieces. These solution pieces are independent to a large extent. More importantly, they can evolve independently of each other without much trouble. Each piece, in networking parlance, is known as *a layer*. We will look at each piece, now onwards referred to as a layer, one after another during this course. Layers are arranged in hierarchical order, lower layer provides a service to a layer above and that layer provides service another layer above it and so on.

There are some reasons for designing networks in form of hierarchical components; i.e. the layers. Here is the list

- A component's work is clearly defined. For example, a layer known as network layer is assigned a job of deciding where to send the outgoing packet. Irrespective of what others do and how others do their job, the network layer continues to make decisions about sending packets on their right route. Similar is the case with other layers.
- 2. When each layer is working independently, it becomes easier to replace that layer by another, more suitable for changed situation or faster or smarter version. For example, IPv4, a typical network layer used by the Internet is being replaced by IPv6. Both are network layer protocols. If you want your computer to start using IPv6 and replace IPv4, nothing on the computer needs to change, neither your FTP program, nor your Ethernet or Wi-Fi card.
- 3. Each layer is designed to take a specific service or providing a service or both. That means, each layer's functions and interfaces are clearly defined. Different vendors competing for designing products get a fair market and thus can produce best products at reasonable prices.
- 4. These layers help the designers divide the service into small manageable components and thus, help them evolve independently. A company researching on Gigabit Ethernet and Terabit Ethernet does not need to bother about what IETF (Internet Engineering Task Force, a standardization body for Internet) is doing for Internet standards. The Ethernet card technically works at lower two layers, popularly known as Physical and Data Link Layers and do not need to worry about network, transport and application layers where IETF works (standardizes). Similarly, other companies which design clients and servers are operating at the application layer, they have nothing to do with Ethernet and other cards however their data, in bits are carried by those cards to the other end.
- 5. Layering enables standardization of specific layers and thus make sure that companies who design their products can interoperate. For example, if we design a new browser and we have followed IETF guidelines for developing a browser, our browser can communicate with any web server of this world.
- 6. If a component is to be replaced, for example, I bought a new 10Gb Ethernet card for my laptop, do I need to upgrade my browser? Should I reinstall my TCP/IP software? Nothing else needs to be changed. It is because when the interface is standardized, replacing a component with another, providing identical service but otherwise different component, is easy and even seamless in most cases.
- 7. This layering and standardization together provide one more advantage. When I develop a browser, and my users start using it, I do not need to compel them to use a specific card or operating system. They can use any brand they like.

However, Layering has some disadvantages as well. Let us look at few major ones.

- 1. The first disadvantage of layering architecture is that it is inefficient when applied to small problems. It is like dividing a small work into pieces. That usually adds to and not reduces the overall work.
- 2. The second problem is to synchronize and regulate the function of all the layers. When synchronization overhead is not acceptable, layering should not be called for. Demand for synchronization reduces the speed too. For example, a data link layer has many incoming packets which it passes on to a network layer. However fast the data link layer works, it cannot work faster than the network layer consumes the data it produces. Another way round, if a transport layer provides data to a network layer for sending it to another network, and the network is congested (there is a traffic jam), the network layer cannot work as fast as the transport layer expects it to and thus it has to slow down.
- 3. Based on above discussion, it is clear that each layer has some speed of processing and the overall speed of the system is equal to the slowest component, i.e. the layer which processes data in the slowest manner.
- 4. When inter-layer communication is minimized to preserve the independence of layers, it becomes more time consuming for a layer to judge the environment.
- 5. The memory usage, in the layered architecture, is more than a normal. In a multilayered architecture, different stages of a process running at different layers (for example, on the Internet, the applications like SMTP, FTP, Telnet, etc. run at the application layer; at the same point of time, the TCP and UDP run at the transport layer and IP runs at the network layer) have to be stored and retrieved repeatedly as and when a particular process running at a specific layer starts, stops, and resumes their operation. Therefore, the system requires more storage space (memory) to execute a large number of processes. When the memory, as well as the processing power, is limited, layering is not a good solution. IoT devices like car sensors, implantable medical devices like insulin pumps, garage door sensors, etc. have very small amount of memory and thus they do not prefer to have layers like conventional networks.

#### Two models, OSI and TCP/IP

Let us try see how these layers are organized. There are two ways described in the literature, the first one being used for pedagogy but never used in practice and another is normally used but one has to retrofit like the first one to study it. We will introduce both of them here.

The OSI layering mechanism was designed by Open System Interconnection group from ISO, hence the name OSI. The OSI model employed a seven-layer scheme which was heavily influenced by a model known as SNA (System Network Architecture) from IBM. The TCP-IP

model is a retrofit to a practical solution provided by the Internet community. The OSI model is almost non-existent in terms of its implementation. Our discussion in the course follows the TCP/IP model for discussion and whenever necessary, we may consider the OSI model as well.

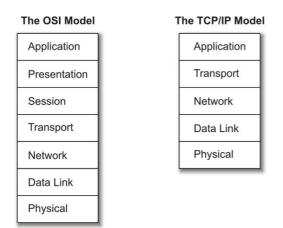


Figure 1.1 OSI and TCP/IP models

### The OSI Model

This model was designed by International Standards Organization in 1983. Some of the leading companies at that time and some of the governments thought of having a standard solution for the network and the services provided.

The layering mechanism of OSI model contained seven layers. They are detailed in Figure 1.1. It contains two extra layers as compared to the TCP/IP model, i.e., the presentation layer and the session layer. One may like to have them, though having other things are much more important.

The presentation layer is designed to check minor differences in the way the data is presented by the sender and the receiver. For example, the sender and the receiver may have different ways of storing integer values in the memory. Some machines store an integer starting with the most significant bit to the least significant bit, while other machines store it starting from the least significant bit to the most significant bit. The role of presentation layer is to convert the data to ensure that it is presented according to the native representation scheme of the sender or the receiver machines.

The session layer is designed to manage sessions between the sender and the receiver. Both the functionalities (of the presentation and session layers) are too small to deserve a layer. The TCP/IP model does not have either of them.

Additionally, the TCP/IP model does not even mention anything about the data link or the physical layer (which exists in OSI model), but all the network cards used for networking have them, so they are mentioned here.

### The TCP-IP Model

This model is used by the networks that we use and the internet that we access. That means it is the real world network. TCP/IP network is based on an approach known as connection oriented service over a connectionless delivery model. Let us try to understand that.

The connection-oriented service demands connection to be established before any communication can begin. The opposite, called connectionless, works in a way that does not require to establish a connection before any data transfer.

The connection-oriented mechanism 's best example is the telephone system. When we call somebody, the call is established first and then the communication begins. The line remains occupied during the call, irrespective of the case that we use the line or otherwise.

The best example of the connectionless system is sending SMS. The sender and receiver do not need to remain sync, the connection is not established, before sender starts sending, the connection is not required to be terminated once the SMS is sent, the receiver might receive multiple SMS sent by the sender in some other order than it is sent.

The TCP/IP model combines the connectionless and connection-oriented approaches in its design. Applications like Telnet, FTP, Web browsing uses a model where connection oriented service is provided by TCP/IP model over a delivery design which is connectionless. Let us try to understand. Look at Figure 1.2.

Sender's TCP layer (transport layer in OSI Parlance), wants to communicate with receiver's TCP layer. This happens in most of the applications that we use daily. The process demands the establishment of the connection beforehand and terminating the connection afterward. We will study how this happens when we look at how TCP functions later in module 24,25 and 26. Interestingly, the TCP gets the job done by instructing a layer below, the IP layer. IP layer, unlike TCP, does not establish a connection but just sends the packet to next router, next router sends it further to next to next router and so on till the receiver accepts the same. Like SMS messages, the receiver might receive the message in any order, might not receive the message without sender learning about the failure and so on. Thus, the IP is said to deploy connectionless delivery mechanism. The combined work of these two protocols, TCP and IP, describes the connection-oriented service over the connectionless delivery system. In fact, the TCP/IP model also provides an option to have Connectionless service over the connectionless delivery system, deployed using UDP/IP protocol. We will brief about UDP in module 25. The TCP/IP and OSI models differ in many ways, the major

difference, though, is the one which we described above, the connection-oriented service over connectionless delivery system. OSI did not do that.

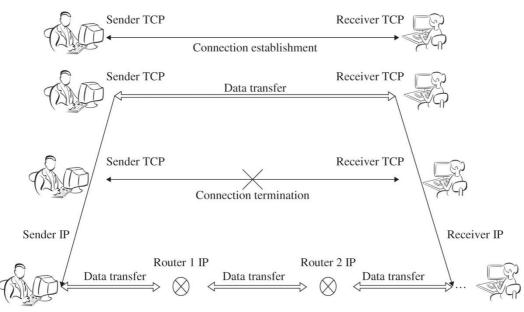


Figure 1.2 Connection oriented service over a connectionless network.

### Functions of each layer

Now let us look at each of the layers, from the bottommost physical layer to the topmost application layer. We will look at the functioning of each layer at length later. Let us begin with the physical layer.

#### The physical layer

The job of the physical layer is to carry bits from one end to another using the communication medium available, i.e., a wired or a wireless connection to transfer the bits to the other end. It is interesting to note that there is more than one mechanism to transfer the bits from one end to another using the same medium. The study of physical layer describes different ways of transferring bits from one end to another and their pros and cons.

There are two basic methods of transferring bits across, first is using analog (curved) waves, popularly known as analog signaling and another using digital (square) signaling. The digital signals are of two different types when discrete voltage levels represent zeros and ones, it is known as digital signaling. As zeros and one are at different levels, when one draws a graph, it draws square figures and thus known as square signals. This layer determines how 1 and 0 are represented when sent across. This layer also manages how the job is done when multiple senders are sending and multiple receivers are receiving. Look at figure 1.3. The data link layer passes bits to the physical layer, physical layer converts that to signals and send the signals to the recipient. The receiving physical layer converts the signal back to bits. There are two basic modes of transmission, wired and wireless. In wired

communication, the physical layer manages to send data over the wire while the wireless case, the data is transmitted over the air. There are two types of cables used in practice. A copper cable (a series of cat-3, cat-5, cat-6 and cat-7 cables), or a fiber optic cable is usually the choice when wired communication is done.

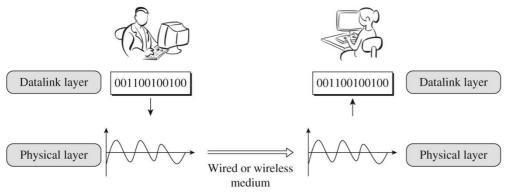


Figure 1.3The work of physical layer

In a wireless mechanism, there are many methods used to transmit. A common method called Direct Sequence Spread Spectrum uses multiple bits to send a single bit. Another method called Orthogonal Frequency Division Multiplexing is also commonly used. However, all these methods use some frequency to transmit the bits over. The transmission technically is, always in form of electromechanical signals. Copper cables send electrons over the media while fiber optic cables are made up of glass and send photons across. We will look at those methods later.

#### The data link layer

The data link layer's job is to send the bits using the physical layer. Additionally, it provides quality control measures by ensuring the bits sent and the bits received remain identical. This is important as there is every chance of the data getting corrupted in the transit. The data link layer provides ways for the sender and the receiver to recognize erroneous or unintended data. To ensure error-free transmission, the data link layer adds additional bits to the data using some algorithm. These additional bits are calculated from the data itself. The calculation procedure is designed in a way such that the additional bits are different for different data. The same algorithm is also applied to the data at the other end and the results are compared. If the results match, then the data is accepted; otherwise, it is rejected. This is known as the *error-detection mechanism*. In some cases, the bits added are designed to not only detect the errors but also to correct them.

Correcting errors is also possible to a limited extent. When the redundant bits are possible to point to the exact place where the error has changed the bit, it is possible for the receiver to change the bit value to the original position. That process is known as error correction.

To perform these actions, the data link layer encapsulates the packet given by network layer into a frame. That means the data unit at data link layer is known as a frame. The frame

contains many fields including sender's and receiver's unique address. Those addresses help the system to determine who is the recipient of the frame and route the frame to that node. On the other hand, the sender address enables the receiver to understand who has sent this frame and so can respond back if need be. Another important part of the frame is collection of fields which can help detect or correct errors.

It is also possible that the receiving data link layer sends back the confirmation in form of some type of acknowledgment. An acknowledgment (or ack for short) indicates that frame is received properly at the other end. Many Data link layers have explicit mechanisms to provide sending and receiving acks and taking appropriate actions based on that. Look at figure 1.4. The data coming from network layer is divided into multiple chunks<sup>1</sup>. The data link layer generates the frame out of it. The frame contains some header as well as the trailer.

Another critical issue is the mismatch in the speeds of the sender and the receiver. The sender might send faster than the receiver can process and thus it is possible that some data is lost due to this problem. Data link layer deploys some mechanisms by which it can request the sender to slow down. That mechanism is popularly known as flow control.

### The Network layer

The network layer's job is to look at the packet, decide where it is heading to (i.e. what is the value in the recipient's address field of the packet), decide the best path to that destination requires it to forward it to which neighbor. That means, the network layer's job is to find next immediate destination based on final destination.

In fact, this requirement demands two distinct jobs to be done. First, every router<sup>2</sup> must know where every other network in the world is, and how to reach to each and every network. Once the router learns that, it can always judge the nearest next immediate neighbor for any destination. The routers store that information in a tabular form, popularly known as a routing table and this process, therefore, is known as routing. The algorithm deployed by the router is known as the routing algorithm<sup>3</sup>.

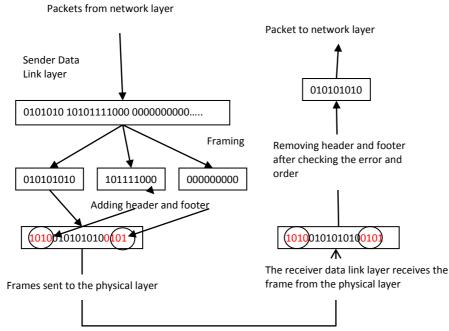
Another job that the network layer is designed to do is called forwarding; i.e. forward the incoming packet to the right neighbor based on its destination address and information available in the routing table. Forwarding is all about extracting destination address from an incoming packet, look into routing table to see what is the ideal next neighbor for that packet and then send it over the line where that neighbor is connected. Figure 1.5 details it.

<sup>&</sup>lt;sup>1</sup> Usually, network layer content is designed based on the capacity of data link layer and thus a single network layer packet usually fits into a single data link layer frame.

<sup>&</sup>lt;sup>2</sup> Router running the network layer which decides the route

<sup>&</sup>lt;sup>3</sup> All these things happen at network layer of the router

Simple! Isn't it? However simple it looks like, forwarding is an extremely complex process as it is to happen in real time. That means the packets are to be processed as fast as they arrive. Consider router having five incoming lines each of which having a capacity of 10Gb. In the worst case, when each of the lines is blasting at the full speed, a router has to process packets 50 Gb/sec which sounds phenomenal but common across routers. Consider the capacity of CISCO CRS -3 type of router's high-end version can crunch 322 Terabits per second.



*Figure 1.4. How data link layer works, at sender as well as receiver* 

Forwarding, like many other things in networks, can be done in two different ways. Connection-oriented and connectionless. In the connection-oriented fashion forwarding, the sender network layer establishes a connection with receiver's network layer before transmission and tears off once it is done. Every packet belonging to that connection travels over that path decided and does not roam around anywhere else. Connectionless forwarding just sends the packets across and hope for the best!

#### The Transport Layer

The transport layer is working as per instructions by the application layer. Suppose we type <u>www.oup.com</u> in our browser. The browser, which is technically a client process running HTTP (Hypertext Transfer Protocol) client, then communicates with the web server located at Oxford University Press website. How does it do it? It instructs the TCP (Transmission Control Protocol) running on your machine to establish the connection with the server at <u>www.oup.com</u> and once the connection is established, sends this URL to that server. The server sends back the home page of the website and the browser displays it in return.

The underlying networks which carry our traffic remain as unreliable as they were in past. The communication lines have improved but there are many wireless links added which drops packets quite frequently and thus the application (like our browser, and OUP's server), needs a reliable connection which delivers the content reliable at the other end despite communication line being unreliable. The transport layer promises reliability and fulfills it by a simple trick.

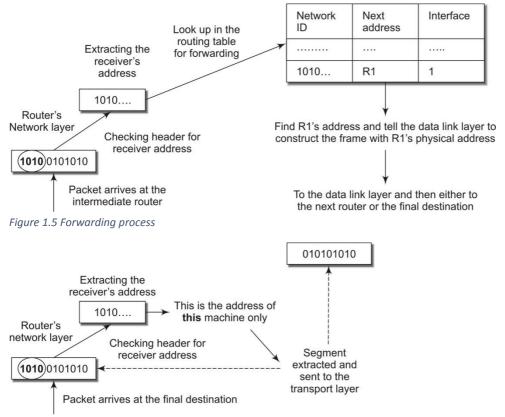


Figure 1.6 when the packet arrives at the final destination

That trick is known as timeout and retransmits with acknowledgment. Every data unit that is being sent, is acked by the receiver. When the data unit (called segment in TCP/IP parlance), is corrupted, the receiver won't send the ack back. Thus either if the data unit itself is lost (in that case the receiver won't receive that segment) or it is corrupted, the sender won't get the ack back. In the case ack is not received in time which is known as Time Out in TCP parlance, the transport layer is designed to send it again. That is known as retransmission. Thus TCP recovers from a lost segment or a corrupted segment without informing the application layer. Whenever we send the URL and receives the page, we do not find anything missing because of this mechanism.

The event Time Out and the process of retransmission are managed using a timer. Whenever a segment is sent, TCP remembers the time and also calculates an expected time of ack coming back. Figure 1.7 describes the process. Seems a great solution! Not always so! Let us see.

Take an example of sending a video to the other end. Suppose in a cartoon movie, Mickey Mouse is shown moving from left to right. There is a creature sitting in the middle of the screen. Frame by frame, Mickey is coming closer to that creature. In one of the frames, Mickey kicks that creature, the next frame shows the creature in the sky, and the next sequence shows the creature falling on top of Mickey. Now assume the frames are sent one by one and the kicking frame is lost. Then the viewers will automatically assume that the creature must have been kicked. It is absolutely fine till now. If the kicking frame is retransmitted, then we have a sequencing problem. If Mickey is shown kicking that creature after the creature falls down, then it will create a confusion in the minds of the viewers as to how Mickey came out from beneath the creature and kicked it.

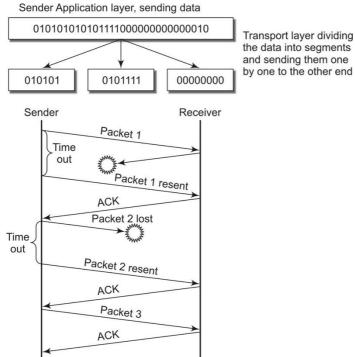


Figure 1.7 Timeout and Retransmission.

So it is better not to transmit the lost frame and keep it lost. Another example in order. Suppose we are sending a message to the server as follows. "Allow Virendra, Jayant Ashwin Disallow Root Bairstow Cook". What is Disallow is little delayed and played after Bairstow? The receiver incorrectly concludes and allow Root and Bairstow. Now the question arises—if we do not need retransmission of lost data, then what is the necessity to have timers and count the number of transmitted frames? If nothing is required, TCP becomes an overhead. Same is the case with sending audio. A word is lost and retransmitted would create a lot of confusion in listener's mind. In fact, the best option is to skip the lost or delayed frame. Human listeners will learn about it intuitively. Retransmitting them later is the last thing that one should do. If ever need, human listener may ask the sender to repeat everything (and not the lost segment!) to clarify any doubt.

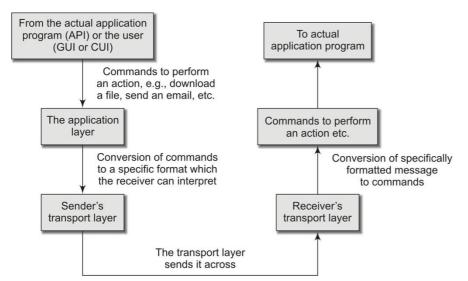
When there is no need for retransmission, what is the need to keep timers, check for each segments time and do all that accounting? Thus TCP is not used for such cases, UDP (User

Datagram Protocol) is a better choice where there is no such formality is maintained. It is a transport protocol with the only option for sending the segment across and forget afterward. In fact, there is one more alternative called SCTP (Stream Control Transfer Protocol) which is more suited for such cases but most of the installations do not provide SCTP. The great thing about TCP/IP model is that it provides all these options while older OSI model did not have any such options.

#### **Application Layer**

All applications, when run, talk to the application layer for all their communication requirements. Consider FTP. When we run an FTP client on our machine and download or upload a file, our client communicates the FTP server at the other end and download and upload files, we execute commands like cd, get, put, mput, mget etc. The commands are processed at the application layer. The client program and the server program both runs at the application layer. When the user types command, the application layer realizes that the communication is to be made and ask TCP to send that command (get) across. TCP uses all its ability to send that command on the server's side. When the FTP server receives that command from its TCP, it processes that command, decide to send that specific file (if exists) and ask the TCP to send that file across.

Thus application layer, the topmost layer, is the one which takes help of transport layer to send and receive messages related to applications they are part of. The transport layer takes the help of the network layer to route that message to the other end, the network layer takes the help of data link layer to make it error free. The data link layer takes the help of physical layer to convert that message into signals to send them across.



*Figure 1.8 The relation between application, application layer and transport layer.*