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Introduction

In the first lecture, we look at major milestones in the history of the Internet and review the original design choices and principles of the Internet architecture. Then, we look at how these initial design choices have formed the shape of the Internet architecture, which looks like an hourglass figure. We learn about a model that can help explain the hourglass shape. Finally, we explore what would happen if we redesigned the Internet architecture from scratch; in other words, what if we adopted a cleanslate design? Towards that goal, we look into some example architecture design approaches that optimize for network control, management and accountability, which were not first-class goals when the Internet was first designed.

The number of devices that connect to the Internet has been exploding, with the number reaching more than 22 billion as of 2019. These devices or end systems can be very diverse, for example: our laptops, mobile devices, tablets, cameras or more recently household devices (refrigerators and temperature control systems). Throughout the course we will be learning about what it takes for two hosts in potentially different networks and also perhaps physically located in different parts of the world to exchange data. We will be learning about the protocols and the infrastructure we need, but also how to overcome the challenges. In this first lecture, we will start by looking at the devices (e.g., bridges and switches) that help provide connectivity between hosts in the same network.

Readings and Additional Resources

Important Readings:

How to Read a Paper.

https://people.cs.umass.edu/~phillipa/CSE390/paper-reading.pdf (Links to an external site.)

How to read a research paper:

https://www.cs.tufts.edu/comp/150PLD/ReadingPapers.pdf (Links to an external site.)

The Design Philosophy of the Darpa Internet Protocols http://ccr.sigcomm.org/archive/1995/jan95/ccr-9501-clark.pdf (Links to an external site.)

The Evolution of Layered Protocol Stacks Leads to an Hourglass-Shaped Architecture

https://www.cc.gatech.edu/~dovrolis/Papers/evoarch.pdf (Links to an external site.)

Books references: If you have access to the Kurose-Ross book and to the Peterson book, below you can find the list of the chapters we have as a reference in this lecture. As we have mentioned in the course schedule, purchasing the books is not required.

- Kurose 1.5.1 (Edition 6): Layered Architecture
- Kurose 1.5.2 (Edition 6): Encapsulation
- Computer Networks: A Systems Approach - Edition 4, Section3 and 3.1 (Interconnecting hosts and networks)
- Computer Networks: A Systems Approach - Edition 4, Section 3.2.1 (Learning Bridges)
- Computer Networks: A Systems Approach - Edition 4, Section 3.2.2 (The looping problem and the spanning tree algorithm)

Optional Readings:

Brief History of the Internet (1997)

https://www.internetsociety.org/wp-content/uploads/2017/09/ISOC-History-of-the-Internet_1997.pdf (Links to an external site.)

Rethinking the design of the Internet

https://dspace.mit.edu/bitstream/handle/1721.1/1519/TPRC_Clark_Blumenthal.pdf?sequence=1&origin=publication_detai
(Links to an external site.)

The End-To-End Argument

http://web.mit.edu/Saltzer/www/publications/endtoend/ANe2ecomment.html (Links to an external site.)

http://web.mit.edu/Saltzer/www/publications/endtoend/endtoend.pdf (Links to an external site.)

Internet Clean-Slate Design: What and Why?

http://citeseerx.ist.psu.edu/viewdoc/download?doi=10.1.1.98.8874&rep=rep1&type=pdf (Links to an external site.)

A Clean Slate 4D Approach to Network Control and Management

https://www.cs.princeton.edu/~jrex/papers/ccr05-4d.pdf (Links to an external site.)

Holding the Internet Accountable

https://conferences.sigcomm.org/hotnets/2007/papers/hotnets6-final71.pdf (Links to an external site.)

An Algorithm for Distributed Computation of a Spanning Tree in an Extended LAN

https://www.it.uu.se/edu/course/homepage/datakom/ht06/slides/sta-perlman.pdf

Why Study Computer Networks?

Internet growth

The Internet is one of the most exciting and influential inventions. Even though it started as a research experiment that escaped from a lab, it eventually evolved into a global communications infrastructure that has been transforming almost all aspects of our lives with tremendous impact. Given the explosion of applications that are becoming available, and the technologies that make it possible for different types of devices to connect to the Internet (for example IoT, vehicles, sensors, home devices, etc), the number of Internet users keeps increasing. As of June 2018, the number of Internet users has been estimated at about 3.2 billion, while the number of estimated Internet users by 2020 is 4 billion.

Networks play an instrumental a role in our society

Indeed, the Internet has been playing a transformative role in our lives. Just to name a few examples, it has been changing the way we do business; for example, e-commerce, advertising and cloud computing applications. It has been changing the way we connect and communicate; for example, e-mail, instant messaging, social networking and virtual worlds applications have been extremely popular. The Internet has even been changing how we fight; for example, we frequently read headline news about large-scale cyber attacks, incidents related to the distribution of fake news, censorship and nationwide attacks. In turn, these developments have law implications both nationwide and on a global scale that we hadn't considered in the past; for example, we have to examine questions such as which countries are responsible for the Internet traffic that is crossing national boundaries, while the end hosts are located in multiple countries across the globe?

Networking is a playground for interdisciplinary research innovations

The Internet's huge transformative role is connected with an ever-expanding and evolving collection of technologies, system and protocol architectures, algorithms, as well as powerful applications. Indeed, the Internet is an amazing "playground" of ongoing cross-disciplinary innovations that are coming from multiple fields such as distributed systems, operating systems, computer architecture, software engineering, algorithms and data structures, graph theory, queuing theory, game theory and mechanism designs stemming from machine learning and AI, cryptography, programming languages, and formal methods, and more.

Networking offers multidisciplinary research opportunities with potential for impact

Networking is a field that offers tremendous opportunities for interdisciplinary research work that sometimes crosses fields that can be very different from each other. For example, in the context of research projects that study how to incentivize Internet providers to keep their networks clean from infected hosts, the research work may span fields such as Internet security, economics, and social sciences. What is perhaps the most exciting aspect about research in the networking field is the ability to design innovative and impactful solutions, and immediately put them to test by leveraging existing platforms.

A Brief History of the Internet

J.C.R. Licklider proposed the "Galactic Network" (1962)

The first vision of a Network - proposed by J.C.R. Licklider as the "Galactic Network" - was at MIT back in 1962. He envisioned that everyone could quickly access data through a set of interconnected computers. Licklider - as the head of the research program at Defense Advanced Research Projects Agency (DARPA) - led a group of researchers to experiment connecting two computers. An MIT researcher, Lawrence G. Roberts, connected one computer in MA to another computer located in CA with a low-speed dial-up telephone line.

The ARPANET (1969)

The results of the first experiments showed that time-shared infrastructure was working sufficiently well at that moment. However, at the same time researchers indicated the need for packet switching technology. Roberts continued developing the computer network concept, which resulted in the first network connecting four nodes (from UCLA, Stanford Research Institute, UCSB and Univ. of Utah, respectively) into the initial ARPANET by the end of 1969.

Network Control Protocol (NCP), an initial ARPANET host-to-host protocol (1970)

As the number of computers that were added to the ARPANET increased quickly, research work proceeded to designing protocols. The initial ARPANET Host-to-Host protocol called Network Control Protocol (NCP) was introduced in 1970, and it allowed the network users to begin developing applications. One of the first applications that launched was email in 1972.

Internetworking and TCP/IP (1973)

At the same time, a DARPA team of researchers led by Bob Kahn introduced the idea of open-architecture networking so that the individual networks could be independently designed and developed in accordance with the specific environment and user requirements of that network. This led researchers to develop a new version of the NCP protocol, which would eventually be called the Transmission Control Protocol / Internet Protocol (TCP/IP). Khan collaborated with Vint Cerf at Stanford and presented the original TCP paper in 1973. The first version of TCP later split its functionalities into two protocols: the simple IP, which provided only for addressing and forwarding of individual packets, and the separate TCP, which focused on service features such as flow control and recovery from lost packets.

The Domain Name System (DNS) (1983) and the World Wide Web (WWW) (1990)

The scale of the Internet was increasing rapidly, and as a result it was no longer feasible to have a single table of hosts to store names and addresses. The Domain Name System (DNS) - which was designed to translate domain names to IP addresses by a scalable distributed mechanism - was introduced by Paul Mockapetris at USC in 1983. More applications sprung up quickly. One of the first and most popular applications was the World Wide Web (WWW), which was introduced by a team of researchers led by Tim Berners-Lee.

Internet Architecture Introduction

After looking at the major milestones in the history of the Internet, let's take a closer look into the current architectural design of the Internet.

Connecting hosts running the same applications but located in different types of networks. A computer network is a complex system that is built on top of multiple components. These components can vary in technologies making up different types of networks that offer different types of applications. For example in the figure below, we have two BitTorrent clients that communicate even though they are using very different networks/technologies (Wifi vs Ethernet). So, how do these technologies and components interconnect and come together to meet the needs of each application? The designers of the network protocols provide structure to the network architecture by organizing the protocols into layers.

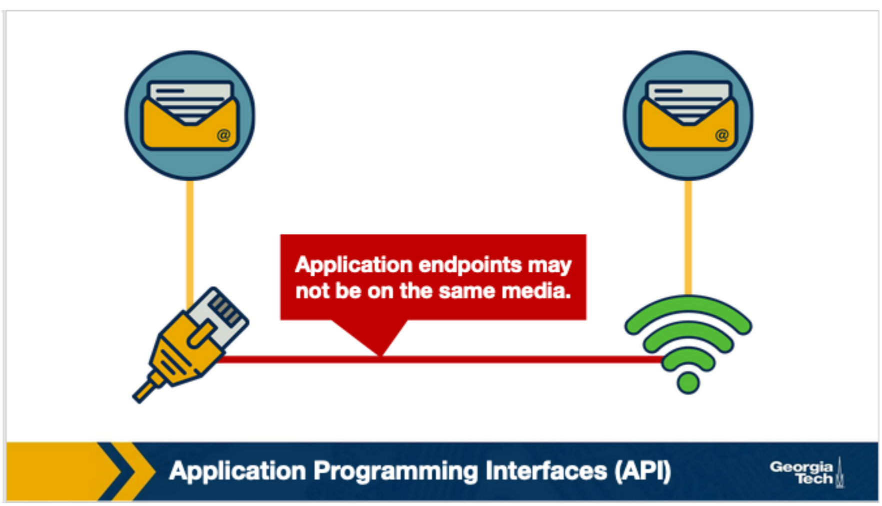


Figure 1: image

Architecture, layers and functionalities. So, the functionalities in the network architecture are implemented by dividing the architectural model into layers. Each layer offers different services.

An analogy. An analogy we can use to explain a layered architecture is the airline system. Let's look first at the actions that a passenger needs to take to move from the origin to the destination place. The passenger purchases the ticket, checks the bags, goes through the gates and after the plane takes off, the passenger travels on the plane to their final destination. At the final destination, the passenger leaves the aircraft, goes through the gate and claims their baggage.

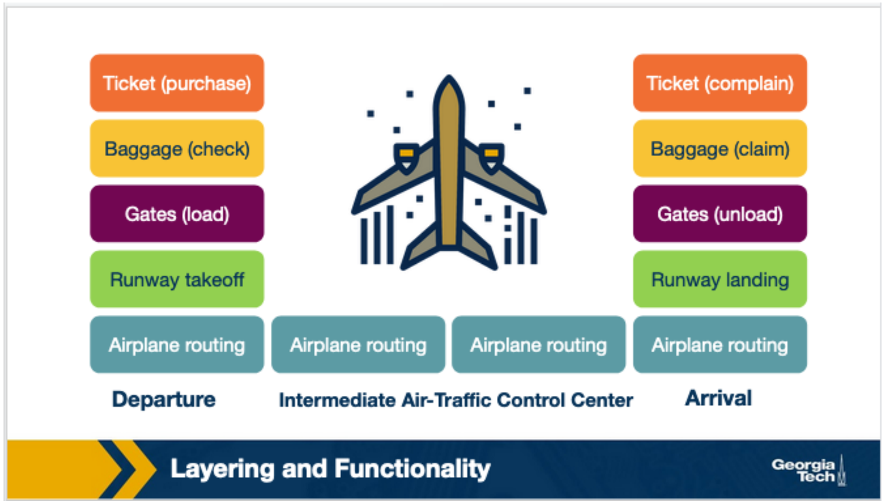


Figure 2: image

We can look at the above picture to identify a structure (or layers) and the services (or functionalities) that are offered at every component of the structure. Dividing the services into layers we get the framework below. We notice that in this framework every layer implements some functionality. Every layer works based on the service provided by the layer below it, and also it provides some service to the layer that is above. The same principle of layers and functionalities is implemented with the model of the Internet architecture.

Layered architecture advantages: scalability, modularity, and flexibility. Some of the advantages of having a layered network stack include scalability, modularity and the flexibility to add or delete components, which makes it easier overall for cost-effective implementations.

The OSI Model

The Internet architecture follows a layered model, where every layer provides some service to the layer above. The International Organization for Standardization (ISO) proposed the seven-layered *OSI model* shown below, which consists of the following layers: application layer, presentation layer, session layer, transport layer, network layer, data link layer, and physical layer.

We will see in later sections a possible explanation about why the Internet architecture came eventually to have this form. Separating the functionalities into layers offers multiple advantages. However, there are disadvantages of the layered protocol stack model, including:

1. Some layers functionality depends on the information from other layers, which can violate the goal of layer separation.
2. One layer may duplicate lower layer functionalities. For example, the functionality of error recovery can occur in lower layers, but also on upper layers as well.
3. Some additional overhead that is caused by the abstraction between layers.

In the following sections, we will go through a brief overview of the layers, and more specifically we will focus on what each layer does (service), how the layer is accessed (interface), how the layer is implemented (example protocols), and how we refer to the packet of information it handles.

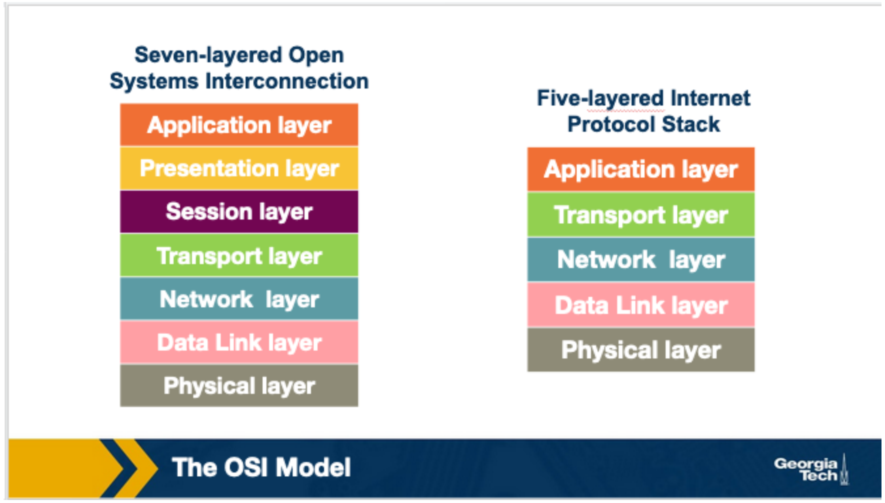


Figure 3: image

Note: The OSI reference model and the Internet architecture model

For completeness, we describe all layers found in the OSI reference model.

However, the *Internet architecture model* has five layers: the application, presentation, and session layers are combined into a single layer, and this combined layer is called the *application layer*. The interface between the application layer and the transport layer are the *sockets*. It is up to the application developer to design the functionality of the overall application.

Application, Presentation, and Session Layers

The Application Layer: The application layer includes multiple protocols, some of the most popular ones include: 1) The HTTP protocol (web), SMTP (e-mail), 2) The FTP protocol (transfers files between two end hosts), and 3) The DNS protocol (translates domain names to IP addresses). So the services that this layer offers are multiple depending on the application that is implemented. The same is true for the interface through which it is accessed, and the protocol that is implemented. At the application layer, we refer to the packet of information as a *message*.

The Presentation Layer: The presentation layer plays the intermediate role of formatting the information that it receives from the layer below and delivering it to the application layer. For example, some functionalities of this layer are formatting a video stream or translating integers from big endian to little endian format.

The Session Layer: The session layer is responsible for the mechanism that manages the different transport streams that belong to the same session between end-user application processes. For example, in the case of teleconference application, it is responsible to tie together the audio stream and the video stream.

Transport and Network Layer

The Transport Layer: The transport layer is responsible for the end-to-end communication between end hosts. In this layer, there are two transport protocols, namely *TCP* and *UDP*. The services that TCP offers include: a connection-oriented service to the applications that are running on the layer above, guaranteed delivery of the application-layer messages, flow control, which in a nutshell matches the sender's and receiver's speed, and a congestion-control mechanism, so that the sender slows its transmission rate when it perceives the network to be congested. On the other hand, the UDP protocol provides a connectionless best-effort service to the applications that are running in the layer above, without reliability, flow or congestion control. At the transport layer, we refer to the packet of information as a *segment*.

The Network Layer: In this layer, we refer to the packet of information as a *datagram*. The network layer is responsible for moving datagrams from one Internet host to another. A source Internet host sends the segment along with the destination address, from the transport layer to the network layer. The network layer is responsible for delivering the datagram to the transport layer in the destination host. The protocols in the network layer are: 1) The IP Protocol, which we often refer to as "the glue" that binds the Internet together. All Internet hosts and devices that have a network layer must run the IP protocol. The IP protocol defines a) the fields in the datagram, and b) how the source/destination hosts and the intermediate routers use these fields, so the datagrams that a source Internet host sends reach their destination. 2) The routing protocols that determine the routes that the datagrams can take between sources and destinations.

Data Link Layer and Physical Layer

The data link layer: In this layer, we refer to the packets of information as *frames*. Some example protocols in this layer include Ethernet, PPP, WiFi. The data link layer is responsible for moving the frames from one node (host or router) to the next node. More specifically, assuming we have a sender and receiver host, the network layer will route the datagram through multiple routers across the path between the sender and the receiver. At each node across this path, the network layer passes the datagram to the data link layer, which in turn delivers the datagram to the next node. Then, at that node, the link layer passes the datagram up to the network layer.

The data link layer offers services that depend on the data link layer protocol that is used over the link. Some example services include reliable delivery, which covers the transmission of the data from one transmitting node, across one link, and finally to the receiving node. We note that this specific type of reliable delivery service is different from the reliable delivery service that is offered by the TCP protocol, which offers reliability from the source host to the destination end host.

The physical layer: The physical layer facilitates the interaction with the actual hardware and is responsible to transfer bits within a frame between two nodes that are connected through a physical link. The protocols in this layer again depend on the link and on the actual transmission medium of the link. One of the main protocols in the data link layer, Ethernet, has different physical layer protocols for twisted-pair copper wire, coaxial cable, and single-mode fiber optics.

Layers Encapsulation

How do the layers and the protocols that run on each layer communicate with each other? To understand the concepts of encapsulation and de-encapsulation, let's take a look at the following diagram which shows the physical path that data take from the sending host to the receiving host.

Encapsulation and De-encapsulation. The sending host sends an application layer message M to the transport layer. The transport layer receives the message, and it appends the transport layer header information (Ht). The application message along with the transport layer header is called segment (or transport-layer segment). The segment thus encapsulates the application layer message. This added information can help the receiving host to a) inform the receiver-side transport layer about which application to deliver the message up to, and b) perform error detection and determine whether bits in the message have been changed along the route.

The segment is then forwarded to network layer which in turn, adds it's own network header information (Hn). The entire combination of the segment and the network header is called datagram. We say that the datagram encapsulates the segment. The header information that the network layer appends includes the source and

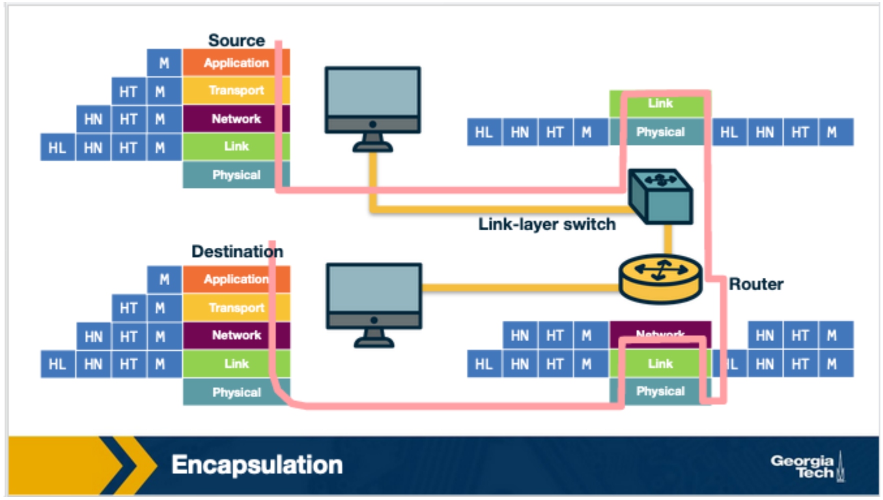


Figure 4: image

destination addresses of the end hosts. The same process continues for the link layer, which in turn it appends its own header information (Hl). The message at the link layer is called a frame, which is transmitted across the physical medium. At each layer the message is a combination of two parts: a) the payload, which is the message from the layer above, and b) the new appended header information. At the receiving end, the process is reversed, with headers being stripped off at each layer. This reverse process is known as de-encapsulation.

Intermediate devices and encapsulation. The path that connects the sending and the receiving hosts may include intermediate layer-3 devices, such as routers, and layer-2 devices such as switches. We will see later how switches and routers work, but for now we note that both routers and layer-2 switches implement protocol stacks similarly to end-hosts. The difference is that routers and layer-2 switches do not implement all the layers in the protocol stack; routers implement layers 1 to 3, and layer-2 switches implement layers 1 to 2. So, going back to our diagram, when the data leave the sending host and they are received by the layer-2 switch, the switch implements the same process of de-encapsulation to process the data and encapsulation to send the data forward to the next device.

A design choice. We note again that end-hosts implement all five layers while the intermediate devices don't. This design choice ensures that the Internet architecture puts much of its complexity and intelligence at the edges of the network while keeping the core simple. Next, we will look deeper into the so-called end-to-end principle.

The End-to-End Principle

The end-to-end (e2e) principle is a design choice that characterized and shaped significantly the current architecture of the Internet. The e2e principle suggests that specific application-level functions usually cannot, and preferably should not be built into the lower levels of the system at the core of the network.

In simple terms, the e2e principle is summarized as: the network core should be simple and minimal, while the end systems should carry the intelligence. As mentioned in the seminal paper "End-to-End Arguments in System Design" by Saltzer, Reed, and Clark: "The function in question can completely and correctly be implemented only with the knowledge and help of the application standing at the endpoints of the communications system. Therefore, providing that questioned function as a feature of the communications systems itself is not possible."

The same paper reasoned that many functions can only be completely implemented at the endpoints of the network, so any attempt to build features in the network to support specific applications must be avoided, or only viewed as a tradeoff. The reason was that not all applications need the same features and network functions to support them. Thus building such functions in the network core is rarely necessary. So, systems designers should avoid building any more than the essential and commonly shared functions into the network.

Many people argue that the e2e principle allowed the internet to grow rapidly, because evolving innovation took place at the network edge in the form of numerous applications and a plethora of services, rather than in the middle of the network, which could be hard to later modify.

What were the designers' original goals that led to the e2e principle? Moving functions and services closer to the applications that use them increases the flexibility and the autonomy of the application designer to offer these services to the needs of the specific application. Thus, the higher-level protocol layers, are more specific to an application. Whereas the lower-level protocol layers are free to organize the lower-level network resources to achieve application design goals more efficiently and independently of the specific application.

Violations of the End-to-End Principle and NAT Boxes

Despite the fact that the e2e principle offers multiple advantages to the Internet and its evolution, there have still been cases where this principle needs to be violated.

Some examples of the e2e violation: Examples include firewalls and traffic filters. The firewalls usually operated at the periphery of a network and they monitor the network traffic that is going through, to allow or drop traffic, if the traffic is flagged as malicious. Firewalls violate the e2e principle since they are intermediate devices that are operated between two end hosts and they can drop the end hosts communication.

Another example of an e2e violation is the Network Address Translation (NAT) boxes. NAT boxes help us as a bandaid measure to deal with the shortage of Internet addresses. Let's see in more detail how a NAT-enabled home router operates. Let's assume we have a home network, where we have multiple devices we want to connect to the Internet. An internet service provider typically assigns a single public IP address (120.70.39.4) to the home router and specifically to the interface that is facing the public global Internet, as shown in the figure below.

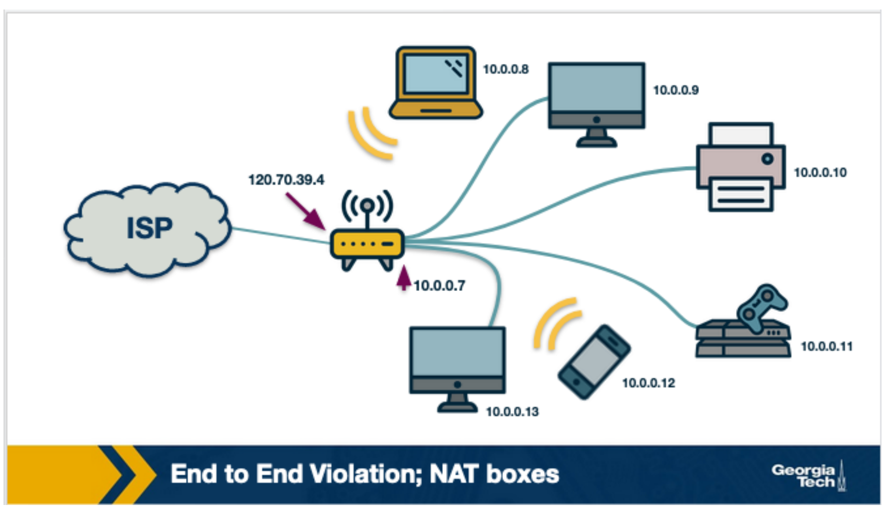


Figure 5: image

The other interface of the NAT-enabled router that is facing the home network (along with all other device-interfaces in the home network) gets an IP address that belongs to the same private subnet. This subnet must belong to the address spaces that are reserved as private, eg 10.0.0/24 or 192.168.0.0/24. This means that the IP addresses that belong to this private subnet only have meaning to devices within that subnet. So we can have hundreds of thousands of private networks with the same address range (eg 10.0.0.0/24). But, these private networks are always behind a NAT, which takes care of the communication between the hosts on the private network and the hosts on the public Internet.

All traffic that leaves the home router and it is destined to hosts in the public Internet must have as the source IP address the IP of the public facing interface of the NAT-enabled router. Similarly, all traffic that enters the home network through the router, must have as the destination address the IP of the public facing interface of the NAT-enabled router. The home router plays the role of a translator maintaining a NAT translation table, and it rewrites the source and destination IP addresses and ports.

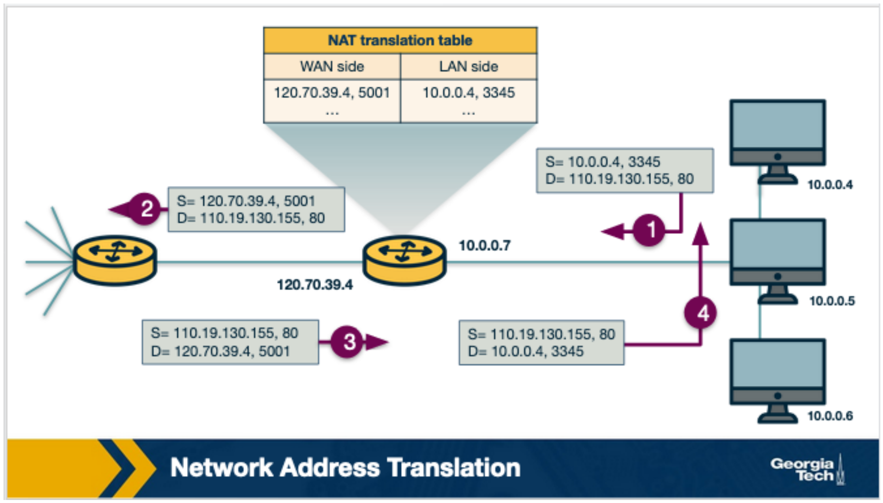


Figure 6: image

The translation table provides a mapping between the public facing IP address/ports, and the IP addresses/ports that belong to hosts inside the private network. For example, let's assume that a host 10.0.0.1 inside the private network, uses port 3345 to send traffic to a host in the public Internet with IP address 128.119.40.186 and port 80. Then the NAT table says that packets with the source IP address of 10.0.0.1 and source port 3345, they should be rewritten to a source address 138.76.29.7 and a source port of 5001 (or any source port number that is not currently used in the NAT translation table). Similarly, packets with a destination IP address of 138.76.29.7 and destination port of 5001, they will be rewritten to destination IP address 10.0.0.1 and destination port 3345.

Why the NAT boxes violate the e2e principle? The hosts that are behind NAT boxes are not globally addressable, or routable. As a result, it is not possible for other hosts on the public Internet to initiate connections to these devices. So, if we have a host behind a NAT and a host in the public Internet, then by default they cannot communicate without the intervention of a NAT box.

There are some workarounds to allow hosts to initiate connections to hosts that behind NATs. Some example tools and protocols include STUN (a tool that allows hosts to discover NATs and the public IP address and port number that the NAT has allocated for the application that the host wants to communicate with), and UDP hole punching (it established bidirectional UDP connections between hosts behind NATs).

The Hourglass Shape of Internet Architecture

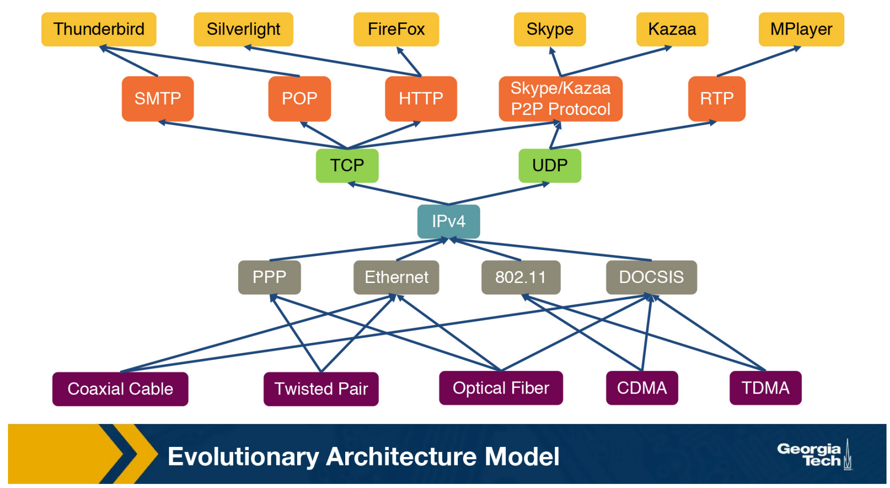
The Internet protocol stack has a layered architecture that resembles an hourglass shape. Was the Internet architecture always shaped like an hourglass, and has there always been a single protocol at the network layer? If we look back in the early nineties, we will see that there were several other network-layer protocols that were competing with IPv4. For example, Novell's IPX and the X.25 network protocol used in Frame Relay. So the network layer did not include only one protocol, but there were multiple protocols that were competing with each other at that time.

Why have there been more frequent innovations at the lower or higher layers of the protocol hourglass? Why have the protocols at the waist of the hourglass (mostly IPv4, TCP, and UDP) been difficult to replace, and have they outcompeted any protocols that offer the same or similar functionalities? Looking ahead, and assuming that we want to design and introduce new and potentially better protocols, how can we make it more likely that the new protocols will outcompete and replaces existing and widely used incumbent protocols?

Researchers have suggested a model called the Evolutionary Architecture model, or EvoArch, that can help to study layered architectures and their evolution in a quantitative manner. Through this model researchers were able to explain how the hierarchical structure of the layer architecture eventually lead to the hourglass shape.

In the next topic, we will talk about the details of the model and how it can help us explain the evolution of the Internet architecture.

Evolutionary Architecture Model



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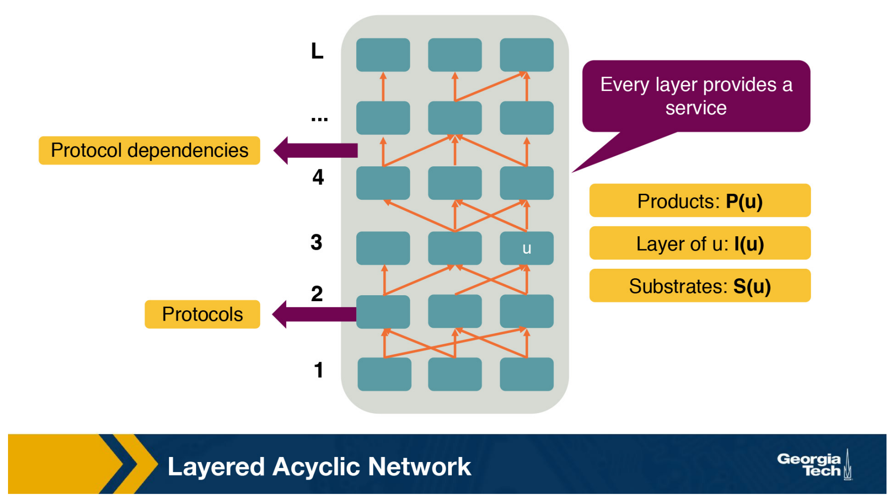


Figure 8: image

In this section, we will talk about a model that attempts to answer our previous questions. Researchers have suggested a model - the Evolutionary Architecture model or EvoArch - that can help to study layered architectures, and their evolution in a quantitative manner. The EvoArch model considers an abstract model of the Internet's protocol stack that has the following components:

- **Layers.** A protocol stack is modeled as a directed and acyclic network with L layers.
- **Nodes.** Each network protocol is represented as a node. The layer of a node u is denoted by $l(u)$.
- **Edges.** Dependencies between protocols are represented as directed edges.
- **Node incoming edges.** If a protocol u at layer l uses the service provided by a protocol w at the lower layer $l-1$, then this is represented by an "upwards" edge from w to u.
- **Node substrates.** We refer to substrates of a node u, $S(u)$, as the set of nodes that u is using their services. Every node has at least one substrate, except the nodes at the bottom layer.
- **Node outgoing edges.** The outgoing edges from a node u terminate at the products of u. The products of a node u are represented by $P(u)$.
- **Layer generality.** Each layer is associated with a probability $s(l)$, which we refer to as layer generality. A node u at layer $l+1$ selects independently each node of layer l as the substrate with probability $s(l)$. The layer generality decreases as we move to higher layers, and thus protocols at lower layers are more general in terms of their functions or provided services than protocols at higher layers. For example, in the case of the Internet protocol stack, layer 1 is very general and the protocols at this layer offer a very general bit transfer service between two connected points, which most higher layer protocols would use.
- **Node evolutionary value.** The value of a protocol node, $v(u)$, is computed recursively based on the products of u. By introducing the evolutionary value of each node, the model captures the fact that the value of a protocol u is driven by the values of the protocols that depend on it. For example, let's consider again the Internet protocol stack. TCP has a high evolutionary value because it is used by many higher layer protocols and some of them being valuable themselves. Let's assume that we introduce a brand new protocol, at the same layer as TCP, that may have better performance or other great new features. The new protocol's evolutionary value will be low if it is not used by important or popular higher layer protocols, regardless of the great new features it may have. So the evolutionary value determines if the protocol will survive the competition with other protocols, at the same layer, that offer similar services.
- **Node competitors and competition threshold.** We refer to the competitors of a node u, $C(u)$, as the nodes at layer l that share at least a fraction c of node u's products. We refer to the fraction c, as the competition threshold. So, a node w competes with a node u, if w shares at least a fraction c of u's products.
- **Node death rate.** The model has a death and birth process in place, to account for the protocols that cease or get introduced respectively. The competition among nodes becomes more intense, and it is more likely that a protocol u dies if at least one of its competitors has a higher value than itself. When a node u dies, then its products also die, if their only substrate is u.
- **Node basic birth process.** The model, in its simplest version, has a basic birth process in place, where a new node is assigned randomly to a layer. The number of new nodes at a given time is set to a small fraction (say 1% to 10%) of the total number of nodes in the network at that time. So, the larger a protocol stack is, then the faster it grows.

Toy example: To illustrate the above model and the parameters, let's consider a toy network example with L equal to 4 layers. The evolutionary value of each node is shown inside each circle. The generality probability for each layer is shown at the left of each layer, and it is denoted as $s(l)$. As we noted earlier, the generality of the layers decreases as we move to higher layers, so on average, the number of products per node decreases as well. Let's further assume that we have a competition threshold $c = \frac{3}{5}$. Nodes u, q and w compete in layer 2. U and q compete, but this is unlikely to cause q to die because u and q have comparable evolutionary values. In contrast, it is likely that w will die because its value is much less than that of its maximum-value competitor, u.

EvoArch iterations: EvoArch is a discrete-time model that is executed over rounds. At each round, we perform the following steps: A) We introduce new nodes, and we place them randomly at layers. B) We examine all layers, from the top to the bottom, and we perform the following tasks: 1) We connect the new nodes that we may have just introduced to that layer, by choosing substrates based on the generality probabilities of the layer below $s(l-1)$, and by choosing products for them based on the generality probability

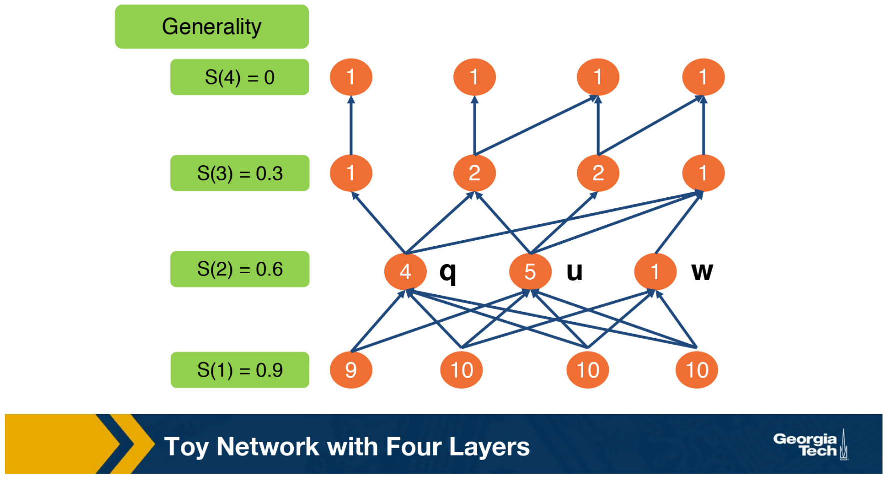


Figure 9: image

of the current layer $s(l)$. 2) We update the value of each node at each layer l, given that we may have new nodes added to the same layer l. 3) We examine all nodes, in order of decreasing value in that layer, and remove the nodes that should die. C) Finally, we stop the execution of the model when the network reaches a given number of nodes.

The figure above shows the width of each layer we execute the EvoArch model for a network of 10 layers over multiple rounds. The main takeaway message from this figure is that the layer width decreases as we move from the bottom layer to a middle layer, around layer 5, and then it increases again as we move towards the top layer.

Implications for the Internet Architecture and future Internet architecture: With the help of the EvoArch model, how can we explain the survival of the TCP/IP stack given that it appeared around the 70s or 80s when the telephone network was very powerful? The EvoArch model suggests that the TCP/IP stack was not trying to compete with the telephone network services. The TCP/IP was mostly used for applications such as FTP, E-mail, and Telnet, so it managed to grow and increase its value without competing or being threatened by the telephone network, at that time that it first appeared. Later it gained even more traction, with numerous and powerful applications relying on it.

IPv4, TCP, and UDP provide a stable framework through which there is an ever-expanding set of protocols at the lower layers (physical and data-link layers), as well as new applications and services at the higher layers. But at the same time, these same protocols have been difficult to replace or even modify significantly. EvoArch provides an explanation for this. A large birth rate at the layer above the waist can cause death for the protocols at the waist if these are not chosen as substrates by the new nodes at the higher layers. The waist of the Internet architecture is narrow, but also the next higher layer (the transport layer) is also very narrow and stable. So, the transport layer acts as an "evolutionary shield" for IPv4, because any new protocols that might appear at the transport layer are unlikely to survive the competition with TCP and UDP which already have multiple products. In other words, the stability of the two transport protocols adds to the stability of IPv4, by eliminating any potential new transport protocols, that could select a new network layer protocol instead of IPv4.

Finally, in terms of future and entirely new Internet architectures, the EvoArch model predicts that even if these brand new architectures do not have the shape of an hourglass initially, they will probably do so as they evolve, which will lead to new ossified protocols. The model suggests that one way to proactively avoid these ossification effects, that we now experience with TCP/IP, a network architect should try to design the functionality of each layer so that the waist is wider, consisting of several protocols that offer largely non-overlapping but general services, so that they do not compete with each other.

Interconnecting Hosts and Networks

We have different types of devices that help to provide connectivity between hosts that are in the same network, or help interconnect networks. These devices offer different services and they operate over different layers.

Repeaters and Hubs: They operate on the physical layer (L1), as they receive and forward digital signals to connect different Ethernet segments. They provide connectivity between hosts that are directly connected (in the same network). The advantage is that they are simple and inexpensive devices, and they can be arranged in a hierarchy. Unfortunately, hosts that are connected through these devices belong to the same collision domain, meaning that they compete for access to the same link.

Bridges and Layer2-Switches: These devices can enable communication between hosts that are not directly connected. They operate on the data link layer (L2) based on MAC addresses. They receive packets and they forward them to reach the appropriate destination. A limitation is the finite bandwidth of the outputs. If the arrival rate of the traffic is higher than the capacity of the outputs then packets are temporarily stored in buffers. But if the buffer space gets full, then this can lead to packet drops.

Routers and Layer3-Switches: These are devices that operate on Layer 3. We will talk more about these devices and the routing protocols on the following lectures.

Learning Bridges

A bridge is a device with multiple inputs/outputs. A bridge transfers frames from an input to one (or multiple) outputs. Though it doesn't need to forward all the frames it receives. In this topic we will talk about how a bridge learns how to perform that task. A learning bridge learns, populates and maintains, a forwarding table. The bridge consults that table so that it only forwards frames on specific ports, rather than over all ports. For example, let's consider the topology on the following figure. When the bridge receives a frame on port 1, with source Host A and destination Host B, the bridge does not have to forward it to port 2.

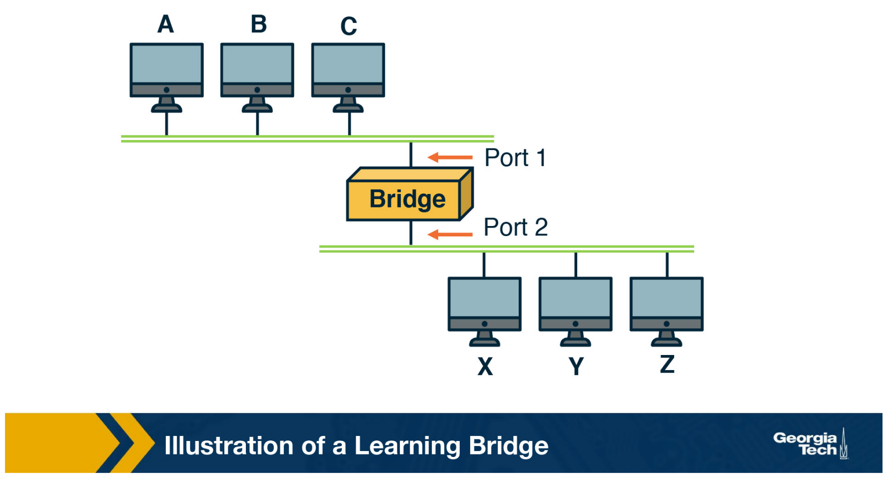
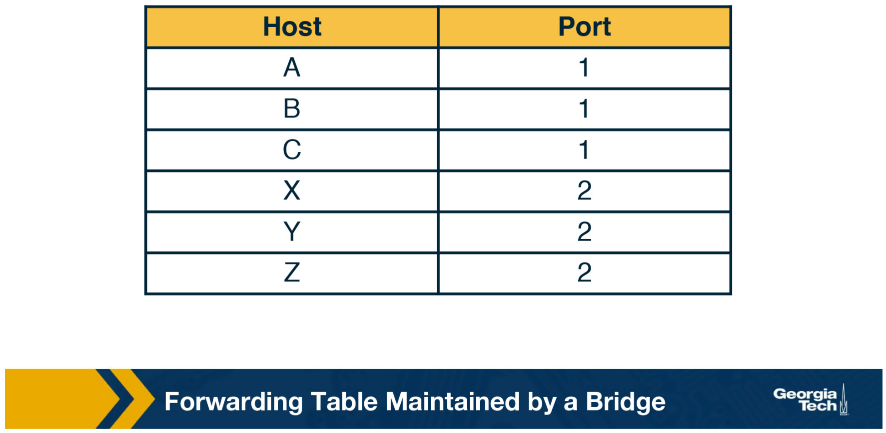


Figure 10: image

So how does the bridge learn? When the bridge receives any frame this is a "learning opportunity" to know which hosts are reachable through which ports. This is because the bridge can view the port over which a frame arrives and the source host. Going back to our example topology, eventually the bridge builds the following forwarding table.



Host	Port
A	1
B	1
C	1
X	2
Y	2
Z	2

Figure 11: image

Looping Problem in Bridges and the Spanning Tree Algorithm

Unfortunately using bridges to connect LANs fails if the network topology results in loops (cycles). In that case, the bridges loop through packets forever! The answer to this problem is excluding links that lead to loops by running the spanning tree algorithm. Let's represent the topology of the network as a graph. The bridges are represented as nodes and the links between the bridges are represented as edges. The goal of the spanning tree algorithm is to have the bridges select which links (ports) to use for forwarding eliminating loops.

Let's take a look at how bridges run this distributed algorithm. Every node (bridge) in the graph has an ID. The bridges eventually select one bridge as the root of the topology. Let's see how this selection happens.

The algorithm runs in "rounds" and at every round each node sends to each neighbor node a configuration message with three fields: a) the sending node's ID, b) the ID of the roots as perceived by the sending node, and c) the number of hops between that (perceived) root and the sending node. At every round, each node keeps track of the best configuration message that it has received so far, and it compares that against the configuration messages it receives from neighboring nodes at that round. At the very first round of the algorithm, every node thinks that it is the root. So for a node with an ID 3 for example, the node sends a configuration message <3, 3, 0> to its neighbors. Note that the distance of the node from itself (perceived root) is 0.

So how does a node compare two configuration messages? Between two configurations, a node selects one configuration as better if: a) The root of the configuration has a smaller ID, or if b) The roots have equal IDs, but one configuration indicates smaller distance from the root, or if c) Both roots IDs are the same and the distances are the same, then the node breaks the tie by selecting the configuration of the sending node that has with the smallest ID. In addition, a node stops sending configuration messages over a link (port), when the node receives a configuration message that indicates that it is not the root, eg when it receives a configuration message from a neighbor that: a) either closer to the root, or b) it has the same distance from the root, but it has a smaller ID.

As an example, let's consider the topology below. By running the above steps on this topology, we note that in the first round B3 receives (B2, B2, 0) and (B5, B5, 0), so it accepts B2 as the root. So in the second round it sends (B3, B2, 1) to its neighbors.

Similarly for B2; In the first round, B2 receives (B3,B3,0) and (B1,B1,0), it accepts B1 as the root. So in the second round B2 sends (B2,B1,1).

Finally, B5 receives configuration messages from B3, B7 and B1. B5 accepts B1 as root and sends (B5, B1, 1) to B3. This results to B3 also accepting B1 as root. In addition, B3 realizes that both its neighbors, namely B2 and B5 are closer to the root (B1) than itself. This causes B3 to not select any of its links (ports). So B3 stops participating in forwarding traffic.

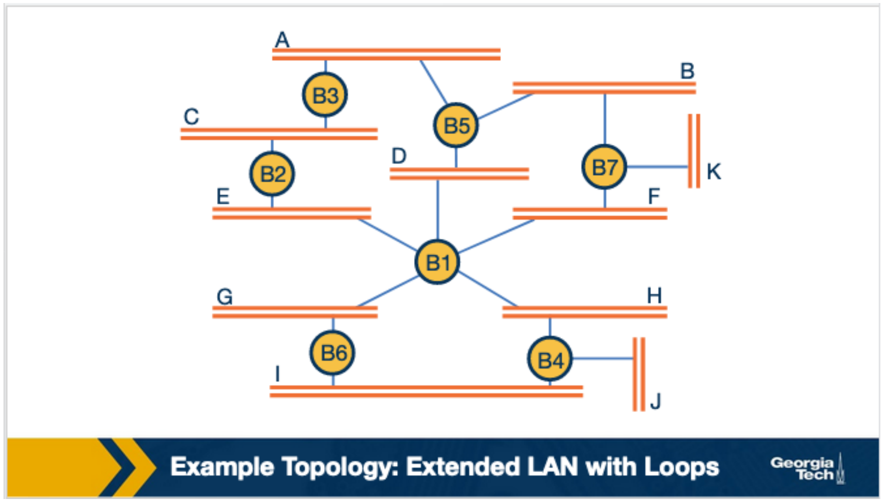


Figure 12: image

The resulting spanning tree is:

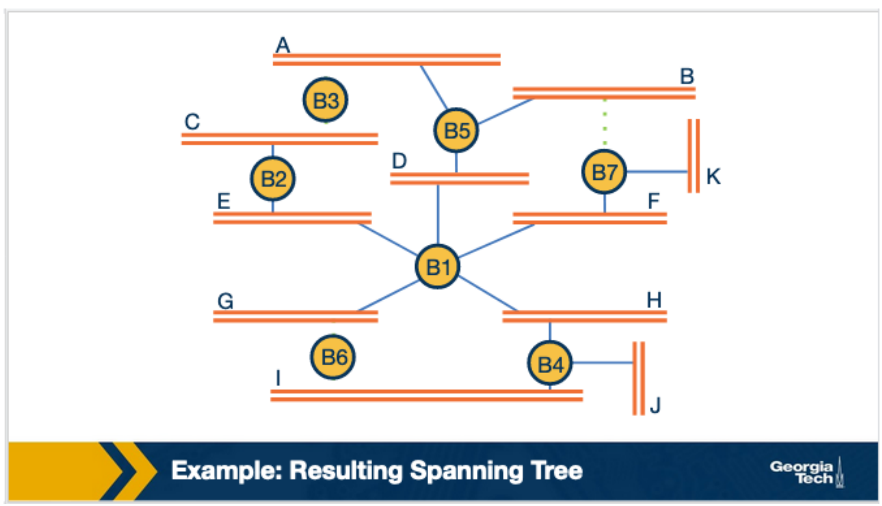


Figure 13: image