6. NETWORKING

Before purchasing equipment or deciding on a hardware platform, you should have a clear idea of the nature of your communications problem. Most likely, you are reading this book because you need to connect computer networks together in order to share resources and ultimately reach the larger global Internet.
The network design you choose to implement should fit the communications problem you are trying to solve.

Do you need to connect a remote site to an Internet connection in the centre of your campus? Will your network likely grow to include several remote sites? Will most of your network components be installed in fixed locations, or will your network expand to include hundreds of roaming laptops and other devices?

In this chapter, we will review the networking concepts that define TCP/IP, the primary family of networking protocols currently used on the Internet.
We will also look at the hardware options that are likely to form the underlying physical layer of your TCP/IP network and end with some examples of wireless configurations. This will prepare you very well for the chapter called **Deployment Planning** later in this book.

TCP/IP refers to the suite of protocols that allow conversations to happen on the global Internet.
By understanding TCP/IP, you can build networks that will scale to virtually any size, and will ultimately become part of the global Internet.

This edition of the book now includes an introduction to IPv6 which is the new numbering system of the Internet.
As it is very likely you will be deploying networks using IPv6, it is highly recommended you become familiar with how this works and also how it can work alongside the older IPv4 networks that will continue to operate on the Internet for some while yet.

1. Introduction

Venice, Italy is a fantastic city to get lost in. The roads are mere foot paths that cross water in hundreds of places, and never go in a simple straight line. Postal carriers in Venice are some of the most highly trained in the world, specialising in delivery to only one or two of the six sestieri (districts) of Venice. This is necessary due to the intricate layout of that ancient city. Many people find that knowing the location of the water and the sun is far more useful than trying to find a street name on a map.

Imagine a tourist who happens to find papier-mâché mask as a souvenir, and wants to have it shipped from the studio in S. Polo, Venezia to their home in London, United Kingdom. This may sound like an ordinary (or even trivial) task, but let's look at what actually happens.

Figure NG 1: Another kind of network mask.

The artist first packs the mask into a shipping box and addresses it to the home of the tourist.

They then hand this to a postal employee in Venice, who attaches some official forms and sends it to a central package processing hub for international destinations.

After several days, the package clears Italian customs and finds its way onto a flight to the UK, arriving at a central import processing depot at Heathrow airport. Once it clears through customs, the package is sent to a distribution point in the city of London, then on to the local district postal processing centre of Camden where the tourist lives.
The package eventually makes its way onto a delivery van which has a route that brings it to the correct house on the correct street in Camden. A member of the family accepts and signs for the package from the delivery van driver and then leaves it in the home studio of the tourist who enjoys unpacking it some time later.

The sorting clerk at the office in Camden neither knows nor cares about how to get to the sestiere of S. Polo, Venezia.
His job is simply to accept packages as they arrive, and deliver them to the correct person in Camden.
Similarly, the postal employee in Venice has no need to worry about how to get to the correct address in London. His job is to accept packages from his local neighborhood and forward them to the next closest hub in the delivery chain.

This is very similar to how Internet routing works. A message is split up into many individual packets, and are labelled with their source and destination.
The computer then sends these packets to a router, which decides where to send them next.

The router needs only to keep track of a handful of routes (for example, how to get to the local network, the best route to a few other local networks, and one route to a gateway to the rest of the Internet). This list of possible routes is called the routing table.
As packets arrive at the router, the destination address is examined and compared against its internal routing table.

If the router has no explicit route to the destination in question, it sends the packet to the closest match it can find, which is often its own Internet gateway (via the default route).
And the next router does the same, and so forth, until the packet eventually arrives at its destination.
Packages can only make their way through the international postal system because we have established a standardised addressing scheme for packages.
For example, the destination address must be written legibly on the front of the package, and include all critical information (such as the recipient's name, street address, city, country, and postal code). Without this information, packages are either returned to the sender or are lost in the system. Packets can only flow through the global Internet because we have agreed on a common addressing scheme and protocol for forwarding packets.
These standard communication protocols make it possible to exchange information on a global scale.

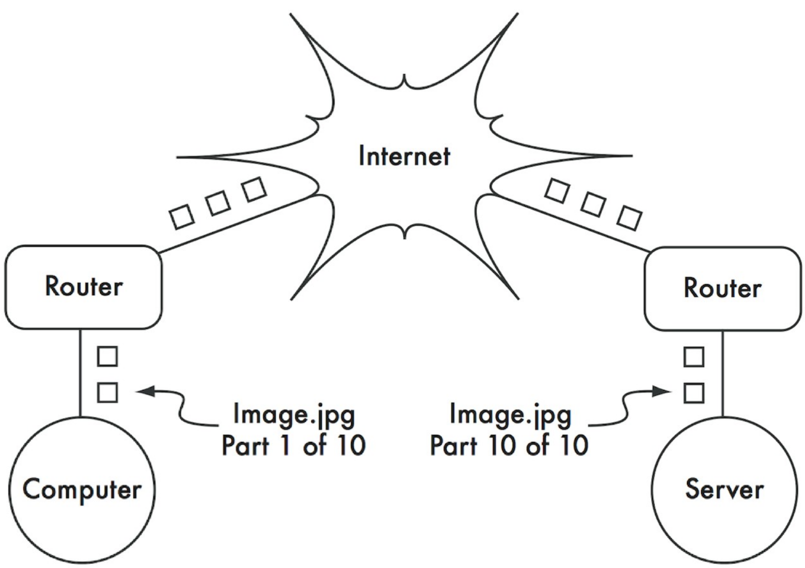


Figure NG 2: Internet networking. Packets are forwarded between routers until they reach their ultimate destination.

2. Cooperative communications

Communication is only possible when the participants speak a common language. But once the communication becomes more complex than a simple conversation between two people, protocol becomes just as important as language.

All of the people in an auditorium may speak English, but without a set of rules in place to establish who has the right to use the microphone, the communication of an individual's ideas to the entire room is nearly impossible. Now imagine an auditorium as big as the world, full of all of the computers that exist.

Without a common set of communication protocols to regulate when and how each computer can speak, the Internet would be a chaotic mess where every machine tries to speak at once. People have developed a number of communications frameworks to address this problem. The most well-known of these is the OSI model.

3. The OSI model

The international standard for Open Systems Interconnection (OSI) is defined by the document ISO/IEC 7498-1, as outlined by the International Standards Organization and the International Electrotechnical Commission. The full standard is available as publication "ISO/IEC 7498-1:1994," available from http://standards.iso.org/ittf/PubliclyAvailableStandards/.

The OSI model divides network traffic into a number of layers. Each layer is independent of the layers around it, and each builds on the services provided by the layer below while providing new services to the layer above. The abstraction between layers makes it easy to design elaborate and highly reliable protocol stacks, such as the ubiquitous TCP/IP stack. A protocol stack is an actual implementation of a layered communications framework. The OSI model doesn't define the protocols to be used in a particular network, but simply delegates each communications "job" to a single layer within a well-defined hierarchy.

While the ISO/IEC 7498-1 specification details how layers should interact with each other, it leaves the actual implementation details up to the manufacturer. Each layer can be implemented in hardware (more common for lower layers) or software.

As long as the interface between layers adheres to the standard, implementers are free to use whatever means are available to build their protocol stack.

This means that any given layer from manufacturer A can operate with the same layer from manufacturer B (assuming the relevant specifications are implemented and interpreted correctly).
Here is a brief outline of the seven-layer OSI networking model:

Layer	Name	Description
7	Application	The Application Layer is the layer that most network users are exposed to; it is the level at which human communication happens. HTTP, FTP, and SMTP are all application layer protocols. The human sits above this layer, interacting with the application.
6	Presentation	The Presentation Layer deals with data representation, before it reaches the application. This would include HTML, MIME encoding, data compression, formatting checks, byte ordering, etc.
5	Session	The Session Layer manages the logical communications session between applications. RPC is an example of a layer five protocol.
4	Transport	The Transport Layer provides a method of reaching a particular service on a given network node. Examples of protocols that operate at this layer are TCP, UDP and SCTP. Some protocols at the transport layer (such as TCP) ensure that all of the data has arrived at the destination, and is reassembled and delivered to the next layer in the proper order. UDP is a "connectionless" protocol commonly used for video and audio streaming and doesnt check arrival of data packets.
3	Network	IP (the Internet Protocol) is the most common Network Layer protocol. This is the layer where routing occurs. Packets can leave the link local network and be retransmitted on other networks. Routers perform this function on a network by having at least two network interfaces, one on each of the networks to be interconnected. Nodes on the Internet are reached by their globally unique IP address. Another critical Network Layer protocol is ICMP, which is a special protocol which provides various management messages needed for correct operation of IP. This layer is also sometimes referred to as the Internet Layer.
2	Data Link	Whenever two or more nodes share the same physical medium (for example, several computers plugged into a hub, or a room full of wireless devices all using the same radio channel) they use the Data Link Layer to communicate. Common examples of data link protocols are Ethernet, Token Ring, ATM, and the wireless networking protocols (IEEE 802.11A/B/G). Communication on this layer is said to be link-local, since all nodes connected at this layer communicate with each other directly. This layer is sometimes known as the Media Access Control (MAC) layer. On Ethernet networks, nodes are referred to by their MAC address. This is a unique 48-bit number assigned to every networking device when it is manufactured.
1	Physical	The Physical Layer is the lowest layer in the OSI model, and refers to the actual physical medium over which communicationstake place. This can be a copper CAT5 cable, a fibre optic bundle, radio waves, or just about any other medium capable of transmitting signals. Cut wires, broken fibre, and RF interference are all physical layer problems.

The layers in this model are numbered one through seven, with seven at the top. This is meant to reinforce the idea that each layer builds upon, and depends upon, the layers below. Imagine the OSI model as a building, with the foundation at layer one, the next layers as successive floors, and the roof at layer seven. If you remove any single layer, the building will not stand. Similarly, if the fourth floor is on fire, then nobody can pass through it in either direction.

The first three layers (Physical, Data Link, and Network) all happen "on the network." That is, activity at these layers is determined by the configuration of cables, switches, routers, and similar devices. A network switch can only distribute packets by using MAC addresses, so it need only implement layers one and two. A simple router can route packets using only their IP addresses, so it needs to implement only layers one through three. A web server or a laptop computer runs applications, so it must implement all seven layers. Some advanced routers may implement layer four and above, to allow them to make decisions based on the higher-level information content in a packet, such as the name of a website, or the attachments of an email.

The OSI model is internationally recognised, and is widely regarded as the complete and definitive network model. It provides a framework for manufacturers and network protocol implementers that can be used to build networking devices which interoperate in just about any part of the world. From the perspective of a network engineer or troubleshooter, the OSI model can seem needlessly complex. In particular, people who build and troubleshoot TCP/IP networks rarely need to deal with problems at the Session or Presentation layers. For the majority of Internet network implementations, the OSI model can be simplified into a smaller collection of five layers.

4. The TCP/IP model

Unlike the OSI model, the TCP/IP model is not an international standard and its definitions vary. Nevertheless, it is often used as a pragmatic model for understanding and troubleshooting Internet networks.
The vast majority of the Internet uses TCP/IP, and so we can make some assumptions about networks that make them easier to understand.

The TCP/IP model of networking describes the following five layers:

Layer	Name
5	Application
4	Transport
3	Internet
2	Data Link
1	Physical

In terms of the OSI model, layers five through seven are rolled into the topmost layer (the Application layer). The first four layers in both models are identical.
Many network engineers think of everything above layer four as "just data" that varies from application to application.
Since the first three layers are interoperable between virtually all manufacturers' equipment, and layer four works between all hosts using TCP/IP, and everything above layer four tends to apply to specific applications, this simplified model works well when building and troubleshooting TCP/IP networks.

We will use the TCP/IP model when discussing networks in this book.

The TCP/IP model can be compared to a person delivering a letter to a city office building.
The person first needs to interact with the road itself (the Physical layer), pay attention to other traffic on the road (the Data Link layer), turn at the correct junction to join another road and arrive at the correct address (the Internet layer), go to the correct floor and room number (the Transport layer), and finally give it to a receptionist who can take the letter from there (the Application layer).

Once they have delivered the message to the receptionist, the delivery person is free to go on their way. The five layers can be easily remembered by using the mnemonic “Please Don’t Look In The Attic,” which of course stands for “Physical / Data Link / Internet / Transport / Application.”

5. The Internet Protocols

TCP/IP is the protocol stack most commonly used on the global Internet. The acronym stands for Transmission Control Protocol (TCP) and Internet Protocol (IP), but actually refers to a whole family of related communications protocols. TCP/IP is also called the Internet protocol suite, and it operates at layers three and four of the TCP/IP model.
In this discussion, we will focus on version six of the IP protocol (IPv6) as since 2012 this is the version to deploy in parallel with the previous version four (IPv4). In 2012, about half of the Internet content is available with a better user experience by using IPv6.
The previous version is explained in this chapter too because some old content or old applications (Skype in 2012) still require IPv4. And indeed many networks that you might have to interconnect to will still have the legacy IPv4 technology deployed for some years to come.

Besides the length of the address, IPv4 and IPv6 are quite similar: they are connectionless network protocols running on the same data-link layer (WiFi, Ethernet...) and serving the same transport protocols (TCP, SCTP, UDP...) In this book, when IP is written without any version, then it means that it applies to both versions. A dual-stack network is a network that runs IPv6 and IPv4 and the same time. It is expected that dual-networks will be the norm at least until 2020 when IPv6-only will become the norm.

IPv6 Addressing

The IPv6 address is a 128-bit number usually written as multiple hexadecimal numbers. In order to make this address human-readable, it is written in chunks of 32 bits or 4 hexadecimal numbers separated by a colon ':'. The hexadecimal number should be written in lowercase but can also be written in uppercase. An example of an IPv6 address is:

2001:0db8:1234:babe:0000:0000:0000:0001

This address corresponds to:

2001	0db8	1234	babe	0	0	0	1

As these addresses are quite long, it is common to remove the leading 0 in each chunk, so, the same address can also be written as:

2001:db8:1234:babe:0:0:0:1

This address can further be simplified by grouping one block of consecutive chunks of '0' into the abbreviated form of '::', the same address becomes then:

2001:db8:1234:babe::1

There are some specific IPv6 addresses:

- ::1 (or 0000:00000:0000:0000:0000:0000:0000:0001) represents the loopback address, this is the node itself when the node wants to send packets to itself;
- :: (all zero) is the undetermined address, to be used by a node when it does not know its global address, for instance when it boots.

IPv6 Prefixes

IPv6 nodes on the same link or network share the same IPv6 prefix, which is defined as the most-significant part of the IPv6 address. The prefix length is usually 64 bit on a LAN. So, our usual address of 2001:db8:babe::1 can be written as 2001:db8:1234:babe::1/64 (the prefix length is added at the end of the address after a '/'). Defining a prefix length on an address actually splits the address in two parts: the prefix itself and the interface identifier (IID).

2001	0DB8	1234	babe	0	0	0	0
Prefix				Interface Identifier			

On a LAN or WLAN, the prefix length must be 64 bits else some protocols will not work correctly. All nodes on the same LAN or WLAN usually share the same prefix but their IID must be unique to avoid confusion.
The analogy with a postal address in big cities is that the street name is the

prefix and the house number is the IID.
The prefix length can be different on links that are neither LAN or WLAN. The network itself is identified by the prefix without any IID but with the prefix length, for example: 2001:db8:1234:babe::/64

IPv4 Addressing

In an IPv4 network, the address is a 32-bit number, normally written as four 8-bit numbers expressed in decimal form and separated by periods. Examples of IPv4 addresses are 10.0.17.1, 192.168.1.1, or 172.16.5.23.

If you enumerated every possible IPv4 address, they would range from 0.0.0.0 to 255.255.255.255.
This yields a total of more than four billion possible IPv4 addresses (255 x 255 x 255 x 255 = 4,228,250,625); although many of these are reserved for special purposes and should not be assigned to hosts.

Some IPv4 addresses that are special:

- 127.0.0.1 represents the loopback address (similar to ::1 for IPv6);
- 0.0.0.0 represents the unspecified address (similar to :: for IPv6).

IPv4 Subnets

By applying a subnet mask (also called a network mask, or simply netmask or even prefix) to an IPv4 address, you can logically define both a host and the network to which it belongs.

Traditionally, subnet masks are expressed using dotted decimal form, much like an IPv4 address. For example, 255.255.255.0 is one common netmask. You will find this notation used when configuring network interfaces, creating routes, etc. However, subnet masks are more succinctly expressed using CIDR notation, which simply enumerates the number of bits in the mask after a forward slash (/).
Thus, 255.255.255.0 can be simplified as /24. CIDR is short for Classless Inter-Domain Routing, and is defined in RFC1518. A subnet mask determines the size of a given network. Using a /24 netmask, 8 bits are reserved for hosts (32 bits total - 24 bits of netmask = 8 bits for hosts). This yields up to 256 possible host addresses (28 = 256). By convention, the first value is taken as the network address (.0 or 00000000), and the

last value is taken as the broadcast address (.255 or 11111111).
This leaves 254 addresses available for hosts on this network.
Subnet masks work by applying AND logic to the 32 bit IPv4 number.
In binary notation, the "1" bits in the mask indicate the network address portion, and "0" bits indicate the host address portion. A logical AND is performed by comparing two bits. The result is "1" if both of the bits being compared are also "1". Otherwise the result is "0".
Here are all of the possible outcomes of a binary AND comparison between two bits.

Bit 1	Bit 2	Result
0	0	0
0	1	0
1	0	0
1	1	1

To understand how a netmask is applied to an IPv4 address, first convert everything to binary. The netmask 255.255.255.0 in binary contains twenty-four "1" bits:

255 255 255 0

11111111.11111111.11111111.00000000

When this netmask is combined with the IPv4 address 10.10.10.10, we can apply a logical AND to each of the bits to determine the network address.

10.10.10.10: 00001010.00001010.00001010.00001010

255.255.255.0: 11111111.11111111.11111111.00000000

10.10.10.0: 00001010.00001010.00001010.00000000

This results in the network 10.10.10.0/24.

This network consists of the hosts 10.10.10.1 through 10.10.10.254, with 10.10.10.0 as the network address and 10.10.10.255 as the broadcast address.
Subnet masks are not limited to entire octets. One can also specify subnet masks like 255.254.0.0 (or /15 CIDR). This is a large block, containing

131,072 addresses, from 10.0.0.0 to 10.1.255.255.
It could be further subdivided, for example into 512 subnets of 256 addresses each. The first one would be 10.0.0.0-10.0.0.255, then 10.0.1.0-10.0.1.255, and so on up to 10.1.255.0-10.1.255.255. Alternatively, it could be subdivided into 2 blocks of 65,536 addresses, or 8192 blocks of 16 addresses, or in many other ways. It could even be subdivided into a mixture of different block sizes, as long as none of them overlap, and each is a valid subnet whose size is a power of two.
While many netmasks are possible, common netmasks include:

CIDR	Decimal	# of Hosts
/30	255.255.255.252	4
/29	255.255.255.248	8
/28	255.255.255.240	16
/27	255.255.255.224	32
/26	255.255.255.192	64
/25	255.255.255.128	128
/24	255.255.255.0	256
/16	255.255.0.0	65 536
/8	255.0.0.0	16 777 216

With each reduction in the CIDR value the IPv4 space is doubled. Remember that two IPv4 addresses within each network are always reserved for the network and broadcast addresses.

There are three common netmasks that have special names. A /8 network (with a netmask of 255.0.0.0) defines a Class A network. A /16 (255.255.0.0) is a Class B, and a /24 (255.255.255.0) is called a Class C. These names were around long before CIDR notation, but are still often used for historical reasons.
In many ways as you can already see IPv6 is easier to plan for than IPv4.

Global IP Addresses

Interconnected networks must agree on an IP addressing plan for IPv6 and IPv4 addresses.

IP addresses must be unique and generally cannot be used in different places on the Internet at the same time; otherwise, routers would not know how best to route packets to them.

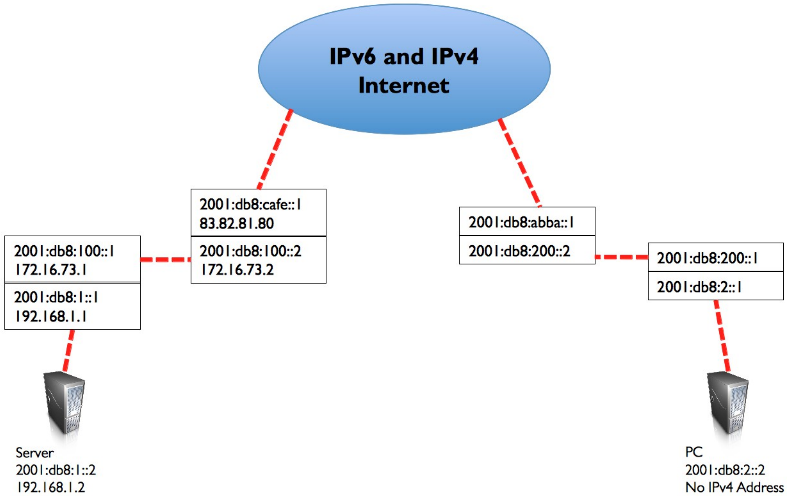


Figure NG 3: With unique IP addresses, ambiguous global routing is impossible. If the PC requests a web page from 2001:db8:1::2, it will reach the correct server.

In order to keep IP addresses unique and globally routable, they are allocated by a central numbering authority that provides a consistent and coherent numbering method. This ensures that duplicate addresses are not used by different networks.
The authority assigns large blocks of consecutive addresses to smaller authorities, who in turn assign smaller consecutive blocks within these ranges to other authorities, or to their customers. The groups of addresses are called subnets or prefixes as we have already mentioned.
A group of related addresses is referred to as an address space.
Both IPv4 and IPv6 addresses are administered by the Internet Assigned Numbers Authority (IANA, http://www.iana.org/).

IANA has divided these address spaces into large subnets, and these subnets are delegated to one of the five regional Internet registries (RIRs),

who have been given authority over large geographic areas.
IP addresses are assigned and distributed by Regional Internet Registrars (RIRs) to ISPs. The ISP then allocates smaller IP blocks to their clients as required. Virtually all Internet users obtain their IP addresses from an ISP.

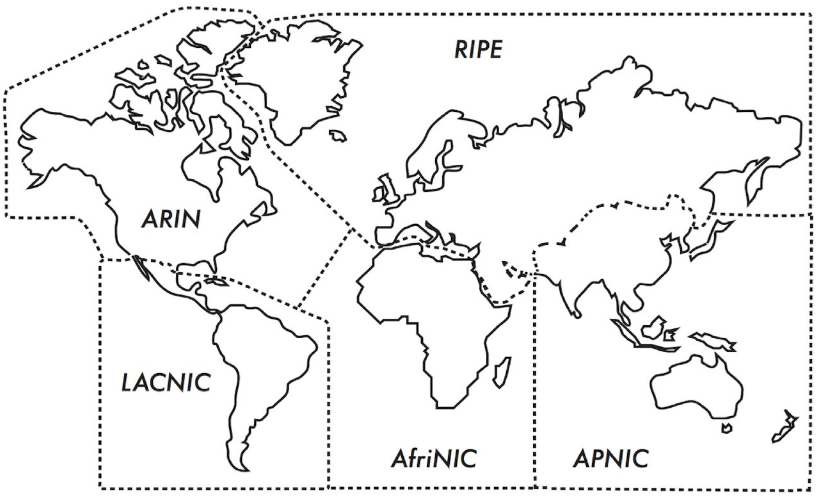


Figure NG 4: Authority for Internet IP address assignments is delegated to the five Regional Internet Registrars.

The five RIRs are:

1) African Network Information Centre
(AfriNIC, http://www.afrinic.net)
2) Asia Pacific Network Information Centre
(APNIC, http://www.apnic.net)
3) American Registry for Internet Numbers
(ARIN, http://www.arin.net)
4) Regional Latin-American and Caribbean IP Address Registry
(LACNIC, http://www.lacnic.net)
5) Réseaux IP Européens
(RIPE NCC, http://www.ripe.net)

Your ISP will assign globally routable IP address space to you from the pool allocated to it by your RIR.

The registry system assures that IP addresses are not reused in any part of the network anywhere in the world.
Once IP address assignments have been agreed upon, it is possible to pass packets between networks and participate in the global Internet.
The process of moving packets between networks is called routing.

Static IP Addresses

A static IP address is an address assignment that never changes.
Static IP addresses are important because servers using these addresses may have DNS mappings pointing towards them, and typically serve information to other machines (such as email services, web servers, etc.).
Blocks of static IP addresses may be assigned by your ISP, either by request or automatically depending on your means of connection to the Internet.

Dynamic IP Addresses

Dynamic IP addresses are assigned by an ISP for non-permanent nodes connecting to the Internet, such as a home computer which is on a dial-up connection or a laptop connecting to a wireless hotspot.
Dynamic IP addresses can be assigned automatically using the Dynamic Host Configuration Protocol (DHCP), or the Point-to-Point Protocol (PPP), depending on the type of Internet connection.

A node using DHCP first requests an IP address assignment from the network, and automatically configures its network interface. IP addresses can be assigned randomly from a pool by your ISP, or might be assigned according to a policy. IP addresses assigned by DHCP are valid for a specified time (called the lease time).

The node must renew the DHCP lease before the lease time expires. Upon renewal, the node may receive the same IP address or a different one from the pool of available addresses.

While DHCP works for IPv6 and IPv4, IPv6 has another primary mechanism which is more commonly used for address assignment - it is called Stateless Address Auto-Configuration (SLAAC) which is the default on routers and hosts running IPv6.

It does not require a DHCP server; the router sends periodically Router Advertisement (RA) messages on all connected (W)LAN's which contain the 64-bit prefix to be used on that (W)LAN; hosts then generate their 64-bit interface identifier (usually a random number or a number based on their MAC address – see further) and build their 128-bit address by concatenating the 64-bit prefix from the RA and the newly created 64-bit IID.

Dynamic addresses are popular with Internet Service Providers, because it enables them to use fewer IP addresses than their total number of customers.
They only need an address for each customer who is active at any one time.
Globally routable IP addresses cost money, and there is now a shortage of IPv4 addresses.
Assigning addresses dynamically allows ISPs to save money, and they will often charge extra to provide a static IP address to their customers.

Private IPv4 addresses

Around 2000, it became clear that there would not be enough IPv4 addresses for everyone; this is the reason that IPv6 was specified and developed.
But there was also a temporary trick as most private networks do not require the allocation of globally routable, public IPv4 addresses for every computer in the organisation.
In particular, computers which are not public servers do not need to be addressable from the public Internet.
Organisations typically use IPv4 addresses from the private address space for machines on the internal network.

There are currently three blocks of private address space reserved by IANA: 10.0.0.0/8, 172.16.0.0/12, and 192.168.0.0/16.
These are defined in RFC1918. These addresses are not intended to be routed on the Internet, and are typically unique only within an organisation or group of organisations that choose to follow the same numbering scheme.
This means that several distinct organisations can use the same addresses as long as they never interconnect their networks directly.

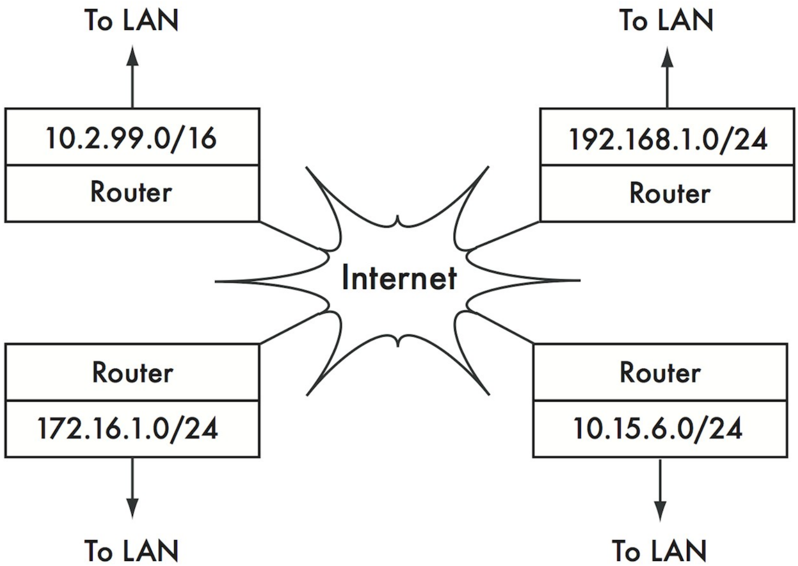


Figure NG 5: RFC1918 private addresses may be used within an organisation, and are not routed on the global Internet.

If you ever intend to link together private networks that use RFC1918 address space, be sure to use unique addresses throughout all of the networks.

For example, you might break the 10.0.0.0/8 address space into multiple Class B networks (10.1.0.0/16, 10.2.0.0/16, etc.).
One block could be assigned to each network according to its physical location (the campus main branch, field office one, field office two, dormitories, and so forth).

The network administrators at each location can then break the network down further into multiple Class C networks (10.1.1.0/24, 10.1.2.0/24, etc.) or into blocks of any other logical size.

In the future, should the networks ever be linked (either by a physical connection, wireless link, or VPN), then all of the machines will be reachable from any point in the network without having to renumber network devices.

Some Internet providers may allocate private addresses like these instead of public addresses to their customers, although this has serious disadvantages.

Since these addresses cannot be routed over the Internet, computers which use them are not really "part" of the Internet, and are not directly reachable from it. In order to allow them to communicate with the Internet, their private addresses must be translated to public addresses.

This translation process is known as Network Address Translation (NAT), and is normally performed at the gateway between the private network and the Internet.

We will look at NAT in more detail later on in this chapter.
As there are huge numbers of IPv6 addresses, there is no need for private IPv6 addresses, although there are Unique Local Addresses (ULA) that are suitable for non connected networks such as labs.

Discovering Neighbours

Imagine a network with three hosts: H_A, H_B, and H_C. They use the corresponding IP addresses A, B and C.
These hosts are part of the same subnet/prefix.
For two hosts to communicate on a local network, they must determine each others' MAC addresses. It is possible to manually configure each host with a mapping table from IP address to MAC address, but it is easier to dynamically discover the neighbour's MAC address through Neighbor Discovery Protocol (NDP) in IPv6 and Address Resolution Protocol (ARP) in IPv4. NDP and ARP work in a very similar way.

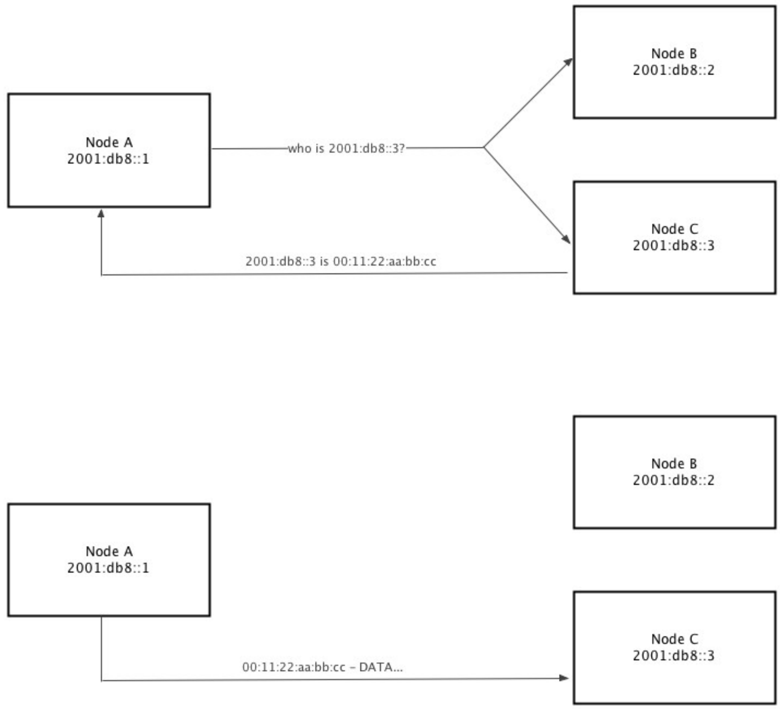


Figure NG 6: IPv6 node A, 2001:db8::1 needs to send data to 2001:db8::3 in the same network (2001:db8::/64 prefix). But it must first ask for the MAC address that corresponds to 2001:db8::3.

When using NDP, node A multicasts to some hosts the question, "Who has the MAC address for the IPv6 2001:db8::3?"

When node C sees a Neighbor Solicitation (NS) for an IPv6 address of its own, it replies with its MAC address with a Neighbor Advertisement (NA) message.

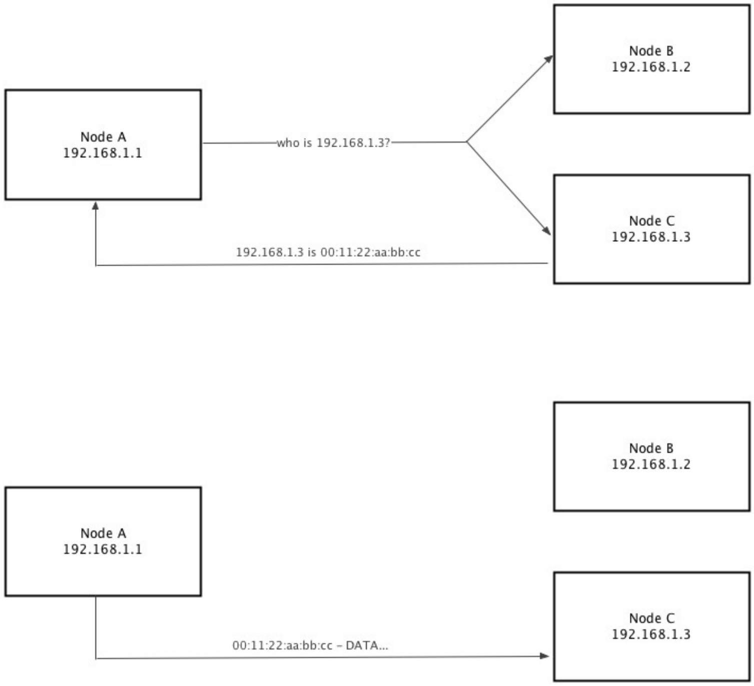


Figure NG 7: IPv4 node A, 192.168.1.1, needs to send data to 192.168.1.3 in the same subnet (192.168.1.0/24). But it must first ask the whole network for the MAC address that corresponds to 192.168.1.3.

When using ARP, node A broadcasts to all hosts the question,
"Who has the MAC address for the IPv4 192.168.1.3?"

When node C sees an ARP request for its own IPv4 address, it replies with its MAC address. Node B will also see the ARP request but will not reply as 192.168.1.3 is not an address of it. This is very similar to NDP for IPv6 except that an IPv4 node has only a single IPv4 address.

Also ARP broadcasts the request, this means that it is received by all IPv4 nodes in the network causing more host CPU utilisation than IPv6 NDP which only multicasts to some hosts.

IP Routing to non-Neighbours

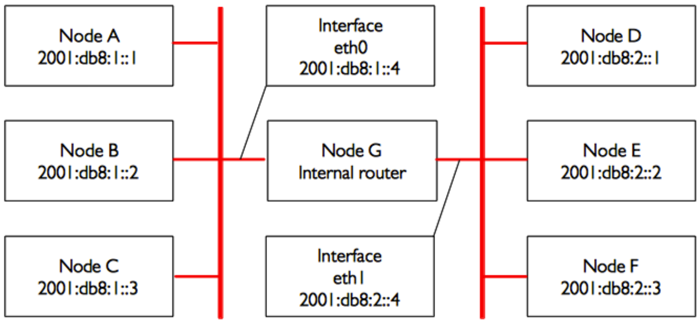


Figure NG 8: Two separate IPv6 networks.

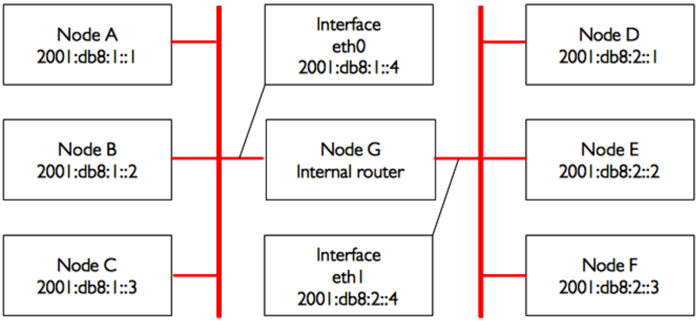
Consider now another network with 3 nodes, D, E, and F, with the corresponding IPv6 addresses 2001:db8:2::1, 2001:db8:2::2, and 2001:db8:2::3.
This is another /64 network, but it is not in the same range as the network on the left hand side.
All three hosts can reach each other directly (first using NDP to resolve the IPv6 address into a MAC address, and then sending packets to that MAC address).

Now we will add node G. This node has two network cards (also called interfaces), with one plugged into each network. The first network card uses the IPv6 address 2001:db8:1::4, on interface eth0 and the other, eth1, uses 2001:db8:2::4.
Node G is now link-local to both networks, and can forward packets between them: node G can route packets between the two networks, it is therefore called a router or sometimes a gateway.
But what if hosts A, B, and C want to reach hosts D, E, and F? They need to know that they should use node G and so they will need to add a route to the other network via host G. For example, hosts A-C would add a static route via 2001:db8:1::4.
In Linux, this can be accomplished with the following command:

 # ip -6 route add 2001:db8:2::/64 via 2001:db8:1::4

...and hosts D-F would add the following:

 # ip -6 route add 2001:db8:1::/64 via 2001:db8:2::4




```
Routing table for node A
# netstat -rn -A inet6
Destination            Next Hop              If
2001:db8:1:/64         ::                    eth0
2001:db8:2:/64         2001:db8:1::4         eth1
```

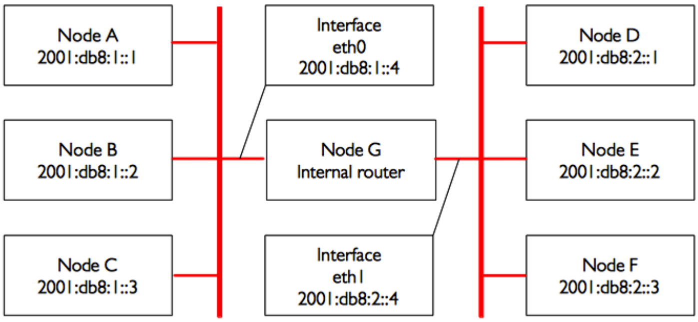
Figure NG 9: Node G acts as a router between the two networks, other hosts use static routes.

The result for node A is shown in Figure NG 9.

Notice that the route is added via the IPv6 address on host G that is link-local to the respective network.

Host A could not add a route via 2001:db8:2::4, even though it is the same physical machine as 2001:db8:1::4 (node G), since that IPv6 is not link-local.
The address of the next hop can be entered either as a global address (2001:db8:2::4) or as a link-local address (fe80::...); it is usually easier to configure a static route with a global address.

In IPv6, the router G also sends a solicitation and periodically router advertisements that contain its own link-local address, hence, all nodes using stateless auto-configuration or DHCP automatically add a default route via the router link-local address as shown in Figure NG 10.



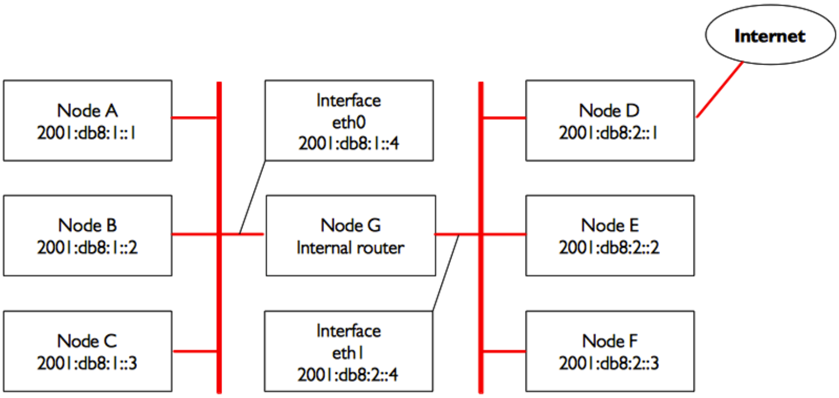

```
Routing table for normal node A(using RA)
# netstat -rn -A inet6
Destination           Next Hop                  If
2001:db8:1:/64        ::                        eth0
::/0                  fe80::21e:79ff:fe1e:d400  eth0
```

Figure NG 10: Node G acts as a router between the two networks, hosts use stateless address autoconfiguration.

This is a very simple routing example, where the destination is only a single hop away from the source. As networks get more complex, many hops may need to be traversed to reach the ultimate destination. Since it isn't practical for every machine on the Internet to know the route to every other, we make use of a routing entry known as the default route (also known as the default gateway).
When a router receives a packet destined for a network for which it has no explicit route, the packet is forwarded to its default gateway.
The default gateway is typically the best route out of your network, usually in the direction of your ISP.

An example of a router that uses a default gateway is shown in Figure NG 11. Figure NG 11 shows the routing table (which is the set of all routes) on the internal router G which includes the two directly connected networks 2001:db8:1::/64 and 2001:db8:2::/64 as well as a route to all other hosts on the Internet ::/0.

A node always uses the most specific route; that is the route with the longest match to the destination, in Figure NG 11 eth0 will be used for destination 2001:db8:1::1 (match length /64) rather than the less specific ::/0 (match length of 0).




```
Routing table for internal router G
# netstat -rn -A inet6
Destination            Next Hop                 If
2001:db8:1:/64         ::                       eth0
2001:db8:2:/64         ::                       eth1
::/0                   2001:db8:2::1            eth1
```

Figure NG 11: Node G is the internal router and uses the Internet router.

A route tells the OS that the desired network doesn't lie on the immediate link-local network, and it must forward the traffic through the specified router.

If host A wants to send a packet to host F, it would first send it to node G. Node G would then look up host F in its routing table, and see that it has a direct connection to host F's network.
Finally, host G would resolve the hardware (MAC) address of host F and forward the packet to it.

Routes can be updated manually, or can dynamically react to network outages and other events.
Some examples of popular dynamic routing protocols are RIP, OSPF, BGP.

Configuring dynamic routing is beyond the scope of this book, but for further reading on the subject, see the resources in **Appendix F**.
IPv4 behaves exactly the same way as depicted in figure NG 12.

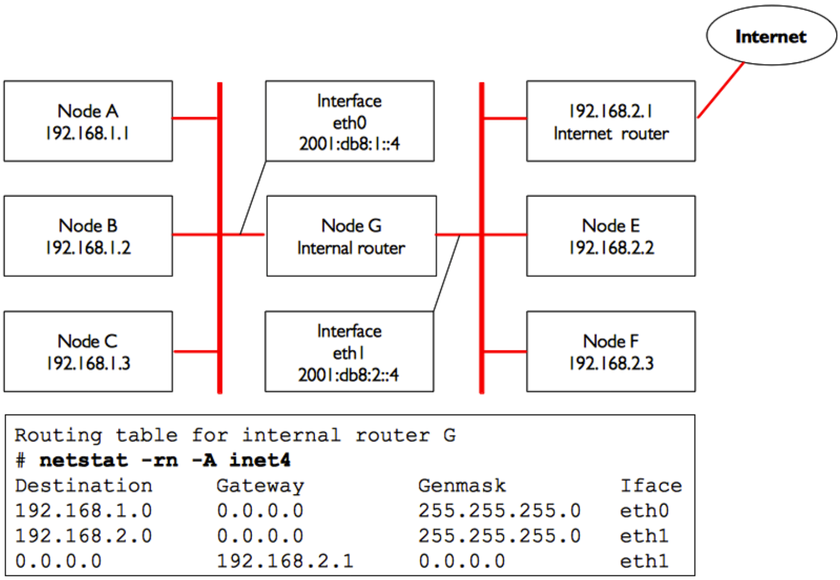


Figure NG 12: Node G is the Internet router on this IPv4 network.

As noted before, most networks and the Internet are dual-stack and all hosts and routers have both IPv4 and IPv6 addresses, this also means that the nodes will have routes for IPv4 and routes for IPv6. For instance, the set of all routes on node G of the previous figures will be:

*# **netstat -rn -A inet6***

Destination	*Next Hop*	*If*
2001:db8:1:/64	*::*	*eth0*
2001:db8:2:/64	*::*	*eth1*
::/0	*2001:db8:2::1*	*eth1*

*# **netstat -rn -A inet4***

Destination	*Gateway*	*Genmask*	*Iface*
192.168.1.0	*0.0.0.0*	*255.255.255.0*	*eth0*
192.168.2.0	*0.0.0.0*	*255.255.255.0*	*eth1*
0.0.0.0	*192.168.2.1*	*0.0.0.0*	*eth1*

Network Address Translation (NAT) for IPv4

In order to reach hosts on the Internet, private addresses must be converted to global, publicly routable IPv4 addresses.
This is achieved using a technique known as Network Address Translation, or NAT.
A NAT device is a router that manipulates the addresses of packets instead of simply forwarding them.
On a NAT router, the Internet connection uses one (or more) globally routed IPv4 addresses, while the private network uses an IPv4 address from the RFC1918 private address range.
The NAT router allows the global address(es) to be shared with all of the inside users, who all use private addresses.
It converts the packets from one form of addressing to the other as the packets pass through it. As far as the network users can tell, they are directly connected to the Internet and require no special software or drivers.
They simply use the NAT router as their default gateway, and address packets as they normally would.
The NAT router translates outbound packets to use the global IPv4 address as they leave the network, and translates them back again as they are received from the Internet.

The major consequence of using NAT is that machines from the Internet cannot easily reach servers within the organisation without setting up explicit forwarding rules on the router.

Connections initiated from within the private address space generally have no trouble, although some applications (such as Voice over IPv4 and some VPN software) can have difficulty dealing with NAT.

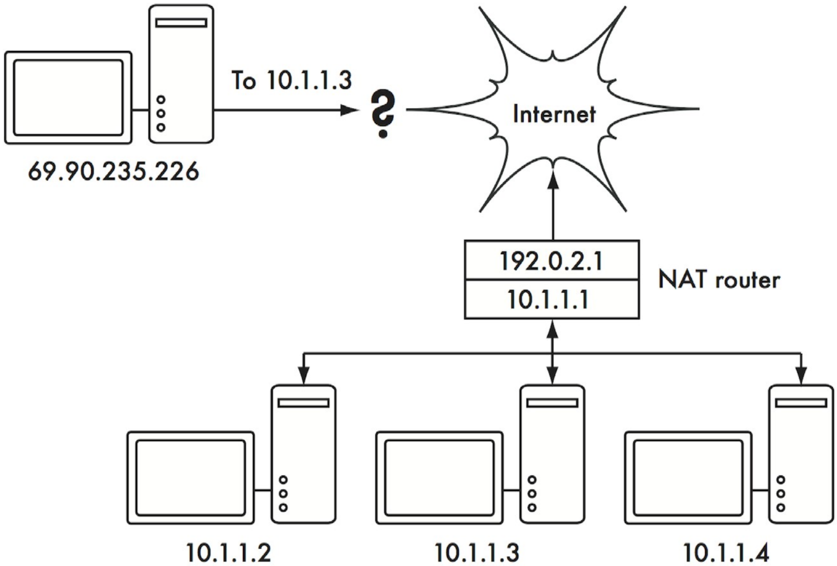


Figure NG 13: Network Address Translation allows you to share a single IPv4 address with many internal hosts, but can make it difficult for some services to work properly.

Depending on your point of view, this can be considered a bug (since it makes it harder to set up two-way communication) or a feature. RFC1918 addresses should be filtered on the edge of your network to prevent accidental or malicious RFC1918 traffic entering or leaving your network.

While NAT performs some firewall-like functions, it is not a replacement for a real firewall as most of the attacks happen now when an internal user visits some web sites with hostile content (called malware for malevolent software).

6. Internet Protocol Suite

Machines on the Internet use the Internet Protocol (IP) to reach each other, even when separated by many intermediary machines.

There are a number of protocols that are run in conjunction with IP that provide features which are as critical to normal operations as IP itself. Every packet specifies a protocol number that identifies the packet as one of these protocols.

The most commonly used protocols are the Transmission Control Protocol (TCP, number 6), User Datagram Protocol (UDP, number 17), and the Internet Control Message Protocol (ICMP, number 1 for IPv4 and number 58 for IPv6). Taken as a group, these protocols (and others) are known as the Internet Protocol Suite, or simply TCP/IP for short.

The TCP and UDP protocols introduce the concept of port numbers. Port numbers allow multiple services to be run on the same IP address, and still be distinguished from each other. Every packet has a source and destination port number. Some port numbers are well-defined standards, used to reach well-known services such as email and web servers. For example, web servers normally listen on TCP port 80 for insecure traffic and on TCP port 443 for encrypted/secure traffic, NTP time servers listen on UDP port 123, DNS domain name servers listen on UDP port 53, and SMTP email servers listen on TCP port 25.

When we say that a service "listens" on a port (such as port 80), we mean that it will accept packets that use its IP as the destination IP address, and 80 as the destination port.
Servers usually do not care about the source IP or source port, although sometimes they will use them to establish the identity of the other side.
When sending a response to such packets, the server will use its own IP as the source IP, and 80 as the source port.
When a client connects to a service, it may use any source port number on its side that is not already in use, but it must connect to the proper port on the server (e.g. 80 for web, 25 for email).

TCP is a session-oriented protocol with guaranteed and ordered delivery and transmission control features (such as detection and mitigation of network congestion, retries, packet reordering and reassembly, etc.).

UDP is designed for connectionless streams of information, and does not guarantee delivery at all, or in any particular order but can be faster so it is often used for real-time protocols such as for timing, voice or video.

The ICMP protocol is designed for debugging and maintenance on the Internet.
Rather than port numbers, it has message types, which are also numbers. Different message types are used to request a simple response from

another computer (echo request), notify the sender of another packet of a possible routing loop (time exceeded), or inform the sender that a packet that could not be delivered due to firewall rules or other problems (destination unreachable).
By now you should have a solid understanding of how computers on the network are addressed, and how information flows on the network between them.

Now let's take a brief look at the physical hardware that implements these network protocols.

7. Physical hardware

Ethernet

Ethernet is the name of the most popular standard for connecting together computers on a Local Area Network (LAN). It is sometimes used to connect individual computers to the Internet, via a router, ADSL modem, or wireless device.
However, if you connect a single computer to the Internet, you may not use Ethernet at all.
The name comes from the physical concept of the ether, the medium which was once supposed to carry light waves through free space. The official standard is called IEEE 802.3.

One widely deployed Ethernet standard is called 100baseT also known as Fast Ethernet.
This defines a data rate of 100 Megabits per second (hence the 100), running over twisted (hence the T) pair wires, with modular RJ-45 connectors on the end.

The network topology is a star, with switches or hubs at the centre of each star, and end nodes (devices and additional switches) at the edges. Servers are also connected using Gigabit Ethernet with a rate of 1 Gigabit per second.
Increasingly Gigabit Ethernet is replacing Fast Ethernet in many networks these days as demand for high volume video and other high data rate applications become more prevalent.

Medium Access Control (MAC) addresses

Every device connected to an Ethernet or WiFi network has a unique MAC address, assigned by the manufacturer of the network card. It serves as a unique identifier that enables devices to talk to each other. However, the scope of a MAC address is limited to a broadcast domain, which is defined as all the computers connected together by wires, hubs, switches, and bridges, but not crossing routers or Internet gateways.
MAC addresses are never used directly on the Internet, and are not transmitted across routers.
MAC addresses for Ethernet and IEEE 802.11 WiFi networks are 48 bits long and look like this - 00:1c:c0:17:78:8c or 40:6c:8f:52:59:41; for the latter MAC address, the first 24 bits 40:6c:8f indicates that Apple assigned this address.

Hubs

Ethernet hubs connect multiple twisted-pair Ethernet devices together. They work at the physical layer (the lowest or first layer). They repeat the signals received by each port out to all of the other ports. Hubs can therefore be considered to be simple repeaters.
Due to this design, only one port can successfully transmit at a time. If two devices transmit at the same time, they corrupt each other's transmissions, and both must back off and retransmit their packets later. This is known as a collision, and each host remains responsible for detecting and avoiding collisions before transmitting, and retransmitting its own packets when needed.

When problems such as excessive collisions are detected on a port, some hubs can disconnect (partition) that port for a while to limit its impact on the rest of the network.

While a port is partitioned, devices attached to it cannot communicate with the rest of the network.
Hubs are limited in their usefulness, since they can easily become points of congestion on busy networks so they are no longer normally deployed in networks nowadays. Its only important to note that a WiFi access point acts as a hub on the radio side.

Switches

A switch is a device which operates much like a hub, but provides a dedicated (or switched) connection between ports.

Rather than repeating all traffic on every port, the switch determines which ports are communicating directly and temporarily connects them together. There can be several such temporary port connections at the same time.

Switches generally provide much better performance than hubs, especially on busy networks with many computers. They are not much more expensive than hubs, and are replacing them in most situations.

Switches work at the data link layer (the second layer), since they interpret and act upon the MAC address in the packets they receive. When a packet arrives at a port on a switch, it makes a note of the source MAC address, which it associates with that port. It stores this information in an internal MAC table often known as Content Addressable Memory (CAM) table. The switch then looks up the destination MAC address in its MAC table, and transmits the packet only on the matching port. If the destination MAC address is not found in the MAC table, the packet is then sent to all of the connected interfaces hoping to reach the right MAC.

Hubs vs. Switches

Hubs are considered to be fairly unsophisticated devices, since they inefficiently rebroadcast all traffic on every port. This simplicity introduces both a performance penalty and a security issue. Overall performance is slower, since the available bandwidth must be shared between all ports. Since all traffic is seen by all ports, any host on the network can easily monitor all of the network traffic.

Switches create temporary virtual connections between receiving and transmitting ports. This yields better performance because many virtual connections can be made simultaneously. More expensive switches can switch traffic by inspecting packets at higher levels (at the transport or application layer), allowing the creation of VLANs, and implementing other advanced features.

A hub can be used when repetition of traffic on all ports is desirable; for example, when you want to explicitly allow a monitoring machine to see all of the traffic on the network. Most switches provide monitor port functionality that enables repeating on an assigned port specifically for this purpose.

Hubs were once cheaper than switches. However, the price of switches has reduced dramatically over the years. Therefore, old network hubs should be replaced whenever possible with new switches.

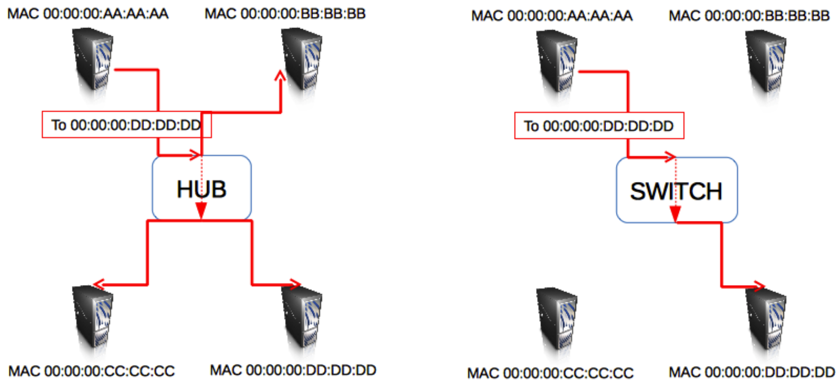


Figure NG 14: A hub simply repeats all traffic on every port, while a switch makes a temporary, dedicated connection between the ports that need to communicate.

Both hubs and switches may offer managed services.
Some of these services include the ability to set the link speed (10baseT, 100baseT, 1000baseT, full or half-duplex) per port, enable triggers to watch for network events (such as changes in MAC address or malformed packets), and usually include port counters for easy bandwidth accounting.

A managed switch that provides upload and download byte counts for every physical port can greatly simplify network monitoring.
These services are typically available via SNMP, or they may be accessed via telnet, ssh, a web interface, or a custom configuration tool.

Routers and Firewalls

While hubs and switches provide connectivity on a local network segment, a router's job is to forward packets between different network segments.
A router typically has two or more physical network interfaces.

It may include support for different types of network media, such as Ethernet, WiFi, optical fibre, DSL, or dial-up.
Routers can be dedicated hardware devices or they can be made from a standard PC with multiple network cards and appropriate software.
Routers sit at the edge of two or more networks. By definition, they have one connection to each network, and as border machines they may take on other responsibilities as well as routing. Many routers have firewall capabilities that provide a mechanism to filter or redirect packets that do not fit security or access policy requirements.
They may also provide Network Address Translation (NAT) services for IPv4.

Routers vary widely in cost and capabilities.
The lowest cost and least flexible are simple, dedicated hardware devices, often with NAT functionality, used to share an Internet connection between a few computers; well known brands include Linksys, D-Link, Netgear.

The next step up is a software router, which consists of an operating system running on a standard PC with multiple network interfaces. Standard operating systems such as Microsoft Windows, Linux, and BSD are all capable of routing, and are much more flexible than the low-cost hardware devices; it is often called Internet Connection Sharing.

However, they suffer from the same problems as conventional PCs, with high power consumption, a large number of complex and potentially unreliable parts, and more involved configuration.

The most expensive devices are high-end dedicated hardware routers, made by companies like Cisco and Juniper.
They tend to have much better performance, more features, and higher reliability than software routers on PCs.
It is also possible to purchase technical support and maintenance contracts for them.

Most modern routers offer mechanisms to monitor and record performance remotely, usually via the Simple Network Management Protocol (SNMP), although the least expensive devices often omit this feature.

Other equipment

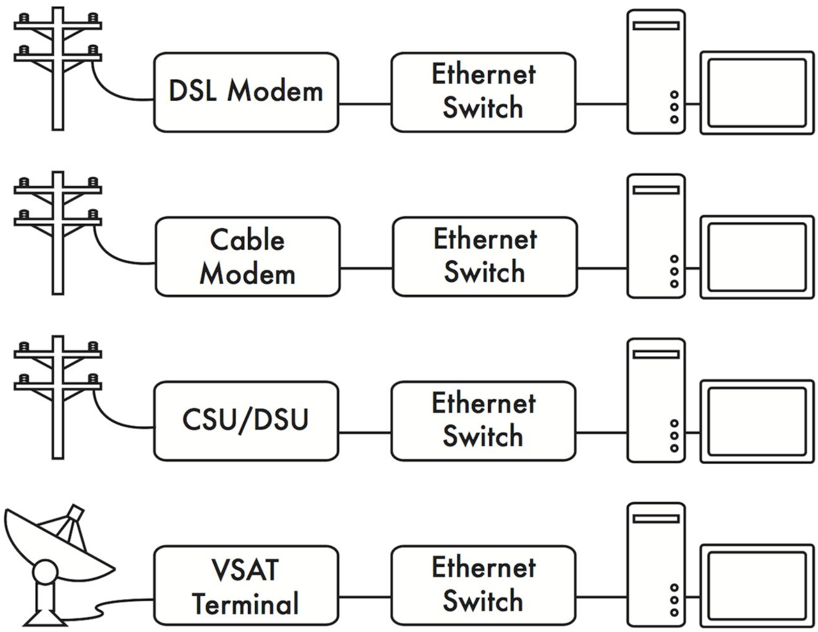


Figure NG 15: Many DSL modems, cable modems, wireless access points, and VSAT terminals terminate at an Ethernet jack.

Each physical network has an associated piece of terminal equipment. For example, VSAT connections consist of a satellite dish connected to a terminal that either plugs into a card inside a PC, or ends at a standard Ethernet connection. DSL lines use a DSL modem that bridges the telephone line to a local device, either an Ethernet network or a single computer via USB. Cable modems bridge the television cable to Ethernet, or to an internal PC card bus.

Standard dialup lines use modems to connect a computer to the telephone, usually via a plug-in card or serial port. And there are many different kinds of wireless networking equipment that connect to a variety of radios and antennas, but nearly always end at an Ethernet jack. The functionality of these devices can vary significantly between manufacturers. Some provide mechanisms for monitoring performance, while others may not.

Since your Internet connection ultimately comes from your ISP, you should follow their recommendations when choosing equipment that bridges their network to your Ethernet network.

8. Putting it all together

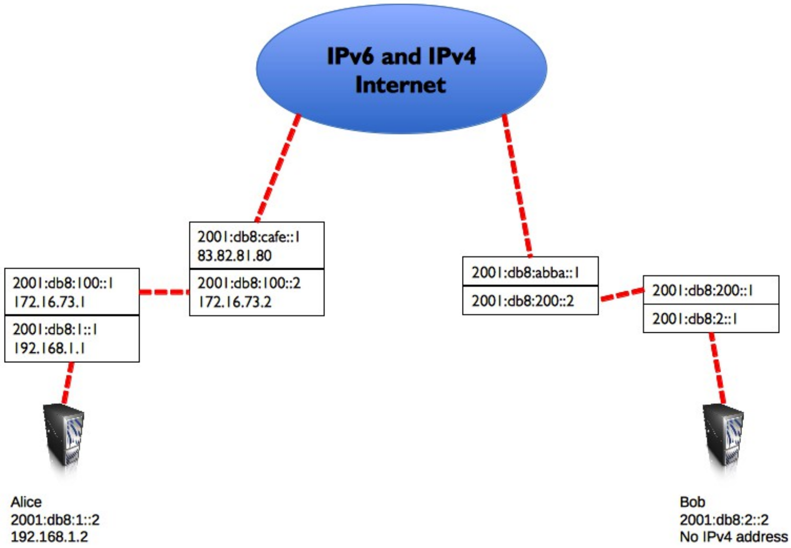


Figure NG 16: Internet networking. Each network segment has a router with two IP addresses, making it "link local" to two different networks. Packets are forwarded between routers until they reach their ultimate destination.

Once all network nodes have an IP address, they can send data packets to the IP address of any other node. Through the use of routing and forwarding, these packets can reach nodes on networks that are not physically connected to the originating node. This process describes much of what "happens" on the Internet.

In this example, you can see the path that the packets take as Alice chats with Bob using an instant messaging service.
Each dotted line represents an Ethernet cable, a wireless link, or any other kind of physical network.
The cloud symbol is commonly used to stand in for "The Internet", and represents any number of intervening IP networks.

Neither Alice nor Bob need to be concerned with how those networks operate, as long as the routers forward IP traffic towards the ultimate destination.
If it weren't for Internet protocols and the cooperation of everyone on the net, this kind of communication would be impossible.

In Figure NG 16, Alice is dual-stack and has IPv4 and IPv6 addresses, and as Bob has only IPv6 addresses, they will communicate by using IPv6 which is the common IP version between them.

9. Designing the physical network

It may seem odd to talk about the "physical" network when building wireless networks.
After all, where is the physical part of the network? In wireless networks, the physical medium we use for communication is obviously electromagnetic energy.
But in the context of this chapter, the physical network refers to the mundane topic of where to put things. How do you arrange the equipment so that you can reach your wireless clients?
Whether they fill an office building or stretch across many miles, wireless networks are naturally arranged in these three logical configurations: point-to-point links, point-to-multipoint links, and multipoint-to-multipoint clouds. While different parts of your network can take advantage of all three of these configurations, any individual link will fall into one of these topologies.

Point-to-point

Point-to-point links typically provide an Internet connection where such access isn't otherwise available. One side of a point-to-point link will have an Internet connection, while the other uses the link to reach the Internet.

For example, a university may have a fast frame relay or VSAT connection in the middle of campus, but cannot afford such a connection for an important building just off campus. If the main building has an unobstructed view of the remote site, a point-to-point connection can be used to link the two together. This can augment or even replace existing dial-up links. With proper antennas and clear line of sight, reliable point-to-point links in excess of thirty kilometres are possible.

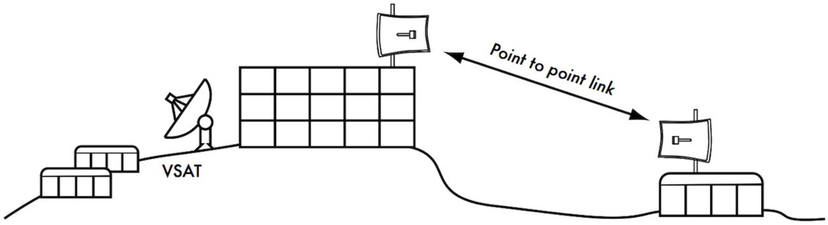


Figure NG 17: A point-to-point link allows a remote site to share a central Internet connection.

Of course, once a single point-to-point connection has been made, more can be used to extend the network even further. If the remote building in our example is at the top of a tall hill, it may be able to see other important locations that can't be seen directly from the central campus. By installing another point-to-point link at the remote site, another node can join the network and make use of the central Internet connection.

Point-to-point links don't necessarily have to involve Internet access. Suppose you have to physically drive to a remote weather monitoring station, high in the hills, in order to collect the data which it records over time. You could connect the site with a point-to-point link, allowing data collection and monitoring to happen in realtime, without the need to actually travel to the site.

Wireless networks can provide enough bandwidth to carry large amounts of data (including audio and video) between any two points that have a connection to each other, even if there is no direct connection to the Internet.

Point-to-multipoint

The next most commonly encountered network layout is point-to-multipoint. Whenever several nodes are talking to a central point of access, this is a point-to-multipoint application. The typical example of a point-to-multipoint layout is the use of a wireless access point that provides a connection to several laptops. The laptops do not communicate with each other directly, but must be in range of the access point in order to use the network.

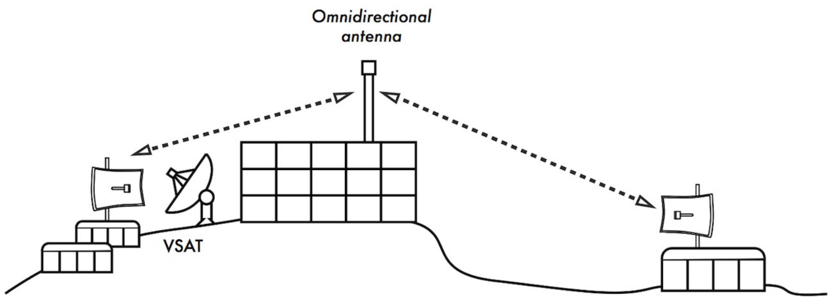


Figure NG 18: The central VSAT is now shared by multiple remote sites. All three sites can also communicate directly at speeds much faster than VSAT.

Point-to-multipoint networking can also apply to our earlier example at the university. Suppose the remote building on top of the hill is connected to the central campus with a point-to-point link.
Rather than setting up several point-to-point links to distribute the Internet connection, a single antenna could be used that is visible from several remote buildings. This is a classic example of a wide area point (remote site on the hill) to multipoint (many buildings in the valley below) connection.

Note that there are a number of performance issues with using point-to-multipoint over very long distance, which will be addressed in the chapter called **Deployment Planning**. Such links are possible and useful in many circumstances, but don't make the classic mistake of installing a single high powered radio tower in the middle of town and expecting to be able to serve thousands of clients, as you would with an FM radio station. As we will see, two-way data networks behave very differently than broadcast radio.

Multipoint-to-multipoint

The third type of network layout is multipoint-to-multipoint, which is also referred to as an ad-hoc or mesh network. In a multipoint-to-multipoint network, there is no central authority. Every node on the network carries the traffic of every other as needed, and all nodes communicate with each other directly.

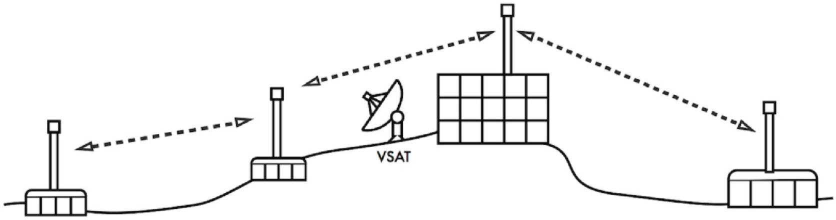


Figure NG 19: A multipoint-to-multipoint mesh. Every point can reach every other at very high speed, or any of them can use the central access point for a VSAT connection to the Internet.

The benefit of this network layout is that even if none of the nodes are in range of a central access point, they can still communicate with each other.
Good mesh network implementations are self-healing, which means that they automatically detect routing problems and fix them as needed. Extending a mesh network is as simple as adding more nodes.
If one of the nodes in the "cloud" happens to be an Internet gateway, then that connection can be shared among all of the clients.

Several disadvantages of this topology include increased complexity and lower performance.

Security in such a network is also a concern, since every participant potentially carries the traffic of every other.
Multipoint-to-multipoint networks tend to be difficult to troubleshoot, due to the large number of changing variables as nodes join and leave the network.
Multipoint-to-multipoint clouds typically have reduced capacity compared to point-to-point or point-to-multipoint networks, due to the additional overhead of managing the network routing and increased contention in the radio spectrum.

Nevertheless, mesh networks are useful in many circumstances.
For more information about them please read the chapter called **Mesh Networking.**

Use the technology that fits

All of these network designs can be used to complement each other in a large network, and additionally they can make use of traditional wired networking techniques whenever possible. Wired networks still often have higher bandwidth capacity than wireless so should be used whenever appropriate or affordable.
But looking at the wireless, it is a common practice, for example, to use a long distance wireless link to provide Internet access to a remote location, and then set up an access point on the remote side to provide local wireless access. One of the clients of this access point may also act as a mesh node, allowing the network to spread organically between laptop users who all ultimately use the original point-to-point link to access the Internet.

This is just one common scenario for wireless deployments, there are many others.

Now that we have a clear idea of how wireless networks are typically arranged, we can begin to understand how communication is possible over such networks.