The most common implementation types of TLS VPN are as follows:

- **TLS portal VPN** An individual uses a single standard TLS connection to a website to securely access multiple network services. The website accessed is typically called a *portal* because it is a single location that provides access to other resources. The remote user accesses the TLS VPN gateway using a web browser, is authenticated, and is then presented with a web page that acts as the portal to the other services.
- **TLS tunnel VPN** An individual uses a web browser to securely access multiple network services, including applications and protocols that are not web-based, through a TLS tunnel. This commonly requires custom programming to allow the services to be accessible through a web-based connection.

Summary of Tunneling Protocols

Layer 2 Tunneling Protocol (L2TP):

- Hybrid of L2F and PPTP
- Extends and protects PPP connections
- Works at the data link layer
- Transmits over multiple types of networks, not just IP
- Combined with IPSec for security

IPSec:

- Handles multiple VPN connections at the same time
- Provides secure authentication and encryption
- Supports only IP networks
- Focuses on LAN-to-LAN communication rather than user-to-user communication
- Works at the network layer and provides security on top of IP

Transport Layer Security (TLS):

- Works at the session layer and protects mainly web and e-mail traffic
- Offers granular access control and configuration
- Easy to deploy since TLS is already embedded into web browsers
- Can only protect a small number of protocol types, thus is not an infrastructure-level VPN solution

Since TLS VPNs are closer to the application layer, they can provide more granular access control and security features compared to the other VPN solutions. But since they are dependent on the application layer protocol, there are a smaller number of traffic types that can be protected through this VPN type.

One VPN solution is not necessarily better than the other; they just have their own focused purposes:

- L2TP is used when a PPP connection needs to be extended through a network.
- IPSec is used to protect IP-based traffic and is commonly used in gateway-to-gateway connections.
- TLS VPN is used when a specific application layer traffic type needs protection.

Secure Protocols

TLS may be one of the most talked-about technologies when it comes to network security. Still, there are other protocols, and other applications of TLS, that you should know. This section addresses each of the main network services, web, DNS, and e-mail. Let's start with how we secure web services.

Web Services

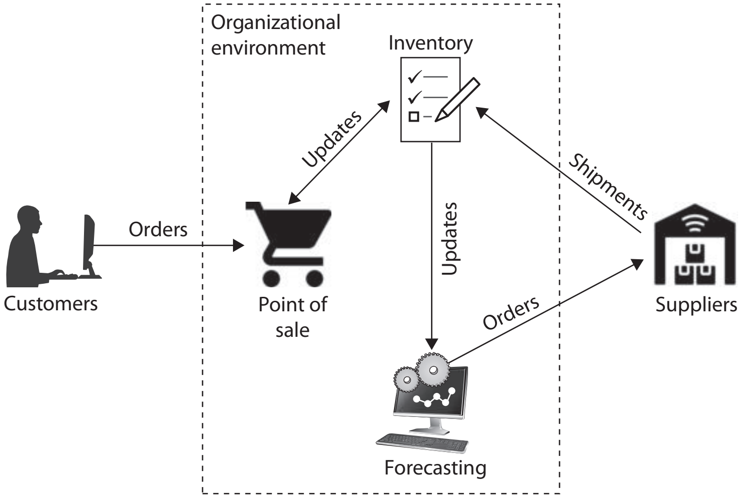
Many people hear the term "web services" and think of websites and the web servers that do the work behind the scenes. In reality, however, this is but a portion of what the term actually covers. A *web service* is a client/server system in which clients and servers communicate using HTTP over a network such as the Internet. Sure, this definition covers static web pages written in HTML being served out of an old Apache server somewhere, but it can also cover much more.

For example, suppose you are a retailer and don't want to pay for a huge storage space for merchandise that may or may not sell anytime soon. You could implement a just-in-time logistics system that keeps track of your inventory and past selling patterns, and then automatically order merchandise so that it arrives just before you start running low. This kind of system is typically implemented using a business-to-business (B2B) web service and is depicted in Figure 13-5. Each icon in the figure represents a distinct web service component.

When we look at web services this way, it becomes clear that we have much more to worry about than simply the interaction between customers and our website. Let's look at ways in which we could implement some of the secure design principles in this example. The following list is meant to be illustrative, not all-inclusive:

- **Least privilege** The forecasting service should have read-only access to some of the data in the inventory system. It doesn't need any additional access.
- **Secure defaults** The inventory service should refuse all connection requests from any endpoint other than those explicitly authorized (point of sale and forecasting). If any other connections are required, those should be added as exceptions after a careful review.

Figure 13-5 Example just-in-time logistics B2B web service



- **Fail securely** The forecasting service has the ability to spend money by placing orders from suppliers. It should not process any order that is malformed or otherwise fails any checks.
- **Separation of duties** The forecasting service can place orders but cannot receive shipments and update inventory. Ordering and receiving should be two separate duties performed by different people/systems to mitigate the risk of fraud.
- **Zero trust** Before any two components collaborate, they should both be required to authenticate with each other and encrypt all communications. This is particularly true (and the authentication protocol should be more rigorous) when dealing with external parties like customers and suppliers.
- **Privacy by design** Customer information should not be shared outside the point-of-sale (PoS) system, particularly since the other two internal systems (inventory and forecasting) communicate with an external third party. This example is overly simplistic, but the point is that customer data should be limited to the components that absolutely need it.
- **Trust but verify** All components (with the possible exception of the user) should generate logs that are sufficient to detect attacks or errors. Ideally, these logs are centrally collected to make them easier to correlate and harder to tamper with.
- **Shared responsibility** The security obligations of the organization and of the supplier should be codified in a legally binding contract and audited periodically.

Again, the list is not exhaustive, but it should give you an idea of how the secure design principles can be applied to a web services scenario. You should be prepared to do likewise with a variety of other scenarios for the CISSP exam.

How are these web services actually delivered? The key is to focus on *what* service is being delivered, and not on *how* it is implemented or *where* it is hosted (as long as it is available). A *service-oriented architecture (SOA)* describes a system as a set of interconnected but self-contained components that communicate with each other and with their clients through standardized protocols. These protocols, called *application programming interfaces (APIs)*, establish a "language" that enables a component to make a request from another component and then interpret that second component's response. The requests that are defined by these APIs correspond to discrete business functions (such as estimated shipping costs to a postal code) that can be useful by themselves or can be assembled into more complex business processes. An SOA has three key characteristics: self-contained components, a standardized protocol (API) for requests/responses, and components that implement business functions.

SOAs are commonly built using web services standards that rely on HTTP as a standard communication protocol. Examples of these are SOAP (which used to stand for the Simple Object Access Protocol) and the Representational State Transfer (REST) architectures. Let's look at these three (HTTP, SOAP and REST) in turn.

Hypertext Transfer Protocol

HTTP is a TCP/IP-based communications protocol used for transferring resources (e.g., HTML files and images) between a server and a client. It also allows clients to send queries to the server. The two basic features of HTTP are that it is connectionless and stateless. Connectionless protocols do not set up a connection (obviously) and instead send their messages in a best-effort manner. They rely on some other protocol (in this case TCP) to ensure the message gets across. *Stateless* means that the server is amnesiac; it doesn't remember any previous conversations with any clients. Thus, whatever is needed for the server to "remember" has to be provided with each request. This is a role commonly played by session identifiers and cookies.

NOTE A cookie is just a small text file containing information that only one website can write or read.

Uniform Resource Identifiers A foundational component of HTTP is the use of the uniform resource identifier (URI), which uniquely identifies a resource on the Internet. A typical URI looks like this: http://www.goodsite.com:8080/us/en/resources/search.php?term=cissp. Let's look at its components in sequence:

1. **Scheme** This is another name for the protocol being used (e.g., HTTP or HTTPS) and ends in a colon (:).
2. **Authority** There are three possible subcomponents here, but the second is the most prevalent:
   - Username (optional) (and optional password, separated by a colon) followed by an at (@) symbol.
   - Host in either hostname (e.g., www.goodsite.com) or IP address format.
   - Port number (optional), preceded by a colon (e.g., :8080). Note that port 80 is assumed for HTTP schemes and port 443 for HTTPS schemes.

3. **Path** The path to the requested resource on the server. If the path is not specified by the client, it is assumed to be a single slash (/), which is the default document at the root of the website (e.g., the homepage). Subdirectories are indicated as they are in Linux/Unix by successive slashes (e.g., /us/en/resources/search.php).
4. **Query (optional)** An attribute-value pair preceded by a question mark (?) (e.g., ?term=cissp). Each additional pair is separated from the previous one by an ampersand (&).

Request Methods HTTP uses a request-response model in which the client requests one or more resources from the server, and the latter provides the requested resources (assuming, of course, they are available to the client). The protocol defines two request methods: GET and POST. The main difference for our discussion is that a GET request must include all parameters in the URI, while POST allows us to include additional information (e.g., parameters) in the body of the request, where it will not be revealed in the URI. So, in the previous example we can safely guess that the method used was GET because we see the search term (cissp) in the URI.

Hypertext Transfer Protocol Secure *HTTP Secure (HTTPS)* is HTTP running over Transport Layer Security (TLS). Ensuring that all your web services require HTTPS is probably the most important security control you can apply to them. Recall that unencrypted requests can provide an awful lot of sensitive data, including credentials, session IDs, and URIs. Ideally, you require TLS 1.3 on all your web servers and ensure they do not allow unencrypted communications (by enforcing secure defaults).

An important consideration before you jump to HTTPS everywhere is whether you want to perform deep packet analysis on all your internal traffic. If you force use of HTTPS, you will need to deploy TLS decryption proxies, which can be pricey and require careful configuration on all your endpoints. The way these proxies work is by performing what is essentially a (benign) man-in-the-middle attack in which they terminate the clients' secure sessions and establish the follow-on session to their intended server. This allows the proxy to monitor all HTTPS traffic, which provides a measure of defense in depth but may pose some challenges to the privacy by design principle. Many organizations deal with this challenge by whitelisting connections to certain types of servers (e.g., healthcare and financial services organizations), while intercepting all others.

SOAP

SOAP is a messaging protocol that uses XML over HTTP to enable clients to invoke processes on a remote host in a platform-agnostic way. SOAP was one of the first SOAs to become widely adopted. SOAP consists of three main components:

- A message envelope that defines the messages that are allowed and how they are to be processed by the recipient
- A set of encoding rules used to define data types
- Conventions regarding what remote procedures can be called and how to interpret their responses

Extensible Markup Language

The term XML keeps coming up for good reasons. Extensible Markup Language is a popular language to use if you want to mark up parts of a text document. If you've ever looked at raw HTML documents, you probably noticed the use of tags such as <title>CISSP</title> to mark up the beginning and end of a page's title. These tags enable both humans and machines to interpret text and process it (such as rendering it in a web browser) as the author intended. Similarly, XML enables the author of a text document to "explain" to a receiving computer what each part of the file means so that a receiving process knows what to do with it. Before XML, there was no standard way to do this, but nowadays there are a number of options, including JavaScript Object Notation (JSON) and YAML Ain't Markup Language (YAML).

SOAP security is enabled by a set of protocol extensions called the Web Services Security (WS-Security or WSS) specification, which provides message confidentiality, integrity, and authentication. Note that, in keeping with HTTP's stateless nature, the focus here is on message-level security. Confidentiality is provided through XML encryption, integrity through XML digital signatures, and single-message authentication through security tokens. These tokens can take on various forms (the specification is intentionally broad here), which include username tokens, X.509 digital certificates, SAML assertions, and Kerberos tickets (we'll cover the last two in Chapter 17).

One of the key features of SOAP is that the message envelope allows the requester to describe the actions that it expects from the various nodes that respond. This feature supports options such as routing tables that specify the sequence and manner in which a series of SOAP nodes will take action on a given message. This can make it possible to finely control access as well as efficiently recover from failures along the way. This richness of features, however, comes at a cost: SOAP is not as simple as its name implies. In fact, SOAP systems tend to be fairly complex and cumbersome, which is why many web service developers prefer more lightweight options like REST.

Representational State Transfer

Unlike SOAP, which is a messaging protocol, Representational State Transfer (REST) is an architectural pattern used to develop web services using a variety of languages. In REST, HTTP is used to provide an API that allows clients to make programmatic requests from servers. For example, a client of a RESTful service could insert a new user record using the HTTP POST method (which lets you send additional information in the body of the request) by sending the following URI: https://www.goodsite.com/UserService/Add/1. The server would know to read the body of the POST to get the new user's details, create it, and then send a HTTP confirmation (or error). As you can see, REST essentially creates a programming language in which every statement is an HTTP URI.

Because every interaction with the system is spelled out in the URI, it is essential to use HTTPS as a secure default communications protocol. Of course, in keeping with the principle of zero trust, we want to authenticate clients and servers to each other, as well as put limits on what resources are available to each client. Another good security practice for RESTful services, which applies to any software system, is to validate all inputs before processing them. This mitigates a large number of possible injection attacks in which the adversary deliberately provides malformed inputs in order to trigger a system flaw.

Domain Name System

We covered the Domain Name System (DNS) in a fair amount of detail back in Chapter 11. Let's return to it now in the context of its role in helping us to secure our networks. Early on in its history, DNS was most commonly targeted by attackers to hijack requests, redirecting the unwitting requesters to malicious hosts instead of the legitimate ones they were seeking. While this is still a concern that we'll address in a bit, we also have to consider the much more common use of DNS to assist threat actors in conducting attacks, rather than being the target of attacks.

Since some of the most problematic adversarial uses of DNS depend on how this system works, let's review the process by which DNS performs recursive queries. Recall from Chapter 11 that a *recursive query* means that the request can be passed on from one DNS server to another one until the DNS server with the correct information is identified. This is illustrated in Figure 13-6. First, the client queries its local DNS server, which may either be an authoritative source for it or have cached it after some other client's request. Failing that, the server will typically start by consulting the root DNS server. The root server (there are actually a few of them for redundancy) will probably say something like "No, but here is the address of the name server for all .com domains." The local server will then query that server, which will probably result in it responding "No, but here is the address of the name server responsible for ironnet.com." Finally, the local server will query that other server, which will respond with an A record containing the IP address of the www host.

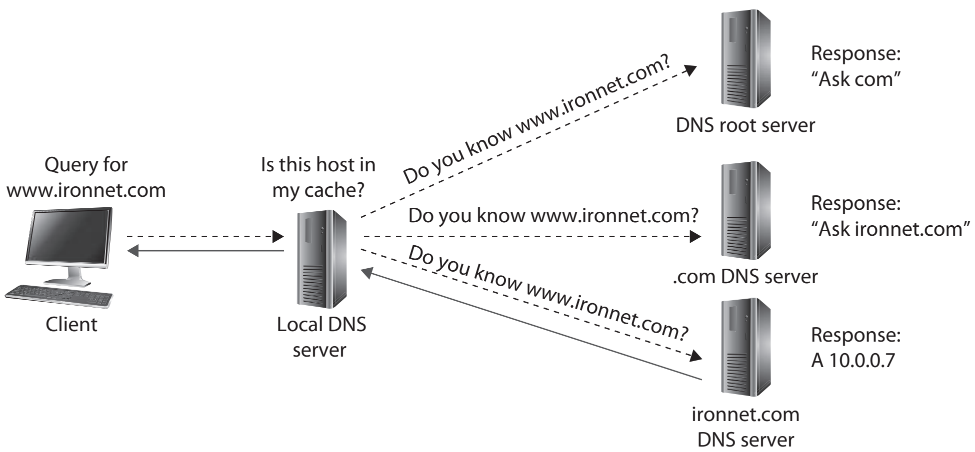


Figure 13-6 A recursive DNS query

Preventing Common DNS Attacks

DNS is the Internet's ubiquitous messenger; its queries and responses go everywhere, and without them the Internet as we know it would not work. Because of its importance to most other network systems, DNS traffic is seldom blocked by firewalls or routers. Attackers quickly figured out that this ubiquity makes DNS a preferred tool to manipulate and use for their own nefarious purposes. Perhaps the cleverest application of DNS for unintended purposes is its use to reach out and touch hosts in ways that are difficult to block using pseudo-randomly generated domain names.

EXAM TIP You will not be tested on the material that covers the following DNS attacks, but note that these attacks are both important to know and illustrative of the challenges we face in securing networks. If you are preparing for the exam only, feel free to move to the "Domain Name System Security Extensions" section.

Domain Generation Algorithms Once malware is implanted on target systems, the adversaries still need to communicate with those hosts. Since inbound connection attempts would easily be blocked at the firewall, most malware initiates outbound connections to the attacker's command and control (C2) infrastructure instead. The problem for the attackers is that if they provide a hostname or IP address in the malware, defenders will eventually find it, share it as an indicator of compromise (IOC), and reduce or negate the effectiveness of the C2 system.

To bypass signature detection by intrusion detection systems (IDSs) and intrusion prevention systems (IPSs) that use these IOCs, malware authors developed algorithms that can generate different domain names in a manner that appears random but produces a predictable sequence of domain names for those who know the algorithm. Suppose I am an attacker and want to hide my real C2 domains to keep them from being blocked or removed. I develop a domain generation algorithm (DGA) that produces a new (seemingly) random domain name each time it is run. Sprinkled somewhere in that (very long) list of domains are the ones I actually want to use. The infected host then attempts to resolve each domain to its corresponding IP address using DNS. Most of the domains do not exist and others may be benign, so either way there is no malicious C2 communications that follow. However, since I know the sequence of domains generated by the DGA and I know how quickly the malware will generate them, I can determine approximately when a particular infected host will query a specific domain. I can then register it the day before and rendezvous with the malware on that domain so I can receive its report and/or issue commands. The defenders won't know which domains are my malicious ones and which are just noise meant to distract them.

Figure 13-7 shows three domains being generated by an infected host. The first two that are queried do not exist, and thus result in an NXDOMAIN response from the server, which means the domain was not found. The third domain resolves to a malicious domain. When the authoritative (malicious) server for that domain receives the request, it knows it comes from a compromised system and sends a response that, when decoded, means "sleep for 7 hours."

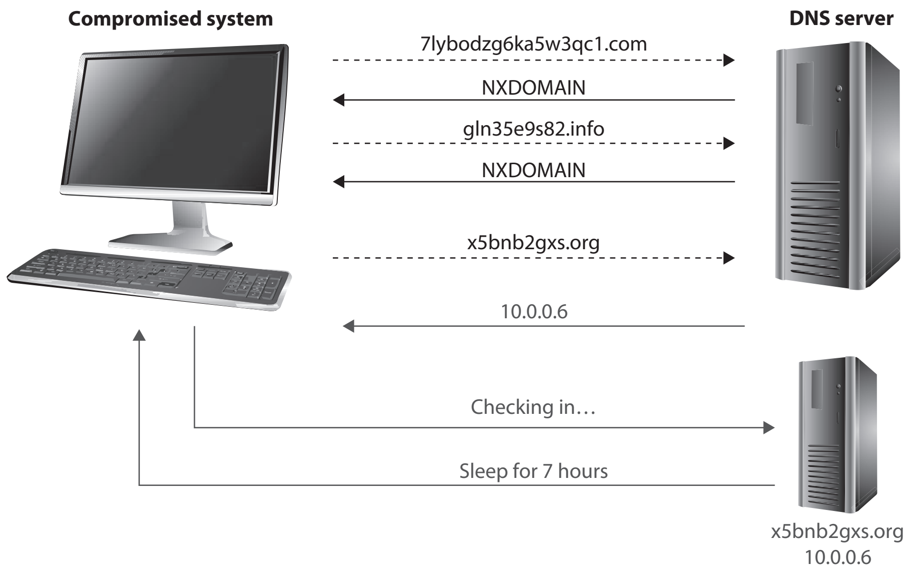


Figure 13-7 DGA in use by a compromised system

How can we detect and stop this kind of adversarial behavior? There are two general approaches. The first is to capture the malware and reverse engineer its DGA. We then play it forward (just like the attacker does) to determine which domains will be generated and when. Knowing this timeline, you can blacklist the domains and use the fact that a host attempted to reach them to infer that the querying system is compromised. Keep in mind that different compromised systems will be generating domain names at different times, so the task is onerous even for organizations that are mature enough to reverse engineer malware in the first place.

The second approach to detecting and stopping the use of DGAs is to analyze the domain names in each query to determine the probability of the query being legitimate. You can see from Figure 13-7 that most domains generated by these algorithms look, well, random. They are not the sort of domain names that you would expect someone to pay money to register. If you find a domain that is highly suspicious, you can investigate the host to see if it is infected, or you could block or monitor the DNS query and response to see if there is anything suspicious in either. For example, in some cases, the response will come as an encoded or encrypted message in a TXT record. This approach is only practical if you have a fairly sophisticated artificial intelligence analysis system that can examine every DNS request and learn over time which ones are likely to be bad.

NOTE There are legitimate uses of DGAs. For example, some systems use them to test whether or not a system can reach the Internet and perhaps track who that system is. This is done by some developers for licensing, updating, or diagnostic purposes.

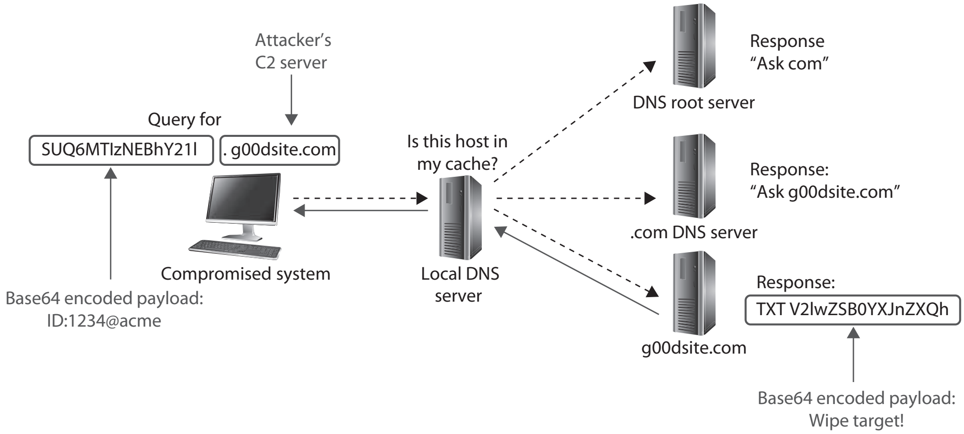


Figure 13-8 Covert communication over a DNS tunnel

DNS Tunneling Malicious use of a DGA can be very hard to stop unless you have advanced capabilities at your disposal. Fortunately, however, this use is limited to simple messaging between a compromised host and an external threat actor. But what if we could use DNS to transfer more information? A lot more? It turns out that data can be hidden in DNS queries using encoded host and other resource labels. *DNS tunneling* is the practice of encoding messages in one or a series of DNS queries or responses for exfiltrating or infiltrating data into an environment.

Figure 13-8 shows a very simple example of DNS tunneling that builds on our discussion of recursive queries in Figure 13-6. In this case, the compromised system wants to check in with its C2 server, so it uses Base64 encoding to obfuscate its message, which contains its identifier. Let's say that this is an infected host in the Acme Corporation, so its ID is 1234@acme. The recursive DNS query eventually is sent to the server that owns the malicious domain g00dsite.com. It decodes the hostname field, sees which of its bots this is from, and decides it is time to wipe the file system on the infected system. This command comes in the form of a TXT response that is also Base64 encoded.

A similar, but much less noticeable, use of DNS tunneling is to slowly exfiltrate data from the compromised system. Since DNS allows names of up to 63 characters between each dot, attackers can break down a longer file (e.g., a secret document) and exfiltrate it in a sequence of DNS queries to the same server or different servers.

Defending against DNS tunneling is similarly difficult to countering DGAs. Again, we could use network detection and response (NDR) solutions that use artificial intelligence to look for this type of behavior. However, because this type of attack (unlike DGAs) tends to rely on just a few domains, we could use domain reputation tools to determine whether any of our systems are making queries for suspicious or malicious domains.

Distributed Denial of Service The third type of DNS attack targets someone else's infrastructure using your DNS servers. An attacker who owns (or can rent) a large army of compromised systems (bots) can use them to overwhelm a target with name resolution responses to queries it didn't send out in the first place. To see how this attack works, we must first consider that DNS is based on UDP, which means spoofing the source address of a query is trivial.

In a *DNS reflection attack*, the threat actor instructs each bot they control to send a query to one of many open DNS servers around the world, while spoofing the source addresses on those queries. Collectively, the responding servers then bombard the target with traffic. If you have a sufficient number of bots and servers doing this quickly enough, the results could take the target system offline. Even if the target is not a DNS server, it still has to process millions (or more) of UDP packets arriving each second, which can overwhelm the typical server. But what if we could amplify the effects?

A *DNS amplification attack* is characterized by small queries that result in very much larger responses. A typical query is about 30 bytes and its response is around 45 bytes on average. The following are three techniques that are used to turn this relatively equal ratio of query to response size by a factor of up to 50 times:

- **DNS ANY** DNS has a (deprecated in 2019, but still used) diagnostic feature that allows a client to request all the information a server has on a given domain name. By sending a query of type ANY, an attacker can cause the server to send all the records in that domain up to the maximum size of a DNS message, which is 512 bytes. Having a 30-byte query produce a 512-byte response is a 17× amplification.
- **EDNS(0)** There are several situations in which the 512-byte limit on DNS messages over UDP becomes problematic. In particular, it is not possible to implement DNS Security Extensions (DNSSEC) with this constraint. Therefore, the Internet Engineering Task Force (IETF) developed EDNS(0), the Extension Mechanisms for DNS, which allows for up to 4096-byte responses. Properly used by an attacker, this new maximum size represents a 136× amplification given a 30-byte query.
- **DNSSEC** One of the most practical ways to exploit the maximum size defined in EDNS(0) is, ironically, using DNSSEC. Going back to Figure 13-6, when the local DNS server requests the A record from the authoritative server for that domain (the bottom left one), it also requests the DNSSEC associated with the zone. This is done to ensure the identity of the authoritative server (and hence the response) but results in a significantly larger response (because it includes a digital signature). So, all an attacker needs to do is find open DNS servers that have DNSSEC enabled and direct the bots at them.

Domain Name System Security Extensions

DNSSEC is a set of standards IETF developed to protect DNS from a variety of attacks. Specifically, DNSSEC is focused on ensuring the integrity of DNS records, not their confidentiality or availability. In plain-old DNS, a client makes a recursive query that, eventually, is responded to by some server that claims to be authoritative and provides an IP address. As we discussed in Chapter 11, however, this led to impersonation attacks where unwitting clients were pointed to malicious hosts. In response to this threat, the IETF came up with DNSSEC.

DNSSEC works by grouping records in a DNS zone according to their name and type (e.g., A, NS, MAIL) into Resource Record Sets (RRSets) that are then digitally signed, with the resulting signature going into a resource record signature (RRSig) record. The corresponding public key is published in a DNSKey record. So, when we want to resolve a fully qualified domain name (FQDN) using DNSSEC, we first retrieve the RRSet containing the name, then we request the RRSig for that set, and finally we verify that the record has not been tampered with. While this approach prevents impersonation and cache poisoning attacks, it has, as we just saw, also opened the door to crippling amplification attacks.

DNS over HTTPS

While DNSSEC ensures the integrity of DNS data, it does nothing to protect the confidentiality or privacy of queries. Sure, you can be confident that the IP address you got back was the right one, but what if anyone on the network can now see that you went to a domain called embarrassingmedicalcondition.com? We know from our discussion of TLS 1.3 earlier in this chapter that this URL will not go out in plaintext over HTTPS (which, by the way, it will in TLS 1.2 and earlier), but it will still be visible before the TLS handshake when the DNS query goes out. This is particularly problematic when we are connected to public networks such as the Wi-Fi network at the local coffee shop.

DNS over HTTPS (DoH) is a (yet to be ratified) approach to protecting the privacy and confidentiality of DNS queries by sending them over HTTPS/TCP/IP instead of unsecured UDP/IP. As of this writing, DoH is available on most platforms, though it is an optional feature that has to be configured. Keep in mind, however, that DoH provides confidentiality but (unlike DNSSEC) not integrity protections. Also, DoH was conceived as a privacy mechanism when using public networks. If you think back to the DNS-enabled attacks we discussed earlier in this chapter (especially DGA and DNS tunneling), DoH would actually make these much harder to detect unless you have a TLS decryption proxy in place. This is one of the reasons why the U.S. NSA recommended in 2021 that DoH not use external resolvers in enterprise networks.

DNS Filtering

Our final topic on securing DNS is perhaps the most obvious. Instead of allowing any DNS request to go out of our organizational networks, what if we first filtered them to block known malicious (or otherwise disallowed) domains from being resolved in the first place? A DNS filter performs a similar role as a web proxy that blocks content that is inappropriate, except that it works on DNS instead of HTTP traffic. There are many commercial solutions that provide this functionality, but keep in mind they should be deployed as part of a broader, defense-in-depth approach to securing DNS.

Electronic Mail

Let's now shift our attention to the third major service (along with web and DNS services) that is required for virtually all major organizations: e-mail. Though it has lost some ground to other business communication platforms such as Slack, Microsoft Teams,

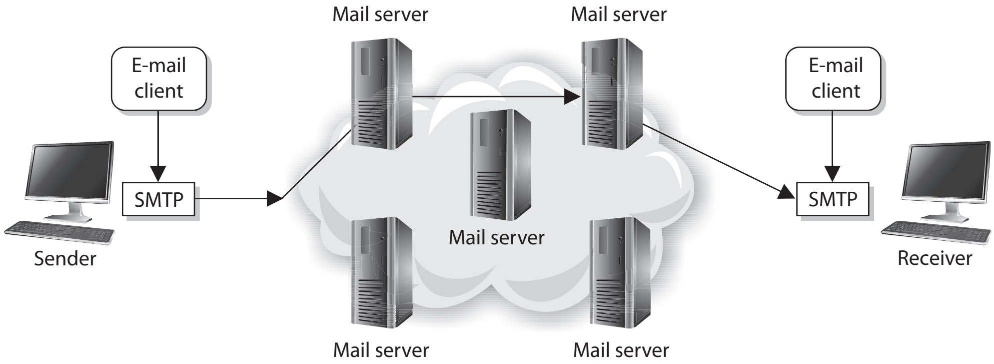


Figure 13-9 SMTP works as a transfer agent for e-mail messages.

and Google Hangouts, e-mail remains a critical service in virtually all organizations. An e-mail message, however, is of no use unless it can actually be sent somewhere. This is where *Simple Mail Transfer Protocol (SMTP)* comes in. In e-mail clients, SMTP works as a message transfer agent, as shown in Figure 13-9, and moves the message from the user's computer to the mail server when the user clicks the Send button. SMTP also functions as a message transfer protocol between e-mail servers. Lastly, SMTP is a message-exchange addressing standard, and most people are used to seeing its familiar addressing scheme: something@somewhere.com.

Many times, a message needs to travel throughout the Internet and through different mail servers before it arrives at its destination mail server. SMTP is the protocol that carries this message, and it works on top of TCP because it is a reliable protocol and provides sequencing and acknowledgments to ensure the e-mail message arrived successfully at its destination.

The user's e-mail client must be SMTP-compliant to be properly configured to use this protocol. The e-mail client provides an interface to the user so the user can create and modify messages as needed, and then the client passes the message off to the SMTP application layer protocol. So, to use the analogy of sending a letter via the post office, the e-mail client is the typewriter that a person uses to write the message, SMTP is the mail courier who picks up the mail and delivers it to the post office, and the post office is the mail server. The mail server has the responsibility of understanding where the message is heading and properly routing the message to that destination.

It is worth noting that basic SMTP doesn't include any security controls. This is why the IETF published Extended SMTP (ESMTP), which, among other features, allows servers to negotiate a TLS session in which to exchange the messages. This implementation, referred to as SMTP Secure (SMTPS), can provide authentication, confidentiality, and integrity protections for mail transfers.

The mail server is often referred to as an SMTP server. The most common SMTP server software in the world is Exim, which is an open-source mail transfer agent (MTA). SMTP works closely with two mail server protocols, POP and IMAP, which are explained in the following sections.

E-mail Threats

E-mail spoofing is a technique used by malicious users to forge an e-mail to make it appear to be from a legitimate source. Usually, such e-mails appear to be from known and trusted e-mail addresses when they are actually generated from a malicious source. This technique is widely used by attackers these days for spamming and phishing purposes. An attacker tries to acquire the target's sensitive information, such as username and password or bank account credentials. Sometimes, the e-mail messages contain a link of a known website when it is actually a fake website used to trick the user into revealing his information.

E-mail spoofing is done by modifying the fields of e-mail headers, such as the From, Return-Path, and Reply-To fields, so the e-mail appears to be from a trusted source. This results in an e-mail looking as though it is from a known e-mail address. Mostly the From field is spoofed, but some scams have modified the Reply-To field to the attacker's e-mail address. E-mail spoofing is caused by the lack of security features in SMTP. When SMTP technologies were developed, the concept of e-mail spoofing didn't exist, so countermeasures for this type of threat were not embedded into the protocol. A user could use an SMTP server to send e-mail to anyone from any e-mail address. We'll circle back to these threats when we describe e-mail security later in this section.

POP

Post Office Protocol (POP) is an Internet mail server protocol that supports incoming and outgoing messages. The current version is 3, so you'll also see it referred to as POP3. A mail server that uses POP, apart from storing and forwarding e-mail messages, works with SMTP to move messages between mail servers. By default, POP servers listen on TCP port 110.

A smaller organization may have only one POP server that holds all employee mailboxes, whereas larger organizations could have several POP servers, one for each department within the organization. There are also Internet POP servers that enable people all over the world to exchange messages. This system is useful because the messages are held on the mail server until users are ready to download their messages, instead of trying to push messages right to a person's computer, which may be down or offline.

The e-mail server can implement different authentication schemes to ensure an individual is authorized to access a particular mailbox, but this is usually handled through usernames and passwords. Connections to these clients can be encrypted using TLS by using the secure version of POP, known as POP3S, which typically listens on port 995.

IMAP

Internet Message Access Protocol (IMAP) is also an Internet protocol that enables users to access mail on a mail server (the default TCP port is 143). IMAP provides all the functionalities of POP, but has more capabilities. If a user is using POP, when he accesses his mail server to see if he has received any new messages, all messages are automatically downloaded to his computer. Once the messages are downloaded from the POP server, they are usually deleted from that server, depending upon the configuration. POP can cause frustration for mobile users because the messages are automatically pushed down to their computer or device and they may not have the necessary space to hold all the messages. This is especially true for mobile devices that can be used to access e-mail servers. This is also inconvenient for people checking their mail on other people's computers. If Christina checks her e-mail on Jessica's computer, all of Christina's new mail could be downloaded to Jessica's computer.

If a user uses IMAP instead of POP, she can download all the messages or leave them on the mail server within her remote message folder, referred to as a mailbox. The user can also manipulate the messages within this mailbox on the mail server as if the messages resided on her local computer. She can create or delete messages, search for specific messages, and set and clear flags. This gives the user much more freedom and keeps the messages in a central repository until the user specifically chooses to download all messages from the mail server.

IMAP is a store-and-forward mail server protocol that is considered POP's successor. IMAP also gives administrators more capabilities when it comes to administering and maintaining the users' messages. Just like SMTP and POP, IMAP can run over TLS, in which case the server listens for connections on TCP port 993.

E-mail Authorization

POP has the capability to integrate *Simple Authentication and Security Layer (SASL)*, a protocol-independent framework for performing authentication. This means that any protocol that knows how to interact with SASL can use its various authentication mechanisms without having to actually embed the authentication mechanisms within its code.

To use SASL, a protocol includes a command for identifying and authenticating a user to an authentication server and for optionally negotiating protection of subsequent protocol interactions. If its use is negotiated, a security layer is inserted between the protocol and the connection. The data security layer can provide data integrity, data confidentiality, and other services. SASL's design is intended to allow new protocols to reuse existing mechanisms without requiring redesign of the mechanisms and allows existing protocols to make use of new mechanisms without redesign of protocols.

The use of SASL is not unique just to POP; other protocols, such as IMAP, Internet Relay Chat (IRC), Lightweight Directory Access Protocol (LDAP), and SMTP, can also use SASL and its functionality.

Sender Policy Framework

A common way to deal with the problem of forged e-mail messages is by using *Sender Policy Framework (SPF)*, which is an e-mail validation system designed to prevent e-mail spam by detecting e-mail spoofing by verifying the sender's IP address. SPF allows administrators to specify which hosts are allowed to send e-mail from a given domain by creating a specific SPF record in DNS. Mail exchanges use DNS to check that mail from a given domain is being sent by a host sanctioned by that domain's administrators.

DomainKeys Identified Mail

We can also leverage public key infrastructure (PKI) to validate the origin and integrity of each message. The *DomainKeys Identified Mail (DKIM)* standard, codified in RFC 6376, allows e-mail servers to digitally sign messages to provide a measure of confidence for the receiving server that the message is from the domain it claims to be from. These digital signatures are normally invisible to the user and are just used by the servers sending and receiving the messages. When a DKIM-signed message is received, the server requests the sending domain's certificate through DNS and verifies the signature. As long as the private key is not compromised, the receiving server is assured that the message came from the domain it claims and that it has not been altered in transit.

Domain-Based Message Authentication

SPF and DKIM were brought together to define the Domain-based Message Authentication, Reporting and Conformance (DMARC) system. DMARC, which today is estimated to protect 80 percent of mailboxes worldwide, defines how domains communicate to the rest of the world whether they are using SPF or DKIM (or both). It also codifies the mechanisms by which receiving servers provide feedback to the senders on the results of their validation of individual messages. Despite significant advances in securing e-mail, phishing e-mail remains one of the most common and effective attack vectors.

Secure/Multipurpose Internet Mail Extensions

Multipurpose Internet Mail Extensions (MIME) is a technical specification indicating how multimedia data and e-mail binary attachments are to be transferred. The Internet has mail standards that dictate how mail is to be formatted, encapsulated, transmitted, and opened. If a message or document contains a binary attachment, MIME dictates how that portion of the message should be handled.

When an attachment contains an audio clip, graphic, or some other type of multimedia component, the e-mail client sends the file with a header that describes the file type. For example, the header might indicate that the MIME type is Image and that the subtype is jpeg. Although this information is in the header, many times, systems also use the file's extension to identify the MIME type. So, in the preceding example, the file's name might be stuff.jpeg. The user's system sees the extension .jpeg, or sees the data in the header field, and looks in its association list to see what program it needs to initialize to open this particular file. If the system has JPEG files associated with the Explorer application, then Explorer opens and presents the image to the user.

Sometimes systems either do not have an association for a specific file type or do not have the helper program necessary to review and use the contents of the file. When a file has an unassociated icon assigned to it, it might require the user to choose the Open With command and choose an application in the list to associate this file with that program. So when the user double-clicks that file, the associated program initializes and presents the file. If the system does not have the necessary program, the website might offer the necessary helper program, like Acrobat or an audio program that plays WAV files.

MIME is a specification that dictates how certain file types should be transmitted and handled. This specification has several types and subtypes, enables different computers to exchange data in varying formats, and provides a standardized way of presenting the data. So if Sean views a funny picture that is in GIF format, he can be sure that when he sends it to Debbie, it will look exactly the same.

Secure MIME (S/MIME) is a standard for encrypting and digitally signing e-mail and for providing secure data transmissions. S/MIME extends the MIME standard by providing support for the encryption of e-mail and attachments. The encryption and hashing algorithms can be specified by the user of the mail application, instead of having it dictated to them. S/MIME follows the Public Key Cryptography Standards (PKCS). It provides confidentiality through encryption algorithms, integrity through hashing algorithms, authentication through the use of X.509 public key certificates, and nonrepudiation through cryptographically signed message digests.

Multilayer Protocols

Not all protocols fit neatly within the layers of the OSI model. This is particularly evident among devices and networks that were never intended to interoperate with the Internet. For this same reason, they tend to lack robust security features aimed at protecting the availability, integrity, and confidentiality of the data they communicate. The problem is that as the Internet of old becomes the Internet of Things (IoT), these previously isolated devices and networks find themselves increasingly connected to a host of threats they were never meant to face.

As security professionals, we need to be aware of these nontraditional protocols and their implications for the security of the networks to which they are connected. In particular, we should be vigilant when it comes to identifying nonobvious cyber-physical systems. In December 2015, attackers were able to cut power to over 80,000 homes in Ukraine apparently by compromising the utilities' supervisory control and data acquisition (SCADA) systems in what is considered the first known blackout caused by a cyberattack. A few years later, in 2017, attackers were able to exploit a previously unknown vulnerability and reprogram a Schneider Electric safety instrumented system (SIS) at an undisclosed target, causing the facility to shut down. At the heart of most SCADA systems used by power and water utilities is a multilayer protocol known as DNP3.

Distributed Network Protocol 3

The *Distributed Network Protocol 3 (DNP3)* is a communications protocol designed for use in SCADA systems, particularly those within the power sector. It is not a general-purpose protocol like IP, nor does it incorporate routing functionality. SCADA systems typically have a very flat hierarchical architecture in which sensors and actuators are connected to remote terminal units (RTUs). The RTUs aggregate data from one or more of these devices and relay it to the SCADA master, which includes a human–machine interface (HMI) component. Control instructions and configuration changes are sent from the SCADA master to the RTUs and then on to the sensors and actuators.

At the time DNP3 was designed, there wasn't a need to route traffic among the components (most of which were connected with point-to-point circuits), so networking was not needed or supported in DNP3. Instead of using the OSI seven-layer model, its developers opted for a simpler three-layer model called the Enhanced Performance Architecture (EPA) that roughly corresponds to layers 2, 4, and 7 of the OSI model. There was no encryption or authentication, since the developers did not think network attacks were feasible on a system consisting of devices connected to each other and to nothing else.

Over time, SCADA systems were connected to other networks and then to the Internet for a variety of very valid business reasons. Unfortunately, security wasn't considered until much later. Encryption and authentication features were added as an afterthought, though not all implementations have been thus updated. Network segmentation is not always present either, even in some critical installations. Perhaps most concerning is the shortage of effective IPSs and IDSs that understand the interconnections between DNP3 and IP networks and can identify DNP3-based attacks.

Controller Area Network Bus

Another multilayer protocol that had almost no security features until very recently is the one that runs most automobiles worldwide. The *Controller Area Network (CAN) bus* is a protocol designed to allow microcontrollers and other embedded devices to communicate with each other on a shared bus. Over time, these devices have diversified so that today they can control almost every aspect of a vehicle's functions, including steering, braking, and throttling. CAN bus was never meant to communicate with anything outside the vehicle except for a mechanic's maintenance computer, so there never appeared to be a need for security features.

As automobiles started getting connected via Wi-Fi and cellular data networks, their designers didn't fully consider the new attack vectors this would introduce to an otherwise undefended system. That is, until Charlie Miller and Chris Valasek famously hacked a Jeep in 2015 by connecting to it over a cellular data network and bridging the head unit (which controls the sound system and GPS) to the CAN bus (which controls all the vehicle sensors and actuators) and causing it to run off a road. As automobiles become more autonomous, security of the CAN bus becomes increasingly important.

Modbus

Like CAN bus, the Modbus system was developed to prioritize functionality over security. A communications system created in the late 1970s by Modicon, now Schneider Electric, Modbus enables communications among SCADA devices quickly and easily. Since its inception, Modbus has quickly become the de facto standard for communications between programmable logic controllers (PLCs). But as security was not built in, Modbus offers little protection against attacks. An attacker residing on the network can simply collect traffic using a tool like Wireshark, find a target device, and issue commands directly to the device.

Converged Protocols

Converged protocols are those that started off independent and distinct from one another but over time converged to become one. How is this possible? Think about the phone and data networks. Once upon a time, these were two different entities and each had its own protocols and transmission media. For a while, in the 1990s, data networks sometimes rode over voice networks using data modems. This was less than ideal, which is why we flipped it around and started using data networks as the carrier for voice communications. Over time, the voice protocols converged onto the data protocols, which paved the way for Voice over IP (VoIP).

IP convergence, which addresses a specific type of converged protocols, is the transition of services from disparate transport media and protocols to IP. It is not hard to see that IP has emerged as the dominant standard for networking, so it makes sense that any new protocols would leverage this existing infrastructure rather than create a separate one.

Technically, the term *converged* implies that the two protocols became one. Oftentimes, however, the term is used to describe cases in which one protocol was originally independent of another but over time started being encapsulated (or tunneled) within that other one.

Encapsulation

We already saw (in Chapter 9) how encapsulation enables the transmission of data down the seven layers of the OSI reference model. We came across encapsulation again earlier in this chapter when we discussed techniques to tunnel (or encapsulate) one protocol's traffic inside some other protocol. The next two sections describe two more examples. It should be obvious that encapsulation can be helpful in architecting our networks, but it can also have significant security implications.

When we covered DNS tunneling, we saw another, less helpful application of encapsulation. Threat actors develop their own protocols for controlling compromised hosts and they can encapsulate those protocols within legitimate systems. It is important, therefore, to not assume that just because we have a network link that should be transporting data of a certain protocol, it won't have something else embedded in it. Whether encapsulation is malicious or benign, the point is that we need to be aware of what traffic should be where and have the means to inspect it to ensure we are not surprised.

Fiber Channel over Ethernet

Fibre Channel (FC) (also called Fiber Channel in the United States) was developed by the American National Standards Institute (ANSI) in 1988 as a way to connect supercomputers using optical fibers. FC is now used to connect servers to data storage devices in data centers and other high-performance environments. One of its best features is that it can support speeds of up to 128 Gbps over distances of up to 500 meters. (Distances of up to 50 km are possible at lower data rates.) While the speed and other features of FC are pretty awesome for data centers and storage area network (SAN) applications, the need to maintain both Ethernet and fiber-optic cabling adds costs and complexity to its use in enterprise environments.

Fibre Channel over Ethernet (FCoE) is a protocol encapsulation that allows FC frames to ride over Ethernet networks. Its use allows data centers to be almost exclusively wired using Ethernet cabling. It is important to note, however, that FCoE rides on top of Ethernet and is, therefore, a non-routable protocol. It is only intended for LAN environments where devices are in close proximity to each other and efficiency is essential.

Internet Small Computer Systems Interface

A much different approach to encapsulation is exemplified by the Internet Small Computer Systems Interface (iSCSI), which encapsulates SCSI data in TCP segments. SCSI is a set of technologies that allows peripherals to be connected to computers. The problem with the original SCSI is that it has limited range, which means that connecting a remote peripheral (e.g., camera or storage device) is not normally possible. The solution was to let SCSI ride on TCP segments so that a peripheral device could be literally anywhere in the world and still appear as local to a computer.

Network Segmentation

Once upon a time, networks were flat (i.e., almost everyone within an organization was in the same layer 2 broadcast domain) so that everyone could easily communicate with everyone else inside the "trusted" perimeter. Network defenses were mostly (sometimes solely) outward-facing. This led to the networks that were "crunchy on the outside but soft and chewy on the inside." Believe it or not, this was the design mantra for many organizations for many years. Eventually, they realized that this design was a really bad idea. For starters, they recognized that at least some attackers will get through their perimeter defenses. Also, they learned that insider threats could be just as dangerous as external ones, and these insiders would have no problem moving through the soft and chewy interior network. Furthermore, they realized that most networks no longer have a neat concept of "inside" and "outside." Instead, organizations increasingly rely on external systems such as those provided by cloud service providers.

Network segmentation is the practice of dividing networks into smaller subnetworks. An example is to divide the network by department, so that the finance department and marketing department are each in their own LAN. If they need to communicate directly, they have to go through a gateway (e.g., a router or firewall) that allows network administrators to block or detect suspicious traffic. This is a classic implementation of the zero trust security design principle.

The decision to segment a network begs a couple of questions. How many subnetworks should we have? Are more subnets better? There really is no one-size-fits-all answer, but generally, the smaller the subnetworks (and the more you have), the better. In fact, many organizations are implementing *micro-segmentation*, which is the practice of isolating individual assets (e.g., data servers) in their own protected network environment. Think of it as a subnet where the only devices are the protected asset and a security gateway.

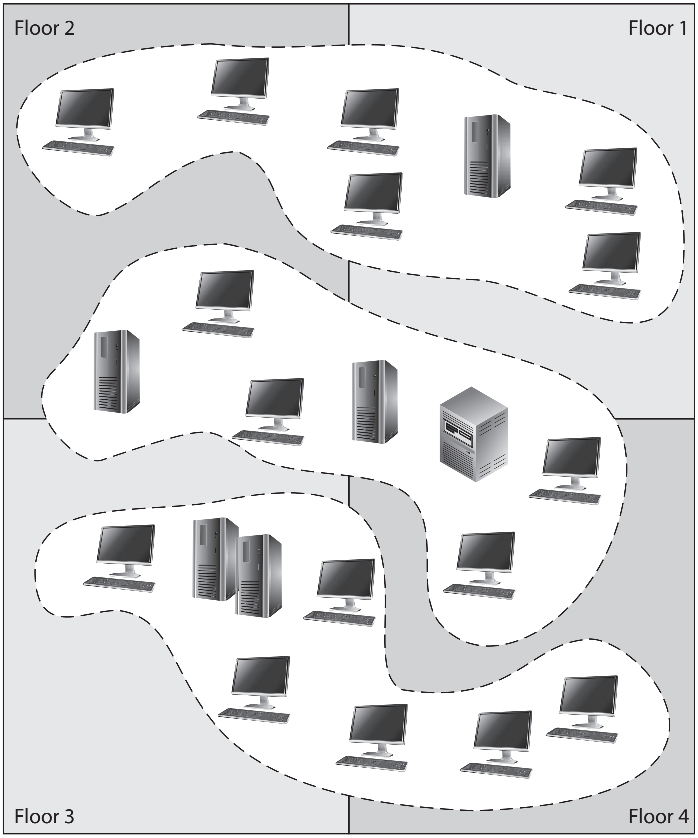
So, how do we go about segmenting networks? We can do it either physically (using devices like switches and routers) or logically (using virtualization software). We'll cover devices in detail in the next chapter, so let's turn our attention to the most important technologies that enable segmentation and micro-segmentation.

VLANs

One of the most commonly used technologies used to segment LANs is the *virtual local area network (VLAN)*. A LAN can be defined as a set of devices on the same layer 2 (data link layer) broadcast domain. This typically means hosts that are *physically* connected to the same layer 2 switches. A VLAN is a set of devices that *behave* as though they were all directly connected to the same switch, when in fact they aren't. This allows you to, for instance, ensure that all members of the finance team are on the same (virtual) LAN, even though they are scattered across multiple countries. The ability to segment networks of users in this manner is critical for both functional and security reasons.

Virtually all modern enterprise-grade switches have the capability to use VLANs. VLANs enable administrators to separate and group computers logically based on resource requirements, security, or business needs instead of the standard physical location of the systems. When repeaters, bridges, and routers are used, systems and resources are grouped in a manner dictated by their physical location. Figure 13-10 shows how computers that are physically located next to each other can be grouped logically into different VLANs. Administrators can form these groups based on the users' and organization's needs instead of the physical location of systems and resources.

Figure 13-10 VLANs enable administrators to manage logical networks.

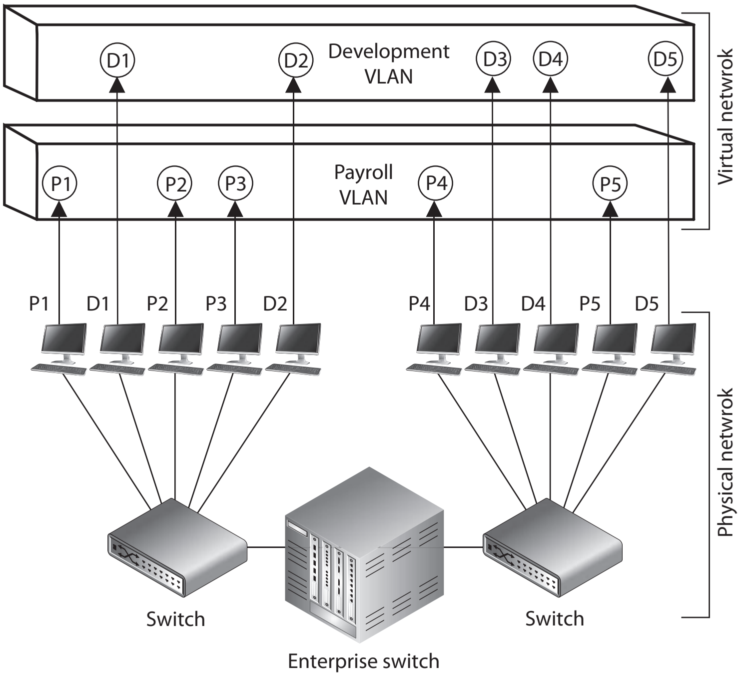


An administrator may want to segment the computers of all users in the marketing department in the same VLAN network, for example, so all users receive the same broadcast messages and can access the same types of resources. This arrangement could get tricky if a few of the users are located in another building or on another floor, but VLANs provide the administrator with this type of flexibility. VLANs also enable an administrator to apply particular security policies to respective zones or segments. This way, if tighter security is required for the payroll department, for example, the administrator can develop a policy, add all payroll systems to a specific VLAN, and apply the security policy only to the payroll VLAN.

A VLAN exists on top of the physical network, as shown in Figure 13-11. Each Ethernet frame is prepended with a VLAN identifier (VID), which is a 12-bit field. This means that we can define up to 4,095 VLANs in the same network. (The first and last VID values are reserved.) If workstation P1 wants to communicate with workstation D1, the message has to be routed—even though the workstations are physically next to each other—because they are on different logical networks.

NOTE The IEEE standard that defines how VLANs are to be constructed and how tagging should take place to allow for interoperability is IEEE 802.1Q.

Figure 13-11 VLANs exist on a higher level than the physical network and are not bound to it.



While VLANs are used to segment traffic, attackers can still gain access to traffic that is supposed to be "walled off" in another VLAN segment. *VLAN hopping attacks* allow attackers to gain access to traffic in various VLAN segments. An attacker can have a system act as though it is a switch. The system understands the tagging values being used in the network and the trunking protocols and can insert itself between other VLAN devices and gain access to the traffic going back and forth. This is called a *switch spoofing attack*. An attacker can also insert VLAN tags to manipulate the control of traffic at the data link layer in what is known as a *double tagging attack*. Proper configuration of all switches mitigates VLAN hopping attacks.

Virtual eXtensible Local Area Network

VLANs, however, have some significant limitations. For starters, remember that you're limited to around 4,000 VLANs because the VID is 12 bits. While this sounds like a lot, it really isn't if you happen to be a cloud-based service provider supporting hundreds of customers. Another challenge is that VLANs are layer 2 constructs separated by layer 3 routers. This means that all the hosts on a given VLAN must be on the same port of the same router. In other words, if the hosts are in different countries, it becomes really hard to join them to the same VLAN.

The *Virtual eXtensible Local Area Network (VxLAN)* is a network virtualization technology that encapsulates layer 2 frames onto UDP (layer 4) datagrams for distribution anywhere in the world. Whereas VLANs have VIDs, VxLANs have a virtual network identifier (VNI) that is 24 bits long, which gives us over 16 million segments. VxLANs are mostly used in cloud environments where hosts and networks are virtualized.

VxLANs are overlay networks on top of UDP/IP underlay networks. Each network switch or router that is part of a VxLAN has a *virtual tunnel end point (VTEP)* that provides the interface between the underlay and overlay networks. When a VTEP receives a frame, it establishes a virtual tunnel on the overlay network connecting it to the destination VTEP just long enough to deliver the frame. The VTEP encapsulates this overlay frame in UDP datagrams that are then passed to the underlay network for delivery.

Software-Defined Networks

Software-defined networking (SDN) is an approach to networking that relies on distributed software to provide unprecedented agility and efficiency. Using SDN, it becomes much easier to dynamically route traffic to and from newly provisioned services and platforms. This means a new server can be quickly provisioned using a cloud service provider in response to a spike in service requests and the underlying network can just as quickly adapt to the new traffic patterns. It also means that a service or platform can be quickly moved from one location to another and the SDN will just as quickly update traffic-flow rules in response to this change. Unsurprisingly, the three biggest drivers to the adoption of SDN are the growth in cloud computing, big data, and mobile computing.

How does SDN differ from traditional networking? Whereas traditional networking relies on network devices that coordinate with one another in a mostly decentralized manner, SDN centralizes the configuration and control of devices. In a decentralized environment, it takes time for routers to converge onto (or agree on) good routes. These devices must normally be manually configured whenever any changes take place, which is also a time-consuming task. In SDN, on the other hand, all changes are pushed out to the devices either reactively (i.e., in response to requests from the devices) or proactively (i.e., because the admins know a change is being made, such as the addition of 100 servers). Because it is centrally controlled, the SDN approach allows traffic to be routed much more efficiently and securely. Perhaps the most important element of SDN is the abstraction of control and forwarding planes.

Control and Forwarding Planes

The *control plane* is where the internetwork routing decisions are being made. Think of this as the part of your router that runs the routing protocol, such as Open Shortest Path First (OSPF). (The analogy is not perfect, but it is useful for now.) The control plane is responsible for discovering the topology of neighboring networks and maintaining a table of routes for outbound packets. Since most networks are pretty dynamic places in which congestion along different routes is always changing, the control plane is a pretty dynamic place as well. New routes are routinely being discovered, just as old routes are dropped or at least flagged as slow or expensive. As you can see, the control plane is mostly interested in effects that are more than one hop away.

The *forwarding plane*, by contrast, is where traffic forwarding decisions are made. Think of this as that part of your router that decides (very quickly) that a packet received on network interface eth0 needs to be forwarded to network interface eth3. How does the forwarding plane decide this? By using the products developed by the control plane. The control plane is the strategic, methodical planner of traffic routing, while the forwarding plane is the tactical, fast executioner of those plans. Unsurprisingly, the forwarding plane is typically implemented in hardware such as an application-specific integrated chip (ASIC).

NOTE Because traditional routing decisions are made by the controller in an SDN architecture, the network devices behave (and are referred to) as switches.

In a traditional network architecture, each networking device has its own control plane and its own forwarding plane, both of which run on some sort of proprietary operating system (e.g., Cisco IOS). The normal way of reconfiguring these traditional devices is via a terminal connection of some sort. This means that an administrator must remotely log into each device in order to change its configuration. Let's suppose that we want to support a distinct QoS for a new user. In order to do this, we'd modify the configuration in each networking device that would be involved in providing services to this user. Even assuming that we are able to do this without making any mistakes, we still face the onerous task of manually changing these parameters whenever the terms of the contract change, or when equipment is replaced or upgraded, or when the network architecture changes. There are exceptions to these challenges, of course, but the point is that making frequent, granular configuration changes is tough.

What About Automation?

One of the challenges of network administration is that most network devices (apart from those that support SDN) do not have comprehensive mechanisms for programmatically and remotely changing the configuration of the device. This is why administrators have to manually log into each device and update the configuration. Reading information is easier because these devices typically support SNMP, but writing meaningful changes to the devices almost always requires manual interaction or some third-party tool that comes with its own set of constraints.

Further complicating the issue of making dynamic changes, vendors typically use their own proprietary operating system, which makes it harder to write a script that makes the same changes to all devices in heterogeneous environments that implement products from multiple vendors. This is the reason why many organizations implement homogeneous network architectures in which all the devices are manufactured by the same vendor. A big downside of this homogeneity is that it leads to vendor lockdown because it is hard (and expensive) to change vendors when that means you must change every single device on your network. Furthermore, homogeneity is bad for security, because an exploit that leverages a vulnerability in a network operating system will likely affect every device in a homogeneous network.

In SDN, by contrast, the control plane is implemented in a central node that is responsible for managing all the devices in the network. For redundancy and efficiency, this node can actually be a federation of nodes that coordinate their activities with one another. The network devices are then left to do what they do best: forward packets very efficiently. So the forwarding plane lives in the network devices and the control plane lives in a centralized SDN controller. This allows us to abstract the network devices (heterogeneous or otherwise) from the applications that rely on them to communicate in much the same way Windows abstracts the hardware details from the applications running on a workstation.

Approaches to SDN

The concept of network abstraction is central to all implementations of SDN. The manner in which this abstraction is implemented, however, varies significantly among flavors of SDN. There are at least three common approaches to SDN, each championed by a different community and delivered primarily through a specific technology:

- **Open** The SDN approach championed by the Open Networking Foundation (ONF) (https://opennetworking.org) is, by most accounts, the most common. It relies on open-source code and standards to develop the building blocks of an SDN solution. The controller communicates with the switches using OpenFlow, a standardized, open-source communications interface between controllers and network devices in an SDN architecture. OpenFlow allows the devices implementing the forwarding plane to provide information (such as utilization data) to the controller, while allowing the controller to update the flow tables (akin to traditional routing tables) on the devices. Applications communicate with the controller using the RESTful or Java APIs.

- **API** Another approach to SDN, and one that is championed by Cisco, is built on the premise that OpenFlow is not sufficient to fully leverage the promise of SDN in the enterprise. In addition to OpenFlow, this approach leverages a rich API on proprietary switches that allows greater control over traffic in an SDN. Among the perceived shortcomings that are corrected are the inability of OpenFlow to do deep packet inspection and manipulation and its reliance on a centralized control plane. This proprietary API approach to SDN is seen as enriching rather than replacing ONF's SDN approach.
- **Overlays** Finally, one can imagine a virtualized network architecture as an overlay on a traditional one. In this approach, we virtualize all network nodes, including switches, routers, and servers, and treat them independently of the physical networks upon which this virtualized infrastructure exists. The SDN exists simply as a virtual overlay on top of a physical (underlay) network.

Software-Defined Wide Area Network

Software-defined wide area networking (SD-WAN) is the use of software (instead of hardware) to control the connectivity, management, and services between distant sites. Think of it as SDN applied to WANs instead of LANs. Similarly to SDN, SD-WAN separates the control plane from the forwarding plane. This means that network links, whether they are leased lines or 5G wireless, are better utilized. Also, since the control plane is centralized, security policies can be consistently applied throughout.

Another advantage of SD-WANs is that they are application-aware, meaning they know the difference between supporting video conferencing (low latency, loss tolerance), supporting file transfers (latency tolerance, loss intolerant), or supporting any other sort of traffic. This means SD-WANs use the right path for the traffic and are able to switch things around as links become congested or degraded.

Chapter Review

Securing our networks is a lot more effective if we first understand the underlying technologies and then apply secure design principles to their selection and integration. This chapter built on the foundations of the previous two chapters to show common approaches to building and operating secure networking architectures. We focused our attention on network encryption and service security techniques but also covered how to deal with dispersed networks and those with cloud service components. A key aspect of our discussion was the application of the secure design principles at multiple points. We'll continue this theme in the next chapter as we talk about securing the components of our networks.

Quick Review

- Link encryption encrypts all the data along a specific communication path.
- End-to-end encryption (E2EE) occurs at the session layer (or higher) and does not encrypt routing information, enabling attackers to learn more about a captured packet and where it is headed.
- Transport Layer Security (TLS) is an E2EE protocol that provides confidentiality and data integrity for network communications.
- Secure Sockets Layer (SSL) is the predecessor of TLS and is deprecated and considered insecure.
- A virtual private network (VPN) is a secure, private connection through an untrusted network.
- The Point-to-Point Tunneling Protocol (PPTP) is an obsolete and insecure means of providing VPNs.
- The Layer 2 Tunneling Protocol (L2TP) tunnels PPP traffic over various network types (IP, ATM, X.25, etc.) but does not encrypt the user traffic.
- Internet Protocol Security (IPSec) is a suite of protocols that provides authentication, integrity, and confidentiality protections to data at the network layer.
- TLS can be used to provide VPN connectivity at layer 5 in the OSI model.
- A web service is client/server system in which clients and servers communicate using HTTP over a network such as the Internet.
- A service-oriented architecture (SOA) describes a system as a set of interconnected but self-contained components that communicate with each other and with their clients through standardized protocols.
- Application programming interfaces (APIs) establish a "language" that enables a system component to make a request from another component and then interpret that second component's response.
- The Hypertext Transfer Protocol (HTTP) is a TCP/IP-based communications protocol used for transferring data between a server and a client in a connectionless and stateless manner.
- A uniform resource identifier (URI) uniquely identifies a resource on the Internet.
- HTTP Secure (HTTPS) is HTTP running over TLS.
- The Simple Object Access Protocol (SOAP) is a messaging protocol that uses XML over HTTP to enable clients to invoke processes on a remote host in a platform-agnostic way.
- SOAP security is enabled by a set of protocol extensions called the Web Services Security (WS-Security or WSS) specification, which provides message confidentiality, integrity, and authentication.

- Representational State Transfer (REST) is an architectural pattern used to develop web services without using SOAP.
- A domain generation algorithm (DGA) produces seemingly random domain names in a way that is predictable by anyone who knows the algorithm.
- DNS tunneling is the practice of encoding messages in one or a series of DNS queries or responses for exfiltrating or infiltrating data into an environment.
- DNS reflection attacks involve sending a query to a server while spoofing the source address to be that of the intended target.
- A DNS amplification attack is characterized by small queries that result in very much larger responses.
- Domain Name System Security Extensions (DNSSEC) is a set of IETF standards that ensures the integrity of DNS records but not their confidentiality or availability.
- DNS over HTTPS (DoH) is a (yet to be ratified) approach to protecting the privacy and confidentiality of DNS queries by sending them over HTTPS/TCP /IP instead of unsecured UDP/IP.
- E-mail spoofing is a technique used by malicious users to forge an e-mail to make it appear to be from a legitimate source.
- Simple Authentication and Security Layer (SASL) is a protocol-independent framework for performing authentication that is typically used in POP3 e-mail systems.
- The Sender Policy Framework (SPF) is an e-mail validation system designed to prevent e-mail spam by detecting e-mail spoofing by verifying the sender's IP address.
- The DomainKeys Identified Mail (DKIM) standard allows e-mail servers to digitally sign messages to provide a measure of confidence for the receiving server that the message is from the domain it claims to be from.
- Domain-based Message Authentication, Reporting and Conformance (DMARC) systems incorporate both SPF and DKIM to protect e-mail.
- Secure MIME (S/MIME) is a standard for encrypting and digitally signing e-mail and for providing secure data transmissions.
- The Distributed Network Protocol 3 (DNP3) is a multilayer communications protocol designed for use in SCADA systems, particularly those within the power sector.
- The Controller Area Network (CAN) bus is a multilayer protocol designed to allow microcontrollers and other embedded devices to communicate with each other on a shared bus.
- Converged protocols are those that started off independent and distinct from one another but over time converged to become one.

- Fibre Channel over Ethernet (FCoE) is a protocol encapsulation that allows Fibre Channel (FC) frames to ride over Ethernet networks.
- The Internet Small Computer Systems Interface (iSCSI) protocol encapsulates SCSI data in TCP segments so that computer peripherals could be located at any physical distance from the computer they support.
- Network segmentation is the practice of dividing networks into smaller subnetworks.
- A virtual LAN (VLAN) is a set of devices that behave as though they were all directly connected to the same switch, when in fact they aren't.
- Virtual eXtensible LAN (VxLAN) is a network virtualization technology that encapsulates layer 2 frames onto UDP (layer 4) datagrams for distribution anywhere in the world.
- Software-defined networking (SDN) is an approach to networking that relies on distributed software to separate the control and forwarding planes of a network.
- Software-defined wide area networking (SD-WAN) is the use of software (instead of hardware) to control the connectivity, management, and services between distant sites in a manner that is similar to SDN but applied to WANs.

Questions

Please remember that these questions are formatted and asked in a certain way for a reason. Keep in mind that the CISSP exam is asking questions at a conceptual level. Questions may not always have the perfect answer, and the candidate is advised against always looking for the perfect answer. Instead, the candidate should look for the best answer in the list.

1. Which of the following provides secure end-to-end encryption?

   A. Transport Layer Security (TLS)

   B. Secure Sockets Layer (SSL)

   C. Layer 2 Tunneling Protocol (L2TP)

   D. Domain Name System Security Extensions (DNSSEC)

2. Which of the following can take place if an attacker is able to insert tagging values into network- and switch-based protocols with the goal of manipulating traffic at the data link layer?

   A. Open relay manipulation

   B. VLAN hopping attack

   C. Hypervisor denial-of-service attack

   D. DNS tunneling

3. Which of the following provides an incorrect definition of the specific component or protocol that makes up IPSec?
   A. Authentication Header protocol provides data integrity, data origin authentication, and protection from replay attacks.
   B. Encapsulating Security Payload protocol provides confidentiality, data origin authentication, and data integrity.
   C. Internet Security Association and Key Management Protocol provides a framework for security association creation and key exchange.
   D. Internet Key Exchange provides authenticated keying material for use with encryption algorithms.
4. Alice wants to send a message to Bob, who is several network hops away from her. What is the best approach to protecting the confidentiality of the message?
   A. PPTP
   B. S/MIME
   C. Link encryption
   D. SSH
5. Which technology would best provide confidentiality to a RESTful web service?
   A. Web Services Security (WS-Security)
   B. Transport Layer Security (TLS)
   C. HTTP Secure (HTTPS)
   D. Simple Object Access Protocol (SOAP)
6. Which of the following protections are provided by Domain Name System Security Extensions (DNSSEC)?
   A. Confidentiality and integrity
   B. Integrity and availability
   C. Integrity and authentication
   D. Confidentiality and authentication
7. Which approach provides the best protection against e-mail spoofing?
   A. Internet Message Access Protocol (IMAP)
   B. Domain-based Message Authentication, Reporting and Conformance (DMARC)
   C. Sender Policy Framework (SPF)
   D. DomainKeys Identified Mail (DKIM)

8. Which of the following is a multilayer protocol developed for use in supervisory control and data acquisition (SCADA) systems?
    A. Controller Area Network (CAN) bus
    B. Simple Authentication and Security Layer (SASL)
    C. Control Plane Protocol (CPP)
    D. Distributed Network Protocol 3 (DNP3)
9. All of the following statements are true of converged protocols *except* which one?
    A. Distributed Network Protocol 3 (DNP3) is a converged protocol.
    B. Fibre Channel over Ethernet (FCoE) is a converged protocol.
    C. IP convergence addresses a specific type of converged protocols.
    D. The term includes certain protocols that are encapsulated within each other.
10. Suppose you work at a large cloud service provider that has thousands of customers around the world. What technology would best support segmentation of your customers' environments?
    A. Virtual local area network (VLAN)
    B. Virtual eXtensible Local Area Network (VxLAN)
    C. Software-defined wide area networking (SD-WAN)
    D. Layer 2 Tunneling Protocol (L2TP)

Answers

1. **A.** TLS and SSL are the only two answers that provide end-to-end encryption, but SSL is insecure, so it's not a good answer.
2. **B.** VLAN hopping attacks allow attackers to gain access to traffic in various VLAN segments. An attacker can have a system act as though it is a switch. The system understands the tagging values being used in the network and the trunking protocols and can insert itself between other VLAN devices and gain access to the traffic going back and forth. Attackers can also insert tagging values to manipulate the control of traffic at this data link layer.
3. **D.** Authentication Header protocol provides data integrity, data origin authentication, and protection from replay attacks. Encapsulating Security Payload protocol provides confidentiality, data origin authentication, and data integrity. Internet Security Association and Key Management Protocol provides a framework for security association creation and key exchange. Internet Key Exchange provides authenticated keying material for use with ISAKMP.
4. **B.** Secure Multipurpose Internet Mail Extensions (S/MIME) is a standard for encrypting and digitally signing e-mail and for providing secure data transmissions using public key infrastructure (PKI).

5. **C.** Either TLS or HTTPS would be a correct answer, but since web services in general and RESTful ones in particular require HTTP, HTTPS is the best choice. Keep in mind that you are likely to come across similar questions where multiple answers are correct but only one is best. SOAP is an alternative way to deliver web services and uses WS-Security for confidentiality.
6. **C.** Domain Name System Security Extensions (DNSSEC) is a set of IETF standards that ensures the integrity and authenticity of DNS records but not their confidentiality or availability.
7. **B.** Domain-based Message Authentication, Reporting and Conformance (DMARC) systems incorporate both SPF and DKIM to protect e-mail. IMAP does not have any built-in protections against e-mail spoofing.
8. **D.** DNP3 is a multilayer communications protocol designed for use in SCADA systems, particularly those within the power sector.
9. **A.** DNP3 is a multilayer communications protocol that was designed for use in SCADA systems and has not converged with other protocols. All other statements are descriptive of converged protocols.
10. **B.** Since there are thousands of customers to support, VxLAN is the best choice because it can support over 16 million subnetworks. Traditional VLANs are capped at just over 4,000 subnetworks, which would not be able to provide more than a few segments to each customer.

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CHAPTER 14

Network Components

This chapter presents the following:

- Transmission media
- Network devices
- Endpoint security
- Content distribution networks

The hacker didn't succeed through sophistication. Rather he poked at obvious places, trying to enter through unlocked doors. Persistence, not wizardry, let him through.

—Clifford Stoll,
The Cuckoo's Egg

In the previous chapter, we covered how to defend our networks. Let's now talk about securing the components of those networks. We need to pay attention to everything from the cables, to the network devices, to the endpoints, because our adversaries will poke at all of it, looking for ways to get in. We (defenders) have to get it right all the time; they (attackers) only need to find that one chink in our armor to compromise our systems. In this chapter, we focus on physical devices. In the next chapter, we'll drill into the software systems that run on them.

Transmission Media

We've already talked a fair bit about the protocols that allow us to move data from point A to point B, but we haven't really covered what actually carries this information. A transmission medium is a physical thing through which data is moved. If we are speaking with each other, our vocal chords create vibrations in the air that we expel from our lungs, in which case the air is the transmission medium. Broadly speaking, we use three different types of transmission media:

- **Electrical wires** Encode information as changes in the voltage level of an electric current. Typically, we use cables, which are two or more wires encased within a sheath.
- **Optical fibers** Transmit data that is encoded in the wavelength (color), phase, or polarization of the light. The light is generated by either an LED or a laser diode. As with electrical wires, we usually bundle multiple fibers into cables for longer distances.

- **Free space** The medium we use for wireless communications, covered in Chapter 12. Any electromagnetic signal can travel through free space even outside our atmosphere. We tend to use mostly radio signals in free space, but every now and then you may encounter a system that uses light, such as infrared laser beams.

Types of Transmission

Physical data transmission can happen in different ways (analog or digital); can use different synchronization schemes (synchronous or asynchronous); can use either one sole channel over a transmission medium (baseband) or several different channels over a transmission medium (broadband); and can take place as electrical voltage, radio waves, or optical signals. These transmission types and their characteristics are described in the following sections.

Analog vs. Digital

A *signal* is just some way of moving information in a physical format from one point to another point. You can signal a message to another person through nodding your head, waving your hand, or giving a wink. Somehow you are transmitting data to that person through your signaling method. In the world of technology, we have specific carrier signals that are in place to move data from one system to another system. The carrier signal is like a horse, which takes a rider (data) from one place to another place. Data can be transmitted through analog or digital signaling formats. If you are moving data through an analog transmission technology (e.g., radio), then the data is represented by the characteristics of the waves that are carrying it. For example, a radio station uses a transmitter to put its data (music) onto a radio wave that travels all the way to your antenna. The information is stripped off by the receiver in your radio and presented to you in its original format—a song. The data is encoded onto the carrier signal and is represented by various amplitude and frequency values, as shown in Figure 14-1.

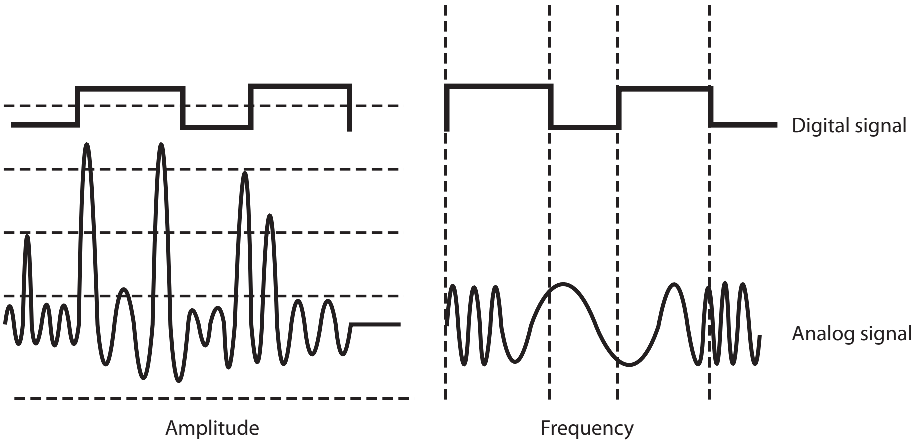

Figure 14-1 Analog signals are measured in amplitude and frequency, whereas digital signals represent binary digits as electrical pulses.

Data being represented in wave values (analog) is different from data being represented in discrete voltage values (digital). As an analogy, compare an analog clock and a digital clock. An analog clock has hands that continuously rotate on the face of the clock. To figure out what time it is, you have to interpret the position of the hands and map their positions to specific values. So you have to know that if the small hand is on the number 1 and the large hand is on the number 6, this actually means 1:30. The individual and specific location of the hands corresponds to a value. A digital clock does not take this much work. You just look at it and it gives you a time value in the format of hour:minutes. There is no mapping work involved with a digital clock because it provides you with data in clear-cut formats.

An analog clock can represent different values as the hands move forward—1:35 and 1 second, 1:35 and 2 seconds, 1:35 and 3 seconds. Each movement of the hands represents a specific value just like the individual data points on a wave in an analog transmission. A digital clock provides discrete values without having to map anything. The same is true with digital transmissions: the values are almost always binary, meaning they are either a 1 or a 0—no need for mapping to find the actual value.

Computers have always worked in a binary manner (1 or 0). When our telecommunication infrastructure was purely analog, each system that needed to communicate over a telecommunication line had to have a modem (modulator/demodulator), which would modulate the digital data into an analog signal. The sending system's modem would modulate the data on to the signal, and the receiving system's modem would demodulate the data off the signal.

Digital signals are more reliable than analog signals over a long distance and provide a clear-cut and efficient signaling method because the voltage is either on (1) or not on (0), compared to interpreting the waves of an analog signal. Extracting digital signals from a noisy carrier is relatively easy. It is difficult to extract analog signals from background noise because the amplitudes and frequencies of the waves slowly lose form. This is because an analog signal could have an infinite number of values or states, whereas a digital signal exists in discrete states. A digital signal is a square wave, which does not have all of the possible values of the different amplitudes and frequencies of an analog signal. Digital systems can implement compression mechanisms to increase data throughput, provide signal integrity through repeaters that "clean up" the transmissions, and multiplex different types of data (voice, data, video) onto the same transmission channel.

Asynchronous vs. Synchronous

Analog and digital transmission technologies deal with the characteristics of the physical carrier on which data is moved from one system to another. Asynchronous and synchronous transmission types are similar to the cadence rules we use for conversation *synchronization*. Asynchronous and synchronous network technologies provide synchronization rules to govern how systems communicate to each other. If you have ever spoken over a satellite phone, you have probably experienced problems with communication synchronization. Commonly, when two people are new to using satellite phones, they do not allow for the necessary delay that satellite communication requires, so they "speak over" one another. Once they figure out the delay in the connection, they resynchronize their timing so that only one person's data (voice) is transmitting at one time, enabling each person to properly understand the full conversation. Proper pauses frame your words in a way to make them understandable.

Synchronization through communication also happens when we write messages to each other. Properly placed commas, periods, and semicolons provide breaks in text so that the person reading the message can better understand the information. If you see "stickwithmekidandyouwillweardiamonds" without the proper punctuation, it is more difficult for you to understand. This is why we have grammar rules. If someone writes a letter to you that starts from the bottom and right side of a piece of paper, and that person does not inform you of this unconventional format, you will not be able to read the message properly, at least initially.

Technological communication protocols also have their own grammar and synchronization rules when it comes to the transmission of data. If two systems are communicating over a network protocol that employs asynchronous timing, they use start and stop bits. The sending system sends a "start" bit, then sends its character, and then sends a "stop" bit. This happens for the whole message. The receiving system knows when a character is starting and stopping; thus, it knows how to interpret each character of the message. This is akin to our previous example of using punctuation marks in written communications to convey pauses. If the systems are communicating over a network protocol that uses synchronous timing, then they don't add start and stop bits. The whole message is sent without artificial breaks, but with a common timing signal that allows the receiver to know how to interpret the information without these bits. This is similar to our satellite phone example in which we use a timing signal (i.e., we count off seconds in our head) to ensure we don't talk over the other person's speech.

If two systems are going to communicate using a synchronous transmission technology, they do not use start and stop bits, but the synchronization of the transfer of data takes place through a timing sequence, which is initiated by a clock pulse.

It is the data link protocol that has the synchronization rules embedded into it. So when a message goes down a system's network stack, if a data link protocol, such as High-level Data Link Control (HDLC), is being used, then a clocking sequence is in place. (The receiving system must also be using this protocol so that it can interpret the data.) If the message is going down a network stack and a protocol such as Asynchronous Transfer Mode (ATM) is at the data link layer, then the message is framed with start and stop indicators.

Data link protocols that employ synchronous timing mechanisms are commonly used in environments that have systems that transfer large amounts of data in a predictable manner (i.e., data center environment). Environments that contain systems that send data in a nonpredictable manner (i.e., Internet connections) commonly have systems with protocols that use asynchronous timing mechanisms.

So, synchronous communication protocols transfer data as a stream of bits instead of framing it in start and stop bits. The synchronization can happen between two systems using a clocking mechanism, or a signal can be encoded into the data stream to let the receiver synchronize with the sender of the message. This synchronization needs to take place before the first message is sent. The sending system can transmit a digital clock pulse to the receiving system, which translates into, "We will start here and work in this type of synchronization scheme." Many modern bulk communication systems,

Asynchronous	Synchronous
Simpler, less costly implementation	More complex, costly implementation
No timing component	Timing component for data transmission synchronization
Parity bits used for error control	Robust error checking, commonly through cyclic redundancy checking (CRC)
Used for irregular transmission patterns	Used for high-speed, high-volume transmissions
Each byte requires three bits of instruction (start, stop, parity)	Minimal protocol overhead compared to asynchronous communication

Table 14-1 Main Differences Between Asynchronous and Synchronous Transmissions

such as high-bandwidth satellite links, use Global Positioning System (GPS) clock signals to synchronize their communications without the need to include a separate channel for timing.

Table 14-1 provides an overview of the differences between asynchronous and synchronous transmissions.

Broadband vs. Baseband

As you read, analog transmission means that data is being moved as waves, and digital transmission means that data is being moved as discrete electric pulses. Synchronous transmission means that two devices control their conversations with a clocking mechanism, and asynchronous means that systems use start and stop bits for communication synchronization. Now let's look at how many individual communication sessions can take place at one time.

A *baseband* technology uses the entire communication channel for its transmission, whereas a *broadband* technology divides the communication channel into individual and independent subchannels so that different types of data can be transmitted simultaneously. Baseband permits only one signal to be transmitted at a time, whereas broadband carries several signals over different subchannels. For example, a coaxial cable TV (CATV) system is a broadband technology that delivers multiple television channels over the same cable. This system can also provide home users with Internet access, but this data is transmitted at a different frequency range than the TV channels.

As an analogy, baseband technology only provides a one-lane highway for data to get from one point to another point. A broadband technology provides a data highway made up of many different lanes, so that not only can more data be moved from one point to another point, but different types of data can travel over the individual lanes.

Any transmission technology that "chops up" one communication channel into multiple channels is considered broadband. The communication channel is usually a specific range of frequencies, and the broadband technology provides delineation between these frequencies and provides techniques on how to modulate the data onto the individual subchannels. To continue with our analogy, we could have one large highway that *could* fit eight individual lanes—but unless we have something that defines

> **How Do These Technologies Work Together?**
> If you are new to networking, it can be hard to understand how the OSI model, analog and digital, synchronous and asynchronous, and baseband and broadband technologies interrelate and differentiate. You can think of the OSI model as a structure to build different languages. If you and Luigi are going to speak to each other in English, you have to follow the rules of this language to be able to understand each other. If you are going to speak French, you still have to follow the rules of that language (OSI model), but the individual letters that make up the words are in a different order. The OSI model is a generic structure that can be used to define many different "languages" for devices to be able to talk to each other. Once you and Luigi agree that you are going to communicate using English, you can *speak* your message to Luigi, and thus your words move over continuous airwaves (analog). Or you can choose to send your message to Luigi through Morse code, which uses individual discrete values (digital). You can send Luigi all of your words with no pauses or punctuation (synchronous) or insert pauses and punctuation (asynchronous). If you are the only one speaking to Luigi at a time, this would be analogous to baseband. If ten people are speaking to Luigi at one time, this would be broadband.

these lanes and have rules for how these lanes are used, this is a baseband connection. If we take the same highway and lay down painted white lines, post traffic signs, add on and off ramps, and establish rules that drivers have to follow, now we are talking about broadband.

A digital subscriber line (DSL) uses one single phone line and constructs a set of high-frequency channels for Internet data transmissions. A cable modem uses the available frequency spectrum that is provided by a cable TV carrier to move Internet traffic to and from a household. Mobile broadband devices implement individual channels over a cellular connection, and Wi-Fi broadband technology moves data to and from an access point over a specified frequency set. The point is that there are different ways of cutting up one channel into subchannels for higher data transfer and that they provide the capability to move different types of traffic at the same time.

Cabling

The different types of transmission techniques we just covered eventually end up being used to send signals over either a cable or free space. We already covered wireless communications in Chapter 12, so let's talk about cabling now.

Electrical signals travel as currents through cables and can be negatively affected by many factors within the environment, such as motors, fluorescent lighting, magnetic forces, and other electrical devices. These items can corrupt the data as it travels through the cable, which is why cable standards are used to indicate cable type, shielding, transmission rates, and maximum distance a particular type of cable can be used.

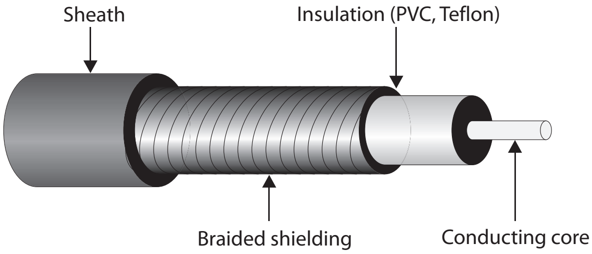

Figure 14-2
Coaxial cable

Coaxial Cable

Coaxial cable has a copper core that is surrounded by a shielding layer and grounding wire, as shown in Figure 14-2. This is all encased within a protective outer jacket. Compared to twisted-pair cable, coaxial cable is more resistant to electromagnetic interference (EMI), provides a higher bandwidth, and supports the use of longer cable lengths. So, why is twisted-pair cable more popular? Twisted-pair cable is cheaper and easier to work with, and the move to switched environments that provide hierarchical wiring schemes has overcome the cable-length issue of twisted-pair cable.

Coaxial cabling is used as a transmission line for radio frequency signals. If you have cable TV, you have coaxial cabling entering your house and the back of your TV. The various TV channels are carried over different radio frequencies. Modems allow us to use some of the "empty" TV frequencies for Internet connectivity.

Twisted-Pair Cable

Twisted-pair cabling has insulated copper wires surrounded by an outer protective jacket. If the cable has an outer foil shielding, it is referred to as *shielded twisted pair (STP)*, which adds protection from radio frequency interference (RFI) and EMI. Twisted-pair cabling, which does not have this extra outer shielding, is called *unshielded twisted pair (UTP)*.

The twisted-pair cable contains copper wires that twist around each other, as shown in Figure 14-3. This twisting of the wires protects the integrity and strength of the signals they carry. Each wire forms a balanced circuit, because the voltage in each pair uses the same amplitude, just with opposite phases. The tighter the twisting of the wires, the more

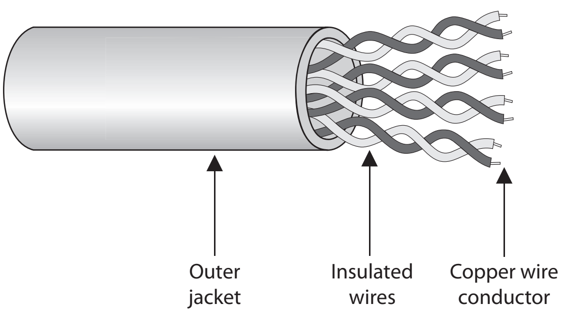

Figure 14-3
Twisted-pair cabling uses copper wires.

UTP Category	Characteristics	Usage
Category 1	Voice-grade telephone cable for up to 1 Mbps transmission rate	No longer in use for data or phones.
Category 2	Data transmission up to 4 Mbps	Historically used in mainframe and minicomputer terminal connections, but no longer in common use.
Category 3	10 Mbps for Ethernet	Used in older 10Base-T network installations and legacy phone lines.
Category 4	16 Mbps	Normally used in Token Ring networks.
Category 5	100 Mbps; two twisted pairs	Sometimes used in legacy 100Base-TX; deprecated in 2001 for data but still used for telephone and video.
Category 5e	1 Gbps; four twisted pairs, providing reduced crosstalk	Widely used in modern networks.
Category 6	1 Gbps, but can support 10 Gbps up to 55 meters	Used in newer network installations requiring high-speed transmission. Standard for Gigabit Ethernet.

Table 14-2 UTP Cable Ratings

resistant the cable is to interference and attenuation. UTP has several categories of cabling, each of which has its own unique characteristics.

The twisting of the wires, the type of insulation used, the quality of the conductive material, and the shielding of the wire determine the rate at which data can be transmitted. The UTP ratings indicate which of these components were used when the cables were manufactured. Some types are more suitable and effective for specific uses and environments. Table 14-2 lists the cable ratings.

Copper cable has been around for many years. It is inexpensive and easy to use. A majority of the telephone systems today use copper cabling with the rating of voice grade. Twisted-pair wiring is the preferred network cabling, but it also has its drawbacks. Copper actually resists the flow of electrons, which causes a signal to degrade after it has traveled a certain distance. This is why cable lengths are recommended for copper cables; if these recommendations are not followed, a network could experience signal loss and data corruption. Copper also radiates energy, which means information can be monitored and captured by intruders. UTP is the least secure networking cable compared to coaxial and fiber. If an organization requires higher speed, higher security, and cables to have longer runs than what is allowed in copper cabling, fiber-optic cable may be a better choice.

Fiber-Optic Cable

Twisted-pair cable and coaxial cable use copper wires as their data transmission media, but fiber-optic cable uses a type of glass that carries light waves, onto which we modulate the data being transmitted. The glass core is surrounded by a protective cladding, which in turn is encased within an outer jacket.

Fiber Components

Fiber-optic cables are made up of a light source, an optical fiber cable, and a light detector.

Light Sources Convert electrical signal into light signal.

- Light-emitting diodes (LEDs)
- Diode lasers

Optical Fiber Cable Data travels as light.

- **Single mode** Small glass core, used for high-speed data transmission over long distances. They are less susceptible to attenuation than multimode fibers.
- **Multimode** Large glass core, able to carry more data than single mode fibers, though they are best for shorter distances because of their higher attenuation levels.

Light Detector Converts light signal back into electrical signal.

Because it uses glass, *fiber-optic* cabling has higher transmission speeds that allow signals to travel over longer distances. Fiber-optic cabling is not as affected by attenuation and EMI when compared to cabling that uses copper. It does not radiate signals, as does UTP cabling, and is difficult to eavesdrop on; therefore, fiber-optic cabling is much more secure than UTP, STP, or coaxial.

Using fiber-optic cable sounds like the way to go, so you might wonder why you would even bother with UTP, STP, or coaxial. Unfortunately, fiber-optic cable is expensive and difficult to work with. It is usually used in backbone networks and environments that require high data transfer rates. Most networks use UTP and connect to a backbone that uses fiber.

NOTE The price of fiber and the cost of installation have been steadily decreasing, while the demand for more bandwidth only increases. More organizations and service providers are installing fiber directly to the end user.

Cabling Problems

Cables are extremely important within networks, and when they experience problems, the whole network could experience problems. This section addresses some of the more common cabling issues many networks experience.

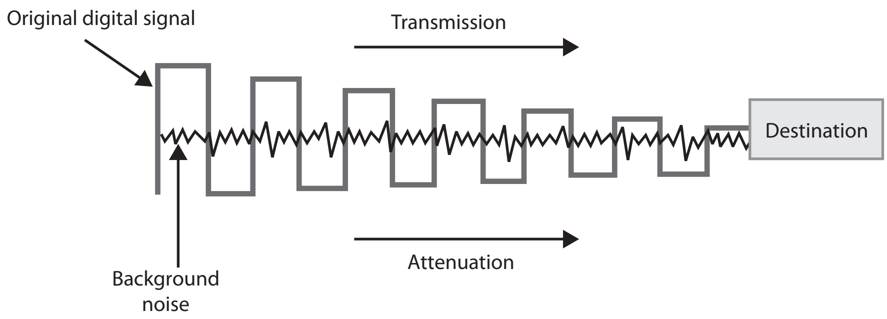

Figure 14-4 Background noise can merge with an electronic signal and alter the signal's integrity.

Noise The term *line noise* refers to random fluctuations in electrical-magnetic impulses that are carried along a physical medium. Noise on a line is usually caused by surrounding devices or by characteristics of the wiring's environment. Noise can be caused by motors, computers, copy machines, fluorescent lighting, and microwave ovens, to name a few. This background noise can combine with the data being transmitted over the cable and distort the signal, as shown in Figure 14-4. The more noise there is interacting with the cable, the more likely the receiving end will not receive the data in the form originally transmitted.

Attenuation *Attenuation* is the loss of signal strength as it travels. This is akin to rolling a ball down the floor; as it travels, air causes resistance that slows it down and eventually stops it. In the case of electricity, the metal in the wire also offers resistance to the flow of electricity. Though some materials such as copper and gold offer very little resistance, it is still there. The longer a wire, the more attenuation occurs, which causes the signal carrying the data to deteriorate. This is why standards include suggested cable-run lengths.

The effects of attenuation increase with higher frequencies; thus, 100Base-TX at 80 MHz has a higher attenuation rate than 10Base-T at 10 MHz. This means that cables used to transmit data at higher frequencies should have shorter cable runs to ensure attenuation does not become an issue.

If a networking cable is too long, attenuation will become a problem. Basically, the data is in the form of electrons, and these electrons have to "swim" through a copper wire. However, this is more like swimming upstream, because there is a lot of resistance on the electrons working in this media. After a certain distance, the electrons start to slow down and their encoding format loses form. If the form gets too degraded, the receiving system cannot interpret the electrons any longer. If a network administrator needs to run a cable longer than its recommended segment length, she needs to insert a repeater or some type of device that amplifies the signal and ensures that it gets to its destination in the right encoding format.

Attenuation can also be caused by cable breaks and malfunctions. This is why cables should be tested. If a cable is suspected of attenuation problems, cable testers can inject signals into the cable and read the results at the end of the cable.

EXAM TIP Most implementations of Ethernet over UTP have a maximum cable length of 100 meters, partly to deal with attenuation.

Crosstalk *Crosstalk* is a phenomenon that occurs when electrical signals of one wire spill over to the signals of another wire. When electricity flows through a wire, it generates a magnetic field around it. If another wire is close enough, the second wire acts as an antenna that turns this magnetic field into an electric current. When the different electrical signals mix, their integrity degrades and data corruption can occur. UTP mitigates crosstalk by twisting the wires around each other. Because crosstalk is greatest wherever wires are parallel to each other, this twisting makes it harder for this condition to exist. Still, UTP is much more vulnerable to crosstalk than STP or coaxial because it does not have extra layers of shielding to help protect against it.

Fire Rating of Cables Just as buildings must meet certain fire codes, so must wiring schemes. A lot of organizations string their network wires in drop ceilings—the space between the ceiling and the next floor—or under raised floors. This hides the cables and prevents people from tripping over them. However, when wires are strung in places like this, they are more likely to catch on fire without anyone knowing about it. Some cables produce hazardous gases when on fire that would spread throughout the building quickly. Network cabling that is placed in these types of areas, called *plenum space*, must meet a specific fire rating to ensure the cable will not produce and release harmful chemicals in case of a fire. A ventilation system's components are usually located in this plenum space, so if toxic chemicals were to get into that area, they could easily spread throughout the building in minutes.

Nonplenum cables usually have a polyvinyl chloride (PVC) jacket covering, whereas plenum-rated cables have jacket covers made of fluoropolymers. When setting up a network or extending an existing network, it is important that you know which wire types are required in which situation.

Cables should be installed in unexposed areas so they are not easily tripped over, damaged, or eavesdropped upon. The cables should be strung behind walls and in the protected spaces, such as in dropped ceilings. In environments that require extensive security, wires can be encapsulated within *pressurized conduits* so if someone attempts to access a wire, the pressure of the conduit changes, causing an alarm to sound and a message to be sent to the security staff. A better approach to high-security requirements is probably to use fiber-optic cable, which is much more difficult to covertly tap.

NOTE While a lot of the world's infrastructure is wired and thus uses one of these types of cables, remember that a growing percentage of our infrastructure is not wired, but rather uses some form of wireless technology (Bluetooth, Wi-Fi, satellite, etc.), particularly to reach end devices.

Bandwidth and Throughput

Whatever type of transmission you use over any given cable, there is a limit to how much information you can encode within it. In computer networks, we use two different but related terms to measure this limit. *Bandwidth* is the amount of information that theoretically can be transmitted over a link within a second. In a perfect world, this is the data transfer capability of a connection and is commonly associated with the number of available frequencies and speed of a link. Data *throughput* is the actual amount of data that can be carried over a real link. Throughput is always less than or equal to a link's bandwidth. In fact, it is most often the case that throughput is notably less than bandwidth. Why?

As mentioned, bandwidth is a theoretical limit determined by analyzing a medium (e.g., category 5 UTP cable) and a physical layer protocol (e.g., 100BaseT Ethernet) and then doing the math to calculate the maximum possible amount of data we could push through it. Now, of course, when you put that medium and protocol into a real environment, a multitude of issues come into play and make it hard to achieve that optimal data rate.

The throughput of our networks is affected by many factors. There could be EMI (or line noise) in the medium, as previously discussed. However, in a well-engineered facility and network, this should not be a big problem. Typically, you'll be more concerned about packet delays and losses. *Latency* is the amount of time it takes a packet to get from its source to its destination. This could be measured as either time to first byte (TTFB) or round-trip time (RTT). Latency can be caused by multiple factors, including

- **Transmission medium** Even though electricity and light move at the speed of light, it still takes time to get from one place to another. If your links are very long, or if the cables have too many imperfections, the medium itself will cause latency.
- **Network devices** Routers and firewalls take some time to examine packets, even if they're just deciding which outbound interface to use. If you have too many rules in your routing or security devices, this is invariably going to introduce delays.

To reduce latency, you should keep your physical links as short as possible. You should also look at how many hops your packets must take to get to their destinations. Virtual LANs (VLANs) can help keep devices that communicate frequently "closer" to each other. For international organizations, using a content distribution network (CDN), which we address later in this chapter, keeps most data close to where it is needed. Finally, the use of proxies can reduce latency by bringing frequently requested data closer to your users.

Another issue that negatively impacts your data throughputs (compared to a link's rated bandwidth) is congestion. Since some links in your network are shared, if you have too many packets moving around, it will inevitably bog things down. You may have a 1-GBps (bandwidth) connection to your home, but if every house in your neighborhood has one too and you all share a 1-GBps link from the local switch to the first router, your throughput will be way lower than advertised unless you log on when everyone else is sleeping. The best way to prevent congestion is through careful design and implementation of your network. Keep your broadcast domains as small as possible, ensure that your shared links are able to support peak traffic rates, and consider prioritizing certain types of traffic so that if your staff decides to livestream news, that doesn't slow down your ability to get real work done.

Network Devices

Several types of devices are used in LANs, MANs, and WANs to provide intercommunication among computers and networks. We need to have physical devices throughout the network to actually use all the protocols and services we have covered up to this point. The different network devices vary according to their functionality, capabilities, intelligence, and network placement. We will look at the following devices:

- Repeaters
- Bridges
- Switches
- Routers
- Gateways
- Proxy servers
- PBXs
- Network access control devices

The typical network has a bunch of these devices, and their purposes and operation can get confusing really quickly. Therefore, we will also look at network diagram techniques that can help us create different (simpler) views into complex environments. We'll also consider operational issues like power requirements, warranties, and support agreements.

Repeaters

A *repeater* provides the simplest type of connectivity because it only repeats electrical signals between cable segments, which enables it to extend a network. Repeaters work at the physical layer and are add-on devices for extending a network connection over a greater distance. The device amplifies signals because signals attenuate the farther they have to travel.

Repeaters can also work as line conditioners by actually cleaning up the signals. This works much better when amplifying digital signals than when amplifying analog signals because digital signals are discrete units, which makes extraction of background noise from them much easier for the amplifier. If the device is amplifying analog signals, any accompanying noise often is amplified as well, which may further distort the signal.

A *hub* is a multiport repeater. A hub is often referred to as a *concentrator* because it is the physical communication device that allows several computers and devices to communicate with each other. A hub does not understand or work with IP or MAC addresses. When one system sends a signal to go to another system connected to it, the signal is broadcast to all the ports, and thus to all the systems connected to the concentrator.

NOTE Hubs are exceptionally rare nowadays but you may still come across them.

Bridges

A *bridge* is a LAN device used to connect LAN segments (or VLAN segments) and thus extends the range of a LAN. It works at the data link layer and therefore works with MAC addresses. A repeater does not work with addresses; it just forwards all signals it receives. When a frame arrives at a bridge, the bridge determines whether or not the MAC address is on the local network segment. If it is not, the bridge forwards the frame to the necessary network segment. A bridge amplifies the electrical signal, as does a repeater, but it has more intelligence than a repeater and is used to extend a LAN and enable the administrator to filter frames to control which frames go where.

When using bridges, you have to watch carefully for *broadcast storms*. While bridges break up a collision domain by port (i.e., computers on the same bridge port are in the same collision domain), all ports are on the same broadcast domain. Because bridges can forward all traffic, they forward all broadcast packets as well. This can overwhelm the network and result in a broadcast storm, which degrades the network bandwidth and performance.

The international standard for bridges on Ethernet networks is IEEE 802.1Q. It describes the principal elements of bridge operation as follows:

- Relaying and filtering frames (based on MAC addresses and port numbers)
- Maintenance of the information required to make frame filtering and relaying decisions (i.e., the forwarding tables)
- Management of the elements listed (e.g., aging off forwarding table entries)

EXAM TIP Do not confuse routers with bridges. Routers work at the network layer and filter packets based on IP addresses, whereas bridges work at the data link layer and filter frames based on MAC addresses. Routers usually do not pass broadcast information, but bridges do pass broadcast information.

Forwarding Tables

A bridge must know how to get a frame to its destination—that is, it must know to which port the frame must be sent and where the destination host is located. Years ago, network administrators had to type route paths into bridges so the bridges had static paths indicating where to pass frames that were headed for different destinations. This was a tedious task and prone to errors. Today, most bridges use *transparent bridging*.

In transparent bridging, a bridge starts to learn about the network's environment as soon as it is powered on and continues to learn as the network changes. It does this by examining frames and making entries in its forwarding tables. When a bridge receives a frame from a new source computer, the bridge associates this new source address and the

> **Connecting Two LANS: Bridge vs. Router**
> What is the difference between two LANs connected via a bridge versus two LANs connected via a router? If two LANs are connected with a bridge, the LANs have been extended because they are both in the same broadcast domain. A router separates broadcast domains, so if two LANs are connected with a router, an internetwork results. An *internetwork* is a group of networks connected in a way that enables any node on any network to communicate with any other node. The Internet is an example of an internetwork.

port on which it arrived. It does this for all computers that send frames on the network. Eventually, the bridge knows the address of each computer on the various network segments and to which port each is connected. If the bridge receives a request to send a frame to a destination that is not in its forwarding table, it sends out a query frame on each network segment except for the source segment. The destination host is the only one that replies to this query. The bridge updates its table with this computer address and the port to which it is connected and forwards the frame.

Many bridges use the *Spanning Tree Protocol (STP)*, which adds more intelligence to the bridges. STP ensures that frames do not circle networks forever, provides redundant paths in case a bridge goes down, assigns unique identifiers to each bridge, assigns priority values to these bridges, and calculates path costs. This creates much more efficient frame-forwarding processes by each bridge. STP also enables an administrator to indicate whether he wants traffic to travel certain paths instead of others. Newer bridges implement the Shortest Path Bridging (SPB) protocol, which is defined in IEEE 802.1aq and is more efficient and scalable than STP.

Switches

Switches are, essentially, multiport bridges that typically have additional management features. Because bridges are intended to connect and extend LANs (and not necessarily individual hosts), they tend to have few ports. However, if you take the exact same functionality and add a bunch of ports to it, you could use the ports to connect to each individual host or to other switches. Figure 14-5 illustrates a typical, hierarchical network configuration in which computers are directly connected to access switches within close proximity (100 m or less). Access switches are, in turn, connected to distribution switches, which usually connect different departments or floors in a building. This distribution layer is a great place to implement access control lists (ACLs) and filtering to provide security. Finally, the upper tier of core switches provides a high-speed switching and routing backbone for the organization and is designed to pass network traffic as fast as possible. In this layer, only switches are connected to each other (i.e., there are no computers directly connected to them).

On Ethernet networks, computers have to compete for the same shared network medium. Each computer must listen for activity on the network and transmit its data when it thinks the coast is clear. This contention and the resulting collisions cause

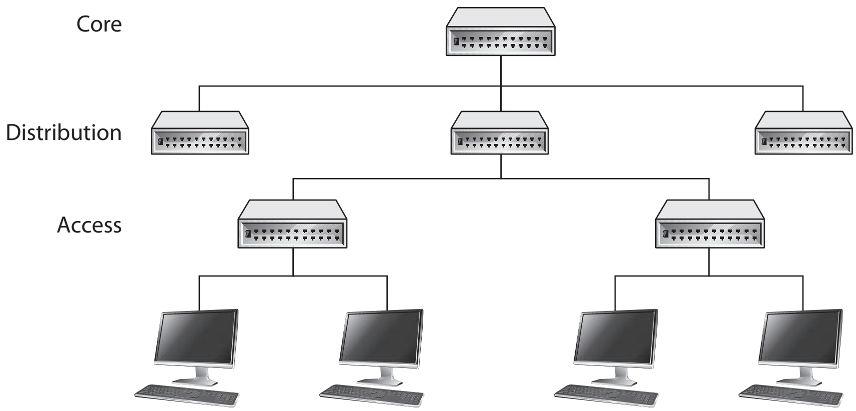

Figure 14-5 Hierarchical model of a switched network

traffic delays and use up precious bandwidth. When switches are used, contention and collisions are not issues, which results in more efficient use of the network's bandwidth and decreased latency. Switches reduce or remove the sharing of the network medium and the problems that come with it.

Since a switch is a multiport bridging device where each port is connected to exactly one other device, each port provides dedicated bandwidth to the device attached to it. A port is bridged to another port so the two devices have an end-to-end private link. The switch employs full-duplex communication, so one wire pair is used for sending and another pair is used for receiving. This ensures the two connected devices do not compete for the same bandwidth.

Basic switches work at the data link layer and forward traffic based on MAC addresses. However, today's layer 3, layer 4, and other layer switches have more enhanced functionality than layer 2 switches. These higher-level switches offer routing functionality, packet inspection, traffic prioritization, and QoS functionality. These switches are referred to as *multilayered switches* because they combine data link layer, network layer, and other layer functionalities.

Multilayered switches use hardware-based processing power, which enables them to look deeper within the frame, to make more decisions based on the information encapsulated within the frame, and then to provide forwarding and traffic management tasks. Usually this amount of work creates a lot of overhead and traffic delay, but multilayered switches perform these activities within an application-specific integrated circuit (ASIC). This means that most of the functions of the switch are performed at the hardware and chip level rather than at the software level, making it much faster than routers.

CAUTION While it is harder for attackers to sniff traffic on switched networks, they should not be considered safe just because switches are involved. Attackers commonly poison cache memory used on switches to divert traffic to their desired location.

Layer 3 and 4 Switches

Layer 2 switches only have the intelligence to forward a frame based on its MAC address and do not have a higher understanding of the network as a whole. A layer 3 switch has the intelligence of a router. It not only can route packets based on their IP addresses but also can choose routes based on availability and performance. A layer 3 switch is basically a router on steroids, because it moves the route lookup functionality to the more efficient switching hardware level.

The basic distinction between layer 2, 3, and 4 switches is the header information the device looks at to make forwarding or routing decisions (data link, network, or transport OSI layers). But layer 3 and 4 switches can use tags, which are assigned to each destination network or subnet. When a packet reaches the switch, the switch compares the destination address with its tag information base, which is a list of all the subnets and their corresponding tag numbers. The switch appends the tag to the packet and sends it to the next switch. All the switches in between this first switch and the destination host just review this tag information to determine which route it needs to take, instead of analyzing the full header. Once the packet reaches the last switch, this tag is removed and the packet is sent to the destination. This process increases the speed of routing of packets from one location to another.

The use of these types of tags, referred to as *Multiprotocol Label Switching (MPLS)*, not only allows for faster routing but also addresses service requirements for the different packet types. Some time-sensitive traffic (such as video conferencing) requires a certain level of service (QoS) that guarantees a minimum rate of data delivery to meet the requirements of a user or application. When MPLS is used, different priority information is placed into the tags to help ensure that time-sensitive traffic has a higher priority than less sensitive traffic, as shown in Figure 14-6.

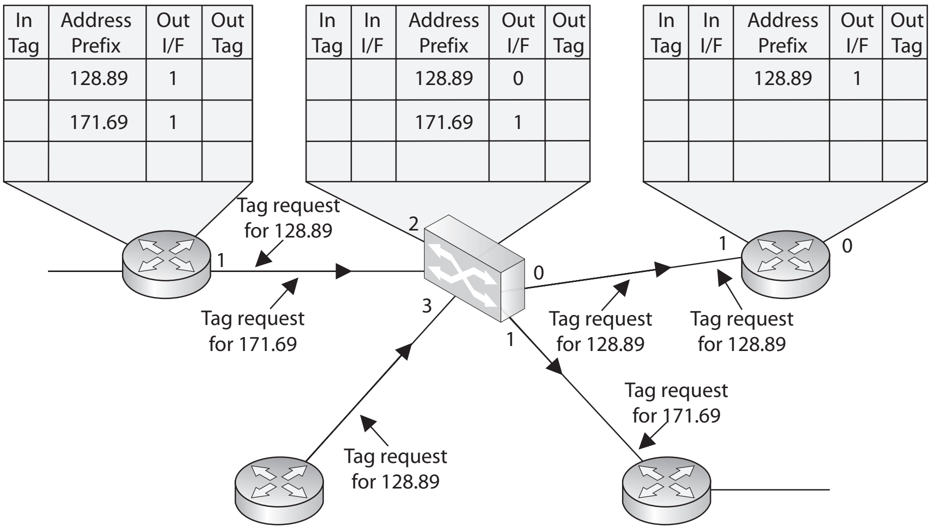

Figure 14-6 MPLS uses tags and tables for routing functions.