Application Delivery Fundamentals Exam

Study Guide

By Philip Jonsson

Contents

1 Overview

Welcome to the Application Delivery Fundamentals Exam – Study Guide. The purpose of this guide is to help you prepare for your exam. The contents of this document are based on the Application Delivery Fundamentals Blueprint and it uses several different sources that are collected from the Internet. This study guide provides students with the foundational knowledge required to pass the exam and a brief view on the TMOS Administration certification.

This study guide is a collection of information and therefore not my own words or texts. I simply compiled the information I found on the different sources. All of the information is referenced at the end of this document.

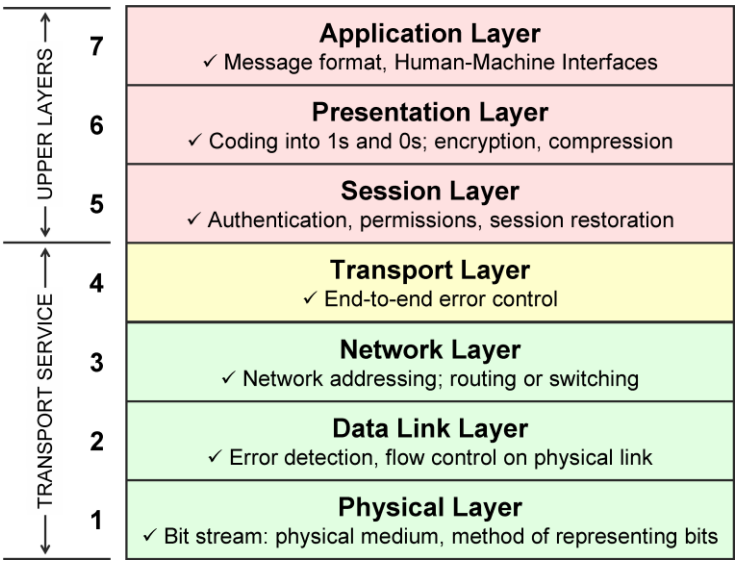
I hope this study guide will help you pass your exam and assist you through F5's many certification paths.

- *Philip Jonsson*

2 The OSI Model

2.1 What is the OSI model?

The term OSI Model is short for Open System Interconnection Basic Reference Model. The OSI Model consists of seven different layers. Each layer of the model is designed so that it can perform a specific task, and facilitate communications between the layer above it and the layer below it. You can see what the OSI Model looks like in Figure B.



(Figure B)

2.2 The Application Layer

The top layer of the OSI model is the Application layer. The first thing that you need to understand about the application layer is that it does not refer to the actual applications that users run. Instead, it provides the framework that the actual applications run on top of.

To understand what the application layer does, suppose for a moment that a user wanted to use Internet Explorer to open an FTP session and transfer a file. In this particular case, the application layer would define the file transfer protocol. This protocol is not directly accessible to the end user. The end user must still use an application that is designed to interact with the file transfer protocol. In this case, Internet Explorer would be that application.

2.3 The Presentation Layer

The presentation layer does some rather complex things, but everything that the presentation layer does can be summed up in one sentence. The presentation layer takes the data that is provided by the application layer, and converts it into a standard format that the other layers can understand. Likewise, this layer con-

verts the inbound data that is received from the session layer into something that the application layer can understand. The reason why this layer is necessary is because applications handle data differently from one another. In order for network communications to function properly, the data needs to be structured in a standard way.

2.4 The Session Layer

Once the data has been put into the correct format, the sending host must establish a session with the receiving host. This is where the session layer comes into play. It is responsible for establishing, maintaining, and eventually terminating the session with the remote host.

The interesting thing about the session layer is that it is more closely related to the application layer than it is to the physical layer. It is easy to think of connecting a network session as being a hardware function, but sessions are established between applications. If a user is running multiple applications, several of those applications may have established sessions with remote resources at any time.

2.5 The Transport Layer

The Transport layer is responsible for maintaining flow control. An operating system allows users to run multiple applications simultaneously and it is therefore possible that multiple applications may need to communicate over the network simultaneously. The Transport Layer takes the data from each application, and integrates it all into a single stream. This layer is also responsible for providing error checking and performing data recovery when necessary. In essence, the Transport Layer is responsible for ensuring that all of the data makes it from the sending host to the receiving host.

2.6 The Network Layer

The Network Layer is responsible for determining how the data will reach the recipient. This layer handles things like addressing, routing, and logical protocols. Since this series is geared toward beginners, I do not want to get too technical, but I will tell you that the Network Layer creates logical paths, known as virtual circuits, between the source and destination hosts. This circuit provides the individual packets with a way to reach their destination. The Network Layer is also responsible for its own error handling, and for packet sequencing and congestion control.

Packet sequencing is necessary because each protocol limits the maximum size of a packet. The amount of data that must be transmitted often exceeds the maximum packet size. Therefore, the data is fragmented into multiple packets. When this happens, the Network Layer assigns each packet a sequence number.

When the data is received by the remote host, that device's Network layer examines the sequence numbers of the inbound packets, and uses the sequence number to reassemble the data and to figure out if any packets are missing.

If you are having trouble understanding this concept, then imagine that you need to mail a large document to a friend, but do not have a big enough envelope. You could put a few pages into several small envelopes, and then label the envelopes so that your friend knows what order the pages go in. This is exactly the same thing that the Network Layer does.

2.7 The Data Link Layer

The data link layer can be sub divided into two other layers; the Media Access Control (MAC) layer, and the Logical Link Control (LLC) layer. The MAC layer basically establishes the computer's identity on the network, via its MAC address. A MAC address is the address that is assigned to a network adapter at the hardware level. This is the address that is ultimately used when sending and receiving packets. The LLC layer controls frame synchronization and provides a degree of error checking.

2.8 The Physical Layer

The physical layer of the OSI model refers to the actual hardware specifications. The Physical Layer defines characteristics such as timing and voltage. The physical layer defines the hardware specifications used by network adapters and by the network cables (assuming that the connection is not wireless). To put it simply, the physical layer defines what it means to transmit and to receive data.

3 Data link layer

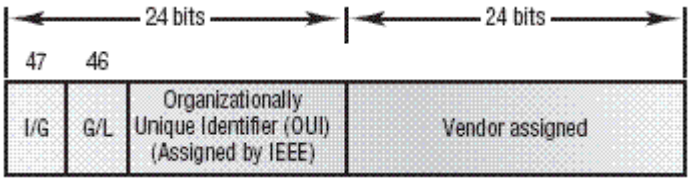
3.1 Ethernet Addressing

Every network device has a unique physical identity that is assigned by the manufacturing vendor is called MAC address or Ethernet address. The MAC address is also known as the hardware address while the IP address is the logical address of the device. The MAC address is defined in the Hexa-decimal format generally. It consists of 6 byte (48 bits) where the first three bytes are used as the identity of the vendor and the last three bytes are used as the node identity. The MAC address works on the MAC sub-layer of the data link layer of the OSI model.

Switches give network managers the ability to increase bandwidth without adding unnecessary complexity to the network. Layer 2 data frames consist of both infrastructure content, such as end user content and MAC Media Access Control address also known as Ethernet address. At Data Link layer, no modification is required to the MAC address of the data frame when going between like physical layer interfaces, such as from Ethernet to Fast Ethernet. However, changes to Media Access Control (MAC) address of the data frames might occur when bridging between unlike media types such as FDDI and Ethernet or Token Ring and Ethernet.

Switches learn the MAC address and build a table on the base of MAC addressing of the LAN segment called MAC Address Table. The Address Resolution Protocol (ARP) is the protocol that resolves the IP addresses into MAC addresses. RARP, the Reverse Address Resolution Protocol is a reverse of ARP and resolves MAC addresses into IP addresses.

The MAC layer of the Gigabit Ethernet is similar to those of standard Ethernet and Fast Ethernet. Media Access Layer of Gigabit Ethernet should maintain full duplex and half duplex broadcasting. The characteristics of Ethernet, such as collision detection, maximum network diameter, repeater rules, MAC addressing and so forth, will be the same of the Gigabit Ethernet. Support for half duplex Ethernet adds frame bursting and carrier extension, two functions not found in Ethernet and Fast Ethernet.



3.2 Address Resolution Protocol (ARP)

ARP defines the exchanges between network interfaces connected to an Ethernet media segment in order to map an IP address to a link layer address on demand. Link layer addresses are hardware addresses (although they are not immutable) on Ethernet cards and IP addresses are logical addresses assigned to machines attached to the Ethernet. Link layer addresses may be known by many different names: Ethernet addresses, Media Access Control (MAC) addresses, and even hardware addresses.

Address Resolution Protocol (ARP) exists solely to glue together the IP and Ethernet networking layers. Since networking hardware such as switches, hubs, and bridges operate on Ethernet frames, they are unaware of the higher layer data carried by these frames. Similarly, IP layer devices, operating on IP packets need to be able to transmit their IP data on Ethernets. ARP defines the conversation by which IP capable hosts can exchange mappings of their Ethernet and IP addressing.

ARP is used to locate the Ethernet address associated with a desired IP address. When a machine has a packet bound for another IP on a locally connected Ethernet network, it will send a broadcast Ethernet frame containing an ARP request onto the Ethernet. All machines with the same Ethernet broadcast address will receive this packet. If a machine receives the ARP request and it hosts the IP requested, it will respond with the link layer address on which it will receive packets for that IP address.

Once the requestor receives the response packet, it associates the MAC address and the IP address. This information is stored in the arp cache.

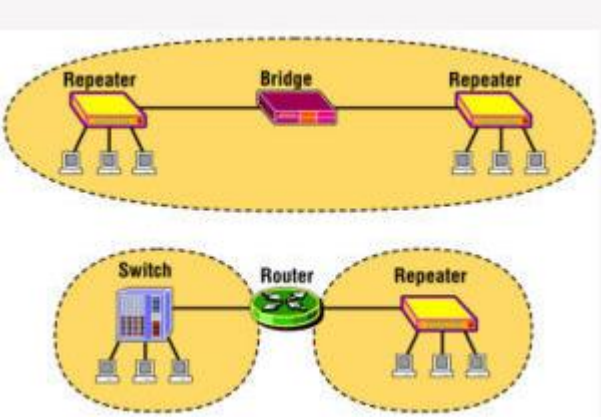
3.3 Broadcast Domains

A broadcast domain is a logical part of a network (a network segment) in which any network equipment can transmit data directly to other equipment or device without going through a routing device (assuming the devices share the same subnet and use the same gateway; also, they must be in the same VLAN).

A more specific broadcast domain definition is the area of the computer network that consists of every single computer or network equipment that can be reached directly by sending a simple frame to the data link layer's broadcast address.

3.4 Details on Broadcast Domains

While any layer 2 device is able to divide the collision domains, broadcast domains are only divided by layer 3 network devices such as routers or layer 3 switches.
Frames are normally addressed to a specific destination device on the network. While all devices detect the frame transmission on the network, only the device to which the frame is addressed actually receives it. A special broadcast address consisting of all is used to send frames to all devices on the network.



The VLAN (Virtual Local Area Network) technology can also create a so-called “virtual” broadcast domain. A network built with switching devices can see each network device as an independent system. These groups of independent systems can be joined into one broadcast domain, even if the computers are not physically connected to each other. This is very useful when administrating large networks where there is the need for better network management and control.

3.5 How to Restrict the Broadcast Domain

Since a broadcast domain is the area where broadcasts can be received, routers restrict broadcasts. If a router receives a broadcast signal, it simply drops it. In other words, the edge or border router connected to the Internet will not up-broadcast or will not relay that broadcast message. This is problematic and not fool-proof either. Suppose two networks exist that are connected to each other through a router and the first network has a running DHCP server that offers IP addresses to networked systems. On the other side, there is no valid DHCP server running on the second network. Offering IP addresses from the first network’s DHCP server to the second network’s systems can be a difficult task to accomplish since DHCP is a broadcast and the router that joins the networks drops the broadcast traffic. This leaves any DHCP request in the second network unanswered. Many router manufacturers provide capabilities for DHCP forwarding to solve this problem. This can be bypassed by connecting the two networks with a well configured, Linux-based, purpose oriented software router. That will handle the job properly and prevent further issues.

3.6 What is a VLAN?

In technical terms, a VLAN is a broadcast domain created by switches. Normally, it is a router creating that broadcast domain. With VLAN’s, a switch can create the broadcast domain.

This works by, you, the administrator, putting some switch ports in a VLAN other than 1, the default VLAN. All ports in a single VLAN are in a single broadcast domain.

Because switches can talk to each other, some ports on switch A can be in VLAN 10 and other ports on switch B can be in VLAN 10. Broadcasts between these devices will not be seen on any other port in any other VLAN, other than 10. However, these devices can all communicate because they are on the same VLAN. Without additional configuration, they would not be able to communicate with any other devices, not in their VLAN.

Are VLANs required?

It is important to point out that you don’t have to configure a VLAN until your network gets so large and has so much traffic that you need one. Many times, people are simply using VLAN’s because the network they are working on was already using them.

Another important fact is that, on a Cisco switch, VLAN’s are enabled by default and ALL devices are already in a VLAN. The VLAN that all devices are already in is VLAN 1. So, by default, you can just use all the ports on a switch and all devices will be able to talk to one another.

When do I need a VLAN?

You need to consider using VLAN's in any of the following situations:

- *You have more than 200 devices on your LAN*
- *You have a lot of broadcast traffic on your LAN*
- *Groups of users need more security or are being slowed down by too many broadcasts?*
- *Groups of users need to be on the same broadcast domain because they are running the same applications. An example would be a company that has VoIP phones. The users using the phone could be on a different VLAN, not with the regular users.*
- *Or, just to make a single switch into multiple virtual switches.*

Why not just subnet my network?

A common question is why not just subnet the network instead of using VLAN's? Each VLAN should be in its own subnet. The benefit that a VLAN provides over a subnetted network is that devices in different physical locations, not going back to the same router, can be on the same network. The limitation of subnetting a network with a router is that all devices on that subnet must be connected to the same switch and that switch must be connected to a port on the router.

With a VLAN, one device can be connected to one switch, another device can be connected to another switch, and those devices can still be on the same VLAN (broadcast domain).

How can devices on different VLAN's communicate?

Devices on different VLAN's can communicate with a router or a Layer 3 switch. As each VLAN is its own subnet, a router or Layer 3 switch must be used to route between the subnets.

What is a trunk port?

When there is a link between two switches or a router and a switch that carries the traffic of more than one VLAN, that port is a trunk port.
A trunk port must run a special trunking protocol. The protocol used would be Cisco's proprietary Inter-switch link (ISL) or the IEEE standard 802.1q.

What do VLAN's offer?

VLAN's offer higher performance for medium and large LAN's because they limit broadcasts. As the amount of traffic and the number of devices grow, so does the number of broadcast packets. By using VLAN's you are containing broadcasts.

VLAN's also provide security because you are essentially putting one group of devices, in one VLAN, on their own network.

3.7 Link Aggregation Control Protocol (LACP)

Link Aggregation binds several full-duplex Ethernet links running at the same speed together into one logical link with a single Media Access Control (MAC) address.

The concept of Ethernet Link Aggregation is known by several different names:

- *IEEE 802.3ad or 802.1ax*
- *Cisco supports this idea under the name EtherChannel*
- *Other vendors use the names "teaming" or "trunking"*

Link Aggregation provides two main advantages: improved reliability through automatic failover and aggregated throughput.

If any of the links lose their connection, the device keeps transmitting outgoing frames on other active links. The link partner does the same for incoming frames. The automatic failover is provided on a per-frame basis, usually without affecting any running workloads. For example, if one link goes down on a four-port aggregate, the remaining three links handle the traffic that would have traveled on the down link. If any links are operational, the aggregate continues to function.

In addition, each operational aggregated link can run at the configured line speed, and outgoing traffic is spread according to a configured preference. Incoming traffic is spread across the links according to configuration at the link partner. Many workloads, especially workloads with multiple parallel TCP or UDP conversations, can take advantage of this configuration to scale their throughput over available ports.

Performance

Incoming and outgoing traffic through an aggregated link requires some more processing than traffic through an individual link. In addition, depending on what combination of resources are in the aggregate, some advanced performance features of the aggregated resources might be disabled. For example, if some aggregated resources support TCP large send offload over IPv6 and others do not, the aggregate does not support TCP large send offload over IPv6.

4 Network Layer

4.1 Understanding IP Addresses

An IP address is an address used in order to uniquely identify a device on an IP network. The address is made up of 32 binary bits, which can be divisible into a network portion and host portion with the help of a subnet mask. The 32 binary bits are broken into four octets (1 octet = 8 bits). Each octet is converted to decimal and separated by a period (dot). For this reason, an IP address is said to be expressed in dotted decimal format (for example, 172.16.81.100). The value in each octet ranges from 0 to 255 decimal, or 00000000 - 11111111 binary.

Here is how binary octets convert to decimal: The right most bit, or least significant bit, of an octet holds a value of 2^0. The bit just to the left of that holds a value of 2^1. This continues until the left-most bit, or most significant bit, which holds a value of 2^7. So if all binary bits are a one, the decimal equivalent would be 255 as shown here:

1	1	1	1	1	1	1	1	
128	64	32	16	8	4	2	1	(128+64+32+16+8+4+2+1=255)

Here is a sample octet conversion when not all of the bits are set to 1.

0	1	0	0	0	0	0	1	
0	64	0	0	0	0	0	1	(0+64+0+0+0+0+0+1=65)

And this is sample shows an IP address represented in both binary and decimal.

10.	**1.**	**23.**	**19**	*decimal*
00001010.	00000001.	00010111.	00010011	*binary*

These octets are broken down to provide an addressing scheme that can accommodate large and small networks. There are five different classes of networks, A to E. This document focuses on addressing classes A to C, since classes D and E are reserved and discussion of them is beyond the scope of this document.

Note: Also note that the terms "Class A, Class B" and so on are used in this document to help facilitate the understanding of IP addressing and subnetting. These terms are rarely used in the industry anymore because of the introduction of classless interdomain routing (CIDR). Although, CIDR is beyond the scope of this document.

Given an IP address, its class can be determined from the three high-order bits. Figure 1 shows the significance in the three high order bits and the range of addresses that fall into each class. For informational purposes, Class D and Class E addresses are also shown.

IP Address Classes

Address Class	1st octet range (decimal)	1st octet bits (green bits do not change)	Network(N) and Host(H) parts of address	Default subnet mask (decimal and binary)	Number of possible networks and hosts per network
A	1-127**	00000000-01111111	N.H.H.H	255.0.0.0	128 nets (2^7) 16,777,214 hosts per net (2^24-2)
B	128-191	10000000-10111111	N.N.H.H	255.255.0.0	16,384 nets (2^14) 65,534 hosts per net (2^16-2)
C	192-223	11000000-11011111	N.N.N.H	255.255.255.0	2,097,150 nets (2^21) 254 hosts per net (2^8-2)
D	224-239	11100000-11101111	NA (multicast)		
E	240-255	11110000-11111111	NA (experimental)		

** All zeros (0) and all ones (1) are invalid hosts addresses.

Network Masks

A network mask helps you know which portion of the address identifies the network and which portion of the address identifies the node. Class A, B, and C networks have default masks, also known as natural masks, as shown here:

Class A:	255.0.0.0
Class B:	255.255.0.0
Class C:	255.255.255.0

An IP address on a Class A network that has not been subnetted would have an address/mask pair similar to: 8.20.15.1 255.0.0.0. To see how the mask helps you identify the network and node parts of the address, convert the address and mask to binary numbers.

8.20.15.1 = 00001000.00010100.00001111.00000001
255.0.0.0 = 11111111.00000000.00000000.00000000

Once you have the address and the mask represented in binary, then identifying the network and host ID is easier. Any address bits which have corresponding mask bits set to 1 represent the network ID. Any address bits that have corresponding mask bits set to 0 represent the node ID.

8.20.15.1 =	00001000.	00010100.	00001111.	00000001
255.0.0.0 =	11111111.	00000000.	00000000.	00000000

	Net id	**Host id**		

Understanding Subnetting

Subnetting allows you to create multiple logical networks that exist within a single Class A, B, or C network. If you do not subnet, you are only able to use one network from your Class A, B, or C network, which is unrealistic.

Each data link on a network must have a unique network ID, with every node on that link being a member of the same network. If you break a major network (Class A, B, or C) into smaller subnetworks, it allows you to create a network of interconnecting subnetworks. Each data link on this network would then have a unique network/subnetwork ID. Any device, or gateway, connecting *n* networks/subnetworks has *n* distinct IP addresses, one for each network / subnetwork that it interconnects.

In order to subnet a network, extend the natural mask using some of the bits from the host ID portion of the address to create a subnetwork ID. For example, given a Class C network of 204.17.5.0 which has a natural mask of 255.255.255.0, you can create subnets in this manner:

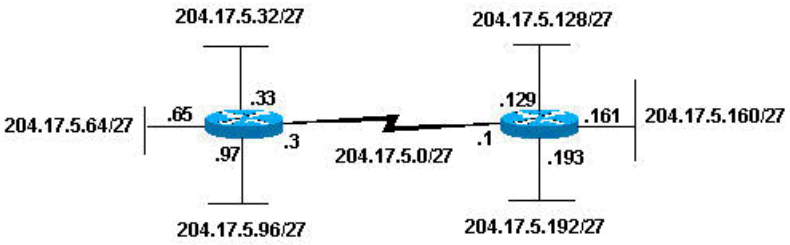
204.17.5.0 -	11001100.	00010001.	00000101.	00000000
255.255.255.224 -	11111111.	11111111.	11111111.	11100000
		----------|sub|----		

By extending the mask to be 255.255.255.224, you have taken three bits (indicated by "sub") from the original host portion of the address and used them to make subnets. With these three bits, it is possible to create eight subnets. With the remaining five host ID bits, each subnet can have up to 32 host addresses, 30 of which can actually be assigned to a device since host ids of all zeros or all ones are not allowed (it is very important to remember this). So, with this in mind, these subnets have been created.

204.17.5.0	255.255.255.224	host address range 1 to 30
204.17.5.32	255.255.255.224	host address range 33 to 62
204.17.5.64	255.255.255.224	host address range 65 to 94
204.17.5.96	255.255.255.224	host address range 97 to 126
204.17.5.128	255.255.255.224	host address range 129 to 158
204.17.5.160	255.255.255.224	host address range 161 to 190
204.17.5.192	255.255.255.224	host address range 193 to 222
204.17.5.224	255.255.255.224	host address range 225 to 254

Note: There are two ways to denote these masks. First, since you are using three bits more than the "natural" Class C mask, you can denote these addresses as having a 3-bit subnet mask. Or, secondly, the mask of 255.255.255.224 can also be denoted as /27 as there are 27 bits that are set in the mask. This second method is used with CIDR. With this method, one of these networks can be described with the notation prefix/length. For example, 204.17.5.32/27 denotes the network 204.17.5.32 255.255.255.224. When appropriate the prefix/length notation is used to denote the mask throughout the rest of this document.

The network subnetting scheme in this section allows for eight subnets, and the network might appear as:



Notice that each of the routers in the figure is attached to four subnetworks, one subnetwork is common to both routers. Also, each router has an IP address for each subnetwork to which it is attached. Each subnetwork could potentially support up to 30 host addresses.

This brings up an interesting point. The more host bits you use for a subnet mask, the more subnets you have available. However, the more subnets available, the less host addresses available per subnet. For example, a Class C network of 204.17.5.0 and a mask of 255.255.255.224 (/27) allows you to have eight subnets, each with 32 host addresses (30 of which could be assigned to devices). If you use a mask of 255.255.255.240 (/28), the break down is:

204.17.5.0 -	11001100.	00010001.	00000101.	00000000
255.255.255.240 -	11111111.	11111111.	11111111.	11110000
		----------|sub|------		

Since you now have four bits to make subnets with, you only have four bits left for host addresses. So in this case you can have up to 16 subnets, each of which can have up to 16 host addresses (14 of which can be assigned to devices).

Take a look at how a Class B network might be subnetted. If you have network 172.16.0.0, then you know that its natural mask is 255.255.0.0 or 172.16.0.0/16. Extending the mask to anything beyond 255.255.0.0 means you are subnetting. You can quickly see that you have the ability to create a lot more subnets than with the Class C network. If you use a mask of 255.255.248.0 (/21), how many subnets and hosts per subnet does this allow for?

172.16.0.0 -	10101100.	00010000.	00000000.	00000000
255.255.248.0 -	11111111.	11111111.	11111000.	00000000
		----------|sub|----------		

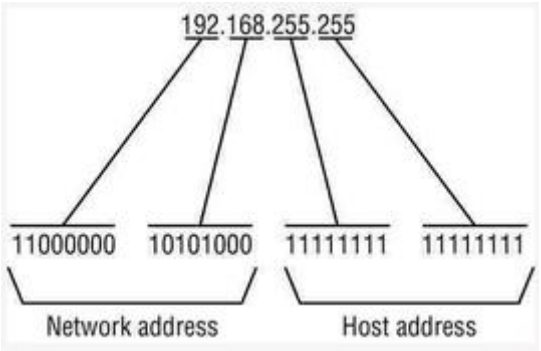
You are using five bits from the original host bits for subnets. This allows you to have 32 subnets (2^5). After using the five bits for subnetting, you are left with 11 bits for host addresses. This allows each subnet so have 2048 host addresses (2^11), 2046 of which could be assigned to devices.

Note: In the past, there were limitations to the use of a subnet 0 (all subnet bits are set to zero) and all ones subnet (all subnet bits set to one). Some devices would not allow the use of these subnets. Cisco Systems devices allow the use of these subnets when the ip subnet zero command is configured.

4.2 Broadcast Address

A broadcast address is an IP address that targets all systems on a specific subnet instead of single hosts. The broadcast address of any IP address can be calculated by taking the bit compliment of the subnet mask, sometimes referred to as the reverse mask, and then applying it with a bitwise OR calculation to the IP address in question.

Some systems that are derived from BSD use zeros-broadcasts instead of ones-broadcasts. This means that when a broadcast address is created, the host area of the IP address is filled while displayed using binary values with zeros instead of ones. Most operating systems use ones-broadcasts. Changing systems to use zeros-broadcasts will break some communications in the wrong environments, so the user should understand his/her needs before changing the broadcast address or type.



Math Example

If a system has the IP address 192.168.12.220 and a network mask of 255.255.255.128, what should the broadcast address for the system be? To do this calculation, convert all numbers to binary values. For bitwise, remember that any two values where at least one value is 1, the result will be 1, otherwise the result is 0.

IP Address:	11000000.10101000.00001100.11011100
Reverse Mask:	00000000.00000000.00000000.01111111
bitwise OR:	————————————————
Broadcast:	11000000.10101000.00001100.11111111

Convert the binary value back to octal and the resulting value is 192.168.12.255.

4.3 The IP routing table

Every computer that runs TCP/IP makes routing decisions. These decisions are controlled by the IP routing table. To display the IP routing table on computers running Windows Server 2003 operating systems, you can type route print at a command prompt.

The following table shows an example of an IP routing table. This example is for a computer running Windows Server 2003, Standard Edition with one 10 megabit/second (Mbit/s) network adapter and the following configuration:

- *IP address: 10.0.0.169*
- *Subnet mask: 255.0.0.0*
- *Default gateway: 10.0.0.1*

Description	Network destination	Netmask	Gateway	Interface	Metric
Default route	0.0.0.0	0.0.0.0	10.0.0.1	10.0.0.169	30
Loopback network	127.0.0.0	255.0.0.0	127.0.0.1	127.0.0.1	1
Local network	10.0.0.0	255.0.0.0	10.0.0.169	10.0.0.169	30
Local IP address	10.0.0.169	255.255.255.255	127.0.0.1	127.0.0.1	30
Multicast addresses	224.0.0.0	240.0.0.0	10.0.0.169	10.0.0.169	30
Limited broadcast address	255.255.255.255	255.255.255.255	10.0.0.169	10.0.0.169	1

The routing table is built automatically, based on the current TCP/IP configuration of your computer. Each route occupies a single line in the displayed table. Your computer searches the routing table for an entry that most closely matches the destination IP address.

Your computer uses the default route if no other host or network route matches the destination address included in an IP datagram. The default route typically forwards an IP datagram (for which there is no matching or explicit local route) to a default gateway address for a router on the local subnet. In the previous example, the default route forwards the datagram to a router with a gateway address of 10.0.0.1.

Because the router that corresponds to the default gateway contains information about the network IDs of the other IP subnets within the larger TCP/IP internet, it forwards the datagram to other routers until the datagram is eventually delivered to an IP router that is connected to the specified destination host or subnet within the larger network.

The following sections describe each of the columns displayed in the IP routing table: network destination, netmask, gateway, interface, and metric.

Network destination

The network destination is used with the netmask to match the destination IP address. The network destination can range from 0.0.0.0 for the default route through 255.255.255.255 for the limited broadcast, which is a special broadcast address to all hosts on the same network segment.

Gateway

The gateway address is the IP address that the local host uses to forward IP datagrams to other IP networks. This is either the IP address of a local network adapter or the IP address of an IP router (such as a default gateway router) on the local network segment.

Interface

The interface is the IP address that is configured on the local computer for the local network adapter that is used when an IP datagram is forwarded on the network.

Metric

A metric indicates the cost of using a route, which is typically the number of hops to the IP destination. Anything on the local subnet is one hop, and each router crossed after that is an additional hop. If there are multiple routes to the same destination with different metrics, the route with the lowest metric is selected.

4.4 IP Routing Protocols

A routing protocol is a set of rules or standard that determines how routers on a network communicate and exchange information with each other, enabling them to select best routes to a remote network. Each router has priority knowledge only of networks attached to it directly. A router running routing protocol shares this information first, among immediate neighbors, then throughout the entire network. This way, routers gain insight knowledge of the topology of the network.

Routing protocols perform several activities, including:

* Network discovery
* Updating and maintaining routing tables

The router which sits at the base of a network maintains a routing table, which is a list of networks and possible routes known by the router. The routing table includes network addresses for its own interfaces, which are the directly connected networks, as well as network addresses for remote networks. A remote network is a network that can only be reached by forwarding the packet to another router.

Remote networks are added to the routing table in two ways:

- *By the network administrator manually configuring static routes.*
- *By implementing a dynamic routing protocol.*

Dynamic Routing protocols are used by routers to share information about the reachability and status of remote networks.

IP Routing Protocols (Dynamic)

There are several dynamic routing protocols for IP. Here are some of the more common dynamic routing protocols for routing IP packets:

- *RIP (Routing Information Protocol)*
- *IGRP (Interior Gateway Routing Protocol)*
- *EIGRP (Enhanced Interior Gateway Routing Protocol)*
- *OSPF (Open Shortest Path First)*
- *IS-IS (Intermediate System-to-Intermediate System)*
- *BGP (Border Gateway Protocol*

Advantages of dynamic routing protocols

- *Dynamic routing protocols update and maintain the networks in their routing tables.*
- *Dynamic routing protocols not only make a best path determination to various networks, they will also determine a new best path if the initial path becomes unusable or there is a change in the topology.*
- *Routers that use dynamic routing protocols automatically share routing information with other routers and compensate for any topology changes without involving the network administrator.*

4.5 IP-Fragmentation

Why does fragmentation occur?

Fragmentation happens when an IP datagram has to travel through a network with a maximum transmission unit (MTU) that is smaller than the size of the IP datagram. Thus, if I send an IP datagram that is bigger than 1500 bytes to an Ethernet network, the datagram needs to be fragmented. The packets are then assembled at the receiving host. Fragmentation can either at the originating sending host or at an intemediate router.

How are the packets reassembled?

Note that with IP fragmentation, packets are not reassembled until they reach the final destination. It is reassembled at the IP layer at the receiving end. This is make fragmentation and reassembly transparent to the protocol layer (TCP and UDP). If one of the packets is lost, the whole packets need to be transmitted again.

Packets are reassembled at the receiving host by associating each fragment with an identical fragment identification number, or frag id for short. The frag ID is actually a copy of the ID field (IP identification number) in the IP header. Besides that, each fragment must carry its "position" or "offset" in the original unfragmented packet. Thus the first fragment will have an offset of 0, since its seat is at the front row and counting starts from 0. Each fragment must also tell the length of data that it carries. This is like the compartments in a train. And finally, each fragment must flag the MF (more fragments) bit if it is not the last fragment.

Fragmenting a Packet

Here is a hypothetical example. Suppose that we want to send a 110 bytes ICMP packet on a network with MTU of 40 (well that's damn small, but this is for illustration purposes). This is a diagram of the original packet:

IP	ICMP	Data
Header |		| Header
20	8	82 (bytes

The packet will be fragmented as shown below.

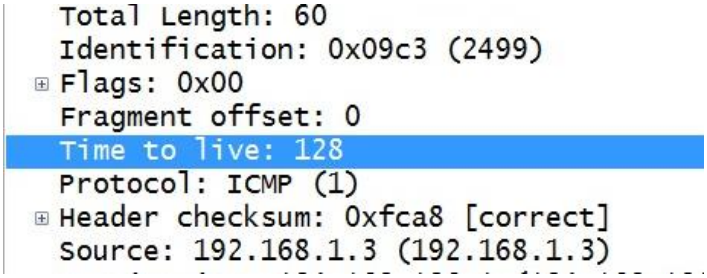
+----------------+----------+----------+
Packet 1 | IP header (20) | ICMP (8) | Data (12) | ID=88, Len=20, Off=0, MF=1
+----------------+----------+----------+
Packet 2 | IP header (20) | Data (20) | ID=88, Len=20, Off=20, MF=1
+--------------------------------------+
Packet 3 | IP header (20) | Data (20) | ID=88, Len=20, Off=40, MF=1
+--------------------------------------+
Packet 4 | IP header (20) | Data (20) | ID=88, Len=20, Off=60, MF=1
+---------------------------+----------+
Packet 5 | IP header (20) | Data (10) | ID=88, Len=10, Off=80, MF=0
+---------------------------+

ID - IP identification number
Len - Data Length (data length does not include IP header)
Off - Offset
MF - More Fragment

Notice that the second packet and subsequent packets contains IP header that is copied from the original packet. There are no ICMP headers, except in the first packet. In a nutshell, the 110 ICMP packet is broke into 5 packet, with total lengths of 40, 40, 40, 40 and 30 bytes each. The ICMP data is broken into lengths of 12, 20, 20, 20, and 10 bytes each.

4.6 What is Time to Live (TTL)

TTL , which stands for Time to Live is a field which is available in the IP header. Take a scenario where a user's pings an IP address from his PC. When the IP Packet is constructed for the ping , the TTL value is embedded inside the IP Packet header. Windows based computers use a default value of 128 for the TTL value.



In the above example, assume a scenario where the destination IP address is not a valid IP address. In this case, the IP Packet would loop on the network hopping between devices. This would eventually slow down the traffic on the links and the IP packet would keep on looping infinitely.

Due to this, the TTL value is used. The TTL value would be decremented when the IP packet crosses routers or networks (layer 3 devices). When the TTL value in the IP header reaches 1, the packet would not be forwarded and dropped and an ICMP destination unreachable message would be sent to the destination, eventually preventing looping of the IP Packet.

4.7 IP version 6

Increasing the IP address pool was one of the major forces behind developing IPv6. It uses a 128-bit address, meaning that we have a maximum of 2^{128} addresses available, or 340,282,366,920,938,463,463,374,607,431,768,211,456, or enough to give multiple IP addresses to every grain of sand on the planet. So our friendly old 32-bit IPv4 dotted-quads don't do the job anymore; these newfangled IPs require eight 16-bit hexadecimal colon-delimited blocks. So not only are they longer, they use numbers and letters. At first glance, those mondo IPv6 addresses look like impenetrable secret code:

2001:0db8:3c4d:0015:0000:0000:abcd:ef12

We'll dissect this in a moment and learn that's it not such a scary thing, but first let's look at the different types of IPv6 addressing.

Under IPv4 we have the old familiar unicast, broadcast and multicast addresses. In IPv6 we have unicast, multicast and anycast. With IPv6 the broadcast addresses are not used anymore, because they are replaced with multicast addressing.

IPv6 Unicast
This is similar to the unicast address in IPv4 – a single address identifying a single interface. There are four types of unicast addresses:

- ***Global unicast*** *addresses, which are conventional, publicly routable address, just like conventional IPv4 publicly routable addresses.*
- ***Link-local*** *addresses are akin to the private, non-routable addresses in IPv4 (10.0.0.0/8, 172.16.0.0/12, 192.168.0.0/16). They are not meant to be routed, but confined to a single network segment. Link-local addresses mean you can easily throw together a temporary LAN, such as for conferences or meetings, or set up a permanent small LAN the easy way.*
- ***Unique local*** *addresses are also meant for private addressing, with the addition of being unique, so that joining two subnets does not cause address collisions.*
- ***Special*** *addresses are loopback addresses, IPv4-address mapped spaces, and 6-to-4 addresses for crossing from an IPv4 network to an IPv6 network.*

If you read about site-local IPv6 addresses, which are related to link-local, these have been deprecated, so you don't need to bother with them.

Multicast

Multicast in IPv6 is similar to the old IPv4 broadcast address a packet sent to a multicast address is delivered to every interface in a group. The IPv6 difference is it's targeted instead of annoying every single host on the segment with broadcast blather, only hosts who are members of the multicast group receive the multicast packets. IPv6 multicast is routable, and routers will not forward multicast packets unless there are members of the multicast groups to forward the packets to. Anyone who has ever suffered from broadcast storms will appreciate this mightily.

Anycast

An anycast address is a single address assigned to multiple nodes. A packet sent to an anycast address is then delivered to the first available node. This is a slick way to provide both load-balancing and automatic failover. The idea of anycast has been around for a long time; it was proposed for inclusion in IPv4 but it never happened.

Several of the DNS root servers use a router-based anycast implementation, which is really a shared unicast addressing scheme. (While there are only thirteen authoritative root server names, the total number of actual servers is considerably larger, and they are spread all over the globe.) The same IP address is assigned to multiple interfaces, and then multiple routing tables entries are needed to move everything along.

IPv6 anycast addresses contain fields that identify them as anycast, so all you need to do is configure your network interfaces appropriately. The IPv6 protocol itself takes care of getting the packets to their final destinations. It's a lot simpler to administer than shared unicast addressing.

Address Dissection

Let's take another look at our example IPv6 address:

2001:0db8:3c4d:0015:0000:0000:abcd:ef12		
_____|_____|_____		
global prefix	subnet	Interface ID

The prefix identifies it as a global unicast address. It has three parts: the network identifier, the subnet, and the interface identifier.

The global routing prefix comes from a pool assigned to you, either by direct assignment from a Regional Internet Registry like APNIC, ARIN, or RIPE NCC, or more likely from your Internet service provider. The subnet and interface IDs are controlled by you, the hardworking local network administrator.

You'll probably be running mixed IPv6/IPv4 networks for some time. IPv6 addresses must have 128 bits. IPv4 addresses are therefore represented like this:

0000:0000:0000:0000:0000:0000:192.168.1.25

Eight blocks of 16 bits each are required in an IPv6 address. The IPv4 address occupies 32 bits, so that is why there are only seven colon-delimited blocks.

The localhost address is 0000:0000:0000:0000:0000:0000:0000:0001.

Naturally we want shortcuts, because these are long and all those zeroes are just dumb-looking. Leading zeroes can be omitted, and contiguous blocks of zeroes can be omitted entirely, so we end up with these:

2001:0db8:3c4d:0015:0:0:abcd:ef12

2001:0db8:3c4d:0015::abcd:ef12

::192.168.1.25

::1

5 Transport Layer

5.1 MTU - Maximum Transmission Unit

The MTU is the maximum size of a single data unit (e.g., a frame) of digital communications. MTU sizes are inherent properties of physical network interfaces, normally measured in bytes. The MTU for Ethernet, for instance, is 1500 bytes. Some types of networks (like Token Ring) have larger MTUs, and some types have smaller MTUs, but the values are fixed for each physical technology.

Higher-level network protocols like TCP/IP can be configured with a maximum packet size, a parameter independent of the physical layer MTU over which TCP/IP runs. Unfortunately, many network devices use the terms interchangeably. On both home broadband routers and Xbox Live enabled game consoles, for example, the parameter called MTU is in fact the maximum TCP packet size and not the physical MTU.

In Microsoft Windows, the maximum packet size for protocols like TCP can be set in the Registry. If this value is set too low, streams of network traffic will be broken up into a relatively large number of small packets that adversely affects performance. Xbox Live, for example, requires the value of MTU (packet size) by at least 1365 bytes. If the maximum TCP packet size is set too high, it will exceed the network's physical MTU and also degrade performance by requiring that each packet be subdivided into smaller ones (a process known as fragmentation). Microsoft Windows computers default to a maximum packet size of 1500 bytes for broadband connections and 576 bytes for dialup connections.

Performance problems may also occur if the TCP "MTU" setting on the home broadband router differs from the setting on individual devices connected to it.

5.2 TCP Maximum Segment Size (MSS)

During session connection establishment, two peers, or hosts, engage in negotiations to determine the IP segment size of packets that they will exchange during their communication. The segment size is based on the MSS option (maximum segment size) value set in the TCP SYN (synchronize) packets that the peers exchange during session negotiation. The MSS field value to be used is largely determined by the maximum transmission unit (MTU) of the interfaces that the peers are directly connected to.

About TCP and MSS

The TCP protocol is designed to limit the size of segments of data to a maximum of number of bytes. The purpose for this is to constrain the need to fragment segments of data for transmission at the IP level. The TCP MSS specifies the maximum number of bytes that a TCP packet's data field, or segment, can contain. It refers to the maximum amount of TCP data in a single IP datagram that the local system can accept and reassemble.

A TCP packet includes data for headers as well as data contained in the segment. If the MSS value is set too low, the result is inefficient use of bandwidth; More packets are required to transmit the data. An MSS value that is set too high could result in an IP datagram that is too large to send and that must be fragmented.

Typically a host bases its MSS value on its outgoing interface's maximum transmission unit (MTU) size. The MTU is the maximum frame size along the path between peers. A packet is fragmented when it exceeds the MTU size. Because of variation of the MTU size of the interfaces of hosts in the path taken by TCP packets between two peers, some packets that are within the negotiated MSS size of the two peers might be fragmented but instead are dropped and an ICMP error message is sent to the source host of the packet.

To diminish the likelihood of fragmentation and to protect against packet loss, you can decrease the TCP MSS.

5.3 TCP (Transmission Control Protocol)

TCP is a subset of the Internet protocol suite, which is often called TCP/IP, although the acronym TCP/IP refers to only two of the many protocols in the Internet protocol suite. Still, most people refer to the Internet protocols as TCP/IP and that style is retained here.

TCP is a connection-oriented protocol that provides the flow controls and reliable data delivery services listed next. These services run in the host computers at either end of a connection, not in the network itself. Therefore, TCP is a protocol for managing end-to-end connections. Since end-to-end connections may exist across a series of point-to-point connections, they are often called virtual circuits.

Quick terminology

Connections	Two computers set up a connection to exchange data. The systems synchronize with one another to manage packet flows and adapt to congestion in the network.
Full-duplex operation	A TCP connection is a pair of virtual circuits (one in each direction). Only the two end systems can use the connection.
Error checking	A checksum technique is used to verify that packets are not corrupted.
Sequencing	Packets are numbered so that the destination can reorder packets and determine if a packet is missing.
Acknowledgements	Upon receipt of one or more packets, the receiver returns an acknowledgement (called an "ACK") to the sender indicating that it received the packets. If packets are not ACKed, the sender may retransmit the packets (or terminate the connection if it thinks the receiver has crashed).
Flow control	If the sender is overflowing the receiver by transmitting too quickly, the receiver drops packets. Failed ACKs alert the sender to slow down or stop sending.
Packet recovery services	The receiver can request retransmission of a packet. Also, if packet receipt is not ACKed, the sender will resend the packets.

Reliable data delivery services are critical for applications such as file transfers, database services, transaction processing, and other mission-critical applications in which every packet must be delivered-guaranteed.

While TCP provides these reliable services, it depends on IP to delivery packets. IP is often referred to as an unreliable or best effort service. While it seems odd to build a network that is unreliable, the original Internet architects wanted to remove as many services from the network itself to support fast packet delivery rather than reliability. Routers do not keep track of packets or do anything to ensure delivery. They just forward packets.

The assumption was that end systems would be relatively smart devices with memory and processors. The end devices could handle all the reliability functions rather than the network. This was actually a radical approach at the time, but the implications have been profound. It meant that end systems would become the focus of application development for the Internet, not the network.

In contrast, the telephone network implements an architecture in which end devices (phones) are dumb and the network is supposedly "smart." The only problem with this model is that you can't run applications on your phone that take advantage of the network. In fact, you are totally dependent on the phone company to deploy new applications (call waiting and caller ID are examples). Compared to the Internet, the phone system is a dinosaur. Consider that the user interface for the Web is a full-color graphical browser, while the interface for the telephone network is a 12-key pad!

While end-systems provide TCP's reliability functions, not all applications need them. For example, there is no need to recover lost packets in a live video stream. By the time they are recovered, the viewer has already seen the barely visible glitch caused by the missing packet. These applications just need speed. So UDP was created to provide an application interface to the network for real-time applications that don't need TCP's extra services. UDP provides a very simple port connection between applications and IP.

5.4 TCP Three-way handshake

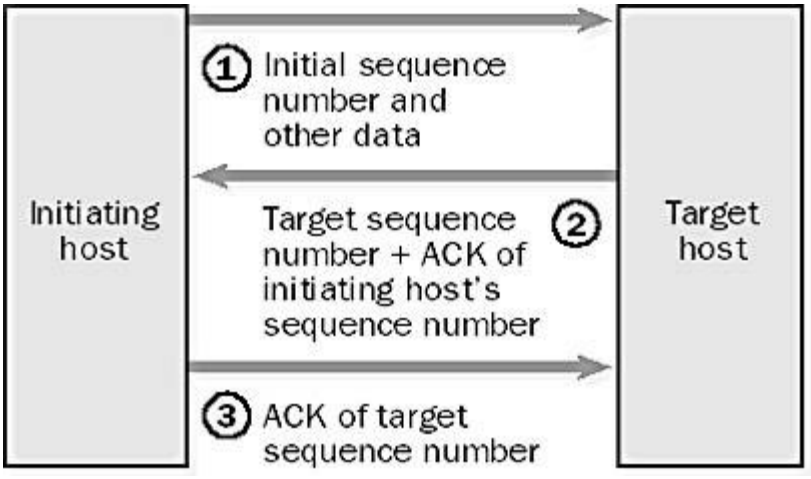
A method of initializing a Transmission Control Protocol (TCP) session between two hosts on a TCP/IP network. The handshake establishes a logical connection between the hosts by synchronizing the sending and receiving of packets and communicating TCP parameters between the hosts.

How the TCP Three-way Handshake works

All TCP communication is connection oriented. A TCP session must be established before the hosts in the connection exchange data. Packets that are transferred between hosts are accounted for by assigning a sequence number to each packet. An ACK, or acknowledgment, is sent after every packet is received. If no ACK is received for a packet, the packet is re-sent. The three-way handshake ensures that the initial request is acknowledged, that the data is sent, and that the data is acknowledged.

These are the three stages of a TCP three-way handshake:

- *The initiating host sends a TCP packet requesting a new session. This packet contains the initiating host's sequence number for the connection. The packet includes information such as a set SYN (synchronization) flag and data about the size of the window buffer on the initiating host.*



- *The target host sends a TCP packet with its own sequence number and an ACK of the initiating host's sequence number.*
- *The initiating host sends an ACK containing the target sequence number that it received.*

NOTE

A similar three-way process is used to terminate a TCP session between two hosts. Using the same type of handshake to end the connection ensures that the hosts have completed their transactions and that all data is accounted for.

5.5 What is UDP and how does it work?

UDP stands for User Datagram Protocol. UDP provides an unreliable packet delivery system built on top of the IP protocol. As with IP, each packet is individual and is handled separately. Because of this, the amount of data that can be sent in a UDP packet is limited to the amount that can be contained in a single IP packet. Thus, a UDP packet can contain at most 65507 bytes (this is the 65535-byte IP packet size minus the minimum IP header of 20 bytes and minus the 8-byte UDP header).

UDP packets can arrive out of order or not at all. No packet has any knowledge of the preceding or following packet. The recipient does not acknowledge packets, so the sender does not know that the transmission was successful. UDP has no provisions for flow control--packets can be received faster than they can be used. We call this type of communication connectionless because the packets have no relationship to each other and because there is no state maintained.

The destination IP address and port number are encapsulated in each UDP packet. These two numbers together uniquely identify the recipient and are used by the underlying operating system to deliver the packet to a specific process (application). Each UDP packet also contains the sender's IP address and port number.

One way to think of UDP is by analogy to communications via a letter. You write the letter (this is the data you are sending); put the letter inside an envelope (the UDP packet); address the envelope (using an IP address and a port number); put your return address on the envelope (your local IP address and port number); and then you send the letter.

Like a real letter, you have no way of knowing whether a UDP packet was received. If you send a second letter one day after the first, the second one may be received before the first. Or, the second one may never be received.

5.6 What is a Network Port and why do I need one?

Any server machine makes its services available using numbered ports -- one for each service that is available on the server. For example, if a server machine is running a Web server and a file transfer protocol (FTP) server, the Web server would typically be available on port 80, and the FTP server would be available on port 21. Clients connect to a service at a specific IP address and on a specific port number.

Once a client has connected to a service on a particular port, it accesses the service using a specific protocol. Protocols are often text and simply describe how the client and server will have their conversation. Every Web server on the Internet conforms to the hypertext transfer protocol (HTTP)

5.7 TCP Timeout and Retransmission

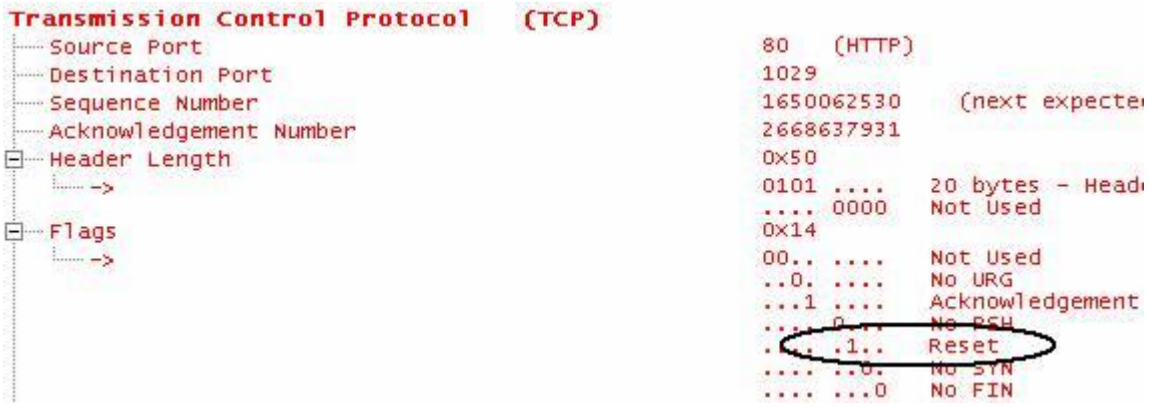
TCP provides a reliable transport layer. One of the ways it provides reliability is for each end to acknowledge the data it receives from the other end. But data segments and acknowledgments can get lost. TCP handles this by setting a timeout when it sends data, and if the data isn't acknowledged when the timeout expires, it retransmits the data. A critical element of any implementation is the timeout and retransmission strategy.

TCP manages four different timers for each connection.

- *A retransmission timer is used when expecting an acknowledgment from the other end.*
- *A persist timer keeps window size information flowing even if the other end closes its receive window.*
- *A keep-alive timer detects when the other end on an otherwise idle connection crashes or reboots.*
- *A 2MSL timer measures the time a connection has been in the TIME_WAIT state.*

5.8 What are TCP RST Packets?

According to RFC 793, which specifies an Option in the Flags portion of the TCP header called Reset (or RST). The Reset bit is designed to allow a station to abort the TCP connection with another station. This can happen for a number of reasons.



If a station involved in a TCP session notices that it is not receiving acknowledgements for anything it sends, the connection is now unsynchronized, and the station should send a reset. This is a half-open connection where only one side is involved in the TCP session. This cannot work by definition of the protocol.

RST packets are a sign that the TCP connections are half open. One station or the other stopped sending information or ACKs for some reason. There are acceptable times for RST packets, however, if there are a large number of RST packets in a conversation, this is definitely something to troubleshoot. Which side is sending the RST? What is causing it to send the RST? Does this happens right away in the TCP setup, or is it later in the session? If later, is there any reason that the station would abort the session in the middle of the data transfer?

5.9 TCP Checksum

The Transmission Control Protocol is designed to provide reliable data transfer between a pair of devices on an IP internetwork. Much of the effort required to ensure reliable delivery of data segments is of necessity focused on the problem of ensuring that data is not lost in transit. But there's another important critical impediment to the safe transmission of data: the risk of *errors* being introduced into a TCP segment during its travel across the internetwork.

Detecting Transmission Errors Using Checksums

If the data gets where it needs to go but is corrupted and we do not detect the corruption, this is in some ways worse than it never showing up at all. To provide basic protection against errors in transmission, TCP includes a 16-bit Checksum field in its header. The idea behind a checksum is very straight-forward: take a string of data bytes and add them all together. Then send this sum with the data stream and have the receiver check the sum. In TCP, a special algorithm is used to calculate this checksum by the device sending the segment; the same algorithm is then employed by the recipient to check the data it received and ensure that there were no errors.

The checksum calculation used by TCP is a bit different than a regular checksum algorithm. A conventional checksum is performed over all the bytes that the checksum is intended to protect, and can detect most bit errors in any of those fields. The designers of TCP wanted this bit error protection, but also desired to protect against other type of problems.

5.10 Flow control

TCP uses an end-to-end flow control protocol to avoid having the sender send data too fast for the TCP receiver to receive and process it reliably. Having a mechanism for flow control is essential in an environment where machines of diverse network speeds communicate. For example, if a PC sends data to a smartphone that is slowly processing received data, the smartphone must regulate the data flow so as not to be overwhelmed.

TCP uses a sliding window flow control protocol. In each TCP segment, the receiver specifies in the receive window field the amount of additionally received data (in bytes) that it is willing to buffer for the connection. The sending host can send only up to that amount of data before it must wait for an acknowledgment and window update from the receiving host.

TCP sequence numbers and receive windows behave very much like a clock. The receive window shifts each time the receiver receives and acknowledges a new segment of data. Once it runs out of sequence numbers, the sequence number loops back to 0.

When a receiver advertises a window size of 0, the sender stops sending data and starts the persist timer. The persist timer is used to protect TCP from a deadlock situation that could arise if a subsequent window size update from the receiver is lost, and the sender cannot send more data until receiving a new window

size update from the receiver. When the persist timer expires, the TCP sender attempts recovery by sending a small packet so that the receiver responds by sending another acknowledgement containing the new window size.

If a receiver is processing incoming data in small increments, it may repeatedly advertise a small receive window. This is referred to as the silly window syndrome, since it is inefficient to send only a few bytes of data in a TCP segment, given the relatively large overhead of the TCP header.

5.11 Congestion control

The final main aspect of TCP is congestion control. TCP uses a number of mechanisms to achieve high performance and avoid congestion collapse, where network performance can fall by several orders of magnitude. These mechanisms control the rate of data entering the network, keeping the data flow below a rate that would trigger collapse. They also yield an approximately max-min fair allocation between flows. Acknowledgments for data sent, or lack of acknowledgments, are used by senders to infer network conditions between the TCP sender and receiver. Coupled with timers, TCP senders and receivers can alter the behavior of the flow of data. This is more generally referred to as congestion control and/or network congestion avoidance.

Modern implementations of TCP contain four intertwined algorithms: Slow-start, congestion avoidance, fast retransmit, and fast recovery (RFC 5681).

In addition, senders employ a retransmission timeout (RTO) that is based on the estimated round-trip time (or RTT) between the sender and receiver, as well as the variance in this round trip time. The behavior of this timer is specified in RFC 6298. There are subtleties in the estimation of RTT. For example, senders must be careful when calculating RTT samples for retransmitted packets; typically they use Karn's Algorithm or TCP timestamps (see RFC 1323). These individual RTT samples are then averaged over time to create a Smoothed Round Trip Time (SRTT) using Jacobson's algorithm. This SRTT value is what is finally used as the round-trip time estimate.

Enhancing TCP to reliably handle loss, minimize errors, manage congestion and go fast in very high-speed environments are ongoing areas of research and standards development. As a result, there are a number of TCP congestion avoidance algorithm variations.

5.12 Delayed Binding

Delayed binding, also called TCP connection splicing, is the postponement of the connection between the client and the server in order to obtain sufficient information to make a routing decision. Some application switches and routers delay binding the client session to the server until the proper handshakes are complete so as to prevent Denial of Service attacks.

6 Application Layer

6.1 The HTTP Protocol

What is HTTP?

HTTP stands for Hypertext Transfer Protocol. It's the network protocol used to deliver virtually all files and other data (collectively called resources) on the World Wide Web, whether they're HTML files, image files, query results, or anything else. Usually, HTTP takes place through TCP/IP sockets.

A browser is an HTTP client because it sends requests to an HTTP server (Web server), which then sends responses back to the client. The standard (and default) port for HTTP servers to listen on is 80, though they can use any port.

What are "Resources"?

HTTP is used to transmit resources, not just files. A resource is some chunk of information that can be identified by a URL (it's the R in URL). The most common kind of resource is a file, but a resource may also be a dynamically-generated query result, the output of a CGI script, a document that is available in several languages, or something else.

While learning HTTP, it may help to think of a resource as similar to a file, but more general. As a practical matter, almost all HTTP resources are currently either files or server-side script output.

6.2 Structure of HTTP Transactions

Like most network protocols, HTTP uses the client-server model: An HTTP client opens a connection and sends a request message to an HTTP server; the server then returns a response message, usually containing the resource that was requested. After delivering the response, the server closes the connection (making HTTP a stateless protocol, i.e. not maintaining any connection information between transactions).

The formats of the request and response messages are similar, and English-oriented. Both kinds of messages consist of:

- *an initial line,*
- *zero or more header lines,*
- *a blank line (i.e. a CRLF by itself), and*
- *an optional message body (e.g. a file, or query data, or query output).*

Put another way, the format of an HTTP message is:

<initial line, different for request vs. response>
Header1: value1
Header2: value2
Header3: value3

<optional message body goes here, like file contents or query data;
it can be many lines long, or even binary data $&*%@!^$@>

Initial lines and headers should end in CRLF, though you should gracefully handle lines ending in just LF. (More exactly, CR and LF here mean ASCII values 13 and 10, even though some platforms may use different characters.)

Initial Request Line

The initial line is different for the request than for the response. A request line has three parts, separated by spaces: **a method name**, **the local path of the requested resource**, and the **version of HTTP** being used. A typical request line is:

GET /path/to/file/index.html HTTP/1.0

Notes:

- *GET is the most common HTTP method; it says "give me this resource". Other methods include POST and HEAD-- more on those later. Method names are always uppercase.*
- *The path is the part of the URL after the host name, also called the request URI (a URI is like a URL, but more general).*
- *The HTTP version always takes the form "**HTTP/x.x**", uppercase.*

Initial Response Line (Status Line)

The initial response line, called the status line, also has three parts separated by spaces: the HTTP version, a response status code that gives the result of the request, and an English reason phrase describing the status code. Typical status lines are:

HTTP/1.0 200 OK

or

HTTP/1.0 404 Not Found

Notes:

- *The HTTP version is in the same format as in the request line, "**HTTP/x.x**".*
- *The status code is meant to be computer-readable; the reason phrase is meant to be human-readable, and may vary.*
- *The status code is a three-digit integer, and the first digit identifies the general category of response:*

  1xx indicates an informational message only

  2xx indicates success of some kind

  3xx redirects the client to another URL

  4xx indicates an error on the client's part

  5xx indicates an error on the server's part

The most common status codes are:

200 OK	The request succeeded, and the resulting resource (e.g. file or script output) is returned in the message body.
301	Moved Permanently
302	Moved Temporarily
303 See Other (HTTP 1.1 only)	The resource has moved to another URL (given by the Location: response header), and should be automatically retrieved by the client. This is often used by a CGI script to redirect the browser to an existing file.
403 Forbidden	The request was a valid request, but the server is refusing to respond to it.
404 Not Found	The requested resource doesn't exist.
500 Internal Server Error	An unexpected server error. The most common cause is a server-side script that has bad syntax, fails, or otherwise can't run correctly.
503 Service Unavailable	The server is currently unavailable (because it is overloaded or down for maintenance). Generally, this is a temporary state.

Header Lines

Header lines provide information about the request or response, or about the object sent in the message body. The header lines are in the usual text header format, which is: one line per header, of the form "Header-Name: value", ending with CRLF. It's the same format used for email and news postings.

6.3 Other HTTP Methods, Like HEAD and POST

Besides GET, the two most commonly used methods are HEAD and POST.

The HEAD Method

A HEAD request is just like a GET request, except it asks the server to return the response headers only, and not the actual resource (i.e. no message body). This is useful to check characteristics of a resource without actually downloading it, thus saving bandwidth. Use HEAD when you don't actually need a file's contents.

The response to a HEAD request must never contain a message body, just the status line and headers.

The POST Method

A POST request is used to send data to the server to be processed in some way, like by a CGI script. A POST request is different from a GET request in the following ways:

- *There's a block of data sent with the request, in the message body. There are usually extra headers to describe this message body, like Content-Type: and Content-Length:.*
- *The request URI is not a resource to retrieve; it's usually a program to handle the data you're sending.*
- *The HTTP response is normally program output, not a static file.*

The most common use of POST, by far, is to submit HTML form data to CGI scripts. In this case, the Content-Type: header is usually application/x-www-form-urlencoded, and the Content-Length: header gives the length of the URL-encoded form data (here's a note on URL-encoding). The CGI script receives the message body through STDIN, and decodes it. Here's a typical form submission, using POST:

POST /path/script.cgi HTTP/1.0
From: frog@jmarshall.com
User-Agent: HTTPTool/1.0
Content-Type: application/x-www-form-urlencoded
Content-Length: 32

home=Cosby&favorite+flavor=flies

You can use a POST request to send whatever data you want, not just form submissions. Just make sure the sender and the receiving program agree on the format.

The GET method can also be used to submit forms. The form data is URL-encoded and appended to the request URI.

6.4 HTTP version 1.1

Like many protocols, HTTP is constantly evolving. HTTP 1.1 has recently been defined, to address new needs and overcome shortcomings of HTTP 1.0. Generally speaking, it is a superset of HTTP 1.0. Improvements include:

- *Faster response, by allowing multiple transactions to take place over a single persistent connection.*
- *Faster response and great bandwidth savings, by adding cache support.*
- *Faster response for dynamically-generated pages, by supporting chunked encoding, which allows a response to be sent before its total length is known.*
- *Efficient use of IP addresses, by allowing multiple domains to be served from a single IP address.*

Host: Header

Starting with HTTP 1.1, one server at one IP address can be multi-homed, i.e. the home of several Web domains. For example, "www.host1.com" and "www.host2.com" can live on the same server.

Several domains living on the same server is like several people sharing one phone: a caller knows who they're calling for, but whoever answers the phone doesn't. Thus, every HTTP request must specify which host name (and possibly port) the request is intended for, with the Host: header. A complete HTTP 1.1 request might be

GET /path/file.html HTTP/1.1
Host: www.host1.com:80
[blank line here]

except the ":**80**" isn't required, since that's the default HTTP port.

Host: is the only required header in an HTTP 1.1 request. It's also the most urgently needed new feature in HTTP 1.1. Without it, each host name requires a unique IP address, and we're quickly running out of IP addresses with the explosion of new domains.

Persistent Connections and the "Connection: close" Header

In HTTP 1.0 and before, TCP connections are closed after each request and response, so each resource to be retrieved requires its own connection. Opening and closing TCP connections takes a substantial amount of CPU time, bandwidth, and memory. In practice, most Web pages consist of several files on the same server, so much can be saved by allowing several requests and responses to be sent through a single persistent connection.

Persistent connections are the default in HTTP 1.1, so nothing special is required to use them. Just open a connection and send several requests in series (called pipelining), and read the responses in the same order as the requests were sent. If you do this, be very careful to read the correct length of each response, to separate them correctly.

If a client includes the "Connection: close" header in the request, then the connection will be closed after the corresponding response. Use this if you don't support persistent connections, or if you know a request will be the last on its connection. Similarly, if a response contains this header, then the server will close the connection following that response, and the client shouldn't send any more requests through that connection.

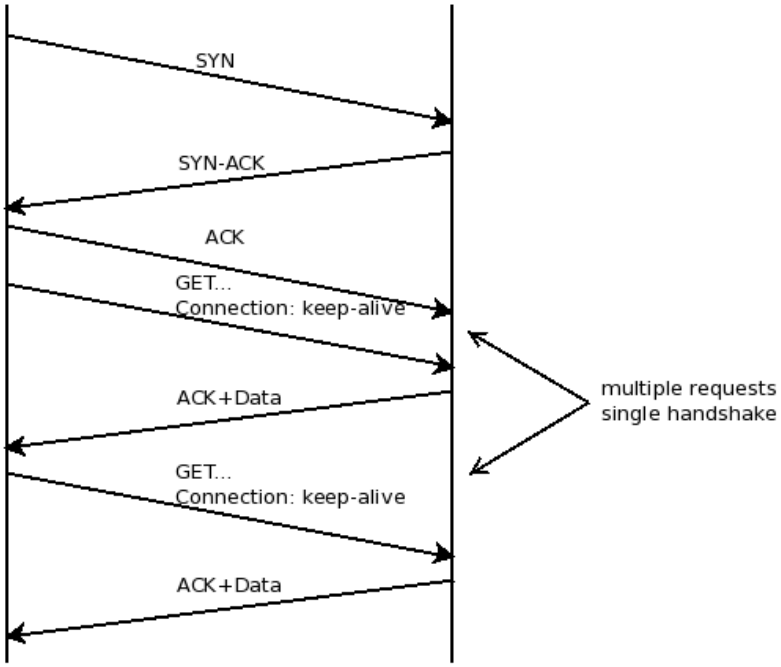
A server might close the connection before all responses are sent, so a client must keep track of requests and resend them as needed. When resending, don't pipeline the requests until you know the connection is persistent. Don't pipeline at all if you know the server won't support persistent connections (like if it uses HTTP 1.0, based on a previous response).

Persistent connection is also called keep-alive connection or HTTP connection reuse.

6.5 Keep-Alive

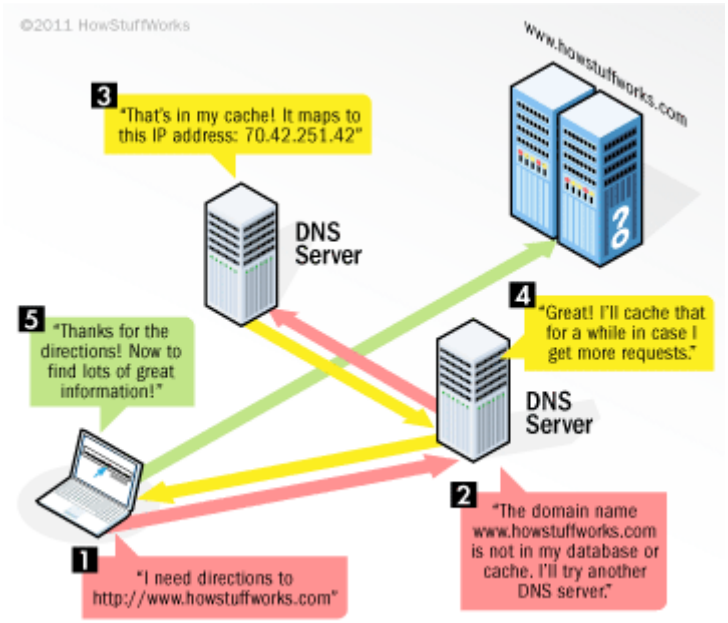
HTTP keep-alive, also called HTTP persistent connection, or HTTP connection reuse, is the idea of using a single TCP connection to send and receive multiple HTTP requests/responses, as opposed to opening a new connection for every single request/response pair.

The Keep-Alive header field and the additional information it provides are optional and do not need to be present to indicate a persistent connection has been established.



6.6 Domain Name System

If you've ever used the Internet, it's a good bet that you've used the Domain Name System, or DNS, even without realizing it. DNS is a protocol within the set of standards for how computers exchange data on the Internet and on many private networks, known as the TCP/IP protocol suite. Its basic job is to turn a user-friendly domain name like "howstuffworks.com" into an Internet Protocol (IP) address like 70.42.251.42 that computers use to identify each other on the network. It's like your computer's GPS for the Internet.



Computers and other network devices on the Internet use an IP address to route your request to the site you're trying to reach. This is similar to dialing a phone number to connect to the person you're trying to call. Thanks to DNS, though, you don't have to keep your own address book of IP addresses. Instead, you just connect through a domain name server, also called a DNS server or name server, which manages a massive database that maps domain names to IP addresses.

Whether you're accessing a Web site or sending e-mail, your computer uses a DNS server to look up the domain name you're trying to access. The proper term for this process is DNS name resolution, and you would say that the DNS server resolves the domain name to the IP address. For example, when you enter "http://www.howstuffworks.com" in your browser, part of the network connection includes resolving the domain name "howstuffworks.com" into an IP address, like 70.42.251.42, for HowStuffWorks' Web servers.

You can always bypass a DNS lookup by entering 70.42.251.42 directly in your browser (give it a try). However, you're probably more likely to remember "howstuffworks.com" when you want to return later. In addition, a Web site's IP address can change over time, and some sites associate multiple IP addresses with a single domain name.

Without DNS servers, the Internet would shut down very quickly. But how does your computer know what DNS server to use? Typically, when you connect to your home network, Internet service provider (ISP) or WiFi network, the modem or router that assigns your computer's network address also sends some important network configuration information to your computer or mobile device.

That configuration includes one or more DNS servers that the device should use when translating DNS names to IP address.

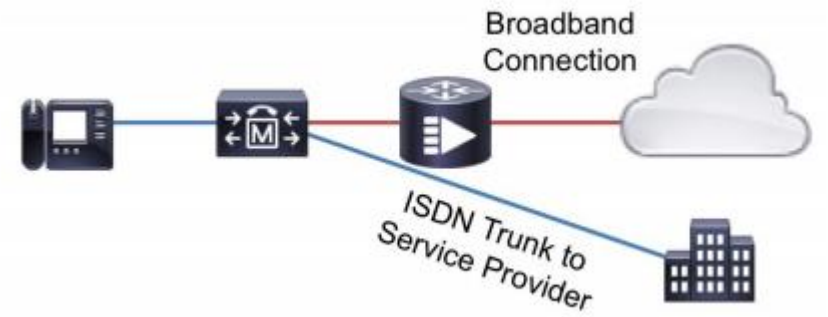
6.7 What Is Session Initiation Protocol or SIP?

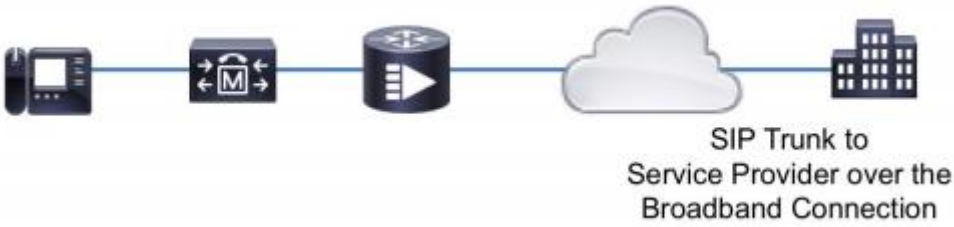
Session Initiation Protocol (SIP) is a communications protocol used for communicating between different devices on a company network, whether on the LAN, the WAN, or across the Internet. An example of this could be a simple two-way phone conversation, using voice over IP (VoIP) on the LAN or WAN or a SIP trunk across the Internet to a service provider. A SIP trunk provides a new way of connecting to a service provider for incoming and outgoing calls; it is a connection over the Internet instead of a traditional telephone connection such as ISDN.

SIP allows you to take full advantage of applications such as video conferencing, presence, and instant messaging. These applications, and others like them, when working together are known as unified communications. Unified communications opens up a new world of possibilities in how you interact with your customers and prospects, giving them a richer experience when dealing with you and your staff.

SIP provides businesses many benefits over older, proprietary telephony solutions, and best of all, it can save you money.

In the past, connections to the service provider (telephone company) were possible only using a dedicated telephone line, such as an ISDN connection.





6.8 What is FTP – File Transfer Protocol?

File Transfer Protocol (FTP) is a standard network protocol used to transfer files from one host to another host over a TCP-based network, such as the Internet. FTP is built on a client-server architecture and uses separate control and data connections between the client and the server. FTP users may authenticate themselves using a clear-text sign-in protocol, normally in the form of a username and password, but can connect anonymously if the server is configured to allow it. For secure transmission that hides (encrypts) the username and password, and encrypts the content, FTP is often secured with SSL/TLS ("FTPS"). SSH File Transfer Protocol ("SFTP") is sometimes also used instead, but is technologically different.

The first FTP client applications were command-line applications developed before operating systems had graphical user interfaces, and are still shipped with most Windows, Unix, and Linux operating systems. Dozens of FTP clients and automation utilities have since been developed for desktops, servers, mobile devices, and hardware, and FTP has been incorporated into hundreds of productivity applications, such as Web page editors.

6.9 Active FTP vs. Passive FTP

One of the most commonly seen questions when dealing with firewalls and other Internet connectivity issues is the difference between active and passive FTP and how best to support either or both of them. Hopefully the following text will help to clear up some of the confusion over how to support FTP in a firewalled environment.

The Basics

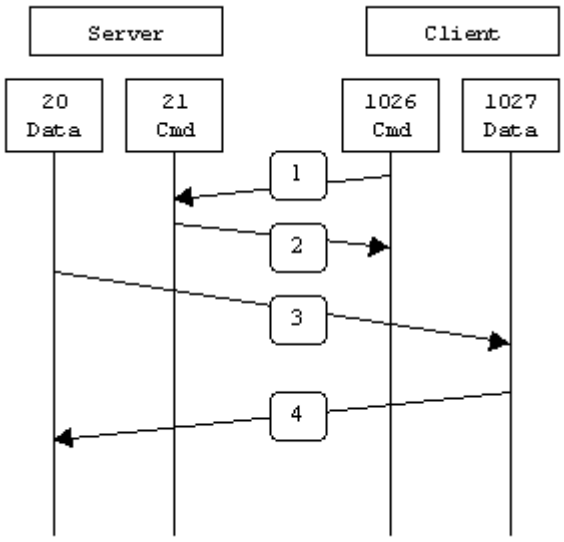
FTP is a TCP based service exclusively. There is no UDP component to FTP. FTP is an unusual service in that it utilizes two ports, a 'data' port and a 'command' port (also known as the control port). Traditionally these are port 21 for the command port and port 20 for the data port. The confusion begins however, when we find that depending on the mode, the data port is not always on port 20.

Active FTP

In active mode FTP the client connects from a random unprivileged port (N > 1023) to the FTP server's command port, port 21. Then, the client starts listening to port N+1 and sends the FTP command PORT N+1 to the FTP server. The server will then connect back to the client's specified data port from its local data port, which is port 20.

From the server-side firewall's standpoint, to support active mode FTP the following communication channels need to be opened:

FTP server's port 21 from anywhere (Client initiates connection)
FTP server's port 21 to ports > 1023 (Server responds to client's control port)
FTP server's port 20 to ports > 1023 (Server initiates data connection to client's data port)
FTP server's port 20 from ports > 1023 (Client sends ACKs to server's data port)
When drawn out, the connection appears as follows:



- *The client's command port contacts the server's command port and sends the command PORT 1027.*
- *The server then sends an ACK back to the client's command port*
- *The server initiates a connection on its local data port to the data port the client specified earlier.*
- *Finally, the client sends an ACK.*

The main problem with active mode FTP actually falls on the client side. The FTP client doesn't make the actual connection to the data port of the server--it simply tells the server what port it is listening on and the server connects back to the specified port on the client. From the client-side firewall this appears to be an outside system initiating a connection to an internal client--something that is usually blocked.

Passive FTP

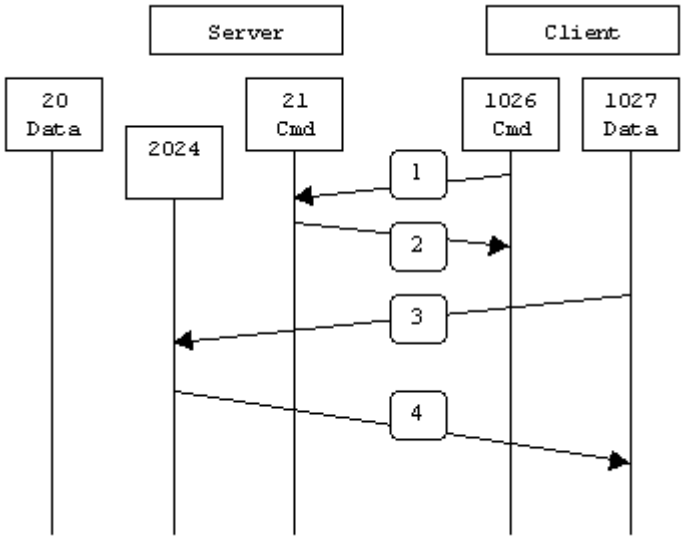
In order to resolve the issue of the server initiating the connection to the client a different method for FTP connections was developed. This was known as passive mode, or PASV, after the command used by the client to tell the server it is in passive mode.

In passive mode FTP the client initiates both connections to the server, solving the problem of firewalls filtering the incoming data port connection to the client from the server. When opening an FTP connection, the client opens two random unprivileged ports locally (N > 1023 and N+1). The first port contacts the server on port 21, but instead of then issuing a PORT command and allowing the server to connect back to its data port, the client will issue the PASV command. The result of this is that the server then opens a random unprivileged port (P > 1023) and sends P back to the client in response to the PASV command. The client then initiates the connection from port N+1 to port P on the server to transfer data.

From the server-side firewall's standpoint, to support passive mode FTP the following communication channels need to be opened:

- *FTP server's port 21 from anywhere (Client initiates connection)*
- *FTP server's port 21 to ports > 1023 (Server responds to client's control port)*
- *FTP server's ports > 1023 from anywhere (Client initiates data connection to random port specified by server)*
- *FTP server's ports > 1023 to remote ports > 1023 (Server sends ACKs (and data) to client's data port)*

When drawn, a passive mode FTP connection looks like this:



- *The client contacts the server on the command port and issues the PASV command.*
- *The server then replies with PORT 2024, telling the client which port it is listening to for the data connection.*
- *The client then initiates the data connection from its data port to the specified server data port.*
- *Finally, the server sends back an ACK to the client's data port.*

While passive mode FTP solves many of the problems from the client side, it opens up a whole range of problems on the server side. The biggest issue is the need to allow any remote connection to high numbered ports on the server. Fortunately, many FTP daemons, including the popular WU-FTPD allow the administrator to specify a range of ports which the FTP server will use.

The second issue involves supporting and troubleshooting clients which do (or do not) support passive mode. As an example, the command line FTP utility provided with Solaris does not support passive mode, necessitating a third-party FTP client, such as ncftp.

6.10 The SMTP Server

Whenever you send a piece of e-mail, your e-mail client interacts with the SMTP server to handle the sending. The SMTP server on your host may have conversations with other SMTP servers to deliver the e-mail. Let's assume that I want to send a piece of e-mail. My e-mail ID is brain, and I have my account on howstuffworks.com. I want to send e-mail to jsmith@mindspring.com. I am using a stand-alone e-mail client like Outlook Express.

When I set up my account at howstuffworks, I told Outlook Express the name of the mail server -- mail.howstuffworks.com. When I compose a message and press the Send button, here's what happens:

- *Outlook Express connects to the SMTP server at mail.howstuffworks.com using port 25.*

- *Outlook Express has a conversation with the SMTP server, telling the SMTP server the address of the sender and the address of the recipient, as well as the body of the message.*

- *The SMTP server takes the "to" address (jsmith@mindspring.com) and breaks it into two parts: the recipient name (jsmith) and the domain name (mindspring.com). If the "to" address had been another user at howstuffworks.com, the SMTP server would simply hand the message to the POP3 server for howstuffworks.com (using a little program called the delivery agent). Since the recipient is at another domain, SMTP needs to communicate with that domain.*

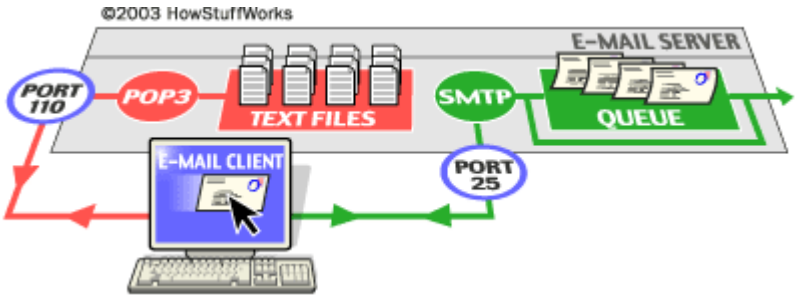
- *The SMTP server has a conversation with a Domain Name Server, or DNS. It says, "Can you give me the IP address of the SMTP server for mindspring.com?" The DNS replies with the one or more IP addresses for the SMTP server(s) that Mindspring operates.*

- *The SMTP server at howstuffworks.com connects with the SMTP server at Mindspring using port 25. It has the same simple text conversation that my e-mail client had with the SMTP server for HowStuffWorks, and gives the message to the Mindspring server. The Mindspring server recognizes that the domain name for jsmith is at Mindspring, so it hands the message to Mindspring's POP3 server, which puts the message in jsmith's mailbox.*

If, for some reason, the SMTP server at HowStuffWorks cannot connect with the SMTP server at Mindspring, then the message goes into a queue. The SMTP server on most machines uses a program called sendmail to do the actual sending, so this queue is called the sendmail queue. Sendmail will periodically try to resend the messages in its queue. For example, it might retry every 15 minutes. After four hours, it will usually send you a piece of mail that tells you there is some sort of problem. After five days, most sendmail configurations give up and return the mail to you undelivered.

The SMTP server understands very simple text commands like HELO, MAIL, RCPT and DATA. The most common commands are:

- ***HELO*** *- introduce yourself*
- ***EHLO*** *- introduce yourself and request extended mode*
- ***MAIL FROM****: - specify the sender*
- ***RCPT TO****: - specify the recipient*
- ***DATA*** *- specify the body of the message (To, From and Subject should be the first three lines.)*
- ***RSET*** *- reset*
- ***QUIT*** *- quit the session*
- ***HELP*** *- get help on commands*
- ***VRFY*** *- verify an address*
- ***EXPN*** *- expand an address*
- ***VERB*** *– verbose*



6.11 How Internet Cookies Work

In April of 2000 I read an in-depth article on Internet privacy in a large, respected newspaper, and that article contained a definition of cookies. Paraphrasing, the definition went like this:

> *"Cookies are programs that Web sites put on your hard disk. They sit on your computer gathering information about you and everything you do on the Internet, and whenever the Web site wants to it can download all of the information the cookie has collected."* *[wrong]*

Definitions like that are fairly common in the press. The problem is, none of that information is correct. Cookies are not programs, and they cannot run like programs do. Therefore, they cannot gather any information on their own. Nor can they collect any personal information about you from your machine.

Here is a valid definition of a cookie: A cookie is a piece of text that a Web server can store on a user's hard disk. Cookies allow a Web site to store information on a user's machine and later retrieve it. The pieces of information are stored as name-value pairs.

For example, a Web site might generate a unique ID number for each visitor and store the ID number on each user's machine using a cookie file.

If you use Microsoft's Internet Explorer to browse the Web, you can see all of the cookies that are stored on your machine. The most common place for them to reside is in a directory called c:windowscookies. When I look in that directory on my machine, I find 165 files. Each file is a text file that contains name-value pairs, and there is one file for each Web site that has placed cookies on my machine.

You can see in the directory that each of these files is a simple, normal text file. You can see which Web site placed the file on your machine by looking at the file name (the information is also stored inside the file). You can open each file by clicking on it.

For example, I have visited goto.com, and the site has placed a cookie on my machine. The cookie file for goto.com contains the following information:

UserID A9A3BECE0563982D www.goto.com/

Goto.com has stored on my machine a single name-value pair. The name of the pair is UserID, and the value is A9A3BECE0563982D. The first time I visited goto.com, the site assigned me a unique ID value and stored it on my machine.

(Note that there probably are several other values stored in the file after the three shown above. That is housekeeping information for the browser.)

The vast majority of sites store just one piece of information -- a user ID -- on your machine. But a site can store many name-value pairs if it wants to.

A name-value pair is simply a named piece of data. It is not a program, and it cannot "do" anything. A Web site can retrieve only the information that it has placed on your machine. It cannot retrieve information from other cookie files, nor any other information from your machine.

How do Web sites use cookies?

Cookies evolved because they solve a big problem for the people who implement Web sites. In the broadest sense, a cookie allows a site to store state information on your machine. This information lets a Web site remember what state your browser is in. An ID is one simple piece of state information -- if an ID exists on your machine, the site knows that you have visited before. The state is, "Your browser has visited the site at least one time," and the site knows your ID from that visit.

Web sites use cookies in many different ways. Here are some of the most common examples:

Sites can accurately determine how many people actually visit the site. It turns out that because of proxy servers, caching, concentrators and so on, the only way for a site to accurately count visitors is to set a cookie with a unique ID for each visitor. Using cookies, sites can determine how many visitors arrive, how many are new versus repeat visitors and how often a visitor has visited. Sites can store user preferences so that the site can look different for each visitor (often referred to as customization). For example, if you visit msn.com, it offers you the ability to "change content/layout/color." It also allows you to enter your zip code and get customized weather information. When you enter your zip code, the following name-value pair gets added to MSN's cookie file:

WEAT CC=NC%5FRaleigh%2DDurham®ION= www.msn.com/

- *Since I live in Raleigh, N.C., this makes sense.*
- *Most sites seem to store preferences like this in the site's database and store nothing but an ID as a cookie, but storing the actual values in name-value pairs is another way to do it.*

E-commerce sites can implement things like shopping carts and "quick checkout" options. The cookie contains an ID and lets the site keep track of you as you add different things to your cart. Each item you add to your shopping cart is stored in the site's database along with your ID value.

When you check out, the site knows what is in your cart by retrieving all of your selections from the database. It would be impossible to implement a convenient shopping mechanism without cookies or something like them.

In all of these examples, note that what the database is able to store is things you have selected from the site, pages you have viewed from the site, information you have given to the site in online forms, etc. All of the information is stored in the site's database, and in most cases, a cookie containing your unique ID is all that is stored on your computer.

6.12 URL - Uniform Resource Locator

URL stands for Uniform Resource Locator. A URL is a formatted text string used by Web browsers, email clients and other software to identify a *network resource* on the Internet. Network resources are files that can be plain Web pages, other text documents, graphics, or programs.
URL strings consist of three parts (substrings):

1. *network protocol*
2. *host name or address*
3. *file or resource location*

These substrings are separated by special characters as follows:

protocol :// host / location

URL Protocol

The 'protocol' substring defines a network protocol to be used to access a resource. These strings are short names followed by the three characters '*://*' (a simple naming convention to denote a protocol definition). Typical URL protocols include *http://*, *ftp://*, and *mailto://*.
URL Host

The 'host' substring identifies a computer or other network device. Hosts come from standard Internet databases such as DNS and can be names or IP addresses. For example, compnetworking.about.com is the host for this Web page.

URL Location

The 'location' substring contains a path to one specific network resource on the host. Resources are normally located in a host directory or folder. For example, */od/internetaccessbestuses/bldef-url.htm* is the location of this Web page including two subdirectories and the file name.
When the location element is omitted such as in *http://compnetworking.about.com/*, the URL conventionally points to the root directory of the host and often a home page (like 'index.htm').

Absolute vs. Relative URLs

Full URLs featuring all three substrings are called *absolute* URLs. In some cases such as within Web pages, URLs can contain only the one location element. These are called *relative* URLs. Relative URLs are used for efficiency by Web servers and a few other programs when they already know the correct URL protocol and host.

7 F5 Solutions and Technology

7.1 BIG-IP Access Policy Manager

Today, business resources, such as applications and data, are accessed inside and outside the traditional business perimeter. Local and remote employees, partners, and customers often access applications without context or security. A central policy control point delivers access based on context and is critical to managing a scalable, secure, and dynamic environment.

BIG-IP® Access Policy Manager™ (APM) is a flexible, high-performance access and security solution that provides unified global access to your applications and network. By converging and consolidating remote access, LAN access, and wireless connections within a single management interface, and providing easy-to-manage access policies, BIG-IP APM helps you free up valuable IT resources and scale cost-effectively

Provide unified global access

BIG-IP APM protects your public-facing applications by providing policy-based, context-aware access to users while consolidating your access infrastructure. It also provides secure remote access to corporate resources from all networks and devices. BIG-IP APM is the most scalable remote access solution and it is IPv6 ready.

Obtain flexibility, high performance,and scalability

BIG-IP APM is available in three deployment options—as an add-on module for
BIG-IP® Local Traffic Manager™ (LTM)for protecting Internet-facing applications, delivered in BIG-IP® Edge Gateway™for accelerated remote access, and run on BIG-IP LTM Virtual Edition to deliver flexible application access in virtualized environments.

Consolidate and simplify

BIG-IP APM helps you consolidate your infrastructure and simplify access management by providing centralized authentication, authorization, and accounting (AAA) control directly on the BIG-IP system. BIG-IP APM integrates with Oracle Access Manager. It simplifies virtual application deployment by supporting Citrix XenApp and XenDesktop, VMware View, Microsoft Remote Desktop Protocol (RDP), and Java RDP, all in one webtop. With BIG-IP APM, you can consolidate and unify elements such as access, security, and policy management to help further reduce costs.

The purpose of the Access Policy Manager is to create a secure access to internal applications by using a single authentication and provide control using a single management interface.

7.2 BIG-IP Application Security Manager

F5 BIG-IP® Application Security Manager™ (ASM) is a flexible web application firewall that secures web applications in traditional, virtual, and private cloud environments. BIG-IP ASM provides unmatched web application and website protection, helps secure deployed applications against unknown vulnerabilities,

and enables compliance for key regulatory mandates—all on a platform that consolidates application delivery with data center firewall capabilities, and network and application access control.

Deliver comprehensive security

BIG-IP ASM blocks web application attacks in minutes to help protect against a broad spectrum of threats, including the latest distributed denial-of-service (DDoS) and SQL injection attacks. It also helps secure interactive web applications that use the latest coding, such as AJAX widgets and JSON payloads. Advanced vulnerability assessment integrations can scan web applications and BIG-IP ASM patches vulnerabilities in minutes to help protect against web threats. BIG-IP ASM stops hackers and attacks from any location and ensures that legitimate users can access applications.

The purpose of the Application Security Manager is to secure web applications using a certified web application firewall and offer threat assessment and visibility.

7.3 BIG-IP Local Traffic Manager

BIG-IP® Local Traffic Manager™ (LTM) turns your network into an agile infrastructure for application delivery. It's a full proxy between users and application servers, creating a layer of abstraction to secure, optimize, and load balance application traffic. This gives you the flexibility and control to add applications and servers easily, eliminate downtime, improve application performance, and meet your security requirements.

Easily deploy applications and ensure availability

BIG-IP LTM gives you the industry's most advanced load balancing and application health monitoring capabilities. F5 iApp™enables you to easily deploy and manage applications using user-defined application centric configuration templates. The associated application related statistics provide complete visibility into the health and performance of your apps for faster troubleshooting and resolution of issues.

Accelerate your applications up to 3x

BIG-IP LTM reduces traffic volumes and minimizes the effect of client connection bottlenecks as well as WAN, LAN, and Internet latency to improve application and replication performance up to 3x. You can achieve additional performance increases with BIG-IP Web Accelerator and BIG-IP WAN Optimization Manager.

Take control over application delivery

The F5 TMOS® platform gives you complete control of the connection, packets, and
payload for applications. Using F5's eventdriven iRules,® you can customize how you intercept, inspect, transform, and direct inbound and outbound application traffic. The F5 iControl® API makes it easy to integrate with third-party management systems.Device Service Clustering provides flexible high-availability scaling and configuration syncing of live application traffic among a cluster of active or standby BIG-IP devices

Reduce servers, bandwidth, and management costs

Advanced TCP connection management, TCP optimization, and server offloading enable you to optimize the utilization of your existing infrastructure—tripling server capacity and reducing bandwidth costs by up to 80 percent. BIG-IP LTM helps you simplify system management by consolidating security, acceleration, and

availability in one easy-to-manage platform. By using fewer servers, less bandwidth, less power, and less cooling, while reducing the time spent managing your infrastructure, you can significantly reduce your operational costs.

The purpose of the Local Traffic Manager is to load balance applications in your environment by using advanced TCP connection management, TCP optimization and server offloading and also provides a high security solution. The LTMs iApps functionality is a powerful set of features that enable you to manage application services rather than individual devices and objects.

7.4 BIG-IP Global Traffic Manager

F5® BIG-IP® Global Traffic Manager™ (GTM) distributes DNS and user application requests based on business policies, data center and network conditions, user location, and application performance. BIG-IP GTM delivers F5's high-performance DNS Services with visibility, reporting, and analysis; scales and secures DNS responses geographically to survive DDoS attacks; delivers a complete, real-time DNSSEC solution; and ensures global application high availability.

Control and ensure app availability
BIG-IP GTM helps you create a strong disaster recovery and business continuity plan by ensuring that users are always connected to a site where the application is available. In addition to performing comprehensive health checking of the entire infrastructure, BIG-IP GTM minimizes downtime and improves the user experience by determining health at the application layer for every user.

Improve application performance
BIG-IP GTM enables you to send users to a site that will give them the best application experience. It uses a range of different load balancing methods and intelligent monitoring for each specific application and user. Traffic is routed according to your business policies and current network and user conditions. BIG-IP GTM includes an accurate, granular geolocation database, giving you control of traffic distribution based on the user's location. By providing persistence for stateful applications, BIG-IP GTM helps you eliminate broken sessions and corrupted data.

The purpose of the Global Traffic Manager is to ensure availability and access to the applications in your environment by using comprehensive health checks and load balancing methods to determine what site the user should access to get the best application experience.

7.5 BIG-IP Enterprise Manager

F5® Enterprise Manager™ significantly reduces the cost and complexity of managing multiple F5 devices. You gain a single-pane view of your entire application delivery infrastructure and the tools you need to automate common tasks, ensure optimized application performance, and improve budgeting and forecasting to meet changing business needs. Enterprise Manager is available as a physical or virtual edition.

Optimize app and device performance
Gain a full picture of your application performance across the entire F5 infrastructure with the Centralized Analytics Module, available through purchase of an add-on license. Analytics provides real-time application performance statistics such as response time, network latency, and more. Use the Performance Monitoring Module for an up-to-date, comprehensive view of traffic and device performance. With Enterprise Manager, you can set thresholds and alerts to react quickly to changing network conditions and user demands.

Troubleshoot more effectively
Full visibility into your application delivery infrastructure helps you quickly isolate current application performance and traffic management problems as well as those that have grown systemically over time. In addition, F5 BIG-IP® iHealth®integration enables you to proactively manage the health of all your BIG-IP®devices. By automating the diagnostic process and checking for known issues and common mistakes, Enterprise Manager can help you stay current with F5 best practices and optimize your ADN infrastructure.

The purpose of the Enterprise Manager is to provide you with a Full visibility and the health of your application delivery infrastructure. It will provide you with real-time statistics and alarms for you to quickly engage and solve the issues in your environment.

7.6 BIG-IP WAN Optimization Manager

BIG-IP® WAN Optimization Manager™ (WOM) overcomes network and application issues on the WAN to ensure that application performance, data replication, and disaster recovery requirements are met. The high throughput and scalable architecture of BIG-IP WOM can dramatically reduce data replication times and enable more efficient use of your existing bandwidth. These advanced optimization services are available as an add-on module on your F5 BIG-IP® Local Traffic Manager™ device or as a standalone appliance or virtual edition.

Improve bandwidth efficiency
BIG-IP WOM includes symmetric adaptive compression, which automatically applies the appropriate compression algorithm to dramatically reduce the amount of traffic that has to be sent between data centers. Another feature is symmetric data deduplication, which eliminates the transfer of redundant data across the WAN to improve response times and throughput while using less bandwidth. To overcome the inherent protocol limitations of TCP, BIG-IP WOM uses adaptive TCP Express™ optimization, which combines persistent sessions, selective acknowledgements, error correction, and optimized TCP windows. This enables BIG-IP WOM to adapt, in real time, to the latency, packet loss, and congestion characteristics of WAN links, to fully utilize available bandwidth.

BIG-IP WOM speeds up data transfers over the WAN to accelerate large file transfers, data replication (for databases, virtual machine live migration, and Microsoft Exchange mailboxes), and more. By optimizing the protocols associated with these applications, including CIFS, MAPI, HTTP, and others, BIG-IP WOM dramatically reduces the effects of latency on applications running over the WAN.

The purpose of the WAN Optimization Manager is to optimize your WAN connection by using a symmetric adaptive compression and symmetric data deduplication. This will improve the respons times and use less bandwidth.

7.7 BIG-IP WebAccelerator

BIG-IP® WebAccelerator™ automates web performance optimization, instantly improving performance for end users and helping you to reduce costs. By offloading your network and servers, BIG-IP WebAccelerator decreases your spending on additional bandwidth and hardware. Users get fast access to applications, and you gain greater revenue and free up IT resources for other strategic projects.

Web Performance Optimization
BIG-IP WebAccelerator solves web content delivery issues by ensuring the best use of bandwidth and preventing repetitive or duplicate data from being served to users. This speeds up the first and repeat visits to portal, CRM, e-learning, and e-commerce sites. The result is significantly decreased download times, reduced bandwidth usage, and lower costs for using enterprise web applications in remote office and mobile deployments.

BIG-IP WebAccelerator uses three layers of web performance optimizations: server and network optimization, Dynamic Content Control, and Dynamic Data Reduction. These optimizations do not require any server side installations, client side software, or changes to users' browsers

Server and Network Optimization
BIG-IP WebAccelerator improves the capacity of application servers and the efficiency of network protocols by offloading intensive processing tasks such as SSL encryption, optimizing application and network protocols, and supporting new emerging protocols.

Dynamic Content Control
Dynamic Content Control (DCC) is a group of capabilities that control users' browser behavior to improve end user experience, ensure the best use of bandwidth, and prevent repetitive or duplicate data from being downloaded. By reducing the amount of conditional requests and data transmitted between the browser and the web application, DCC reduces the effects of WAN latency and errors.

Dynamic Data Reduction
Dynamic Data Reduction (DDR) reduces bandwidth utilization and improves page load times by reducing the amount of data that needs to traverse the WAN or Internet. F5 BIG-IP WebAccelerator offers the following DDR functions:

- *Image Optimization*
- *Dynamic Caching*
- *Dynamic Compression*

The purpose of the WebAccelerator is to improve the user experience by optimizing performance that in the long-run will decrease of cost for additional network and server hardware. WebAccelerator improve performance by using Server and Network Optimization, Dynamic Content Control and Dynamic Data Reduction.

7.8 BIG IP ARX

F5® ARX® file virtualization devices enable you to dramatically simplify data management and reduce storage costs. By introducing intelligent file virtualization into the file storage infrastructure, ARX eliminates the disruption associated with storage administration and automates many storage management tasks. The result is a dramatic improvement in cost, agility, and business efficiency.

Decrease backup time and costs
Many companies back up all of their files repeatedly, even though only a small
fraction are modified on a regular basis. ARX automated tiering policies decrease the amount of unchanging or non-critical data being backed up regularly, enabling you to dramatically reduce backup and recovery times, backup media consumption, and backup costs, without sacrificing data protection.

Gain storage flexibility and choice
ARX offers you the flexibility to manage your storage infrastructure in the way that best meets your business needs. With ARX, you can move data easily whenever and wherever you want, including between heterogeneous storage devices, without affecting users. Powerful policies help you perform a range of data management tasks—whether it's moving entire file systems or even individual files—automatically and non-disruptively.

The purpose of ARX is to provide a flexible and efficient way of backing up data but also the management of files and filesystems.

8 BIG IP - iRules

8.1 What is an iRule?

An iRule is a script that you write if you want to make use of some of the extended capabilities of the BIG-IP that are unavailable via the CLI or GUI. iRules allow you to more directly interact with the traffic passing through the device. Using iRules, you can send traffic not only to pools, but also to individual pool members, ports, or URIs. And directing traffic to a desired pool is only the beginning. You can parse the entire header and payload of the data as it is being passed through the BIG-IP and, at wire speed, execute an entire script of commands on that traffic. The commands at your disposal range from logging to redirecting traffic, from modifying the URI or port to actually rewriting the payload itself.

The iRules you create can be simple or sophisticated, depending on your content-switching needs. The following shows an example of a simple iRule.

```
1 when CLIENT_ACCEPTED {
2   if { [IP::addr [IP::client_addr] equals 10.10.10.10] } {
3     pool my_pool
4   }
5 }
```

This iRule is triggered when a client-side connection has been accepted, causing the LTM system to send the packet to the pool my_pool, if the client's address matches 10.10.10.10.

Using a feature called the Universal Inspection Engine (UIE), you can write an iRule that searches either a header of a packet, or actual packet content, and then directs the packet based on the result of that search. iRules can also direct packets based on the result of a client authentication attempt.

iRules can direct traffic not only to specific pools, but also to individual pool members, including port numbers and URI paths, either to implement persistence or to meet specific load balancing requirements.

The syntax that you use to write iRules is based on the Tool Command Language (Tcl) programming standard. Thus, you can use many of the standard Tcl commands, plus a robust set of extensions that the LTM system provides to help you further increase load balancing efficiency.

The advantages of iRules is that you extend the capabilities of the BIG-IP that is not available through the CLI or the GUI.

8.2 How does an iRule work?

To start at the beginning, as it were, an iRule is first and foremost a configuration object, in F5 terms. This means that it is a part of your general bigip.conf along with your pools, virtual servers, monitors, etc. It is entered into the system either via the GUI or CLI, generally speaking. There is also an iRules Editor available for download on DevCentral that is a windows tool for editing and deploying/testing iRules which can be extremely useful. Unlike most configuration objects, though, an iRule is completely user generated and cus-

tomizable. An iRule is a script, at its core after all. Regardless of how an iRule gets there, be it UI, CLI or Editor, once an iRule is part of your config, it is then compiled as soon as that configuration is saved.

One of the gross misconceptions about iRules is that, as with most interpreted scripting languages such as TCL, and interpreter must be instantiated every time an iRule is executed to parse the code and process it. This is not true at all, because every time you save your configuration all of your iRules are pre-compiled into what is referred to as "byte code". Byte code is mostly compiled and has the vast majority of the interpreter tasks already performed, so that TMM can directly interpret the remaining object. This makes for far higher performance and as such, increase scalability.

Now that the iRule is saved and pre-compiled, it must then be applied to a virtual server before it can affect any traffic. An iRule that is not applied to a virtual server is effectively disabled, for all intents and purposes. Once you've applied an iRule to a given virtual server, however, it will now technically be applied against all traffic passing through that virtual server. Keep in mind though, that this does not necessarily mean that all traffic passing through the virtual server in question will be affected. IRules are most often very selective in which traffic they affect, be it to modify, re-route or otherwise. This is done through both logical constructs within the iRules, but also through the use of events within the iRule itself.

Events are one of the ways in which iRules have been made to be network aware, as a language. An event, which we'll dig into in much more detail in the next installment of this series, is a way of executing iRules code at a given point in time within the flow of a networking session. If I only want to execute a section of code once for each new connection to the virtual server to which my iRule is applied, I could easily do so by writing some simple code in the appropriate event. Events are also important because they indicate at which point in the proxy chain (sometimes referred to as a hud chain) an iRule executes. Given that BIG-IP is a bi-directional proxy, it is important for iRules to execute on not only the right side of the proxy, but at the right moment in the network flow.

So now you have an iRule added to your configuration, it has been automatically pre-compiled to byte code when the configuration was saved, you have it applied to the appropriate virtual server, and the code within the iRule calls out the desired event in which you want your code to execute; now is when the magic happens, as it were. This is where the massive collection of iRules commands comes into play. From header modification to full on payload replacement to creating a socket connection to an outside system and making a request before processing traffic for your virtual, there are very few limitations to what can be achieved when combining the appropriate series of iRules commands. Those commands are then processed by TMM, which will affect whatever change(s) it needs to the traffic it is processing for the given session, depending on what you've designed your iRule to do. The true power of iRules largely comes into play thanks to the massive array of custom commands that we've built into the language, allowing you to leverage your BIG-IP to the fullest.

8.3 When would I use an iRule?

The ideal time to use an iRule is when you're looking to add some form of functionality to your application or app deployment, at the network layer, and that functionality is not already readily available via the built in configuration options in your BIG-IP. Whether it's looking to perform some kind of custom redirect or logging specific information about users' sessions or a vast array of other possibilities, iRules can add valuable

business logic or even application functionality to your deployment. iRules have a single point of management, your BIG-IP, as opposed to being distributed to every server hosting whichever application you're trying to modify or affect. This can save valuable management time, and can also be a large benefit in time to deployment. It is often far easier to deploy an iRule or an iRule change than it is to modify your application for a quick fix.

As an example, one of the most common uses of iRules when it was first introduced was to redirect all traffic from HTTP (port 80) to HTTPS (port 443) without affecting either the host or the requested URI for the connection. This was (and still is, pictured below) a very simple iRule, but it wasn't at the time a feature available in the standard configuration options for BIG-IP.

```
1: when HTTP_REQUEST {
2:   HTTP::redirect "https://[HTTP::host][HTTP::uri]"
3: }
```

8.4 When would I not use an iRule?

The above example of an HTTP to HTTPS redirect iRule actually depicts perfectly when to not use an iRule, because that functionality was so popular that it has since been added as a profile option directly in the BIG-IP configuration. As such, it is more appropriate and technically higher performance, to use that feature in the profile as opposed to writing an iRule to perform the same task. A general rule of thumb is: Any time you can do something from within the standard config options, profiles, GUI or CLI – do it there first. If you're looking to perform a task that can't be accomplished via the "built-in" means of configuration, then it is a perfect time to turn to iRules to expand the possibilities.

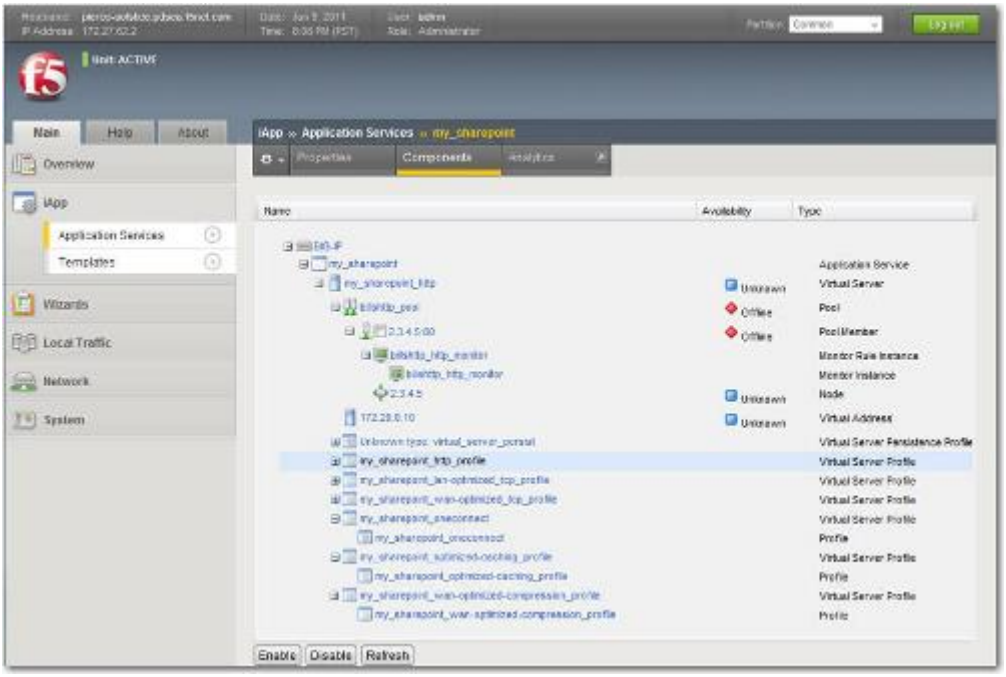
The main reason for this is performance. iRules are extremely high performance as a rule, if written properly. But there is always a slight benefit in performance when you can run functionality directly from built in, core features as opposed to a custom created script, even an iRule. Also, though, it is easier to maintain a feature built into the product through upgrades, rather than re-testing and managing an iRule that could be easily replaced with a few configuration options.

9 BIG IP - iApps

9.1 What's an iApp?

An iApp is a user-customizable framework for deploying applications. It consists of three components: Templates, Application Services, and Analytics. An iApp Template is where the application is described and the objects (required and optional) are defined through presentation and implementation language. An iApp Application Service is the deployment process of an iApp Template which bundles all of the configuration options for a particular application together. You would have an iApp Application Service for SharePoint, for example. iApp Analytics include performance metrics on a per-application and location basis. Benefits of using iApp:

- *User-customizable*
- *Easy editing of configurations and cleanup*
- *Reentrancy*
- *Configuration encapsulation*
- *Cradle-to-grave configuration management*
- *Strictness protects against accidental changes to the configuration*
- *Operational tasks and health status for App objects displayed on App-specific component view (see right)*
- *Copy/Import/Export capability*
- *Community support for DevCentral hosted templates*



9.2 When do you use an iApp?

Two key tenets of iApps are visibility and control. iApps provide unprecedented levels of availability and control across the entire infrastructure; but this is dependent upon their placement at a strategic point of control, where all application delivery logic can be applied dynamically to all bi-directional application traffic with a coordinated mediation layer between the users, the applications, and the data. This layer presents a consistent interface and set of services that can be used for all applications and data regardless of their current location. This layer uses an intelligent understanding of context—who is accessing what from where and why—to determine the optimal connection among users, applications, and data. Finally, it provides a deep understanding of context to inform the underlying application and data elements about the current resource requirements and to instruct the infrastructure to adapt in real time to ensure optimized application and data access.

This strategic flow of application information becomes part of an iApp configuration. Initial configuration with an iApp Template is based on how the application should be delivered in an ideal architecture; the application is managed throughout the delivery lifecycle through iApp Templates and BIG-IP device service clusters, providing the what and where; and finally iApp Analytics brings the why portion, enabling dynamic reconfiguration as the application delivery environment changes and reporting that information back to IT for analysis. iApps are in strategic points of control—the only place that allows such granular and broad application delivery management.

9.3 About iApp Templates

iApps templates create configuration-specific forms used by application services to guide authorized users through complex system configurations. The templates provide programmatic, visual layout and help information. Each new application service uses one of the templates to create a screen with fields and guide the user through the configuration process and creates the configuration when finished.

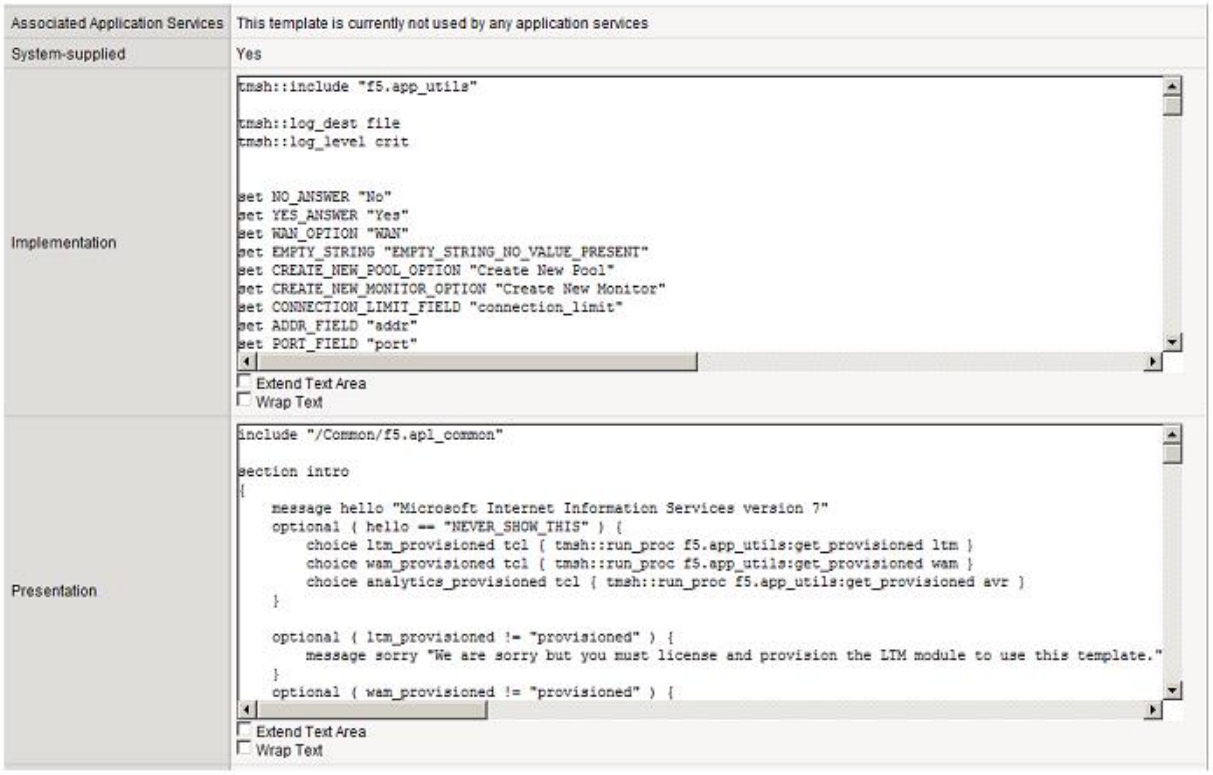
The templates allow users to customize by either modifying an existing template or creating one from scratch. Users can create scratch-built templates using either the iApps Templates screen or any text-editing software

The BIG-IP® system comes with several system-supplied templates that you can use as-is to create your application services. You can also use the system-supplied templates as a starting point for your own templates, or you can write templates from scratch using, variously, tmsh and Tcl for the back-end template implementation section.

Template sections

Templates have three sections; presentation, implementation, and help.

- *The **presentation section** collects user entries.*
- *The **implementation section** uses user entries to build a configuration that will control traffic.*
- *The **help section** documents the template and its presentation to users when creating an application service.*



10 BIG IP - iControl

10.1 What is iControl?

iControl is the first open API that enables applications to work in concert with the underlying network based on true software integration.

Utilizing SOAP/XML to ensure open communications between dissimilar systems, iControl helps F5 customers, leading independent software vendors (ISVs), and Solution Providers realize new levels of automation and configuration management efficiency. Whether monitoring network-level traffic statistics, automating network configuration and management, or facilitating next generation service-oriented architectures, iControl gives organizations the power and flexibility to ensure that applications and the network work together for increased reliability, security, and performance. Further, iControl has proven itself as a valuable technology that can help reduce the cost of managing complex environments.

Dissecting with examples

"iControl is the first open API"

We don't make this statement lightly. iControl was introduced in 2001 which may not seem too long ago, but back then the only way to access a network device was via it's SNMP mib. While that is great for using stock reporting tools like HP OpenView, but try telling a web developer to pull out his ASP.NET SNMP toolkit and write a deployment application... Over the years, there have been several companies out there who have tried to follow our lead, but none to date has an interface that is as extensive and standards compliant as iControl.

"... that enables applications to work in concert with the underlying network based on true software integration".

Here's where the heart of iControl lies. "Applications working in concert with the underlying network". iControl is not only about automation, but real-time dynamic integration between the applications and the network. Management interfaces are most often designed to allow for monitoring and manual configuration control.

Imagine this scenario: Your child is sick in bed with a 104 deg F temperature, and you were helpless to do anything about it until your child's doctor called you when his schedule permitted. Not too good huh? Well, that's what's going on with standard monitoring systems. An outside entity is "polling" your servers on some interval and guessing based on a limited set of information how it's "health" is.

iControl gives the server applications the tools to "call the doctor" (ie, to make a method call to the network device) and make an active change in its health.

"Utilizing SOAP/XML to ensure open communications between dissimilar systems, iControl helps F5 customers, leading independent software vendors (ISVs), and Solution Providers realize new levels of automation and configuration management efficiency."

This is where the fact that iControl is standards based plays in. Since iControl was built on SOAP with the encoding options compatible with most platforms, it works with, as far as we know, all major Web Service based toolkits such as .Net, Java, perl, php, python, ruby, and many others. I would venture to state that almost any company out there has at least one developer proficient in at least one of the above languages ensuring that there is already in-house staff that has the skills to build iControl applications. Try to say the same thing for SNMP!

"Whether monitoring network-level traffic statistics, automating network configuration and management, or facilitating next generation service-oriented architectures, iControl gives organizations the power and flexibility to ensure that applications and the network work together for increased reliability, security, and performance."

We'll have to take this piece by piece. For "monitoring network-level traffic", with the numerous Statistics interfaces in the iControl API, one can get very fine-tuned data on how the network objects are performing - from throughput rates to connection counts, the Statistics interfaces have it all.

As for "automating network configuration and management", nearly 100% of the features in the administration GUI is available from an iControl method, so tasks from adding users, to uploading ssl certificates can be automated, thus allowing organizations to make better use of those network administrators.

"Facilitating next generation service oriented architectures" is directed at how we envision the DataCenter of the future. Grid Networks, Dynamic Services and Service Oriented Architectures are going to become more prevalent and iControl's ability to dynamic manipulate the network makes this DataCenter of the future a feasible goal.

"... work together for increased reliability, security, and performance". With regards to reliability, we have conducted surveys and it is generally agreed that organizations have errors due to "fat fingers" when managing network configurations - "Oh, you meant to take down node 10.10.10.10 and not 10.10.10.01 - oops...". By Automating processes, the odds of typing in a wrong network address, or hitting "Delete" instead of "Enable" goes drastically down and network reliability goes up up and up. As for Security, by limited the amount of users that have access to the network directly, one can mitigate the risk of admin accounts to your devices being leaked. That as well as the fact that iControl is backed by 128bit SSL Encryption, will let you rest assured that your networks are strongly protected. Lastly regarding performance, by allowing the network to "heal" itself, and by enabling systems to automate provisioning of network resources, one can build a system that will be allowed to adapt to stresses at real time, leading to performance increases.

In short terms some say iControl is an API, some say it's a set of web services, some say it's a frameworks, and even some say it's everything including the kitchen sink. Well, most of these are correct. At its core, iControl consists of a client application (something you would write) and a server component (something we provide on our devices). On the client side, iControl is distrubuted to the users in the form of a SDK that includes API documentation on the methods as well as a set of WSDL documents describing the services. On the server side, iControl is a feature on the F5 devices that process the client requests.

10.2 How iControl Works

Before we go any further, I'm going to describe all the technical terms that I'll be using so hopefully when you read further in the article, you understand what I'm talking about.

API (Application Programming Interface)	Basically a way for one application to talk to another.
XML (eXtensible Markup Language)	A text based format with a bunch of less-than, greater-than symbols and text along with a bunch of rules on how they are formatted.
SOAP (Simple Object Access Protocol)	Is an XML based messaging system most commonly runs over the HTTP or HTTPs protocols.
Web Services	A Web 2.0 buzzword for a set of SOAP based services.
WSDL (Web Service Definition Language)	A XML file format specification that allows for one to fully describe the format of their SOAP methods including requests, responses, parameters, return values, and other transport layer settings.
SOAP Toolkit	A language dependent set of libraries uses to create and interpret SOAP responses to/from native code. Some examples are SOAP::Lite for perl, Apache Axis for Java, and the .NET Framework for VB and C#.

The Client Application

To build a client application, one would look at their development environment and determine which language is best suited for their application. One could use Perl or PowerShell for a console based application, C# or VB for Windows, Java for web server applications, etc. Of course most of these languages will work in most of these examples, so it's really up to the user to determine which is best for their needs.

Once the language decision is made, and the associated toolkit is installed and configured, it's time to start implementing the client code. This can be either done automatically by the tools that support client side proxy generation from the iControl WSDL files (.NET, Axis, ...) or manually with the ones that are more dynamic in nature (Perl, Ruby, Python, ...). In this article, I will focus on the languages with native support for

WSDL. In .NET applications, you would use the "wsdl.exe" command line tool to parse the WSDL document and create a set of C# or VB client proxy code that you can use within your programs. In Java, the WSDL2Java tool included with Apache Axis will accomplish the same goals. One could go through this effort if one wanted to, but the smart kids out there are all making use of the iControl Assembly for .NET and Java that have been produced by F5. If you go this route, all you have to do is plug the iControl.dll (or iControl.jar for java) into your project and you are ready to start writing code. The initialization of the client proxy library is configured with the address of the F5 device, port 443 for the HTTPs connection, the URI of the iControl Portal process (/iControl/iControlPortal.cgi) and the user credentials. The authentication mechanism is the same that is used for the Management GUI: HTTP Authenticate headers encrypted over SSL. At this point, one can start making iControl management calls with the generated proxy client code.

The Server Components

The second part of the iControl communication channel is the server side web services. These are implemented in a "Portal" process that runs under that administrative management port on the F5 devices. This Portal, listens for incoming secure and encrypted iControl calls, parses the incoming XML-based SOAP Message into native code, performs the relevant internal operations, and then generates and returns a XML-based SOAP Response to the client application.

iControl means a lot o things to a lot of people, but ultimately it's about allowing automated remote management of F5 devices. As a user, all you need to worry about is the client components. Get the WSDL (or better yet, the iControl assemblies), plug them into your client application, initialize the connection information, and start building your app.

11 The Concise Guide to Proxies

We often mention that the benefits derived from some application delivery controllers are due to the nature of being a full proxy. And in the same breath we might mention reverse, half, and forward proxies, which makes the technology sound more like a description of the positions on a sports team than an application delivery solution. So what does these terms really mean? Here's the lowdown on the different kinds of proxies in one concise guide.

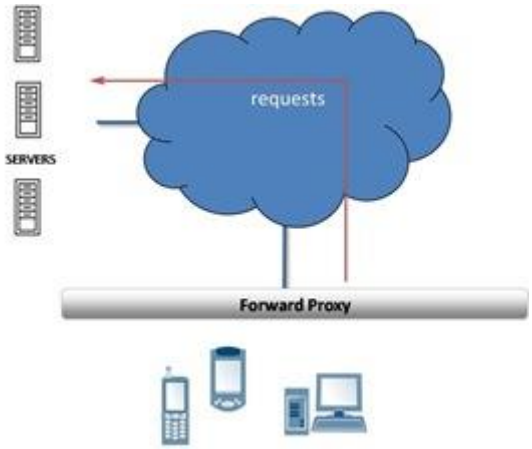
11.1 Proxies

Proxies (often called intermediaries in the SOA world) are hardware or software solutions that sit between the client and the server and do something to requests and sometimes responses. The most often heard use of the term proxy is in conjunction with anonymzing Web surfing. That's because proxies sit between your browser and your desired destination and proxy the connection; that is you talk to the proxy while the proxy talks to the web server and neither you nor the web server know about each other.

Proxies are not all the same. Some are half proxies, some are full proxies; some are forward and some are reverse. Yes, that came excruciatingly close to sounding like a Dr. Seuss book.

11.2 Forward Proxies

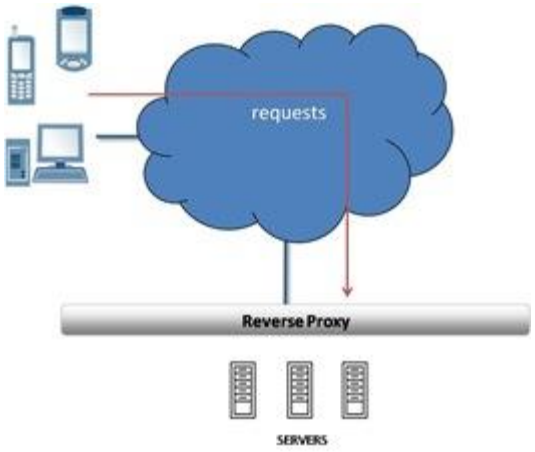
Forward proxies are probably the most well-known of all proxies, primarily because most folks have dealt with them either directly or indirectly. Forward proxies are those proxies that sit between two networks, usually a private internal network and the public Internet. Forward proxies have also traditionally been employed by large service providers as a bridge between their isolated network of subscribers and the public Internet, such as CompuServe and AOL in days gone by. These are often referred to as "mega-proxies" because they managed such high volumes of traffic.



Forward proxies are generally HTTP (Web) proxies that provide a number of services but primarily focus on web content filtering and caching services. These forward proxies often include authentication and authorization as a part of their product to provide more control over access to public content. If you've ever gotten a web page that says "Your request has been denied by blah blah blah. If you think this is an error please contact the help desk/your administrator" then you've probably used a forward proxy.

11.3 Reverse Proxies

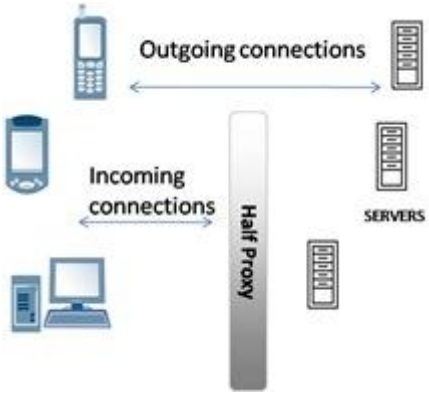
A reverse proxy is less well known, generally because we don't use the term anymore to describe products used as such. Load balancers (application delivery controllers) and caches are good examples of reverse proxies. Reverse proxies sit in front of web and application servers and process requests for applications and content coming in from the public Internet to the internal, private network. This is the primary reason for the name "reverse" proxy - to differentiate it from a proxy that handles outbound requests.



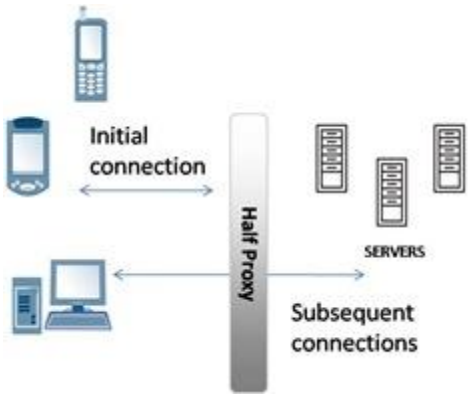
Reverse proxies are also generally focused on HTTP but in recent years have expanded to include a number of other protocols commonly used on the web such as streaming audio (RTSP), file transfers (FTP), and generally any application protocol capable of being delivered via UDP or TCP.

11.4 Half Proxies

Half-proxy is a description of the way in which a proxy, reverse or forward, handles connections. There are two uses of the term half-proxy: one describing a deployment configuration that affects the way connections are handled and one that describes simply the difference between a first and subsequent connections.



The deployment focused definition of half-proxy is associated with a direct server return (DSR) configuration. Requests are proxied by the device, but the responses do not return through the device, but rather are sent directly to the client. For some types of data - particularly streaming protocols - this configuration results in improved performance. This configuration is known as a half-proxy because only half the connection (incoming) is proxied while the other half, the response, is not.



The second use of the term "half-proxy" describes a solution in which the proxy performs what is known as delayed binding in order to provide additional functionality. This allows the proxy to examine the request before determining where to send it. Once the proxy determines where to route the request, the connection between the client and the server are "stitched" together. This is referred to as a half-proxy because the initial TCP handshaking and first requests are proxied by the solution, but subsequently forwarded without interception.

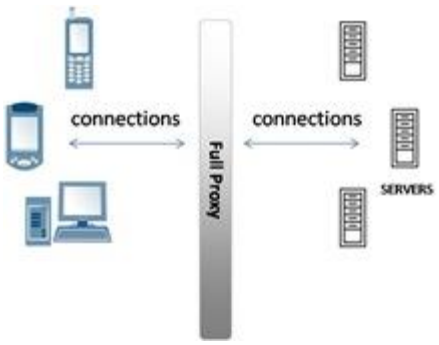
Half proxies can look at incoming requests in order to determine where the connection should be sent and can even use techniques to perform layer 7 inspection, but they are rarely capable of examining the responses. Almost all half-proxies fall into the category of reverse proxies.

11.5 Full Proxies

Full proxy is also a description of the way in which a proxy, reverse or forward, handles connections. A full proxy maintains two separate connections - one between itself and the client and one between itself and the destination server. A full proxy completely understands the protocols, and is itself an endpoint and an originator for the protocols. Full proxies are named because they completely proxy connections - incoming and outgoing.

Because the full proxy is an actual protocol endpoint, it must fully implement the protocols as both a client and a server (a packet-based design does not). This also means the full proxy can have its own TCP connection behavior, such as buffering, retransmits, and TCP options. With a full proxy, each connection is unique; each can have its own TCP connection behavior. This means that a client connecting to the full proxy device would likely have different connection behavior than the full proxy might use for communicating with servers. Full proxies can look at incoming requests and outbound responses and can manipulate both if the solution allows it.

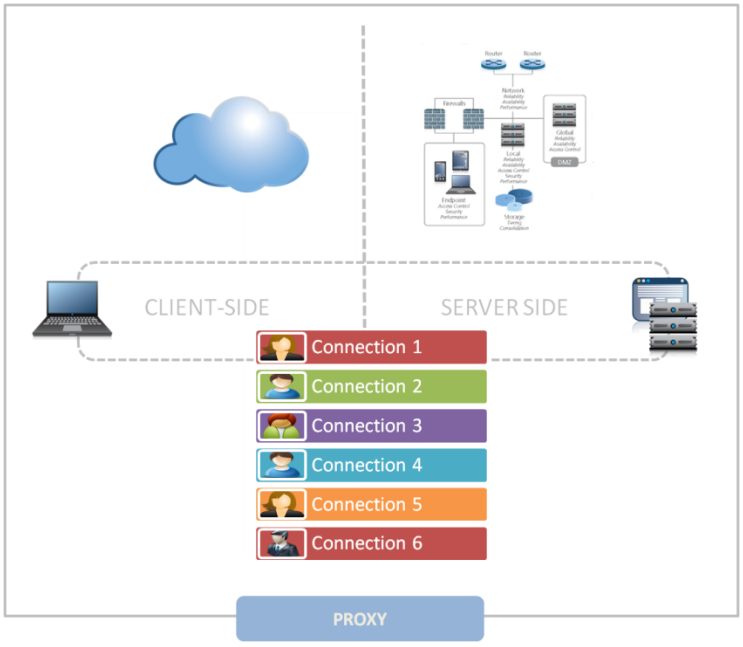
Many reverse and forward proxies use a full proxy model today. There is no guarantee that a given solution is a full proxy, so you should always ask your solution provider if it is important to you that the solution is a full proxy.

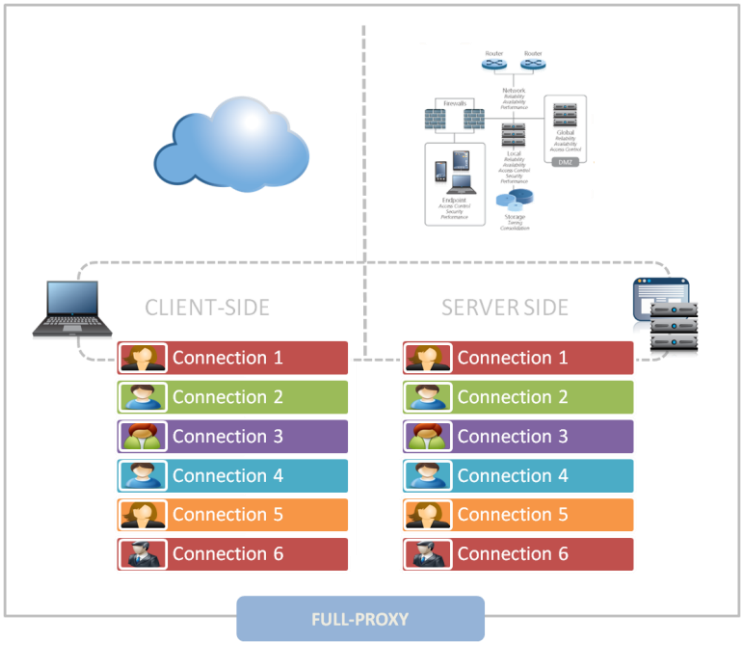


11.6 Full proxy architecture – what do they mean?

The reason there is a distinction made between “proxy” and “full-proxy” stems from the handling of connections as they flow through the device. All proxies sit between two entities – in the Internet age almost always “client” and “server” – and mediate connections. While all full-proxies are proxies, the converse is not true. Not all proxies are full-proxies and it is this distinction that needs to be made when making decisions that will impact the data center architecture.

A full-proxy maintains two separate session tables – one on the client-side, one on the server-side. There is effectively an “air gap” isolation layer between the two internal to the proxy, one that enables focused profiles to be applied specifically to address issues peculiar to each “side” of the proxy. Clients often experience higher latency because of lower bandwidth connections while the servers are generally low latency because they’re connected via a high-speed LAN. The optimizations and acceleration techniques used on the client side are far different than those on the LAN side because the issues that give rise to performance and availability challenges are vastly different.





A full-proxy, with separate connection handling on either side of the “air gap”, can address these challenges. A proxy, which may be a full-proxy but more often than not simply uses a buffer-and-stitch methodology to perform connection management, cannot optimally do so. A typical proxy buffers a connection, often through the TCP handshake process and potentially into the first few packets of application data, but then “stitches” a connection to a given server on the back-end using either layer 4 or layer 7 data, perhaps both. The connection is a single flow from end-to-end and must choose which characteristics of the connection to focus on – client or server – because it cannot simultaneously optimize for both.

The second advantage of a full-proxy is its ability to perform more tasks on the data being exchanged over the connection as it is flowing through the component. Because specific action must be taken to “match up” the connection as its flowing through the full-proxy, the component can inspect, manipulate, and otherwise modify the data before sending it on its way on the server-side. This is what enables termination of SSL, enforcement of security policies, and performance-related services to be applied on a per-client, per-application basis.

This capability translates to broader usage in data center architecture by enabling the implementation of an application delivery tier in which operational risk can be addressed through the enforcement of various policies. In effect, we’re created a full-proxy data center architecture in which the application delivery tier as a whole serves as the “full proxy” that mediates between the clients and the applications.

The full-proxy data center architecture

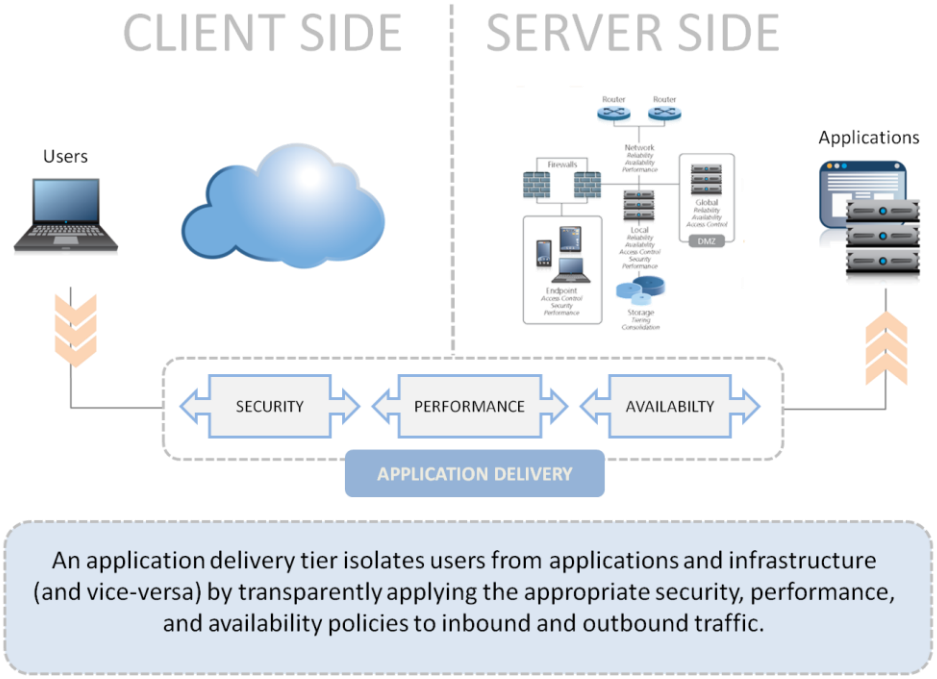
A full-proxy data center architecture installs a digital "air gap” between the client and applications by serving as the aggregation (and conversely disaggregation) point for services. Because all communication is funneled through virtualized applications and services at the application delivery tier, it serves as a strategic point of control at which delivery policies addressing operational risk (performance, availability, security) can be enforced.

A full-proxy data center architecture further has the advantage of isolating end-users from the volatility inherent in highly virtualized and dynamic environments such as cloud computing . It enables solutions such as those used to overcome limitations with virtualization technology, such as those encountered with pod-architectural constraints in VMware View deployments. Traditional access management technologies, for example, are tightly coupled to host names and IP addresses. In a highly virtualized or cloud computing environment, this constraint may spell disaster for either performance or ability to function, or both. By implementing access management in the application delivery tier – on a full-proxy device – volatility is managed through virtualization of the resources, allowing the application delivery controller to worry about details such as IP address and VLAN segments, freeing the access management solution to concern itself with determining whether this user on this device from that location is allowed to access a given resource.

Basically, we’re taking the concept of a full-proxy and expanded it outward to the architecture. Inserting an “application delivery tier” allows for an agile, flexible architecture more supportive of the rapid changes today’s IT organizations must deal with.

Such a tier also provides an effective means to combat modern attacks. Because of its ability to isolate applications, services, and even infrastructure resources, an application delivery tier improves an organizations' capability to withstand the onslaught of a concerted DDoS attack. The magnitude of difference between the connection capacity of an application delivery controller and most infrastructures (and all servers) gives the entire architecture a higher resiliency in the face of overwhelming connections. This ensures better availability and, when coupled with virtual infrastructure that can scale on-demand when necessary, can also maintain performance levels required by business concerns.

A full-proxy data center architecture is an invaluable asset to IT organizations in meeting the challenges of volatility both inside and outside the data center.



12 Packet-based vs. Proxy-based

12.1 What is a packet-based design?

A network device with a packet-based (or packet-by-packet) design is located in the middle of a stream of communications, but is not an endpoint for those communications; it just passes the packets through. Often a device that operates on a packet-by-packet basis does have some knowledge of the protocols flowing through it, but is far from being a real protocol endpoint.

The speed of these devices is primarily based on not having to understand the entire protocol stack, shortcutting the amount of work needed to handle traffic. For example, with TCP/IP, this type of device might only understand the protocols well enough to rewrite the IP addresses and TCP ports; only about half of the entire stack.

As networks became more complex and the need for intelligence increased, more advanced packet-based designs began to emerge (including the BIG-IP® products from F5). These devices knew TCP/IP well enough to understand both TCP connection setup and teardown, modify TCP/IP headers, and even insert data into TCP streams.

Because these systems could insert data into TCP streams and modify the content of the stream, they also had to rewrite the TCP sequence (SEQ) and acknowledgment (ACK) values for packets going back and forth from the client and server. F5's BIG-IP products understood TCP/IP and HTTP well enough to identify individual HTTP requests and could send different requests to different servers, reusing connections the BIG-IP device already had open.

While all of this is possible using a very sophisticated packet-by-packet architecture (BIG-IP devices are some of the most sophisticated of such designs to date), it required a very complex state tracking engine to understand the TCP/IP and HTTP protocols well enough to rewrite header contents, insert data, and maintain its own connections to clients and servers. Despite this increasing complexity, packet-based designs are still less complex and faster than traditional proxy-based designs, as they have the advantage of only requiring a small percentage of the logic required for a full proxy.

12.2 What is a proxy-based design (full proxy)?

A full-proxy design is the opposite of a packet-by-packet design. Instead of having a minimal understanding of the communications streaming through the device, a full proxy completely understands the protocols, and is itself an endpoint and an originator for the protocols. The connection between a client and the full proxy is fully independent of the connection between the full proxy and the server; whereas in a packet-by-packet design, there is essentially a direct communication channel between the client and the server (although the device in the middle may manipulate the packets going back and forth).

Because the full proxy is an actual protocol endpoint, it must fully implement the protocols as both a client and a server (a packet-based design does not). This also means the full proxy can have its own TCP connec-

tion behavior, such as buffering, retransmits, and TCP options. With a full proxy, each connection is unique; each can have its own TCP connection behavior. This means that a client connecting to the full-proxy device would likely have different connection behavior than the full proxy might use for communicating with the backend servers.

Therefore, a full proxy allows for the optimization of every connection uniquely, regardless of the original source and the final destination. Further, a full proxy understands and processes each protocol as a real client or server would, using layers. Using HTTP as an example, first the IP protocol is processed, then TCP, then HTTP; and each layer has no knowledge of the lower layers.

12.3 Redefining the Solution

It is common knowledge that proxy-based solutions, or at least the intelligence offered by them, were the ultimate solution. However, the vastly superior performance of packet-by-packet designs more than made up for their limited intelligence. For a while, this was an acceptable trade-off for most enterprise networks. As the need for increased intelligence grows, packet-based solutions are quickly experiencing the same performance restrictions that proxy-based solutions have always suffered from. And the development complexity of packet-based solutions is quickly approaching that of proxy-based designs as well. Despite dramatic increases in hardware and software power, packet-by-packet designs are unable to keep up with the need for intelligence and performance. It is no longer acceptable to have to choose between them. While packet-based solutions had their time, that time is gone. It is now readily apparent that short-cutting intelligence in lieu of performance did not adequately provide a viable solution. The real solution is to build a proxy-based solution with the performance of the packet-based solution.

13 High availability

13.1 Single device

When you are running the BIG-IP system as a single device (as opposed to a unit of a redundant system), high availability refers to core services being up and running on that device, and VLANs being able to send and receive traffic.

13.2 Redundant system configuration

When you are running the BIG-IP system as a unit of a redundant system configuration, high availability refers to core system services being up and running on one of the two BIG-IP systems in the configuration. High availability also refers to a connection being available between the BIG-IP system and a pool of routers, and VLANs on the system being able to send and receive traffic.

A redundant system is a type of BIG-IP system configuration that allows traffic processing to continue in the event that a BIG-IP system becomes unavailable. A BIG-IP redundant system consists of two identically-configured BIG-IP units. When an event occurs that prevents one of the BIG-IP units from processing network traffic, the peer unit in the redundant system immediately begins processing that traffic, and users experience no interruption in service.

You can configure the units of a redundant system to run in one of two redundancy modes: ***active/standby*** or ***active-active***.

To enable a unit to fail over to its peer unit, you must first specify the type of failover that you want the redundant system to use. The two possible failover types are hard-wired failover and network-based failover.

- *Hard-wired failover*

  When you configure hard-wired failover, you enable failover by using a failover cable to physically connect the two redundant units. This is the default setting. For the procedure on configuring hard-wired failover, see the appropriate platform guide.

- *Network failover*

  When you configure network failover, you enable failover by configuring your redundant system to use the network to determine the status of the active unit. You can use network failover in addition to, or instead of, hard-wired failover.

13.3 Understanding active-standby redundancy

An active-standby pair is a pair of BIG-IP devices configured so that one device is actively processing traffic while the other device remains ready to take over if failover occurs. The two devices synchronize their configuration data and can fail over to one another in the event that one of the devices becomes unavailable.

First, you can run the Setup utility on each device to configure base network components (that is, a management port, administrative passwords, and the default VLANs and their associated self IP addresses). Continue running it on each device to establish a trust relationship between the two devices, and create a Sync-Failover type of device group that contains two member devices.

After the Setup utility is run on both devices, each device contains the default traffic group that the BIG-IP system automatically created during setup. A traffic group represents a set of configuration objects (such as floating self IP addresses and virtual IP addresses) that process application traffic. This traffic group actively processes traffic on one of the two devices, making that device the active device. When failover occurs, the traffic group will become active on (that is, float to) the peer BIG-IP device.

By default, the traffic group contains the floating self IP addresses of the default VLANs. Whenever you create additional configuration objects such as self IP addresses, virtual IP addresses, and SNATs, the system automatically adds these objects to the default traffic group.

13.4 What is failover?

Failover is a process that occurs when one system in a redundant system becomes unavailable, thereby causing the peer unit to assume the processing of traffic originally targeted for the unavailable unit. To facilitate coordination of the failover process, each unit has a unit ID (1 or 2).

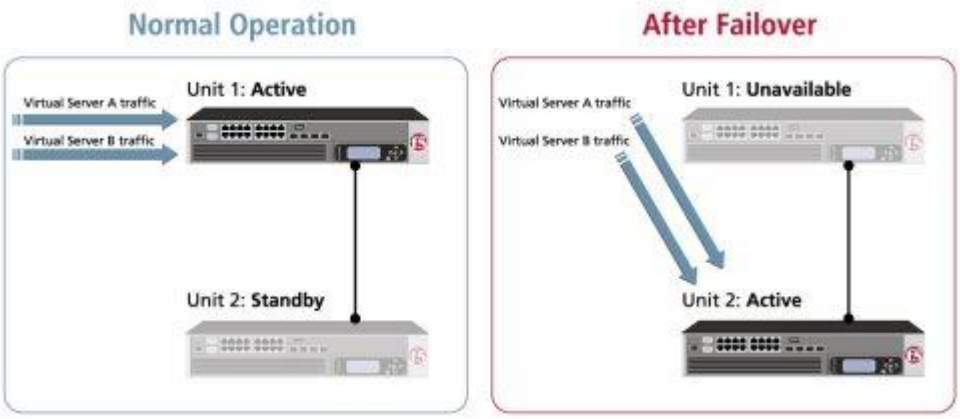
An essential element to making failover successful is a feature called configuration synchronization. Configuration synchronization, or ***ConfigSync***, is a process where you replicate one units main configuration file on the peer unit. Because data is shared in this way, a unit can process the other units traffic when failover occurs.

By default, the way that a BIG-IP unit monitors the status of its peer, in order to detect that failover is required, is through a hard-wired connection between the two BIG-IP units. With proper configuration, however, you can cause each BIG-IP unit to monitor peer status by way of a TCP/IP network connection instead.

13.5 Understanding failover in active/standby mode

When a redundant system is in active/standby mode, one unit is active, that is, accepting and processing connections on behalf of the redundant system, while the other unit is idle (that is, in a standby state). When failover occurs, the standby unit becomes active, and it normally stays active until failover occurs again, or until you force it into a standby state. Forcing the unit into a standby state automatically causes the other system to become active again, if possible.

For example, you can configure unit 1 to process traffic for virtual servers A and B. The standby unit monitors the active unit, and if communications fail, the standby unit initiates a failover and becomes the active unit. The newly-active unit then begins processing traffic for both virtual servers.
You can see an active/standby configuration, first as it behaves normally, and then after failover has occurred, by viewing the figure.



As you can see in the figure, unit 1 is in an active state, and unit 2 is in a standby state. With this configuration, failover causes the following to occur:

- *Unit 2 switches to an active state.*
- *Unit 2 begins processing the connections that would normally be processed by its peer.*

When the failed unit becomes available again, you can force a unit to change its state from active to standby or from standby to active, thereby initiating failback. Failback on an active/standby system causes a unit to relinquish any processing that it is doing on behalf of its peer, and return to a standby state. A redundant system in active/standby mode is the most common type of redundant system.

13.6 Understanding active-active redundancy

For certain network environments, you might want to configure an active-active configuration. The basic configuration procedure is similar to the configuration procedure for an active-standby configuration, except that you must set the redundancy mode on both units to Active.

The remainder of this appendix explains some concepts about active-active configurations, the procedures for controlling failback and changing the redundancy mode, and some special tasks and considerations.

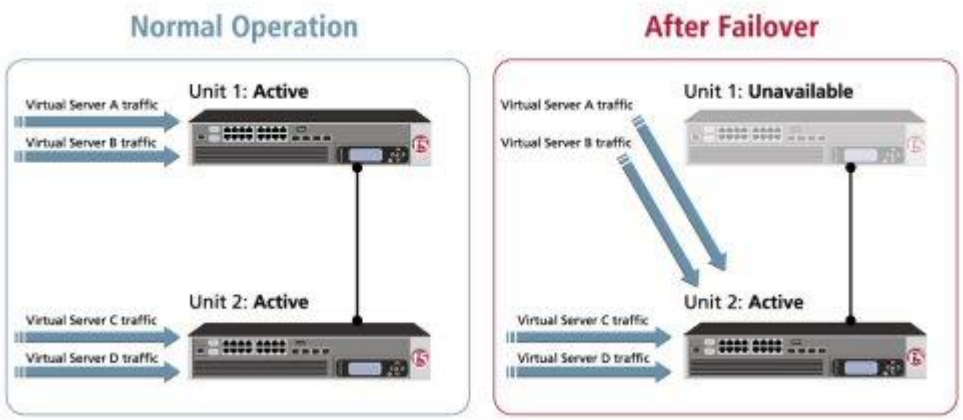
Important: For active-active systems, you must configure network failover. Hard-wired failover alone is not sufficient.

13.7 Understanding failover in active-active mode

An active/active configuration offers the same protection from interruption of service as an active/standby configuration, and can offer decreased latency because traffic is handled by both systems simultaneously. However, to plan for the possibility of failure, each system in an active/active configuration should process no more than 50% of the load that would be carried in an active/standby configuration.

A common active-active configuration is one in which each unit processes connections for different virtual servers. For example, you can configure unit 1 to process traffic for virtual servers A and B, and configure unit 2 to process traffic for virtual servers C and D. If unit 1 becomes unavailable, unit 2 begins processing traffic for all four virtual servers.

You can see an active-active configuration, first as it behaves normally, and then after failover has occurred, by viewing the Figure



The figure shows an active-active configuration in which units 1 and 2 are both in active states. With this configuration, failover causes the following to occur:

- *Unit 2 (already in an active state) begins processing the connections that would normally be processed by unit 1.*
- *Unit 2 continues processing its own connections, in addition to those of unit 1.*

Part of configuring an active-active configuration is to assign a unit ID to each self IP address and virtual address that processes application traffic. For example, if unit 1 normally processes traffic for virtual servers A and B, then you must assign unit ID 1 to the floating self IP addresses pertaining to virtual servers A and B, as well as to the virtual addresses for virtual servers A and B. You can do this using the Configuration utility, by displaying the properties of a floating self IP address or virtual address and then selecting the relevant unit ID.

Likewise, if unit 2 normally processes traffic for virtual servers C and D, then you must assign unit ID 2 to the floating self IP addresses pertaining to virtual servers C and D, as well as to the virtual addresses for virtual servers C and D.

Later, if unit 1 fails over to unit 2, each floating self IP address and virtual address retains its own unit ID regardless of the actual unit on which those objects are running. That is, the floating self IP addresses and virtual addresses for virtual servers A and B retain their unit ID 1 designation while running on unit 2.

When unit 1 becomes available again, you can initiate failback, which, in this case, means that the currently active unit relinquishes any processing that it is doing on behalf of its peer, and continues to operate in an active state, processing its own connections. From this point on, each unit in the active-active system handles its own unit-specific processing.

13.8 Controlling failback for active-active mode

With an active-active configuration, when a failover condition occurs, one unit takes over processing for the other. Then, by default, when the failed unit becomes available again, failback occurs automatically. That is, the recovered unit resumes processing for its own virtual servers, with no user intervention required.
There are two cases, however, where failback does not occur automatically, and you must initiate failback manually:

- *If you manually initiated failover using the bigpipe failover standby command. In this case, failback can only occur when you manually initiate it.*

- *If you configured the bigdb key Failover.ManFailBack key to require manually-initiated failback at all times. By default this key is set to disable, which ensures that the failback process is automatic. If you set this key to enable, you must always initiate failback manually, even when the failover event occurred automatically.*

  In either case, when a failed unit becomes available again, the peer unit continues to process connections for both units until you manually initiate failback.

13.9 Understanding static self IP addresses for network failover

A **static self IP address** is an IP address that you assign to a BIG-IP system VLAN. On any BIG-IP device, you normally assign a unique static self IP address to VLAN **internal**, and another self IP address to VLAN **external**, when you run the Setup utility on that device.

When you set up an active/standby system configuration that uses network failover, F5 recommends that you create an additional VLAN (with a static self IP address) on each unit of the redundant system, to be used specifically for failover communication. Consequently, during normal redundant-system operation prior to failover, the two units use these static self IP addresses to continually communicate with one another about system status. These static self IPs are known as **failover addresses**, that is, addresses that the two units use to communicate with one another, before, during, and after failover.

Additionally, you can specify another static self IP address for that failover VLAN, on each unit that the BIG-IP system, that the system can use for network mirroring.

13.10 Understanding network mirroring

The failover process of a redundant system ensures that a BIG-IP system is always available to process connections with no discernible interruption in service at the time of the failure event. However, failover does not preserve the open connections at the moment of failover; connections are dropped when an active unit becomes unavailable, unless you have configured network mirroring.

The network mirroring feature on the BIG-IP system duplicates a units state (that is, real-time connection and persistence information) on the peer unit. When network mirroring is enabled, failover can be so seamless that file transfers can proceed uninterrupted and your servers can continue with their tasks at the time of failover.

13.11 Understanding fail-safe

Fail-safe is the ability of a BIG-IP system to monitor certain aspects of the system or network, detect interruptions, and consequently take some action, such as rebooting or initiating failover to the peer unit. Fail-safe can apply to: system services, traffic between the BIG-IP system and a gateway router, or VLAN traffic, and pertains to both single devices and redundant system configurations.

14 Persistent and Persistence, What's the Difference?

The English language is one of the most expressive, and confusing, in existence. Words can have different meaning based not only on context, but on placement within a given sentence. Add in the twists that come from technical jargon and suddenly you've got words meaning completely different things. This is evident in the use of persistent and persistence.

While the conceptual basis of persistence and persistent are essentially the same, in reality they refer to two different technical concepts.

Both persistent and persistence relate to the handling of connections. The former is often used as a general description of the behavior of HTTP and, necessarily, TCP connections, though it is also used in the context of database connections. The latter is most often related to TCP/HTTP connection handling but almost exclusively in the context of load-balancing.

14.1 Persistent

Persistent connections are connections that are kept open and reused. The most commonly implemented form of persistent connections is HTTP, with database connections a close second.

Persistent HTTP connections were implemented as part of the HTTP 1.1 specification as a method of improving the efficiency of HTTP in general. Before HTTP 1.1 a browser would generally open one connection per object on a page in order to retrieve all the appropriate resources. As the number of objects in a page grew, this became increasingly inefficient and significantly reduced the capacity of web servers while causing browsers to appear slow to retrieve data. HTTP 1.1 and the Keep-Alive header in HTTP 1.0 were aimed at improving the performance of HTTP by reusing TCP connections to retrieve objects. They made the connections persistent such that they could be reused to send multiple HTTP requests using the same TCP connection.

Similarly, this notion was implemented by proxy-based load-balancers as a way to improve performance of web applications and increase capacity on web servers. Persistent connections between a load-balancer and web servers is usually referred to as TCP multiplexing. Just like browsers, the load-balancer opens a few TCP connections to the servers and then reuses them to send multiple HTTP requests.

Persistent connections, both in browsers and load-balancers, have several advantages:

- *Less network traffic due to less TCP setup/teardown. It requires no less than 7 exchanges of data to set up and tear down a TCP connection, thus each connection that can be reused reduces the number of exchanges required resulting in less traffic.*

- *Improved performance. Because subsequent requests do not need to setup and tear down a TCP connection, requests arrive faster and responses are returned quicker. TCP has built-in mechanisms, for example window sizing, to address network congestion. Persistent connections give TCP the time to adjust itself appropriately to current network conditions, thus improving overall performance. Non-persistent connections are not able to adjust because they are open and almost immediately closed.*
- *Less server overhead. Servers are able to increase the number of concurrent users served because each user requires fewer connections through which to complete requests.*

14.2 Persistence

Persistence, on the other hand, is related to the ability of a load-balancer or other traffic management solution to maintain a virtual connection between a client and a specific server.

Persistence is often referred to in the application delivery networking world as "stickiness" while in the web and application server demesne it is called "server affinity". Persistence ensures that once a client has made a connection to a specific server that subsequent requests are sent to the same server. This is very important to maintain state and session-specific information in some application architectures and for handling of SSL-enabled applications.

When the first request is seen by the load-balancer it chooses a server. On subsequent requests the load-balancer will automatically choose the same server to ensure continuity of the application or, in the case of SSL, to avoid the compute intensive process of renegotiation. This persistence is often implemented using cookies but can be based on other identifying attributes such as IP address. Load-balancers that have evolved into application delivery controllers are capable of implementing persistence based on any piece of data in the application message (payload), headers, or at in the transport protocol (TCP) and network protocol (IP) layers.

Some advantages of persistence are:

- *Avoid renegotiation of SSL. By ensuring that SSL enabled connections are directed to the same server throughout a session, it is possible to avoid renegotiating the keys associated with the session, which is compute and resource intensive. This improves performance and reduces overhead on servers.*

- *No need to rewrite applications. Applications developed without load-balancing in mind may break when deployed in a load-balanced architecture because they depend on session data that is stored only on the original server on which the session was initiated. Load-balancers capable of session persistence ensure that those applications do not break by always directing requests to the same server, preserving the session data without requiring that applications be rewritten.*

14.3 Summary

So persistent connections are connections that are kept open so they can be reused to send multiple requests, while persistence is the process of ensuring that connections and subsequent requests are sent to the same server through a load-balancer or other proxy device.

Both are important facets of communication between clients, servers, and mediators like load-balancers, and increase the overall performance and efficiency of the infrastructure as well as improving the end-user experience.

15 Synchronizing configuration data

15.1 What is configuration synchronization?

When you synchronize data from one unit to another, you are giving the target unit the data that it needs to assume traffic processing for its peer when failover occurs. Examples of configuration data that a target unit receives during configuration synchronization are virtual servers, SNATs, and floating IP addresses.

When you have a redundant system configuration, it is essential that each unit shares, or synchronizes, its current configuration data with its peer unit. If a unit does not share its configuration data with its peer, the surviving unit cannot process traffic for that peer unit. For this reason, you must synchronize configuration data when you initially configure the redundant system, and then repeatedly, on an ongoing basis. The need to repeatedly synchronize data is because a units configuration data typically changes over time during normal system maintenance, such as when you add a virtual server or create a new profile, and the unit must share those changes with its peer.
With respect to configuration synchronization, you can use the Configuration utility to:

- *View or specify the peer IP address to use for synchronization*
- *Enable or disable encryption of configuration data prior to synchronization*
- *Enable or disable the global display of synchronization status*
- *Specify synchronization direction, that is, whether to synchronize data to the peer unit or from the peer unit*

15.2 Understanding configuration synchronization

You can synchronize configuration data from the current system to the peer system, or from the peer system to the current system. The method you choose depends on which units data has most recently changed and which unit you are currently configuring.

Normally, the BIG-IP system uses the peers failover address as the address to which to synchronize the configuration. As an option, however, you can specify a peer address other than the peers failover address.
When you perform configuration synchronization, the BIG-IP system copies a .ucs file containing configuration data from one unit to the other. You must synchronize the configuration data before you can put any redundant system into operation.

Prior to synchronizing configuration data, the BIG-IP system creates a backup configuration file on the target system, named **cs_backup.ucs**, as a preventative measure.

You synchronize configuration data using the ConfigSync screen that is available from the Redundancy Properties screen of the Configuration utility. The ConfigSync screen contains several settings. Two of these settings, ConfigSync User and Synchronize, cause the actual configuration synchronization between redundant units to occur.

16 Difference between Server and Client

Computers are needed in businesses of different sizes. Large computer setups that include networks and mainframes are used in large businesses. A computer network used in these types of businesses has a client-server architecture or two-tier architecture. The main purpose of this architecture is the division of labor which is required in large organizations.

16.1 Server

In client-server environment, the server computer acts as the “brains” of the business. A very large capacity computer is used as a server. There can be a mainframe also as it stores a wide variety of functionalities and data.

Generally, applications and data files are stored on the server computer. Employee computers or work-stations access these applications and files across the network. For example, an employee can access com-pany’s data files stored on the server, from his/her client computer.

In some cases, employees may access only specific applications from their client machine. Application server is the name given to this type of server. The client-server architecture is fully utilized in this type of environ-ment as employees have to login from their client machine in order to access the application stored on the server. For example, these kinds of applications include graphic design programs, spreadsheets and word processors. The client- server architecture is illustrated in each case.

Apart from the storage medium, the server also acts as a processing power source. The client machines get their processing power from this server source. By doing so, no extra hardware for the client is needed and it utilizes greater processing power of the server.

16.2 Client

In client- server architecture, the client acts a smaller computer that is used by the employees of the organi-zation in order to perform their day to day activities. The employee uses the client computer in order to ac-cess the data files or applications stored on the server machine.

The rights authorized to the client machine can be different. Some employees have the access to data files of the organization while other may only access the applications present on the server.

Apart from using the applications and data files, the client machine can also utilize the processing power of the server. In this case, the client computer is plugged-in to the server and the server machine handles all the calculations. In this way, the large processing power of the server can be utilized without any addition of hardware on the client side.

The best example of client- server architecture is WWW or World Wide Web. Here the client is the browser installed on each computer and the information about different pages is stored on the server side from which the client or the user can access it.

Difference between client and server

•Client is a smaller computer through which the information or application stored on the server is accessed by the user whereas server is a powerful computer that stores the data files and applications.

•In some cases, the client may utilize the greater processing power of the server machine.

•In some cases, the client side may have a better graphical user interface or GUI as compared to the server side.

17 Positive & Negative Security model

After many years of purely negative security provided by anti-virus scanners, IDS/IPS, and antispam engines, it's refreshing to hear that the positive security model—the basis for tried and true security devices like network firewalls and ACLs—is coming back in vogue. Most recently, this positive policy re-emergence has revolved around the Web Application Firewall (WAF) and application security market. Yet with the positive security positioning comeback carries with it a very interesting point of detail: although many in the WAF space argue that the positive model is preferable, nearly all application security providers still rely on a partially negative solution. While acknowledging that a positive security model is the preferable model to secure web applications, many practitioners and vendors advocate a bilateral approach of both positive and negative security.

As the application security market continues to evolve and define itself, there continues to be diverging views on which security methodology is the best option. In reality, enterprise security decisions are highly dependent on many factors, most of which are more business than technology oriented. Implementing an application security solution that is both secure and practical—while still allowing for the fluid nature of protecting dynamic applications—requires taking the best pieces of technology and business analysis and synthesizing them into an effective and efficient security solution.

17.1 What Does "Good" Security Cost?

In theory, the best security is impenetrable, but practical security does not function as a control group. In a business environment, security is a multivariate problem. What is the performance of the security? How easy is it to deploy? What impact will adding security have on the cost per transaction? Is it more expensive to build an impenetrable security system or risk covering the cost of a public breach? The quality of the security is always questioned as well, but it's never the only question. Many security-related questions come from the balance sheets, not the security engineers. Approaching security from a technical standpoint alone does not help the business; it hurts it.

Businesses constantly analyze their economic model to generate better operational efficiencies and a greater return on investment; the entire business intelligence market exists for this purpose. The driving force behind any IT security decision is an evaluation of a situation's potential risks versus the investment necessary to circumvent these risks. In the same vein, a business' security efforts should address a business problem; namely, to increase operational efficiencies. Security breaches can mar this efficiency, hurting a company's value, either in real dollars, operational downtime, or loss of customer trust. In truth, the motivation behind every IT decision (including security decisions) is a business decision. This has been stated and fully advocated by Gartner as well:

> *Jay Heiser, a Gartner vice president, said the fundamental problem with a purely technical approach is that IT security professionals have no understanding of business. Speaking at [the] Gartner IT Security Summit in London, Heiser said businesses must now mature and appoint individuals who understand the complexities of business, rather than the simplicities of security.*

When IT decisions become business decisions, blurring the distinction between secure value and business value, theoretical security and applied, or practical, security begin to separate. The theoretical approach places security on a singular plane, untouched by other business factors; applied security is comprised of the measure and level of actual security safeguards and implementations needed to accomplish business goals.

For a security product to be a functional part of a business' IT infrastructure, the product's applied security must be given more attention than the product's theoretical security. A solution that only strives to provide ultimate security will almost always be replaced with a solution designed to apply "good enough" security and increase business efficiencies than one intending to recreate Fort Knox. Theoretical security is a check-box criteria, applied security becomes more of a buy-and-use criteria. Applied security exists at an equilibrium point between total security (theoretical security) and total functionality (no security). The choice between these two—and what to sacrifice—is made based on the most operationally efficient method for achieving that prescribed balance. To better evaluate the root ROI question when dealing with security products, the next logical question becomes "Which model, positive or negative, provides this equilibrium in the most operationally efficient manner?

17.2 Positive vs. Negative Application Security

The two approaches to security most often mentioned in the context of application security—positive and negative—are diametrically opposed in all of their characteristic behaviors, but they are structured very similarly. Both positive and negative security approaches operate according to an established set of rules. Access Control Lists (ACLs) and signatures are two implementation examples of positive and negative security rules, respectively. Positive security moves away from "blocked," end of the spectrum, following an "allow only what I know" methodology. Every rule added to a positive security model increases what is classified as known behavior, and thus allowed, and decreases what is blocked, or what is unknown. Therefore, a positive security model with nothing defined should block everything and relax (i.e., allow broader access) as the acceptable content contexts are defined.

At the opposite end of the spectrum, negative security moves towards "blocked what I know is bad," meaning it denies access based on what has previously identified as content to be blocked, running opposite to the known/allowed positive model. Every rule added to the negative security policy increases the blocking behavior, thereby decreasing what is both unknown and allowed as the policy is tightened. Therefore, a negative security policy with nothing defined would grant access to everything, and be tightened as exploits are discovered. Although negative security does retain some aspect of known data, negative security knowledge comes from a list of very specific repositories of matching patterns. As data is passed through a negative security policy, it is evaluated against individual known "bad" patterns. If a known pattern is matched, the data is rejected; if the data flowing through the policy is unidentifiable, it is allowed to pass. Negative security policies do not take into account how the application works, they only notice what accesses the application and if that access violates any negative security patterns.

Discussions on preferred security methods typically spawn very polarized debates. Tried and true security engineers might ardently argue the merits of the positive security model because it originates from the most "secure" place—"Only allow what I know and expect." Many business pundits would argue that the negative model is the best as it starts in the most "functional" place—

"Block what I know is bad and let everything unknown through." Both groups are correct and yet both opinions become irrelevant when projected onto applied security, because both positive and negative security is theoretical. Applied security falls somewhere in the middle of the spectrum, providing a practical balance. At some point, as the negative approach is tightened, it will take on characteristics of a more positive model, inching towards a more complete security approach.

Likewise, as a positive security model is loosened to accommodate new application behaviors, it will take on some aspects of a more negative approach, such as implementing data pattern matching, to block the more predictable attacks. As a positive policy continues to relax, it will move closer towards complete functionality. The point at which these two opposing concepts begin to overlap is where applied security starts to take shape.

This "meet in the middle" idea suggests that from an applied security standpoint, both models are capable of achieving the same delicate balance between "security" and "functionality." The difference between these models stems from where each begins and where they collide. This can be as simple as the number of rules required to meet the end goal.

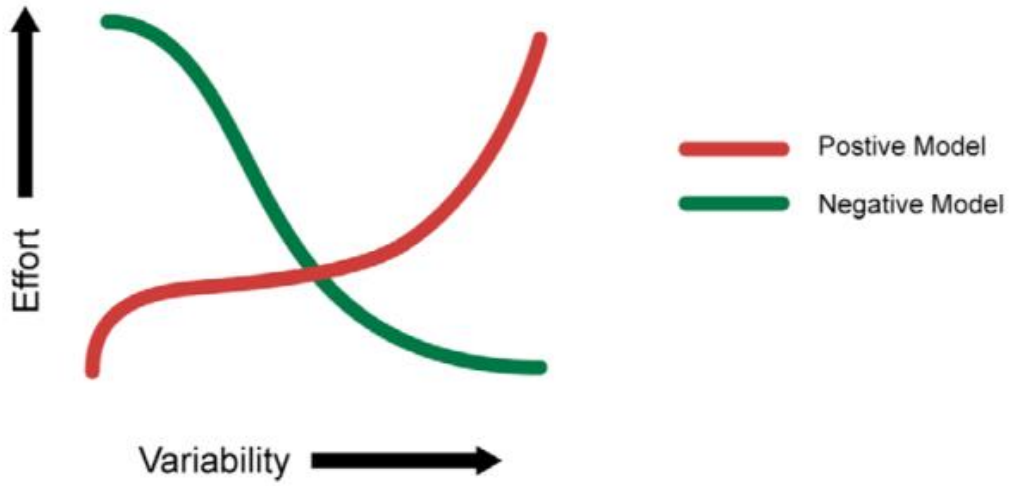
It is clear then that, from an operational efficiency standpoint, the undiluted concepts of neither the positive nor the negative approach intrinsically provides more efficiency than the other. In some cases, the positive approach generates the least number of rules while in other cases the negative approach generates the least. It would also appear that it is the nature of the applied policy and/or the content itself which might determine the best approach. What then, are the qualities of the policy or content which makes one approach more efficient over the other?

17.3 Factors of an Effective Applied Security Model

Implementing a successful application security architecture is not as easy as deciding how much negative security and how much positive security to mix together into a hypothetical applied security blender. By design, application security devices have to have some level of application knowledge, such as the type of content delivered by the application, which is accessing any point within the application, and how to map specific policy criteria to this information. Very specific application awareness of this nature is essential in building an efficient applied security policy.

17.4 The Effect of Content Variability

Within the scope of application security, Content Variability is a measure of the content that needs to be secured and includes a number of different component pieces: the number of objects, the number of types of content, frequency of content change, and the nature of the content. A site that only has five specific objects is much less variable than a site with 500 specific objects. Within those objects, the cohesiveness of the content type is also a factor; if all 500 objects share a common format, they are less variable than a site with where all 500 objects are unique. Obviously, a site that changes only once a year is much less variable than one that changes daily. Finally, the nature of that content—for example, whether it is dynamically generated or static is a contributing factor. Essentially, variability is a measure of the site complexity. The idea of Content Variability is a single measurable value based on all of these factors. The variability of the content dictates the amount of effort needed to achieve the prescribed applied security from the chosen model.



As depicted in the diagram, the higher the variability of the content, the easier it is to define a policy using the negative security model. As the complexity of the known content increases, it is easier to describe what isn't allowed rather than what is. Conversely, the opposite effect is true of the positive model; the more variable the site content, more effort is required to define those elements that are allowed. For example, let us assume that we have 10 different types of content within our site out of a possible 100 different types of content known. Because the site exhibits little variability, or is more cohesive, it is much easier to define the 10 allowable types of content than to define the 90 types of restricted content; a positive model is much more appropriate in this case.

On the other hand, if the site is less cohesive, perhaps representing of 90 of the 100 different types, it now becomes more efficient to define the 10 restricted content types than it is to define the 90 allowed ones; thus a negative model is more efficient. Once again, both models are equally successful at producing a desired level of security, but the variability of the content determines which is more efficient in a given scenario. And as we map the concept of content variability back to applied security, it becomes obvious that we will take the necessary aspects from the negative security model and couple those with what is required from the positive model.

The most successful implementation will come from a joint applied security policy, addressing both the security and the business needs at same time.

17.5 Rule Specificity

As the content variability affects the ability to create and maintain a security policy, the same is true of the specificity of rules used to build that policy. Rule Specificity conveys the level of detail of the protection mechanism implemented for any particular rule. For example, a rule that blocks Unicode attacks may block them from any application on one end of the spectrum all the way to only protecting Unicode directory traversal attacks against IIS5 on the other end. Depending upon the specificity of a rule, many things may be allowed with a single rule (positive security) or disallowed with a single rule (negative security). But as is the problem with theoretical security, Rule Specificity itself is not an exact science.

A rule that is not specific enough may block too much, creating unnecessary false positives (blocking access that shouldn't be blocked); a rule that is too specific may not block enough, creating false negatives. Content variability also impacts the efficiency of a policy by altering the level of specificity in the rules themselves. As the variability of the content increases, the ability to specifically stipulate what content is or isn't allowed becomes more time consuming. In an ideal world, every rule would be as specific as possible for the particular application it was designed to protect, avoiding false positives and false negatives. Similarly, the level of rule specificity within an application security policy can vary greatly depending on the content variability experienced by the application.

17.6 Order of Precedence

A third factor in implementing an efficient applied security policy is the order of precedence: defining which parts of the security policy are enacted before other parts of the policy. This concept is often seen in pro-

grammatic search algorithms: “match first” or “match any.” Using a combination of negative patterns and positive policy rules with varying degrees of specificity is bound to create many conflicts. In order to arbitrate these conflicts an order of precedence for all rules must be defined and followed for the policy to remain coherent. This is a critical decision point for application security, because the policy must decide if it should implement a more funneled approach (parsing through the policy to weed out what doesn’t match) or if it should look for the most restrictive implementation first. This order of precedence may be solved by choosing the most specific rule, whether it is positive or negative, and opening up access as data moves further through the policy.

Alternately, the order may be based on implementing a given ruleset: for example, all traffic may be pattern matched first and if there are any positive matches, the data is rejected, regardless of which specific pattern was matched. No matter which method is chosen, if the policy is implemented with an incorrect order of precedence, access to the application could be blocked by a policy that tightens first. Likewise, a policy that applies rules too loosely may allow unintended access to the application.

And as precedence is factored into the applied security equation, traffic volumes must also be taken into consideration. A two percent false positive error rate may be an acceptable metric in an applied security policy of an application that handles 100 connections/day, but unacceptable for a 10 million connection/day application. Regardless of the precedence methodology used it should be well defined and easy to follow to make a policy easy to audit and manage.

17.7 Conclusion—Best Practices

The problem with a purely positive policy is simply that it’s merely the most appropriate model for about half of the situations in which it’s deployed. The other half are unnecessarily weighed down by the fact that a negative model would be much more efficient. That is why, as a matter of best practice, every security solution should support a weighted balance of both the positive and negative methodologies. In the strictest sense of the term, negative security provides the best applied security out of the box due to the effort applied by the security vendor before the product is shipped. Focusing on known security vulnerabilities, this will block the most attacks, despite content variability. However, this does not provide security against unknown attacks or allow specific functions to be allowed. For that, positive security is required. To lessen the amount of effort needed for a given application, positive security templates should be provided by the application vendors themselves to complement the negative security.

If the goal of applied security is to reach a pre-defined posture in the most efficient manner, then the choice of model is directly related to the variability of the content itself. Somewhere between total security and total functionality is where the desired applied security level exists, and—theoretically—either security model is capable of achieving this goal. But as stated above, theoretical security can only exist in a vacuum. Applied security is a business choice and concept that moves security into real-world implementations to attain the most efficient, functional method. Neither positive nor negative security models alone can deliver the most economical solution in every situation or environment. Applied together, however—and merged with the business needs and requirements—a holistic view of both approaches can help delineate between theoretical security and applied security, enabling businesses to realize the greatest ROI from any security policy implementation.

18 Public Key Infrastructure

18.1 What is a digital signature?

A digital signature is basically a way to ensure that an electronic document (e-mail, spreadsheet, text file, etc.) is authentic. Authentic means that you know who created the document and you know that it has not been altered in any way since that person created it.

Digital signatures rely on certain types of encryption to ensure authentication. Encryption is the process of taking all the data that one computer is sending to another and encoding it into a form that only the other computer will be able to decode. Authentication is the process of verifying that information is coming from a trusted source.

These two processes work hand in hand for digital signatures.

There are several ways to authenticate a person or information on a computer:

Password - The use of a user name and password provide the most common form of authentication. You enter your name and password when prompted by the computer. It checks the pair against a secure file to confirm. If either the name or password do not match, then you are not allowed further access.

Checksum - Probably one of the oldest methods of ensuring that data is correct, checksums also provide a form of authentication since an invalid checksum suggests that the data has been compromised in some fashion. A checksum is determined in one of two ways. Let's say the checksum of a packet is 1 byte long, which means it can have a maximum value of 255. If the sum of the other bytes in the packet is 255 or less, then the checksum contains that exact value. However, if the sum of the other bytes is more than 255, then the checksum is the remainder of the total value after it has been divided by 256. Look at this example:

- *Byte 1 = 212*
- *Byte 2 = 232*
- *Byte 3 = 54*
- *Byte 4 = 135*
- *Byte 5 = 244*
- *Byte 6 = 15*
- *Byte 7 = 179*
- *Byte 8 = 80*
- *Total = 1151. 1151 divided by 256 equals 4.496 (round to 4). Multiply 4 X 256 which equals 1024. 1151 minus 1024 equals checksum of 127*

CRC (Cyclic Redundancy Check) - CRCs are similar in concept to checksums but they use polynomial division to determine the value of the CRC, which is usually 16 or 32 bits in length. The good thing about CRC is that it is very accurate. If a single bit is incorrect, the CRC value will not match up. Both checksum and CRC are good for preventing random errors in transmission, but provide little protection from an intentional attack on your data. The encryption techniques below are much more secure.

Private key encryption -Private key means that each computer has a secret key (code) that it can use to encrypt a packet of information before it is sent over the network to the other computer. Private key requires that you know which computers will talk to each other and install the key on each one. Private key encryption is essentially the same as a secret code that the two computers must each know in order to decode the information. The code would provide the key to decoding the message. Think of it like this. You create a coded message to send to a friend where each letter is substituted by the letter that is second from it. So "A" becomes "C" and "B" becomes "D". You have already told a trusted friend that the code is "Shift by 2". Your friend gets the message and decodes it. Anyone else who sees the message will only see nonsense.

Public key encryption - Public key encryption uses a combination of a private key and a public key. The private key is known only to your computer while the public key is given by your computer to any computer that wants to communicate securely with it. To decode an encrypted message, a computer must use the public key provided by the originating computer and its own private key.

The key is based on a hash value. This is a value that is computed from a base input number using a hashing algorithm. The important thing about a hash value is that it is nearly impossible to derive the original input number without knowing the data used to create the hash value. Here's a simple example:

Input number 10667
Hashing Algorithm = Input # x 143
Hash Value = 1525381

You can see how hard it would be to determine that the value of 1525381 came from the multiplication of 10667 and 143. But if you knew that the multiplier was 143, then it would be very easy to calculate the value of 10667. Public key encryption is much more complex than this example but that is the basic idea. Public keys generally use complex algorithms and very large hash values for encrypting: 40-bit or even 128-bit numbers. A 128-bit number has a possible 2128 different combinations. That's as many combinations as there are water molecules in 2.7 million olympic size swimming pools. Even the tiniest water droplet you can image has billions and billions of water molecules in it!

Digital certificates - To implement public key encryption on a large scale, such as a secure Web server might need, requires a different approach. This is where digital certificates come in. A digital certificate is essentially a bit of information that says the Web server is trusted by an independent source known as a **Certificate Authority**. The Certificate Authority acts as the middleman that both computers trust. It confirms that each computer is in fact who they say they are and then provides the public keys of each computer to the other.

The Digital Signature Standard (DSS) is based on a type of public key encryption method that uses the Digital Signature Algorithm (DSA). DSS is the format for digital signatures that has been endorsed by the US government. The DSA algorithm consists of a private key that only the originator of the document (signer) knows and a public key.

18.2 What is Encryption and How Does It Work?

The Early Days of Encryption

The ancient Greeks used a tool called a Scytale to help encrypt their messages more quickly using a transposition cipher—they would simply wrap the strip of parchment around the cylinder, write out the message, and then when unwound wouldn't make sense.

This encryption method could be fairly easily broken, of course, but it's one of the first examples of encryption actually being used in the real world.

Julius Caesar used a somewhat similar method during his time by shifting each letter of the alphabet to the right or left by a number of positions—an encryption technique known as Caesar's cipher. For instance, using the example cipher below you'd write "GEEK" as "JHHN".

Plain: ABCDEFGHIJKLMNOPQRSTUVWXYZ
Cipher: DEFGHIJKLMNOPQRSTUVWXYZABC

Since only the intended recipient of the message knew the cipher, it would be difficult for the next person to decode the message, which would appear as gibberish, but the person that had the cipher could easily decode and read it.

Other simple encryption ciphers like the Polybius square used a polyalphabetic cipher that listed each letter with the corresponding numeric positions across the top and side to tell where the position of the letter was.

	1	2	3	4	5
1	A	F	E	D	M
2	P	B	C	L	N
3	Q	G	J	K	O
4	R	H	I	S	T
5	W	V	U	X	Y

Using a table like the one above you would write the letter “G” as “23”, or “GEEK” as “23 31 31 43”.

Whether your encryption program is stand-alone or built into your e-mail app, the encryption process is the same: Data passes through a mathematical formula called an algorithm, which converts it into encrypted data called ciphertext. These formulas require one variable from you--called a key--which makes it difficult, if not impossible, for anyone else to crack the encryption.

There are two types of encryption: symmetric and asymmetric (also called public key). With symmetric encryption, you run a file through the program and create a key that scrambles the file. Then you e-mail the encrypted file to the recipient and separately transmit the decoding key (which could be a password or another data file). Running the same encryption application, the recipient uses the decoding key to unscramble the message. Symmetric encryption is fast but not as safe as asymmetric encryption because someone could intercept the key and decode the messages. But because of its speed, it's commonly used for e-commerce transactions.

Asymmetric encryption is more complex--and more secure. Two related keys are required: a public key and a private key. You make your public key available to anyone who might send you encrypted information. That key can only encode data; it cannot decode it. Your private key stays safe with you. When people wish to send you encrypted information, they encrypt it using your public key. When you receive the ciphertext, you decrypt it with your private key. Asymmetric encryption's added safety comes at a price: More computation is required, so the process takes longer.

Symmetric and asymmetric encryption use different algorithms to produce ciphertext. In symmetric encryption, the algorithm divvies up data into small chunks called blocks. It then switches letters around, changes the information in each block into numbers, compresses and expands the data, and runs those numbers through mathematical formulas that include the key. Then the algorithm repeats the process, sometimes dozens of times over. An asymmetric encryption's algorithm, on the other hand, treats the text as though it were a very large number, raises it to the power of another very large number, and then calculates the remainder after dividing it with a third very large number. Finally, the remainder number is converted back into text. Encryption programs can use the same algorithms differently, which is why the recipient needs to use the same application to decode the message that you used to encode it.

Keys are the final piece in the encryption puzzle. Keys vary in length and, consequently, in strength. The reason: The longer the key, the greater the number of possible combinations. For example, if your encryption program uses 128-bit keys, your particular key could be any of more than 3.4 trillion billion billion billion--or 2 to the power of 128--possible combinations of zeros and ones. A hacker is more likely to win the lottery than to crack that level of encryption using the brute-force method (systematically trying key combinations until they find the right one). By comparison, encryption experts can crack the average 40-bit symmetric key in about six hours on a typical home PC using brute force. However, even 128-bit encryption is vulnerable to some extent; pros have some sophisticated techniques that can help them crack even the toughest codes.

The Eagle Flies at Midnight

Encryption technology has been big with the military ever since 479 B.C.: According to the historian Herodotus, secret communiques, scratched into wooden tablets that were then covered with wax, tipped off Spar-

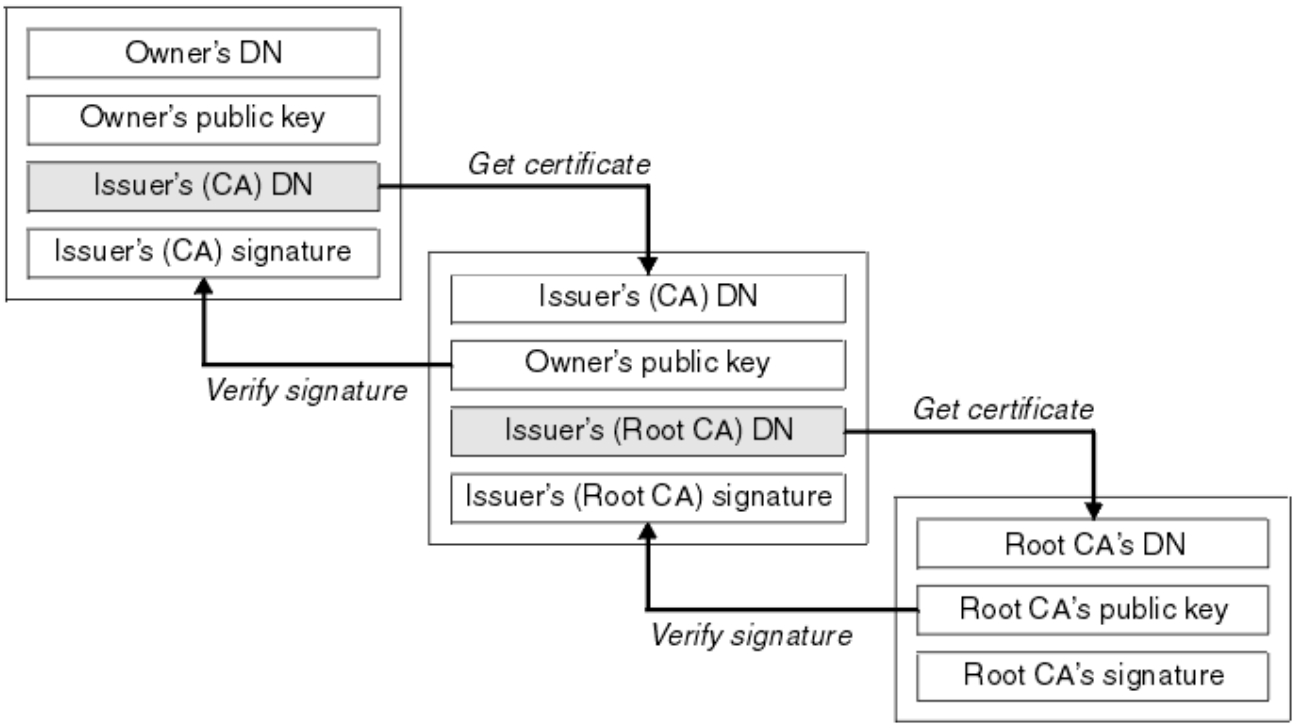
tan leaders to an imminent Persian invasion. Corporate IS types have taken advantage of it for years, as well. But home users are increasingly using encryption tools, whether they know it or not.

For example, Microsoft's Internet Explorer and Netscape's Communicator have built-in encryption tools for e-commerce transactions. Without any input from the user, Secure Sockets Layer (known as SSL), a symmetric protocol, scrambles credit card numbers as they travel from the user's desktop to the Web site's server. By default, browsers come with 40-bit encryption, although you have the option of downloading versions that support 128-bit encryption.

You can also take a more active role in protecting your data. Popular e-mail applications, including Microsoft Outlook Express and Lotus Notes, now let users encrypt their messages. Many e-mail applications provide an asymmetric protocol called Secure Multipurpose Internet Mail Extensions, though few people take advantage of it. S/MIME requires a certificate (a digital ID that resides in your e-mail application), which you have to buy from companies such as VeriSign for $15 per year.

18.3 Certificate Chain

When you receive the certificate for another entity, you might need to use a certificate chain to obtain the root CA certificate. The certificate chain, also known as the certification path, is a list of certificates used to authenticate an entity. The chain, or path, begins with the certificate of that entity, and each certificate in the chain is signed by the entity identified by the next certificate in the chain. The chain terminates with a root CA certificate. The root CA certificate is always signed by the CA itself. The signatures of all certificates in the chain must be verified until the root CA certificate is reached. Figure 1 illustrates a certification path from the certificate owner to the root CA, where the chain of trust begins.


Owner's DN
Owner's public key
Issuer's (CA) DN
Issuer's (CA) signature
Get certificate
Verify signature
Issuer's (CA) DN
Owner's public key
Issuer's (Root CA) DN
Issuer's (Root CA) signature
Get certificate
Verify signature
Root CA's DN
Root CA's public key
Root CA's signature

19 Authentication

19.1 What Is Authentication?

Authentication is the process of determining if a user or identity is who they claim to be. Authentication is accomplished using something the user knows (e.g. password), something the user has (e.g. security token) or something of the user (e.g. biometric).

The authentication process is based on a measure of risk. High risk systems, applications and information require different forms of authentication that more accurately confirm the user's digital identity as being who they claim to be than would a low risk application, where the confirmation of the digital identity is not as important from a risk perspective. This is commonly referred to as "stronger authentication".

Authentication processes are dependent upon identity verification and registration processes. For example, when Jane Doe is hired at an enterprise, she provides the enterprise with information and tokens of who she is (e.g. name, address, driver's license, birth certificate, a SSN number, a passport, etc.). The enterprise may choose to immediately accept this information or, it may instead chose to run background checks on Jane to see if she is who she claims to be and determine if she has any criminal record. When the checks come back favorably, the enterprise will accept her identity and enter her into their systems. The identity registration process will usually involve issuing Jane with enterprise authentication mechanisms such as id and password, security token, digital certificate and/or registering some of her biometrics.

The authentication process is totally dependents on the identity validation and registration process used for Jane. If Jane presents false tokens, which are accepted by the enterprise, then the person acting as Jane will be positively authenticated every time, even though she is not the real Jane Doe. Authentication security therefore is only as good as the weakest link in the chain.

19.2 Password Authentication

Password authentication is the most common method of authentication. It is also the least secure. Password authentication requires the identity to input a user id and a password in order to login. Password length, type of characters used and password duration are password management is now critical concern in enterprises. The ability to easily crack passwords has resulted in high levels of identity theft. As a result, the high risk of passwords means most enterprises now deploy a layered security strategy. A user enters in their id and password for initial login to gain access to only low risk information and applications with other forms of authentication required for higher risk information and applications.

19.3 Single Sign On Authentication

Single Sign On (SSO), Reduced Sign On (RSO), or Enterprise Single Sign On (ESSO) is the ability to reduce the number of id's and passwords a user has to remember. In most enterprises, a strong business case can be made to implement single sign on by reducing the number of password related help desk calls. SSO is also the architecture to require stronger forms of authentication for higher risk information and applications. Thus a user may login using their id and password to gain general low risk access to an enterprise. The SSO software enables them to not have to use multiple IDs and passwords. However, when the user tries to access more sensitive information and applications, the single sign on software will require the identity to input stronger authentication such as a security token, a digital certificate and/or a biometric.

19.4 Multi-factor Authentication

Multifactor authentication (MFA) is a security system in which more than one form of authentication is implemented to verify the legitimacy of a transaction. The goal of MFA is to create a layered defense and make it more difficult for an unauthorized person to access a computer system or network.

Multifactor authentication is achieved by combining two or three independent credentials: what the user knows (knowledge-based authentication), what the user has (security token or smart card) and what the user is (biometric verification). Single-factor authentication (SFA), in contrast, only requires knowledge the user possesses. Although password-based authentication is well-suited for website or application access, it is not secure enough for online financial transactions.

19.5 Authentication, Authorization, and Accounting (AAA)

Authentication, authorization, and accounting (AAA) is a term for a framework for intelligently controlling access to computer resources, enforcing policies, auditing usage, and providing the information necessary to bill for services. These combined processes are considered important for effective network management and security.

As the first process, authentication provides a way of identifying a user, typically by having the user enter a valid user name and valid password before access is granted. The process of authentication is based on each user having a unique set of criteria for gaining access. The AAA server compares a user's authentication credentials with other user credentials stored in a database. If the credentials match, the user is granted access to the network. If the credentials are at variance, authentication fails and network access is denied.

Following authentication, a user must gain authorization for doing certain tasks. After logging into a system, for instance, the user may try to issue commands. The authorization process determines whether the user has the authority to issue such commands. Simply put, authorization is the process of enforcing policies: determining what types or qualities of activities, resources, or services a user is permitted. Usually, authorization occurs within the context of authentication. Once you have authenticated a user, they may be authorized for different types of access or activity.

The final plank in the AAA framework is accounting, which measures the resources a user consumes during access. This can include the amount of system time or the amount of data a user has sent and/or received during a session. Accounting is carried out by logging of session statistics and usage information and is used for authorization control, billing, trend analysis, resource utilization, and capacity planning activities.

Authentication, authorization, and accounting services are often provided by a dedicated AAA server, a program that performs these functions. A current standard by which network access servers interface with the AAA server is the Remote Authentication Dial-In User Service (RADIUS).

19.6 SAML Authentication

What is SAML ?
SAML - Security Assertion Markup Language

SAML, developed by the Security Services Technical Committee of "Organization for the Advancement of Structured Information Standards" (OASIS), is an XML-based framework for exchanging user authentication, entitlement, and attribute information. SAML is a derivative of XML. The purpose of SAML is to enable Single Sign-On for web applications across various domains.

Why SAML ?
There are four 'drivers' behind the creation of the SAML standard:

Limitations of Browser cookies: Most existing Single-Sign On products use browser cookies to maintain state so that re-authentication is not required. Browser cookies are not transferred between DNS domains. So, if you obtain a cookie from www.abc.com, then that cookie will not be sent in any HTTP messages to www.xyz.com. This could even apply within an organization that has separate DNS domains. Therefore, to solve the Cross-Domain SSO (CDSSO) problem requires the application of different technology. All SSO products solve the CDSSO problem by different techniques.

SSO Interoperability: How products implement SSO and CDSSO are completely proprietary. If you have an organization and you want to perform SSO across different DNS domains within the same organization or you want to perform CDSSO to trading partners, then you will have to use the same SSO product in all the domains.

Web Services: Security within Web Services is still being defined. Most of the focus has been on how to provide confidentiality and authentication/integrity services on an end-to-end basis. The SAML standard provides the means by which authentication and authorization assertions can exchanged between communicating parties.

Federation: The need to simplify identity management across organizational boundaries, allowing users to consolidate many local identities into a single (or at least a reduced set).

20 Security solutions

20.1 IPsec – IP Security

Virtual private networks (VPNs), initially based on the IPSec protocol, were originally developed for site-to-site communications between branch offices. These site-to-site VPNs were an economical way to extend the corporate network to remote offices over the public Internet, avoiding the high cost of private wide area network (WAN) connections. The resulting secure connection between trusted private networks offered access similar to that of the corporate network. As companies broadened their use of VPNs to meet other remote access needs, proprietary extensions had to be added to the IPSec standard, or to vendor implementations of the protocol, to address the complexity of adding individual end users to the remote access equation.

An IPSec VPN works by establishing a tunnel over the Internet to connect users outside a corporate firewall or gateway to the internal network. It requires compatible hardware or software—almost always from a single vendor—on both ends of the tunnel. With IPSec, the corporate IT department dictates the technology used on both ends of the tunnel. Although this can work well for systems managed by the IT department, few companies are willing or able to fully control or trust the end point environments of remote devices used by teleworkers, business partners or customers.

At one time, a traditional Internet Protocol Security (IPSec) virtual private network (VPN) was the only option for secure remote access. However, because IPSec solutions were designed for trusted site-to-site connectivity and not with a highly-mobile workforce in mind, IPSec solutions had limitations for supporting untrusted end point locations that were not directly managed by IT. In response to increasing user demands for remote access, a new kind of VPN emerged—SSL VPNs. These new VPNs, based on the Secure Sockets Layer (SSL) protocol that safeguards the world of e-commerce, quickly became the leading option for remote access.

As organizations grow and become more mobile, many are shifting to SSL VPNs, as they offer “everywhere” access while retaining complete control and security. Recent advances in SSL VPN technology offer many benefits for both users and companies. When compared to IPSec, SSL VPNs are typically less costly to manage, eliminate concerns related to open-by-default tunnels and offer a more flexible experience for employees and business partners using untrusted end point environments.

20.2 Why should you use IPsec?

IPSec VPNs are best suited for point-to-point access. Open tunneling protects data between two private networks or between IT-managed machines and a private network. IPSec is a perfectly viable solution when a permanent connection is required between two specific locations, for example between a branch or remote office and a corporate headquarters. It can also be used successfully to provide access to a small finite number of remote workers using tightly-controlled corporate-issued laptops.

Many existing IPSec implementations can continue to work well for these use cases for which they were originally deployed. IT might consider keeping IPSec in these limited areas and extend remote access to other areas, such as trusted partners or extranet users, via a parallel SSL VPN solution. While a parallel VPN implementation is a viable choice for some enterprises, transitioning all access use cases through a single SSL VPN gateway might ultimately cost less and be easier to manage.

While many organizations still implement IPSec solutions today, however, for secure remote access the momentum has clearly shifted to SSL VPNs. Some organizations replace older versions of IPSec with newer versions that better streamline the provisioning of agents, or provide elements of end point control. Nevertheless, these augmented IPSec VPNs still may not be as flexible or robust as SSL VPN solutions.

With increased access from unmanaged end point devices, end point control becomes a key risk factor. For managed devices, some IPSec solution providers suggest keeping IPSec and adding a network access control (NAC) solution. However, this greatly adds to the costs and complexity of administering and maintaining a separate appliance to achieve end point control, and still does not provide granular access controls down to the application layer, essentially allowing the remote device to be a node on the network.

20.3 Replacing IPsec

The ascendancy of IPSec technology as an innovative remote access solution peaked nearly a decade ago. IPSec VPNs are no longer an effective remote access solution when comparing costs of IT overhead and the desire for granular access controls for highly portal devices with the current demands of an increasingly mobile workforce. With early IPSec implementations, the considerable overhead involved in provisioning, maintaining and supporting dedicated IPSec clients was tolerated because IPSec access tended to be restricted only to relatively few managed-device use cases. In recent years, however, since broadband has become widespread and laptops have become cheaper, there has been greater incentive for IT to deploy more laptops and other mobile devices to more users across the enterprise, increasing the overhead needed to support distributed-client IPSec VPNs. While these devices are more likely to be transported beyond the physical office to be used at home or other remote sites, IPSec still views them as nodes on the network, regardless of location.

Workers are also now accessing corporate resources from more end point devices that are not directly managed by IT, such as home computers, WiFi-enabled laptops, PDAs, smartphones and public kiosks. While most workers today are not full-time teleworkers, many commonly perform teleworking functions, such as sending and receiving e-mail and attachments from home before or after work hours, on weekends, while on the road or while on vacation. In addition, business partners need limited access to specific network resources which introduces additional remote access challenges to the IT department in today's world of outsourced supply chains. By providing employees and business partners with wider access to business tools and information, the proliferation of unmanaged end point devices has directly resulted in increased productivity. But it has also greatly increased the complexity for IT in controlling remote access, thereby minimizing the viability of distributed-client IPSec VPNs as an efficient remote access solution.

But still IPsec tunnels are still commonly used in site-to-site communications.

20.4 SSL VPN

SSL is the standard protocol for managing the security of message transmission on the Internet. SSL is a higher-layer security protocol than IPSec, working at the application layer rather than at the network layer. By operating at the application layer, SSL can provide the highly granular policy and access control required for secure remote access. Because SSL is included in all modern browsers, SSL VPNs can empower today's mobile workforce with clientless remote access—while saving IT departments the headache of installing and managing the complexity of IPSec clients. By extending the workplace to home PCs, kiosks, PDAs, and other unmanaged devices, SSL VPN solutions increase workforce productivity, resulting in a greater return on investment. And by eliminating the need to deploy and support "fat" clients, SSL VPN reduces IT overhead, resulting in a lower total cost of ownership.

An SSL VPN uses SSL to provide end users with authorized and secure access for Web, client/server and file share resources. SSL VPNs deliver user-level authentication, ensuring that only authorized users have access to the specific resources allowed by the company's security policy. SSL VPNs start with providing access via a Web browser, removing the need for IT to provision clients to the end point device. For advanced access, agents may be required but SSL VPNs allow IT to have agents provisioned and activated within the context of the Web browser where Active X or Java based "thin" clients are transparently pushed through the browser, Alternatively, most SSL VPNs allow IT to pre-provision the agents directly to a user's device, allowing the user to directly access the SSL VPN without having to open a Web browser.

Potential Benefits of Transitioning to an SSL VPN:

- *Increased productivity: SSL VPNs work in more places, including home PCs, kiosks,* PDAs and unmanaged devices over wired and wireless networks.
- *Lower costs: SSL VPNs are clientless or use lightweight Web-delivered clients rather* than "fat" IPSec clients, reducing management and support calls.
- *Broadened security: SSL VPNs provide granular access and end point control to* managed and non-managed devices

20.5 Why you should transition to SSL VPN

Today's modern mobile workforce demands more secure access to more resources from more remote devices and platforms than ever before. Corporate boundaries are blurring, with partners, vendors and consultants playing as vital a role in daily operations as employees do. These changes suggest the need for an inverted model for the corporate network, evolving from the traditional enclosed-perimeter model to a distributed global network that connects employees, partners and customers over multiple Internet, intranet and VoIP channels. IT managers must now assume that any user and device is a potential risk point, whether the user is accessing remotely or plugged directly into the LAN. Disaster recovery and business continuity initiatives pose additional incentive to provide remote access from any end point location. Policy based granular access control becomes imperative.

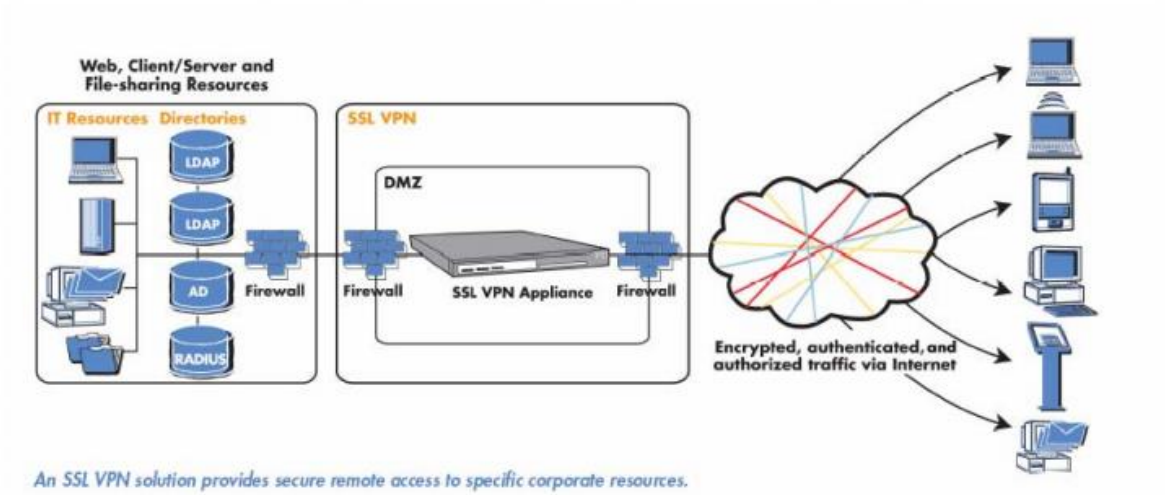
Securing inverted networks with granular access control is an ideal use case for SSL VPN technology. SSL based access control appliances are the key to achieving application access control. SSL VPN solutions can detect what is running on the end point device, protect applications with granular access control based on user identity and device integrity and connect users securely and easily to applications on any device.

Because SSL is part of any Web browser, SSL VPN solutions provide clientless and Web-delivered thin client access that significantly increases the number of points from which employees, partners and customers can access network data. SSL VPN solutions greatly simplify the connection process for mobile 7 users, seamlessly traversing NAT, firewalls and proxy servers. SSL VPN solutions reduce IT support costs, lowering total cost of ownership. SSL VPN clientless access minimizes the IT overhead involved in provisioning, configuring and maintaining an enterprise remote access solution. Alternatively, certain SSL

VPN tunnel solutions provide a complete "in-office" experience by deploying an auto-updating, Webdelivered thin client, eliminating the need for direct IT intervention. SSL VPN solutions also streamline administration costs by controlling all access to enterprise resources via a single, centralized gateway.

SSL VPN solutions also provide greater security compared to IPSec. Since SSL is an application layer protocol, an SSL VPN is inherently better-suited for securing application-based remote access. SSL VPN solutions provide secure, granular access controls, ensuring that users gain access only to the designated resources or applications specific to their needs and according to security policy.

With SSL VPN solutions, end-user access to any given resource is restricted unless authorized. As a result, SSL VPN technology provides the granular access control that requires all users, regardless of location, to be granted explicit permission to access specific network resources. With SSL VPN technology, access control to applications and networks can be as general or specific as required to meet regulatory compliance and corporate security mandates.



An SSL VPN solution provides secure remote access to specific corporate resources.

21 F5 Platforms

21.1 Hardware Platforms

F5 builds two types of hardware platforms. These include application delivery switches and chassis.

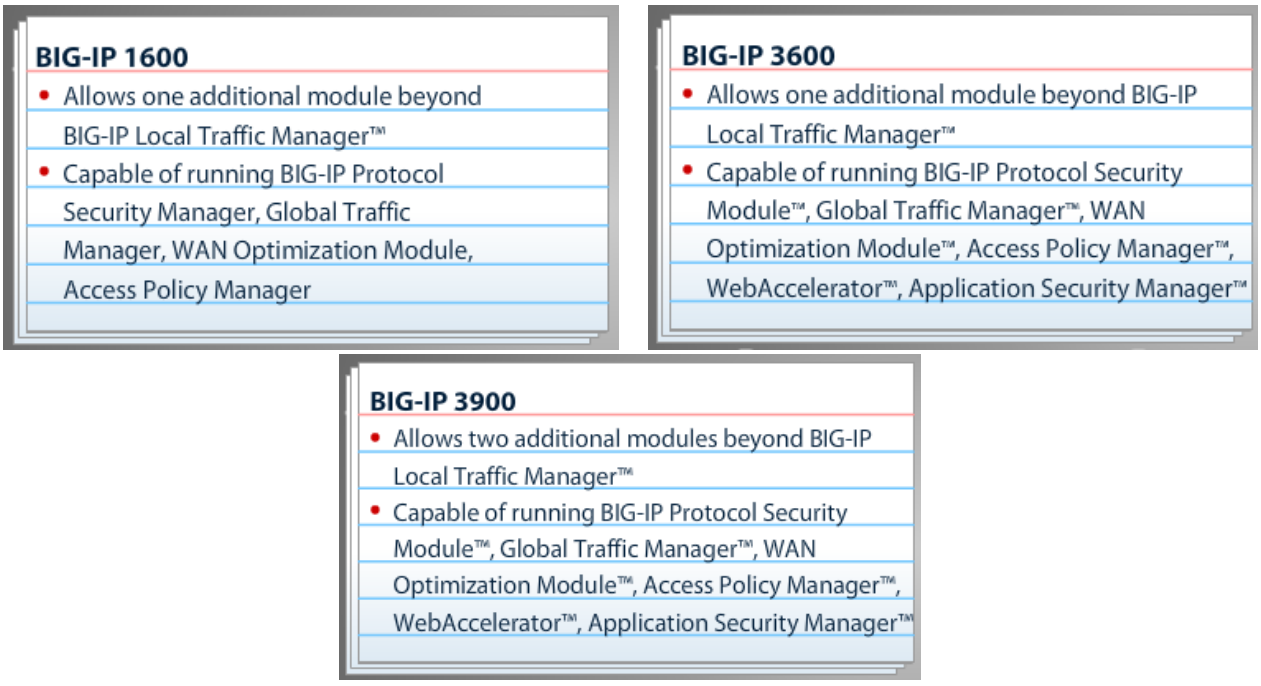
An application delivery switch (sometimes just referred to as an appliance) has a fixed number of network ports and performance. A chassis gives customers the option to purchase additional blades that can be inserted into the chassis when needed. This allows on-demand scaling of both the number of network ports and performance. F5 offers multiple application delivery switches in two value propositions: ***High Performance Meets High Value*** and ***Unified Application Delivery***.

F5 offers the VIPRION chassis as the world's first on-demand ADC.

High Performance Meets High Value is the first series of platforms offered by F5. It includes the BIG-IP 1600, 3600, and 3900. All of these platforms are based on a 1U size. Although this is the base series of platforms, each device still has a very high capacity. High performance per dollar is achieved with one gigabyte of software compression and SSL offload. These platforms also provide advanced security and acceleration options. They are reliable and adaptable with options for dual power and DC power as well as front-to-back cooling.

This series of platforms includes a single enterprise-grade hard drive and CompactFlash for customers who require no spinning medium. All devices have max RAM installed, which is a great feature because it does not have to be upgraded. The platforms in this series are reliable and adaptable with a front-to-back cooling feature and have a DC power option. Each platform is standard equipped with a single hot-pluggable power source and has the capability for a dual power option. The dual-core CPU or quad-core CPU will be used for clustered multiprocessing. This means multiple instances of TMOS can run on a device.

Different modules can run on each platform.



Unified Application Delivery

The next series of platforms is Unified Application Delivery. It includes the BIG-IP 6900, 8900, 8950, and 11050. This series is designed for high throughput and multiple modules.
All of the Unified Application Delivery platforms achieve high performance for effective consolidation. The BIG-IP 6900 and 8900 have anywhere from 6 to 12 gigabytes per second of throughput on Layer 7. These platforms also have hardware SSL and compression offload as well as 10 gig networking functions.

Running multiple modules and unifying application delivery functions onto one device is also possible with these platforms.

VIPRION is the 4-slot chassis made by F5. Compared to an appliance, VIPRION provides more scalability and modularity. It is the world's first on-demand ADC to provide zero reconfiguration and on-demand scaling. Customers can add up to four blades as their workload requirements increase.

VIPRION does not require reconfiguration because the installation of a new blade does not require that the chassis be reconfigured. When a new blade is installed, it automatically copies the configuration from the master blade so that the customer is always running the most updated configuration. The addition of new blades allows automatic scaling of performance and does not require an overprovision of components or software.

It is also possible to install Local Traffic Manager and Application Security Manager modules on each blade. Once a customer purchases the license and the first blade, there is no need to purchase additional licenses when purchasing additional blades.

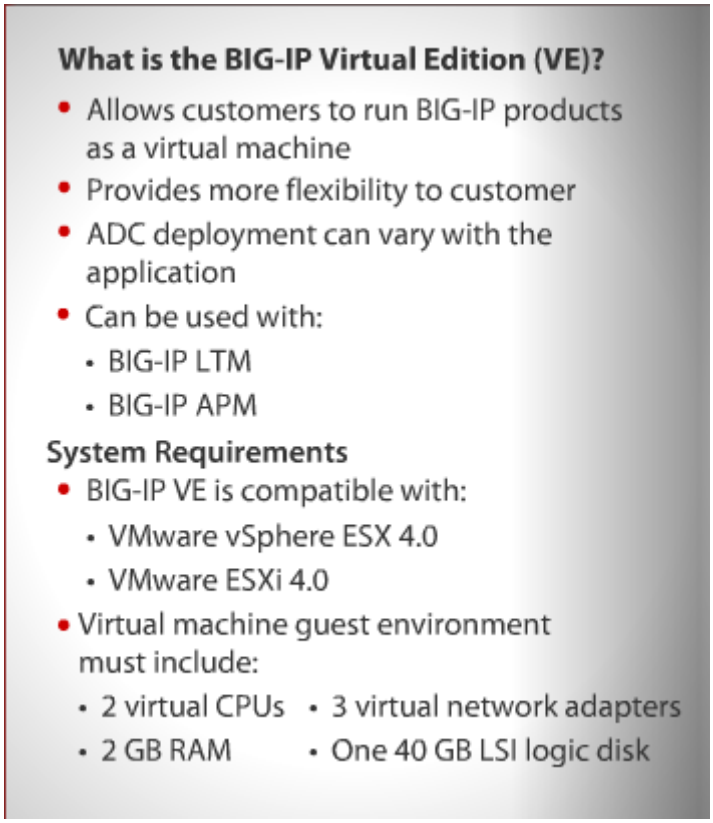
21.2 BIG-IP Products as Virtual Editions

The BIG-IP Virtual Edition, often referred to as VE, lets customers run BIG-IP products as virtual machines. Separating the hardware from the software provides the customer with greater flexibility and enables them to choose how they want to deploy their application delivery controller based on what is best for their application.

For example, if a customer needs lots of SSL performance, the physical hardware is better suited for their needs. However, if a customer is deploying BIG-IP products in a cloud environment, in which they do not own a physical device, then the Virtual Edition would make more sense.

Currently, the Virtual Edition can only be used with certain BIG-IP products.

Here are the specific system requirements for VE.



There are four types of licenses for BIG-IP LTM Virtual Edition. They include a trial edition, lab edition, and two different production licenses.

The trial edition is intended for customers who are considering VE, but want to try it before they purchase it. It is a hook to get them interested. Customers can go to F5.com/trial to download a free version that is limited to 90 days. This version is not intended for production environments and is hard coded to one megabyte. It's important to review the release notes to understand which features are not available.

The lab edition is a full version of the product and contains the tools customers need to build an application lab for their application developers. The developers can code against the ADC to test the product and features such as iRules and iControl. However, this edition is intended for functional testing only.

The last two licenses are both production editions. The production edition is available in either 200 megabytes throughput or one gigabyte throughput. This gives the customer the flexibility to choose the best license for different use cases. BIG-IP modules that are available in a virtual edition, such as APM, can be added to either production license.

Existing F5 customers who already own physical devices find the Virtual Edition to be extremely useful when building their labs.

Building a physical lab takes time and money. Customers have to wait for products to ship and find someone in the organization to set it up. Even if the equipment is already there, it is often allocated to another purpose, and is not available to the application developer.

By comparison, deployment of VE is a simple process. A virtual environment can be built in a few hours.

The simplified deployment of VE lets customers quickly build a development environment, saving time and money.

Finally, a common trend is to move applications to external cloud providers. Customers cannot control the physical hardware that a cloud provider will use. However, VE can be bundled with applications so that the customer can control traffic internally on the external cloud.

21.3 Virtual Appliances VS Hardware Appliances

Different situations call for the BIG-IP Virtual Edition and the physical BIG-IP hardware.

The Virtual Edition should be used for discrete workloads on commodity hardware. It also provides flexible and quick deployment options and failure isolation.

Physical hardware provides a number of important benefits and is necessary in many situations. F5 hardware is purposefully built to provide high performance for application delivery. Using an F5 physical platform also offers a single-vendor solution. If a customer is running an F5 Virtual Edition on an HP server and an issue arises, troubleshooting between vendors is more difficult. When F5 hardware runs F5 software, this is avoided.

Additionally, a BIG-IP physical device can run all BIG-IP modules and perform hardware SSL offload and compression. It provides customers with all other features that may be important to their deployment, such as always-on management, certifications, special-purpose FPGAs, and improved security.

22 Acceleration Techniques

22.1 Optimizing TCP

Although TCP is ubiquitous today, the protocol has undergone many updates to help overcome limitations that existed in earlier versions. An acceleration device can help optimize TCP by implementing features that may not be present in either a client or server's TCP implementation.

An acceleration device can also decrease the number of server-side TCP connections required to service client requests. Additionally, it can help accelerate HTTP traffic by increasing the number of simultaneous client-side TCP connections a browser can open while downloading a web page.

22.2 General TCP Optimizations

Because it operates as a proxy, an acceleration device may be able to implement features missing from a client or server that can help speed application delivery. The acceleration device may be able to leverage optimizations natively supported by particular client or server operating systems and is likely to be able to implement optimizations that are not operating-system specific. The benefit to high speed, high latency WAN connections is that the acceleration device can perform TCP window scaling to improve performance. To overcome packet loss, the acceleration device can implement selective TCP acknowledgements (SACK) and advanced congestion control algorithms to prevent TCP from reducing throughput.

These are only two examples. Some acceleration devices implement hundreds of improvements to TCP in order to help it perform better.

22.3 Decreasing Server-side TCP Connections

Reducing server-side connection processing can dramatically improve application performance and reduce the number of servers required to host an application. TCP connection setup and teardown requires significant overhead, particularly for servers. As the number of open server connections increases, maintaining the open connections while simultaneously opening new connections can severely degrade server performance and therefore, user response time.

Although multiple transactions (for example, file transfers) can occur within a single TCP connection, a connection is generally between one client and one server. Normally, a connection closes either when a server reaches a defined transaction limit or when a client has transferred all needed files from that server.

Because an acceleration device operates as a proxy, it can aggregate, or "pool," TCP server-side connections by combining many separate transactions, potentially from many users, through fewer (or one) TCP connections. The acceleration device opens new server-side connections only when necessary, and instead reuses existing connections for requests from other users whenever possible.

22.4 Increasing Client-side TCP Connections

By default, most web browsers limit the maximum number of simultaneous HTTP/HTTPS connections that the browser can open to one URL. For example, Microsoft Internet Explorer v7 and below limit the maximum number of simultaneous connections to two per domain. Earlier versions of Firefox limit the browser to eight connections per domain. Given that a web page can contain dozens of objects, this limitation can greatly slow page loading times.

For example, suppose a user running Internet Explorer v7 requests a page from a web server that returns a response containing a list of the 30 objects that make up the web page. Further assume that that all objects are accessed through the domain, www.example.com. The browser opens two connections to www.example.com, requests one object at a time per connection (the limit imposed by TCP), and then reuses the two connections until all files have been downloaded or the connection reaches the server's transaction limit. If the connection suffers high latency, round trip time is high and download speed can be greatly reduced.

If the server terminates the connection after reaching a pre-defined transaction limit, the browser opens another connection to that URL. This process continues until the page downloads completely. Operating this way needlessly increases the page load time.

Some acceleration devices can "spoof" a browser by modifying the URLs in an HTTP response to speed page downloading. The modified URLs must first be defined in DNS to point to the same IP address. When examining the server response, the modified names appear to the browser to be different servers, so the web browser opens parallel connections to these altered URLs rather than serially downloading the objects from one URL.

22.5 HTTP Protocol and Web Application Optimizations

HTTP protocol optimizations maintain high user performance levels by optimally tuning each HTTP session. For example, some web applications are unable to return an HTTP 304 status code (Not Modified) in response to a client request rather than returning the entire object. Because an acceleration device proxies connections and caches content, it may be able to note when there is no change to a requested object and return the 304 response instead. This enables the browser to load the content from its own cache, even in conditions where the web application is hard-coded to re-send the object.

Some acceleration devices can additionally examine and change server responses to provide better browser and server performance. For example, some off-the-shelf and custom applications add a no-cache header to some objects, which directs a browser not to cache an object, rather to download the object from the origin web server every time. The purpose of the no-cache header is to ensure a browser always downloads dynamic (changing) data.

However, applications in some cases mark static data like a company logo as being non-cacheeable. Some acceleration devices can re-write the server response to mark the object as being cacheable and supply a more realistic expiration date. This feature can help remedy problems with off-the-shelf or custom-developed applications where code cannot easily be modified.

22.6 Compression

Compression is one of the oldest acceleration techniques, having been around for decades. Gzip, the most common compression algorithm, is implemented in virtually every web browser and server. Compression algorithms such as gzip are good at finding small, repeating patterns and reducing the characters required to send them. Besides web servers and browsers, acceleration devices implement compression. This is done for two reasons: first to offload compression overhead from web servers and second, to enable the acceleration device to perform other optimizations that improve performance for an HTTP/HTTPS stream.

Compression can be computationally expensive, especially for algorithms that provide high compression levels. These algorithms are of limited use with high speed communication, where delays must be minimized to maintain rapid user response times. More effective compression algorithms are therefore limited to low-speed communications where more time is available to perform compression processing without degrading user throughput and hence, response times. Fortunately, compression hardware assist is now available in some acceleration devices that can achieve compression rates in excess of 1 Gbps.

22.7 Caching

Caching involves storing data close to users and re-using the data during subsequent requests. Caching usually takes one of three forms. The first is the classic approach taken by web browsers and web applications. In this case, the web application code running on a server instructs a browser to cache an object marked as static for a specific time period. During that time period, the browser reads the object from cache when building a web page until the content expires. The client then reloads the content. Caching prevents the browser from having to waste time and bandwidth by always accessing data from a central site. This is the most common form of caching in use today.

The second form involves deploying an acceleration device in a data center to offload requests for web application content from web servers. This method operates asymmetrically, with the acceleration device caching objects from web servers and delivering them directly to users. Some acceleration devices cache static content only, while some additionally can process HTTP responses, include objects referenced in a response, and send the included objects as a single object to a browser. This not only offloads web server processing but also offloads web browser processing too. A side benefit to this approach is that as the acceleration device is typically in the data center and connected to higher-speed connections, the acceleration device can both assemble the objects from instructions in the HTTP response and deliver them using fewer objects and with fewer transactions.

Operating in this manner, caching can dramatically reduce server TCP and application processing, improve web page loading time, and hence reduce the need to regularly expand the number of web servers required to service an application.

The third form of caching involves using symmetric acceleration devices to cache and serve content to users at the remote site. The remote acceleration device serves content locally whenever possible, which reduces both response time and network utilization. This form of caching can be deployed not only for HTTP, but also for other protocols as well.

Caching has its limitations. First, if the client-side acceleration device serves content regardless of whether it is in contact with its remote peer, the client side device must implement access control to prevent unauthorized access to an object. Second, the client-side device may serve older, stale versions of content that change after the connection between the devices is broken. While this typically is not an issue with static web content, it can have significant impact on files that regularly change. When both issues are addressed, remote caching can greatly improve application performance, especially for web applications and static files used with other applications.

22.8 HTTP Pipelining

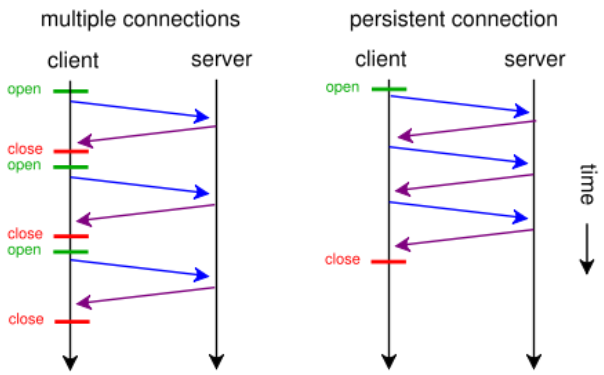
Everyone wants web sites and applications to load faster, and there's no shortage of folks out there looking for ways to do just that. But all that glitters are not gold and not all acceleration techniques actually do all that much to accelerate the delivery of web sites and applications. Worse, some actual incur risk in the form of leaving servers open to exploitation.

A brief history

Back in the day when HTTP was still evolving, someone came up with the concept of persistent connections. See, in ancient times – when administrators still wore togas in the data center – HTTP 1.0 required one TCP connection for every object on a page. That was okay, until pages started comprising ten, twenty, and more objects. So someone added an HTTP header, Keep-Alive, which basically told the server not to close the TCP connection until (a) the browser told it to or (b) it didn't hear from the browser for X number of seconds (a time out). This eventually became the default behavior when HTTP 1.1 was written and became a standard.

I told you it was a brief history.

This capability is known as a persistent connection, because the connection persists across multiple requests. This is not the same as pipelining, though the two are closely related. Pipelining takes the concept of persistent connections and then ignores the traditional request – reply relationship inherent in HTTP and throws it out the window.

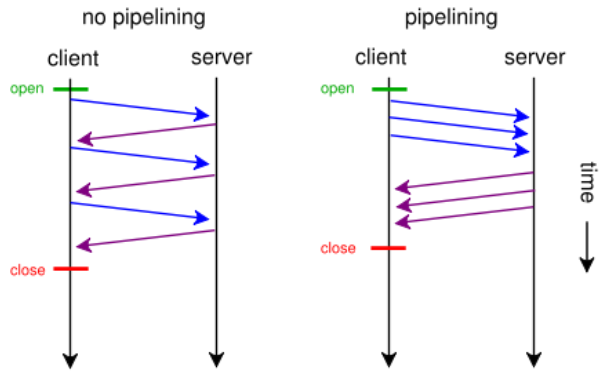


The general line of thought goes like this:

"Whoa. What if we just shoved all the requests from a page at the server and then waited for them all to come back rather than doing it one at a time? We could make things even faster!"

Tada! HTTP pipelining.

In technical terms, HTTP pipelining is initiated by the browser by opening a connection to the server and then sending multiple requests to the server without waiting for a response. Once the requests are all sent then the browser starts listening for responses. The reason this is considered an acceleration technique is that by shoving all the requests at the server at once you essentially save the RTT (Round Trip Time) on the connection waiting for a response after each request is sent.



Why it just doesn't matter anymore (and maybe never did)

Unfortunately, pipelining was conceived of and implemented before broadband connections were widely utilized as a method of accessing the Internet. Back then, the RTT was significant enough to have a negative impact on application and web site performance and the overall user-experience was improved by the use of pipelining. Today, however, most folks have a comfortable speed at which they access the Internet and the RTT impact on most web application's performance, despite the increasing number of objects per page, is relatively low.

There is no arguing, however, that some reduction in time to load is better than none. Too, anyone who's had to access the Internet via high latency links can tell you anything that makes that experience faster has got to be a Good Thing. So what's the problem?

The problem is that pipelining isn't actually treated any differently on the server than regular old persistent connections. In fact, the HTTP 1.1 specification requires that a "server MUST send its responses to those requests in the same order that the requests were received." In other words, the requests are return in serial, despite the fact that some web servers may actually process those requests in parallel. Because the serv-

er MUST return responses to requests in order that the server has to do some extra processing to ensure compliance with this part of the HTTP 1.1 specification. It has to queue up the responses and make certain responses are returned properly, which essentially negates the performance gained by reducing the number of round trips using pipelining.

Depending on the order in which requests are sent, if a request requiring particularly lengthy processing – say a database query – were sent relatively early in the pipeline, this could actually cause a degradation in performance because all the other responses have to wait for the lengthy one to finish before the others can be sent back.

Application intermediaries such as proxies, application delivery controllers, and general load-balancers can and do support pipelining, but they, too, will adhere to the protocol specification and return responses in the proper order according to how the requests were received. This limitation on the server side actually inhibits a potentially significant boost in performance because we know that processing dynamic requests takes longer than processing a request for static content. If this limitation were removed it is possible that the server would become more efficient and the user would experience non-trivial improvements in performance. Or, if intermediaries were smart enough to rearrange requests such a way that their execution were optimized then we'd maintain the performance benefits gained by pipelining. But that would require an understanding of the application that goes far beyond what even today's most intelligent application delivery controllers are capable of providing.

The silver lining

At this point it may be fairly disappointing to learn that HTTP pipelining today does not result in as significant a performance gain as it might at first seem to offer (except over high latency links like satellite or dial-up, which are rapidly dwindling in usage). But that may very well be a good thing.

As miscreants have become smarter and more intelligent about exploiting protocols and not just application code, they've learned to take advantage of the protocol to "trick" servers into believing their requests are legitimate, even though the desired result is usually malicious. In the case of pipelining, it would be a simple thing to exploit the capability to enact a layer 7 DoS attack on the server in question. Because pipelining assumes that requests will be sent one after the other and that the client is not waiting for the response until the end, it would have a difficult time distinguishing between someone attempting to consume resources and a legitimate request.

Consider that the server has no understanding of a "page". It understands individual requests. It has no way of knowing that a "page" consists of only 50 objects, and therefore a client pipelining requests for the maximum allowed – by default 100 for Apache – may not be seen as out of the ordinary. Several clients opening connections and pipelining hundreds or thousands of requests every second without caring if they receive any of the responses could quickly consume the server's resources or available bandwidth and result in a denial of service to legitimate users.

So perhaps the fact that pipelining is not really all that useful to most folks is a good thing, as server administrators can disable the feature without too much concern and thereby mitigate the risk of the feature being leveraged as an attack method against them.

Pipelining as it is specified and implemented today is more of a security risk than it is a performance enhancement. There are, however, tweaks to the specification that could be made in the future that might make it more useful. Those tweaks do not address the potential security risk, however, so perhaps given that there are so many other optimizations and acceleration techniques that can be used to improve performance that incur no measurable security risk that we simply let sleeping dogs lie.

23 Appendix

23.1 References

Electronic
World Wide Web page, **Networking Basics: Part 17 - The OSI Model,** http://www.windowsnetworking.com/articles-tutorials/netgeneral/Networking-Basics-Part17.html [2013-04-21]
World Wide Web page, **BIG-IP Access Policy Manager Data Sheet,** http://www.f5.com/pdf/products/big-ip-access-policy-manager-ds.pdf [2013-04-21]
World Wide Web page, **BIG-IP Application Security Manager Product Overview,** http://www.f5.com/pdf/products/big-ip-application-security-manager-overview.pdf [2013-04-21]
World Wide Web page, **BIG-IP Local Traffic Manager Product Overview,** http://www.f5.com/pdf/products/big-ip-local-traffic-manager-overview.pdf [2013-04-21]
World Wide Web page, **BIG-IP Global Traffic Manager Product Overview,** http://www.f5.com/pdf/products/big-ip-global-traffic-manager-overview.pdf [2013-04-21]
World Wide Web page, **Enterprise Manager Product Overview,** http://www.f5.com/pdf/products/enterprise-manager-overview.pdf [2013-04-21]
World Wide Web page, **BIG-IP WAN Optimization Manager Product Overview,** http://www.f5.com/pdf/products/big-ip-wan-optimization-manager-overview.pdf [2013-04-21]
World Wide Web page, **BIG-IP WebAccelerator Data Sheet,** http://www.f5.com/pdf/products/big-ip-webaccelerator-ds.pdf [2013-04-21]
World Wide Web page, **ARX Series Product Overview,** http://www.f5.com/pdf/products/arx-series-overview.pdf [2013-04-21]
World Wide Web page, **What is an iRule?,** https://devcentral.f5.com/wiki/iRules.WhatIsAnIRule.ashx [2013-04-21]
World Wide Web page, **#The101: iRules – Introduction to iRules,** https://devcentral.f5.com/tech-tips/articles/-the101-irules-ndash-introduction-to-irules [2013-04-21]
World Wide Web page, **iApp Wiki Home,** https://devcentral.f5.com/wiki/iApp.HomePage.ashx [2013-04-21]
World Wide Web page, **F5 iApp Whitepaper,** http://www.f5.com/pdf/white-papers/f5-iapp-wp.pdf [2013-04-21]
World Wide Web page, **iControl 101 - #01 - iControl Marketing Dissected,** https://devcentral.f5.com/tech-tips/articles/icontrol-101-01-icontrol-marketing-dissected [2013-04-24]
World Wide Web page, **iControl 101 - #02 - How iControl Works,** https://devcentral.f5.com/tech-tips/articles/icontrol-101-02-how-icontrol-works [2013-04-24]
World Wide Web page, **The Full-Proxy Data Center Architecture,** https://devcentral.f5.com/blogs/us/the-full-proxy-data-center-architecture [2013-04-24]
World Wide Web page, **The Concise Guide to Proxies,** https://devcentral.f5.com/blogs/us/the-concise-guide-to-proxies [2013-04-24]
World Wide Web page, **Redefining the Solution,** http://www.f5.com/pdf/white-papers/tmos-wp.pdf [2013-04-27]
World Wide Web page, **Additional Considerations for Redundant System Configurations,** http://support.f5.com/kb/en-us/products/big-

ip_ltm/manuals/product/tmos_management_guide_10_1/tmos_appendix_b_redundancy.html [2013-04-27]

World Wide Web page, **Configuring High Availability,** http://support.f5.com/kb/en-us/products/big-ip_ltm/manuals/product/tmos_management_guide_10_0_0/tmos_high_avail.html#1026652 [2013-04-27]

World Wide Web page, **Persistent and Persistence, What's the Difference?,** https://devcentral.f5.com/blogs/us/persistent-and-persistence-whats-the-difference [2013-04-27]

World Wide Web page, **Synchronizing configuration data,** http://support.f5.com/kb/en-us/products/big-ip_ltm/manuals/product/tmos_management_guide_10_0_0/tmos_high_avail.html#1026652 [2013-04-29]

World Wide Web page, **Difference Between Client and Server Systems,** http://www.differencebetween.com/difference-between-client-and-server-systems/ [2013-04-29]

World Wide Web page, **Applied Application Security— Positive & Negative Efficiency,** http://www.f5.com/pdf/white-papers/applied-app-security-wp.pdf [2013-04-29]

World Wide Web page, **What is a digital signature?** http://www.howstuffworks.com/digital-signature.htm [2013-04-29]

World Wide Web page, **HTG Explains: What is Encryption and How Does It Work?,** http://www.howtogeek.com/howto/33949/htg-explains-what-is-encryption-and-how-does-it-work/ [2013-04-29]

World Wide Web page, **How It Works: Encryption,** http://www.pcworld.com/article/15230/article.html [2013-04-30]

World Wide Web page, **How certificate chains work,** http://publib.boulder.ibm.com/infocenter/wmqv6/v6r0/index.jsp?topic=%2Fcom.ibm.mq.csqzas.doc%2Fsy10600_.htm [2013-04-30]

World Wide Web page, **The Business Of Authentication,** http://www.authenticationworld.com/ [2013-04-30]

World Wide Web page, **Multifactor Authentication (MFA),** http://searchsecurity.techtarget.com/definition/multifactor-authentication-MFA [2013-04-30]

World Wide Web page, **Multifactor Authentication (MFA),** http://searchsecurity.techtarget.com/definition/authentication-authorization-and-accounting [2013-04-30]

World Wide Web page, **Acceleration-101,** http://www.f5.com/pdf/white-papers/acceleration-101-wp.pdf [2013-05-01]

World Wide Web page, **HTTP Pipelining: A security risk without real performance benefits,** https://devcentral.f5.com/blogs/us/http-pipelining-a-security-risk-without-real-performance-benefits [2013-05-01]

World Wide Web page, **Ethernet at the Data Link Layer,** http://www.tech-faq.com/ethernet-at-the-data-link-layer.html [2013-05-01]

World Wide Web page, **2.1. Address Resolution Protocol (ARP),** http://linux-ip.net/html/ether-arp.html [2013-05-01]

World Wide Web page, **Broadcast Domain,** http://www.tech-faq.com/broadcast-domain.html [2013-05-01]

World Wide Web page, **What is a VLAN? How to Setup a VLAN on a Cisco Switch,** http://www.petri.co.il/csc_setup_a_vlan_on_a_cisco_switch.htm [2013-05-01]

World Wide Web page, **IP Addressing and Subnetting for New Users,** http://www.cisco.com/en/US/tech/tk365/technologies_tech_note09186a00800a67f5.shtml [2013-05-08]

World Wide Web page, **Broadcast Address,** http://www.tech-faq.com/broadcast-address.html [2013-05-08]
World Wide Web page, **The IP routing table,** http://technet.microsoft.com/en-us/library/cc779122(v=ws.10).aspx [2013-05-08]
World Wide Web page, **IP Routing Protocols,** http://www.orbit-computer-solutions.com/Routing-Protocols.php [2013-05-08]
World Wide Web page, **Understanding IP Fragmentation,** http://news.hitb.org/node/4005 [2013-05-08]
World Wide Web page, **What is Time to Live (TTL),** http://tcpipguru.com/what-is-time-to-live-ttl/ [2013-05-08]
World Wide Web page, **MTU,** http://compnetworking.about.com/od/networkprotocols/g/mtu-maximum.htm [2013-05-10]
World Wide Web page, **TCP Maximum Segment Size (MSS),** http://www.juniper.net/techpubs/software/jseries/junos93/jseries-config-guide-basic/tcp-maximum-segment-size-mss.html [2013-05-10]
World Wide Web page, **TCP (Transmission Control Protocol),** http://www.linktionary.com/t/tcp.html [2013-05-10]
World Wide Web page, **TCP THREE-WAY HANDSHAKE,** http://www.thenetworkencyclopedia.com/d2.asp?ref=1931 [2013-05-10]
World Wide Web page, **What is UDP and how does it work?,** http://www.jguru.com/faq/view.jsp?EID=9472 [2013-05-10]
World Wide Web page, **How Internet Infrastructure Works,** http://computer.howstuffworks.com/internet/basics/internet-infrastructure10.htm [2013-05-10]
World Wide Web page, **How Internet Infrastructure Works,** http://computer.howstuffworks.com/internet/basics/internet-infrastructure10.htm [2013-05-10]
World Wide Web page, **TCP Timeout and Retransmission,** http://www.pcvr.nl/tcpip/tcp_time.htm [2013-05-10]
World Wide Web page, **What are TCP RST Packets? - Protocol Expert,** http://myaccount.flukenetworks.com/fnet/en-us/supportAndDownloads/KB/IT+Networking/protocol+expert/What_are_TCP_RST_Packets_-_Protocol_Expert [2013-05-11]
World Wide Web page, **TCP and IP Options,** http://www.windowsecurity.com/articles-tutorials/windows_networking/TCP-IP-Options.html [2013-05-11]
World Wide Web page, **TCP Checksum Calculation and the TCP "Pseudo Header",** http://www.tcpipguide.com/free/t_TCPChecksumCalculationandtheTCPPseudoHeader.htm [2013-05-11]
World Wide Web page, **A TCP Tutorial,** http://www.ssfnet.org/Exchange/tcp/tcpTutorialNotes.html [2013-05-11]
World Wide Web page, **HTTP Made Really Easy,** http://www.jmarshall.com/easy/http/ [2013-05-11]

World Wide Web page, **Using SIP Trunking with Your Company's Phone System Can Save You a Bundle and Improve Your Customer Sevice** http://www.cisco.com/en/US/prod/collateral/voicesw/ps6788/vcallcon/ps11370/what_sip.pdf [2013-05-11]

World Wide Web page, **How Domain Name Servers Work,** http://www.howstuffworks.com/dns.htm [2013-05-11]

World Wide Web page, **File Transfer Protocol (FTP),** http://www.tcpipguide.com/free/t_FileTransferProtocolFTP.htm [2013-05-11]

World Wide Web page, **How E-mail Works,** http://computer.howstuffworks.com/e-mail-messaging/email3.htm [2013-05-11]

World Wide Web page, **How Internet Cookies Work,** http://www.howstuffworks.com/cookie.htm [2013-05-11]

World Wide Web page, **Module 12: Platforms (Technical Sales Accreditation) - English,** https://f5.learn.com/learn6.asp?sessionid=3-702E36DB-2E09-499E-86D0-C884784A87B7&DCT=1&courseid=607 [2013-05-11]

World Wide Web page, **Understand IPv6 Addresses,** http://www.enterprisenetworkingplanet.com/netsp/article.php/3633211/Understand-IPv6-Addresses.htm [2013-05-18]

World Wide Web page, **List of HTTP status codes,** http://support.microsoft.com/kb/943891 [2013-05-18]

World Wide Web page, **Hypertext Transfer Protocol -- HTTP/1.1,** http://www.w3.org/Protocols/HTTP/1.1/draft-ietf-http-v11-spec-01.html#Keep-Alive [2013-05-18]

World Wide Web page, **SAML Authentication,** https://adminconsole.wiki.zoho.com/accounts/SAML-Authentication.html [2013-05-25]