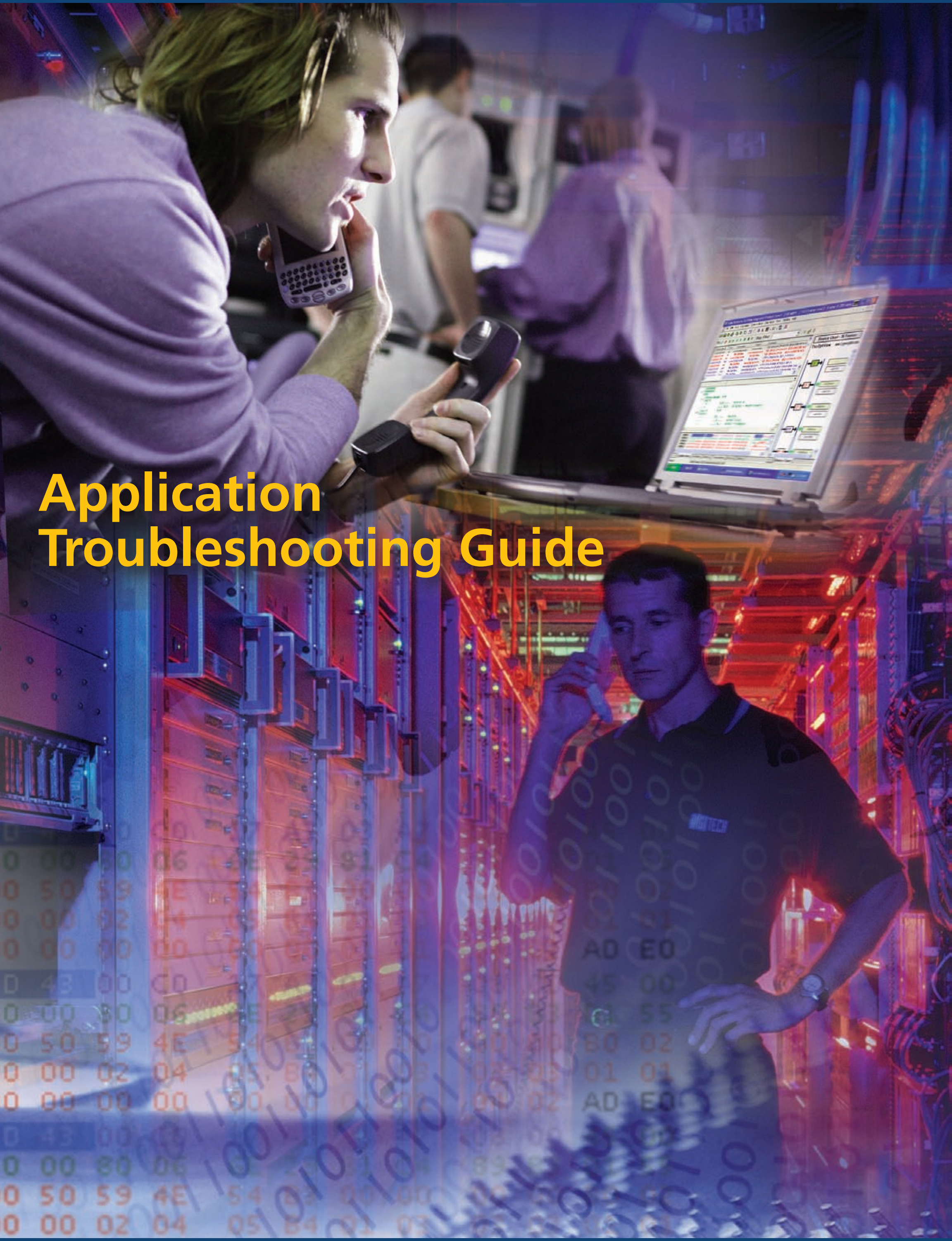
FLUKE networks®



Application Troubleshooting Guide

NETWORK SUPERVISION

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Introduction

Controversy began early on when computers were connected to each other and programmers began to write applications that sent information back and forth between them. The controversy is whether the apparent slowness in performance of the application is due to how the program is executing, or the slowness in the network. When networks were first used, links were very slow when compared to direct connections between computers, printers and disk drives. So, networks often were blamed for the waiting time for the system to produce output.

However, today's networks operate at vastly higher speeds. When users get frustrated with the performance of an application, it is difficult to isolate whether the lack of performance is in the processing in the network attached devices, or in the network itself. In this document, we will address this problem. In order to do this we will need to discuss how applications work. In particular, we will focus on how they use or fail to use the network. This will include both failure to send information in a timely manner and failure to respond to requests coming from the network. We'll also give you a guide in what to look for when troubleshooting applications.

Background

Companies have become almost totally dependent on computers to conduct their business. How many times have you been in a retail store or government office and heard a staff member say, "We can't help you because the system is down." Companies depend on computers for a variety of tasks. Revenue and expenditures are recorded in transaction processing systems.

Orders are taken and inventories are adjusted in similar systems. Most written communications are now done by email rather than by letters that are typed and mailed. Even instant written communications such as instant messaging is moving from fad status to serious message communications where business is negotiated and deals are closed.

Beginning about 2000, businesses began to seriously evolve voice communications over to data networks with VoIP (Voice over IP). Most communications carriers and a majority of large enterprise companies use VoIP exclusively for voice messaging. With the explosion of YouTube and Internet TV, video is beginning to come into enterprise networks as well. Very recently, a new area is collaboration. Under the label of Web 2.0, it represents a group of tools that allow users to share and exchange voice, data and video for the purpose of collaborating on projects. These tools and services are rapidly becoming popular with lawyers, marketing groups, consulting firms and service providers such as architects and engineers.

In TCP/IP networks, applications can be distinguished by the protocol over which they run. Nearly 90% run over TCP (transmission control protocol). That is, they send and receive their messages in packets that have been built by TCP. The other 10% use the UDP (user datagram protocol). These two protocols were devised nearly forty years ago but have proven to be extremely robust and enduring. While modifications in TCP have been made over the years, its essential operation has remained almost constant. In our document we'll focus on TCP applications, since they represent the large majority of the instances in which users are reporting slowness in application performance.

The TCP Protocol

If you segment the traffic on a corporate network into categories determined by the purpose of that traffic, you would probably find a break down something like this: email (10%), transaction processing (25%), http web traffic (35%), broadcast and control (10%) and other traffic (20%). The only part of the mix that is based primarily on UDP is in the “other” category. The remainder use mostly TCP.

TCP is designed to be adaptive to the network. This means when it works on behalf of the application, it modifies how it is interacting with the network. This is based on how the network seems to be responding to the requests and commands it is sending into the network. If the network seems to be fast, it will become more aggressive in sending information. If it senses a slow-down or problem with packets being dropped, it will quickly decrease the rate at which it sends packets.

TCP has three main functions: (1) It handles the rate of flow between the two end applications. (2) It assures reliability for the delivery of all data. (3) It provides for the detection of errors and correction of data by using retransmission of packets that are corrupted or dropped.

TCP is based on a well documented algorithm (formula or operational procedure). However, individual operating systems such as Microsoft® Windows® XP and a particular version of Linux make slight variations in how they implement the algorithm. A detailed understanding of the algorithm is not necessary to troubleshoot a network. Yet, its effect is profound and later, we'll see some examples of how it works.

The adaptive nature of TCP is what makes it very useful. It can be used over 56 kbps dial-up links and 10 Gbps links. The application programmer doesn't have to deal with that significant difference in link speeds - TCP deals with it. On the other hand, this adaptive nature of TCP means that slowness in the application performance might be the way the application is coded or the way in which the application interacts with the network. If the application processing is slow, the data to be transmitted isn't sent to the TCP stack in a timely manner. This also means TCP can't deliver packets to the network. In the other case TCP senses a slow network and decreases the rate at which it offers packets to the network.

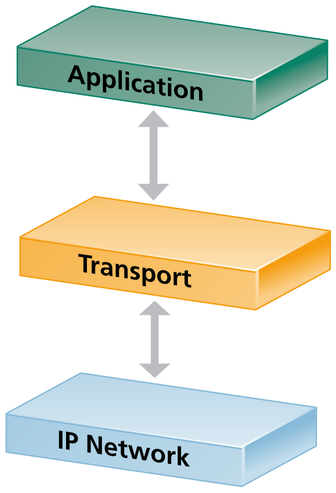


Figure 1 TCP Holds the Application and Network Together

So, when TCP is the protocol being used, both network problems and server problems are hidden by TCP's adaptive nature.

The Life of a Packet

Let's consider a relatively simple model of client/server interaction where TCP is being used. Let's say we have data to send from the client to the server. (It works almost identically when the server sends data to the client.) The client application creates a block of data and sends it to the TCP software in the client's computer. It's placed in an output buffer and timers are started that are associated with that block of data. TCP finds out from the operating system how much data it can send in each outgoing TCP block called a *segment*. Usually it is 1460 bytes maximum. This segment is delivered to the IP software and the IP addresses of the sender and receiver are added. Then the IP software contacts the operating system to say an outgoing message is ready to be sent. When the operating system gives the okay, the outgoing packet is moved to the network interface card (NIC) to be sent. In this entire process, the performance of the client processor is critical to how fast this all can take place.

After the packet is sent from the interface card, the network performance is the critical factor. Packets can follow a short, fast route, a slow, long route or any combination of links. It is common for a packet to traverse 10-20 individual network links when going over an Internet connection. Like the adage about the strength of a chain depending on the weakest link, the speed through the network depends on the slowest link. More often than not, the slowest link is the access link into the network and the egress link out of the network.

At the server, the TCP segment is received on the NIC, delivered to the IP software and the process described above is reversed. During this process, it is the speed of the processing in the server that affects how quickly the data is received and the acknowledgement is prepared to be sent back.

In conclusion, there are three critical phases necessary for the successful delivery of the application message to the server's application:

1. Processing of the message in the client to prepare it to be sent.
2. Network delivery.
3. Processing in the server when the message arrives.

Other Troubleshooting Factors

There are some other things that make it difficult to troubleshoot TCP applications. The TCP protocol isn't very well understood. Just recently information about TCP's operation has begun to appear in networking textbooks and industry whitepapers. Yet, it is the single layer of the seven layer model which has a significant influence on the performance of the application-to-network interface. Because it isn't well understood, network engineers and technicians usually don't look at it as a source of explaining what is causing a problem. Often, they look at server configurations, cabling issues, or system memory as explanations when actually these have little relationship to the underlying problem.

Also, TCP isn't an especially efficient protocol. For years, data communications experts considered an overhead amount of 20% or more to be excessive. But, TCP routinely operates with 50% or more overhead when the actual application data bytes and the total bytes are used to calculate the overhead.

Finally, as we will learn later in this document, TCP reacts very aggressively to packets that are dropped within the network. When a single packet is dropped, a file transfer may decrease its delivery rate by 70% or more.

Why UDP is so Different

While we won't examine UDP in detail in this paper, it is worthwhile to consider how it compares with TCP. That will amplify what we have learned about TCP.

UDP makes no provision for flow control, error detection or correction, or the reliability of data delivery. These responsibilities are pushed up to the application. Therefore, since UDP involves much less processing, poor performance is more often dependent on the performance of the network or the slowness of the processor unit. If the processor is slow, other applications will show the same slowness. This makes the isolation of the cause of slowness less complicated than in the case of TCP applications.

Why is Understanding Troubleshooting Important

When applications appear to be slow, frustrated users waste time and energy. Slow applications cause lost time which, in turn, means lost money. Also, IT staff time is misspent. In some instances, help desk or PC analysts expend much of their time rebooting client stations or checking IP configurations and cabling connections.

When an effort is made to learn about troubleshooting applications and the skills are applied to the system to improve the performance of both the network and the applications, it's an investment in the company. It pays off in dividends the same way that saving on shipping costs, reducing fuel consumption or eliminating any other inefficiency, pay off.

But troubleshooting TCP applications isn't easy. Unlike voice or video, which are typically UDP based, application performance degradation isn't obvious. If the sound is bad or the TV screen shows distortion, everyone sees it immediately. But with TCP applications like transaction processing systems, performance tends to be judged based on how the system behaved yesterday. If it is slower today than it was yesterday, users begin to complain. Even worse, if the application degrades slowly, the change may not be detected at all and the company begins to lose money through the inefficiency.

Understanding Types of Applications

There are many examples of applications that will fall under our scrutiny. Web applications include both applications that actually communicate over the Internet and any that use a browser such as Internet Explorer® or Firefox.® The latter are often called **web enabled applications**. It is becoming more common for applications to be described as "browser-based." This generally means that the user begins by opening a browser and connecting to the server application from within that browser window. Most of us have used these applications when we do on-line banking, order a movie, or reserve an airline ticket. What is important to us is that the underlying traffic will be transmitted between the client and the server using HTTP (hypertext transfer protocol) or HTTPS (secure hypertext transfer protocol). Both of these are TCP based application program interfaces.

SQL (structured query language) **applications** involve calls from a client application to a server database application for records or tables from a SQL database. These can be, or don't have to be, web enabled and use HTTP or HTTPS. For example, the client application can present a result to a user that asks for a field to be completed on the screen. After filling in a value, it is sent to the server to do a computation. The server takes the result and determines that another record needs to be returned to the client. All of this interaction takes place under the control of the TCP protocol and will be affected by TCP's operation, the individual processors' performances and the network's performance.

Closely associated to database activity are **transaction processing** systems. These often overlap SQL database applications.

In transaction processing systems, the emphasis is on taking relatively small amounts of data and doing a calculation in a short period of time. ERP systems that have individual components such as an accounts payable module, inventory module, or order entry module are examples of transaction processing systems. When a user changes a customer's telephone number, a transaction has occurred. The data to make such a change and the information that shows that the change occurred are transferred between the client and the server under the control of TCP.

Imaging systems, such as those that inventory photos or x-rays, are also very often TCP based. They are usually very large files and moving them from a server to a client is a bulk transfer. Such bulk transfers are the reason TCP was originally designed. One major application program interface that is used to do this is FTP (file transfer protocol). It can be executed from the command line, from within a file transfer program or from within a browser. A user clicks on a thumbnail of the image they want to view and the FTP protocol is invoked to retrieve the image from the server. FTP is one of the best examples of simple application of the TCP protocol. The client identifies the file to be retrieved (in this case an image), and a separate connection is opened for the transfer. The emphasis in the operation is to send as much data as possible as quickly as the network can handle it. These applications can saturate links on networks more quickly than almost any other type of application. As a result, they appear to be more sensitive to the method of operation dictated within TCP. Small network problems become very evident in a file transfer, especially if it is a large file.

Data warehousing systems are also dependent on bulk file transfers. A data warehouse is generally considered to be a highly organized, indexed repository of a company's electronic documents. When a document needs to be retrieved, the client application indicates the document or documents or parameters that uniquely define the information needed. When the server application receives this information, it retrieves the information and sends it as a bulk file transfer. Consequently they are nearly always TCP based. And, as a result, they behave very much like other file transfer based processes.

While **VoIP** payload is transferred between the phones using UDP, the call set-up often uses TCP. In the exchange that takes place when a phone goes off-hook, only small amounts of data are transferred. However, the process can involve ARP, DNS and the other process that typically define a connection to a server. Poor performance or failure in this connection won't affect the quality of the call. It will be apparent in the time it takes to make the connection or the lack of signaling tones to which we have become accustomed. For example, the user might dial but not hear either a busy signal or a ringing tone.

Another category of applications that use TCP are **CRM (customer relationship management) systems**. They are usually a combination of transaction oriented and database systems. They also depend on TCP.

Before we delve more deeply into the application flow process, we need to consider one more item.

Application Flow

We’re going to consider six issues that affect application flow:

1. DNS lookups
2. ARP resolution
3. Establishing the TCP connection
4. Sender/receiver (interaction)
5. Data flow
6. Closing the TCP connection

In this section we will not consider UDP applications because they typically represent 15% or less of the traffic on the network and it is easier to isolate the cause of the poor performance. We will discuss UDP later in the paper.

The Amazing Sequence of Events in Connecting to a Server

Very few of us ever stop to consider what is actually taking place when we click on an icon that tells our application to get something from a server. But understanding the process can be very helpful. Let’s start when our user sits down at their desk to begin working with the application. We’ll assume the computer was turned off over night and the first thing they want to do is check the company’s home page for notifications about weather related cancellations.

Here is a typical sequence:

1. Computer off.
2. Computer powered up.
3. OS loaded, NIC detected.
4. TCP/IP stack checked: IP address, mask, default router (gateway) and DNS server known.
5. If DHCP is enabled, the DHCP server is contacted to receive these values.
6. User indicates to start connection to server (e.g. open IE).
7. IE request is created for the default home page (e.g. www.google.com).
8. IP software looks up a DNS server IP address (which we assume is not local).
9. IP software sends ARP request to local net to get MAC address of router.
10. The router receives the broadcast, recognizes that it is the target of the query and sends a response containing its MAC address onto the network as a broadcast.
11. IP software sends IP packet with DNS request to router, which forwards to DNS server.
12. DNS server returns IP address for www.google.com.
13. TCP/IP stack sends connection request to server at Google.™
14. Network delivers connection request.
15. Connection established.
16. (Story to be continued later).

It's rather easy to see that there is much that can go wrong. We'll be referring back to this sequence later in our discussion.

With the dependence that companies have on TCP based applications, we now turn our attention to the six phases of the TCP connection between the client and server applications.

DHCP

There are two ways a user can make sure that a client has the appropriate configuration parameters to connect to the network. One way is to assign the values through the operating system. In Windows,® a screen in the Network Settings allows entry of the IP address, subnet mask, default router, and DNS server. On the other hand, in the same window, the user can check the radio buttons to obtain these automatically. In this case, the *dynamic host configuration protocol (DHCP)* will be used. Figure 2 illustrates the process.

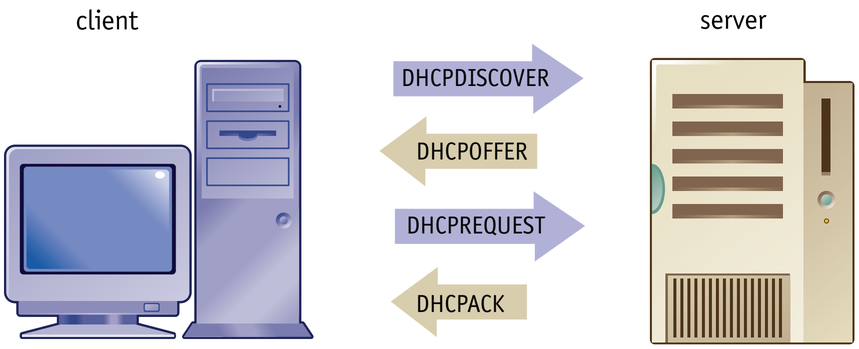


Figure 2 DHCP Process

To begin, the client sends a local broadcast indicating it needs service from a DHCP server. This frame is called a *DHCP Discover*. Typically, all DHCP servers that hear the broadcast will respond with a broadcast, which is called a *DHCP Offer*. The offer will contain the configuration parameters the client needs. More than one offer may be received since there may be multiple DHCP servers. The client chooses one of the offers by sending a broadcast called a *DHCP Request*. The servers that sent the offers that weren't accepted realize they are no longer part of the process. The server that made the accepted offer completes the assignment of the parameters by sending a broadcast called the *DHCP Acknowledgement*.

In the event that the DHCP server is on a different network than the client, the local router that separates their networks must know to forward DHCP broadcasts. This forwarding action is called DHCP relay. Routers do not normally forward broadcasts.

DNS Lookups

Before, we briefly mentioned DNS. But now let's consider it in more detail. DNS refers to both the protocol *domain name services* and the device called the *domain name server*. While it seems to contain a redundancy, a DNS server is a domain name services server. The purpose of DNS is to match IP addresses of hosts to the name that resides in that host. For example, if Apple Baker Company, Inc. has the domain name abc.com registered with the Internet community and they have also registered the address 221.221.13.0, we say that abc.com maps to 221.221.13.0.

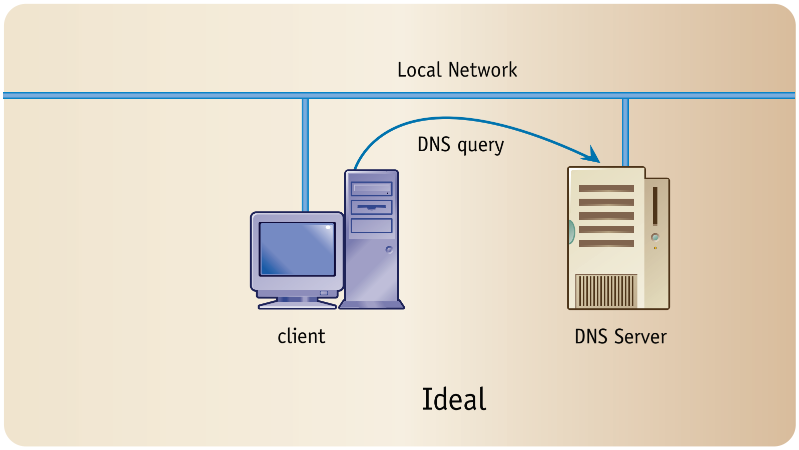
221.221.13.0 **<->** abc.com

When a process queries with a name and learns the associated address, we say the name was *resolved*. Most often names are resolved to provide the address. However, it is sometimes necessary to resolve the address to learn the name. This is known as a reverse lookup. DNS server can do both of these. The DNS server is a computer with a stored table that contains the mapping (or association) of many names and IP addresses. Note that the DNS table does not contain the hardware addresses of the device that has a particular IP address. That's in a different table and we'll consider that later.

In the sequence of events on page 14, the client browser knew the home page was www.google.com. In steps 10 and 11, you see that the client browser asked to be connected to the server at Google™ But the client's TCP/IP software didn't know the IP address for www.google.com. As a result a query was sent to the DNS server and the IP address was returned.

On some rare occasions, another method is used to resolve names to addresses. Clients will store a host name table. This table acts like a local DNS table and also the administrator of the local computer to create fixed, permanent associations between names and IP addresses. This is occasionally used in manufacturing operations where there is a need, or desire, to avoid a name server. This technique is not adequate for situation where the user might connect to the public Internet because the names that need to be resolved are unpredictable.

Now that we know something about how DNS works, let's consider what can go wrong. First, it is important to understand that applications are rarely written to send messages or data to an IP address. It is almost always sent to a server with a name. So, DNS is almost always invoked. Second, the address of the DNS server that is configured in the client or offered by DHCP is often the wrong server or one that isn't easily reached as in Figure 3. If it is the wrong server, the name query will be relayed to another server, and potentially to more servers until it reaches one that can resolve the name. Consequently, to the client application, it appears the query was handled slowly by the local DNS server, when, in fact, the query was resolved by a device that was much further away. In addition, for efficiency and redundancy purposes, many large enterprises have many DNS servers. In one large university, there are ten DNS servers. Users are emailed a list and they may pick two to configure their client. In a case like this, the likelihood that the user picks the best two servers, assuming there are two best, is less than 3%! So, in most cases if there are many DNS servers, you can probably assume the best two haven't been configured in the client.



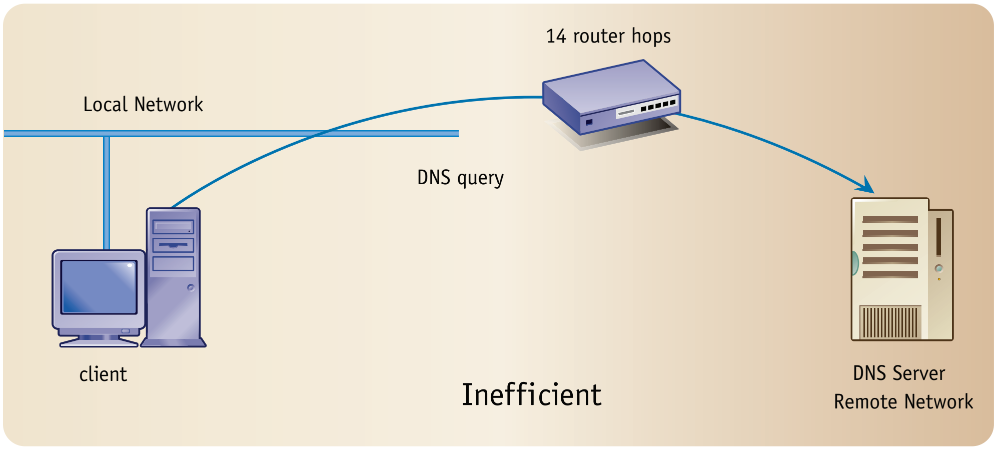
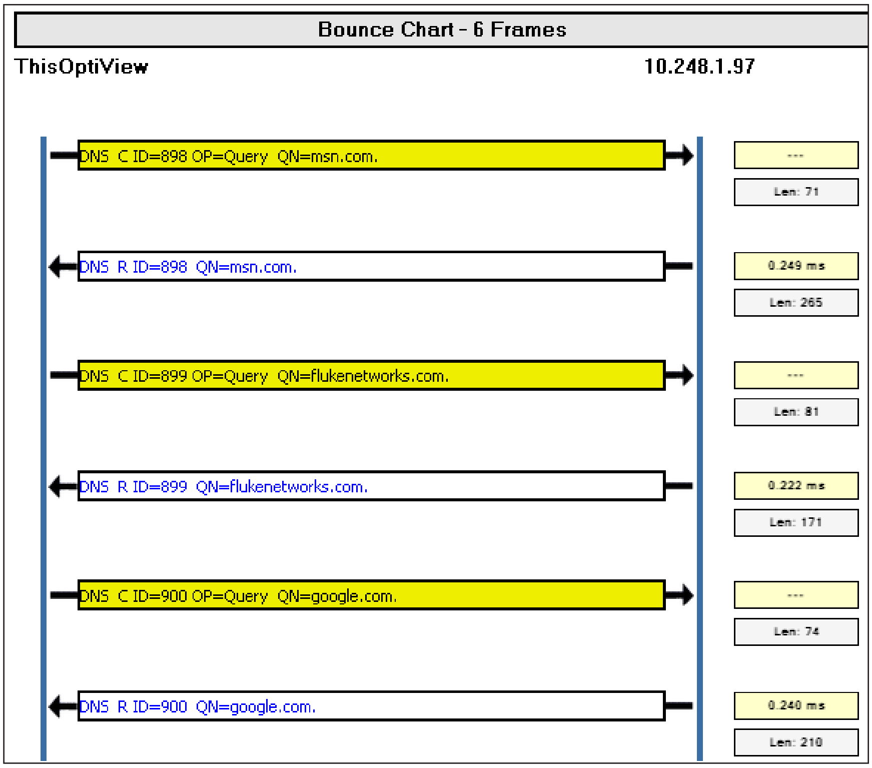


Figure 3 The Location of DNS

The packet that contains the query name is called a command. The response, coming from the DNS server that contains the IP address, is called the response. In Figure 4 you can see three DNS commands and three corresponding responses.



Bounce Chart - 6 Frames

ThisOptiView 10.248.1.97

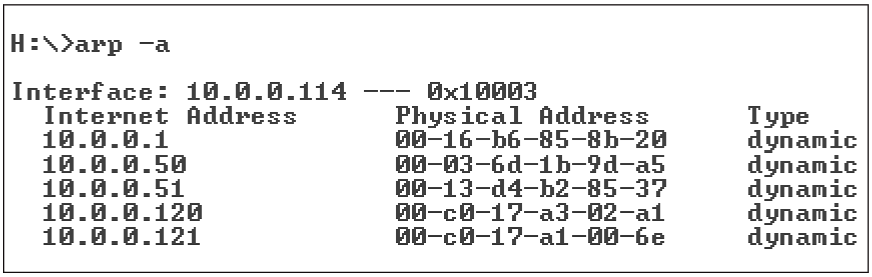
```
DNS  C ID=898 OP=Query  QN=msn.com.              ---        Len: 71
DNS  R ID=898  QN=msn.com.                       0.249 ms   Len: 265
DNS  C ID=899 OP=Query  QN=flukenetworks.com.    ---        Len: 81
DNS  R ID=899  QN=flukenetworks.com.             0.222 ms   Len: 171
DNS  C ID=900 OP=Query  QN=google.com.           ---        Len: 74
DNS  R ID=900  QN=google.com.                    0.240 ms   Len: 210
```

Figure 4 DNS Commands and Responses

So, in conclusion, DNS can be slow due to network slowness, improper or poor choice of DNS server, or a lost query. In the latter case, usually the client automatically sends a second or third request after it times out on the first request. DNS can also fail, if the query contains a name that can't be recognized or if no DNS server can be found. Of these two possibilities, the second is more common because the administrator of the computer lists a nonexistent DNS server or provides an invalid address in the TCP/IP configuration. For example, you might be surprised by how many users configure their TCP/IP stack by inserting the company's email or web server in the DNS server field.

ARP Broadcasting

ARP (address resolution protocol) is used when a device knows the IP address but doesn't know the MAC address of the same device. In our description of connection to the server on page 14, step 9 indicates what happens. The client was configured with the IP address of the router. But sending the packet requires the MAC (hardware) address of the device. The ARP broadcast contains the query and the response to the query fills the need and the addresses are resolved. When the client learns the MAC address, it will temporarily store it in a table called its arp cache (it is not common to use upper case letters in this case.)



```
H:\>arp -a

Interface: 10.0.0.114 --- 0x10003
  Internet Address      Physical Address      Type
  10.0.0.1              00-16-b6-85-8b-20     dynamic
  10.0.0.50             00-03-6d-1b-9d-a5     dynamic
  10.0.0.51             00-13-d4-b2-85-37     dynamic
  10.0.0.120            00-c0-17-a3-02-a1     dynamic
  10.0.0.121            00-c0-17-a1-00-6e     dynamic
```

Figure 5 The ARP Command

Figure 5 shows the arp table of a client. This table is available in Windows® clients by typing the command *arp –a* on the command line. The entries in the arp table are usually stored for about two minutes in Windows.® Then it is discarded. Other operating systems store the entries for different lengths of time. All operating systems supporting TCP/IP will have such a table. The arp table is a good place to look to first to see if a device knows the address of a particular local station. It is important to remember that the arp table contains only devices from the *local* network (subnet).

Devices beyond a router will not be listed because they are on another network.

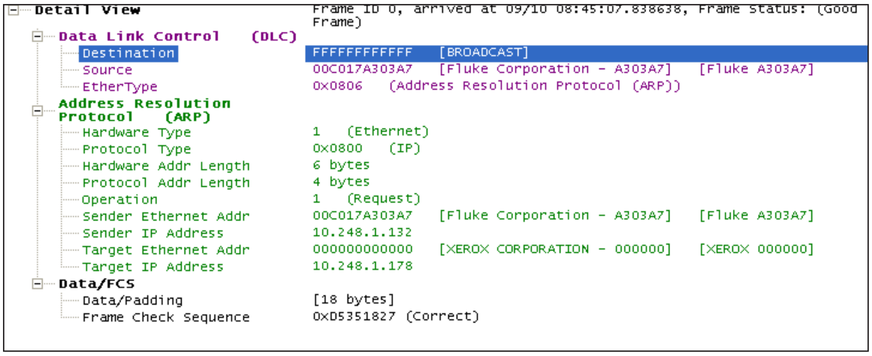
Another view of this problem is illustrated here:

```
        IP A  <-->  IP B
     MAC x         MAC y
```

Suppose the device with IP address A wants to send a message to the device with IP address B. It knows its own MAC address but not the MAC address of the other station. Its application directed the message to be sent to IP address B. Or, possibly, its application directed it be sent to a name at B and DNS provided the IP address B. Either way, it still needs the MAC address of B to complete its task. ARP will be invoked and the entry B and y will be added to the arp table for temporary storage.

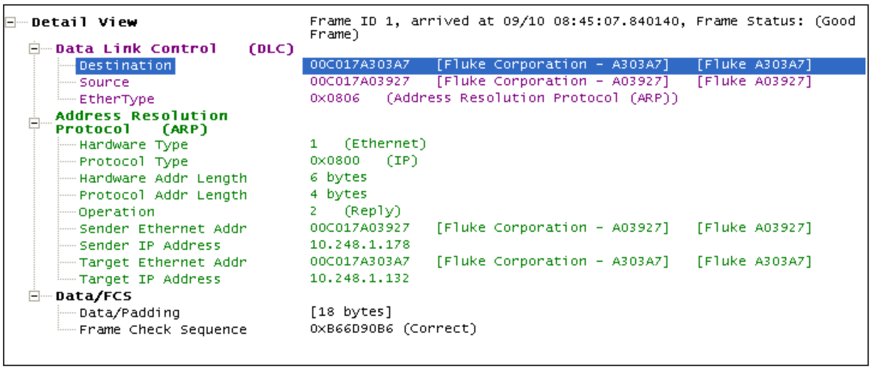
The ARP protocol is a necessary part of TCP network (in the case of IPv4). ARP is also used for some more discretionary tasks. Sometimes referred to as *discovery* techniques, ARP is used when a device like a server wants to see who is actually on a local network. By sequentially sending an ARP query to every possible address and then watching who responds, it can build a list of devices that are actually functioning on the network. Windows® browsing and many network troubleshooting tools implement this technique.

Figures 6 and 7 show an ARP query and the corresponding response.



```
Detail View                                  Frame ID 0, arrived at 09/10 08:45:07.838638, Frame Status: (Good
                                             Frame)
  Data Link Control    (DLC)
    Destination                              FFFFFFFFFFFF    [BROADCAST]
    Source                                   00C017A303A7    [Fluke Corporation - A303A7]    [Fluke A303A7]
    EtherType                                0x0806    (Address Resolution Protocol (ARP))
  Address Resolution
  Protocol    (ARP)
    Hardware Type                            1    (Ethernet)
    Protocol Type                            0x0800    (IP)
    Hardware Addr Length                     6 bytes
    Protocol Addr Length                     4 bytes
    Operation                                1    (Request)
    Sender Ethernet Addr                     00C017A303A7    [Fluke Corporation - A303A7]    [Fluke A303A7]
    Sender IP Address                        10.248.1.132
    Target Ethernet Addr                     000000000000    [XEROX CORPORATION - 000000]    [XEROX 000000]
    Target IP Address                        10.248.1.178
  Data/FCS
    Data/Padding                             [18 bytes]
    Frame Check Sequence                     0xD5351827 (Correct)
```

Figure 6 The ARP Query



```
Detail View                                  Frame ID 1, arrived at 09/10 08:45:07.840140, Frame Status: (Good
                                             Frame)
  Data Link Control    (DLC)
    Destination                              00C017A303A7    [Fluke Corporation - A303A7]    [Fluke A303A7]
    Source                                   00C017A03927    [Fluke Corporation - A03927]    [Fluke A03927]
    EtherType                                0x0806    (Address Resolution Protocol (ARP))
  Address Resolution
  Protocol    (ARP)
    Hardware Type                            1    (Ethernet)
    Protocol Type                            0x0800    (IP)
    Hardware Addr Length                     6 bytes
    Protocol Addr Length                     4 bytes
    Operation                                2    (Reply)
    Sender Ethernet Addr                     00C017A03927    [Fluke Corporation - A03927]    [Fluke A03927]
    Sender IP Address                        10.248.1.178
    Target Ethernet Addr                     00C017A303A7    [Fluke Corporation - A303A7]    [Fluke A303A7]
    Target IP Address                        10.248.1.132
  Data/FCS
    Data/Padding                             [18 bytes]
    Frame Check Sequence                     0xB66D90B6 (Correct)
```

Figure 7 The ARP Response

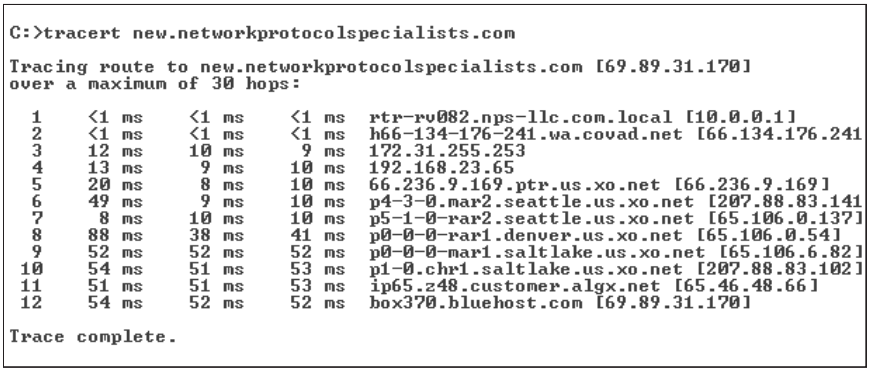
The Internet standard that specifies the ARP protocol also makes provision for a process called reverse ARP. This allows a device to broadcast a MAC address to find out who has the corresponding IP address. While it might seem this could be useful, it is rarely used in practice.

The Route to the Server

Suppose that the client has now obtained the IP address of the server and knows it isn't on the local network. The client also knows the MAC address of the router. The client has a connection request that needs to reach the server. Of course it will send it to the router which will in turn forward it to another router, and so forth until it arrives at the router on the server's local network. Keep in mind, that when it reaches the destination network, that router may need to invoke ARP to get the server's MAC address. But how many steps (links) will the connection request pass through and how fast and reliable is each link? And, is the route selected the optimal route? Usually that is determined by two things: how the network was designed and what routing protocol was used. If a particular link is slow and was used to create the path from the client to the server, a response to the connection request will be slow. Had a faster link been used to create the path the response will be quicker.

In today's IP networks, routers establish a set of links between the potential sending and receiving networks using a *routing protocol*. Investigating how such protocols work is beyond the scope of this document, but it is important to understand a few things that are common to all routing protocols. First, they are dynamic and automatic. That is, they will build a set of links among themselves so that all devices can be reached by other devices. Second, they periodically update each other so that if a link fails, a new set of paths are created. Finally, they all make provisions for *static routes*, or routes that are manually created and remain fixed under all circumstances. Consequently, the choice of the route from the client to the server is made by the network and not the client or the server. Some routing protocols allow routers to select links based on the quality of the links. Other routing protocols use only the criterion of how many links remain to the destination network.

Probably the most widely used tool for investigating the route between two devices is the application *trace route*, which is built into nearly all TCP/IP stacks. While we'll discuss trace route later, it is helpful to look at the output created by using it. In Figure 8 you can see that the route to new.networkprotocolspecialists.com was traced from the host that issued the command. Each line that is numbered shows a routing device that the trace route packets followed. The next three entries after the router number are the measurements of time it took to traverse the link to that router. So, for example, between the device in step 5 and the device in step 6, it took 49 ms., 9 ms., and finally 10 ms. to reach the router at 207.88.83.141.



```
C:>tracert new.networkprotocolspecialists.com

Tracing route to new.networkprotocolspecialists.com [69.89.31.170]
over a maximum of 30 hops:

  1    <1 ms    <1 ms    <1 ms  rtr-rv082.nps-llc.com.local [10.0.0.1]
  2    <1 ms    <1 ms    <1 ms  h66-134-176-241.wa.covad.net [66.134.176.241
  3    12 ms    10 ms     9 ms  172.31.255.253
  4    13 ms     9 ms    10 ms  192.168.23.65
  5    20 ms     8 ms    10 ms  66.236.9.169.ptr.us.xo.net [66.236.9.169]
  6    49 ms     9 ms    10 ms  p4-3-0.mar2.seattle.us.xo.net [207.88.83.141
  7     8 ms    10 ms    10 ms  p5-1-0-rar2.seattle.us.xo.net [65.106.0.137]
  8    88 ms    38 ms    41 ms  p0-0-0-rar1.denver.us.xo.net [65.106.0.54]
  9    52 ms    52 ms    52 ms  p0-0-0-mar1.saltlake.us.xo.net [65.106.6.82]
 10    54 ms    51 ms    53 ms  p1-0.chr1.saltlake.us.xo.net [207.88.83.102]
 11    51 ms    51 ms    53 ms  ip65.z48.customer.algx.net [65.46.48.66]
 12    54 ms    52 ms    52 ms  box370.bluehost.com [69.89.31.170]

Trace complete.
```

Figure 8 Trace Route

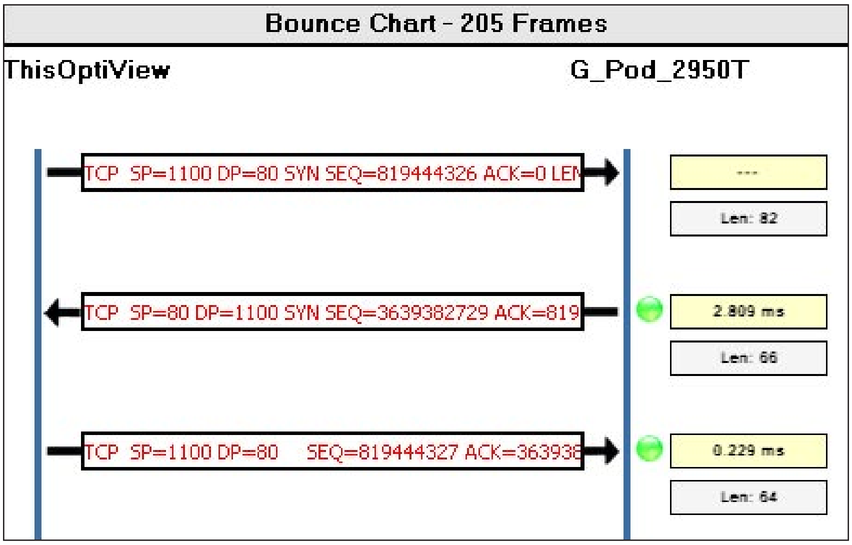
Establishing the Connection to the Server

The method of creating the connection for all TCP applications is the same. It is widely known as the three-way handshake. Here's what happens.

Before data can be transferred over the connection, certain parameters of operation must be agreed upon by the sending and receiving TCP software modules. Recall that TCP is responsible for flow control, reliable delivery, and error control. In order to agree on how this will be handled, each TCP stack exchanges certain information during this three-way handshake. For example, in order to establish the maximum number of bytes that will be permitted in each packet, the sender includes a statement of the intended MTU (maximum transmission unit). While this is normally 1460 bytes, it isn't mandatory that value be used. If the other TCP module receives this indication and doesn't like the value and thinks it should be lower, it simply doesn't acknowledge that it received it. The sender must take the hint and lower the value.

A second thing that must be established is how many bytes the receiver can receive in total in its receive buffer. This value, called the *window advertisement*, is also indicated in the first packet exchanged in the handshake. The other end records this value and any adjustments it receives so that it never overwhelms its partner with too much data. The exchange of these two values provide for flow control.

Another value that must be established is the point at which each end will begin to count bytes of data. This is a random value called the *initial sequence number* and each announces their value in the first two steps of the handshake. From that initial value, every byte of data transferred increases the sequence value by exactly one. So, for example, if the initial sequence number for a device is 36,100,000 and it transfers 50 bytes of data, the next sequence value it will use is 36,100,050.



TCP SP=1100 DP=80 SYN SEQ=819444326 ACK=0 LEN

TCP SP=80 DP=1100 SYN SEQ=3639382729 ACK=819

TCP SP=1100 DP=80 SEQ=819444327 ACK=363938

Figure 9 The Three-way Handshake

Now, let's get back to the three-way handshake. We're assuming the client is opening a connection with the server. As we discuss this you can refer to Figure 9, which shows the process. Again, its purpose is to exchange these values and acknowledge that they were received. The first step in the process is for the client device connection request to send a packet to the server called the SYN packet (SYN is short for synchronization. They are synchronizing values). That packet contains several important numbers and indicators. First it contains the *port* number for the process (application) it wants the connection to be with.

Port numbers for server processes generally are *well-known* or *registered*. That means that everyone agrees on which application process the number is associated with. For example, email often uses ports 25 or 110. HTTP uses port 80. There are hundreds of others. It's a good idea to learn most of the common ones. Like so many other network topics, using your favorite search engine and the phrase "TCP port numbers" will lead you to many lists that are available. In this step, the client will normally use a randomly chosen port. Windows® often selects a value in the range of 2000-2500.

The SYN packet also contains the sender's window advertisement. Also, it contains a request for the maximum transmission unit (MTU). Finally, it indicates the fact that it is a SYN packet by setting a single bit equal to one. That particular bit is never set to one unless SYN is being indicated.

In the second step in the three-way handshake, the server sends a response called the SYN/ACK packet to the client. It contains its window advertisement, initial sequence number and the port numbers of the two application processes. It also takes the sequence number it received, increases the value by one and returns it as its *acknowledgement value*. This is how it tells the other TCP entity that it is now synchronized to its sequence numbers. It will also change the value of a single bit, the acknowledgement bit to one. Therefore, in its packet both the SYN bit and the ACK bit are one and that is why the packet is called the SYN/ACK packet.

In the third and last step of the handshake, the client sends a packet with the sequence number increased by one (That's cheating because it didn't actually send data bytes. It's the only exception to the rule about sequence numbers.) This third packet allows the client to acknowledge that it received the server's acceptance of the opening of the connection. The connection is now prepared to allow data to flow between the two applications represented by the port numbers.

A common metric of performance of the network application is the round trip time (RTT). This is the time from when the SYN is sent and the SYN/ACK is returned. This value is one of the things the sending TCP stack uses to determine how much data should be offered to the network. If the RTT is low, the network appears to be fast. If the RTT is high, the network or the server appears to be slow and the TCP stack will adjust accordingly.

In Figure 10, an HTTP Get command is sent to the server in the first step. 65 ms. later that packet is acknowledged. This time is the RTT. It is important to note that this value isn't the application response time. This is the acknowledgement of the TCP protocol in the server. The application responds in the third step shown in Figure 10.

Sender/Receiver Interaction

The most common model for interaction in the TCP/IP environment is the client server model. In this model, the client makes a request for some form of service and the server fulfills this request. For example, the client browser may request the objects on a home page. An ERP client may request a field from within a data record on the server. As illustrated before, a client process may invoke FTP to receive a large file. In each case, the client is making a request that it hopes the server will fulfill. The server applications that fulfill these requests are sometimes called *services*. So, we may say that a particular service is or isn't available on the server. The way a client knows whether or not a service (process) is available is by the response to the SYN packet. If the server responds to the SYN with a SYN/ACK, it is informing the client that the service is available. For example, that's why requests for web pages are sent to port 80. That port identifies a server that is acting as a web server. While it might also act as an email server, if it doesn't respond to SYNs sent to port 25, it is informing the client it doesn't do email services using the SMTP (simple mail transfer protocol) system. That service is not available.

So, after a connection is established, a client makes a request for information. Say the request is for a web page element. The server may acknowledge receipt of the request with an ACK packet containing no data. Then, it may send the file containing the requested information. In fact that is what is being illustrated in Figure 10.

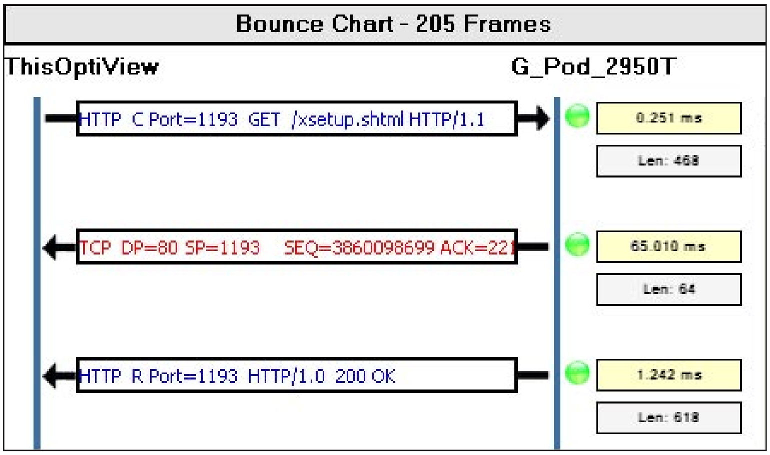

Bounce Chart - 205 Frames
ThisOptiView
G_Pod_2950T
HTTP C Port=1193 GET /xsetup.shtml HTTP/1.1
0.251 ms
Len: 468
TCP DP=80 SP=1193 SEQ=3860098699 ACK=221
65.010 ms
Len: 64
HTTP R Port=1193 HTTP/1.0 200 OK
1.242 ms
Len: 618

Figure 10 A Request and the Response

TCP acknowledgements are governed by a fairly complicated set of rules. However, there is a basic set of behaviors we need to understand. First, not every packet needs to be acknowledged. The TCP stack, written for the operating system involved, may implement a procedure in which every packet is acknowledged, only occasional packets are acknowledged, or some combination of the two rules. What is rather typical of all TCP stacks is that data is sent in a TCP segment and then the receiver waits until one of two things occurs. Either it receives another segment, or 200 ms. transpire. If the 200 ms. pass and no additional segments are received, it will acknowledge the one it received. If a second or third segment is received, it may acknowledge them at any point that it is ready to do so. Sometimes when the packets are examined, you will see a very homogeneous pattern: two segments sent, acknowledgement received, two more sent, acknowledgement received and so forth.

However, sometimes, acknowledgements will be received while the sender is sending a block of four or more segments. The behavior of the receiver – to wait 200 ms. to see if there is more data-is called *delayed acknowledgement*. Its purpose was to cut down on the number of acknowledgements. But it has become controversial because it affects the senders estimate of the real RTT.

The Flow of Data

TCP has a long list of rules that the sending and receiving protocol software must follow. The method used in the complete implementation of the protocol is beyond the scope of this document. However, here are a few of the rules that help in understanding the relationship between how TCP behaves and the network performance:

- Delayed acknowledgements, described above.
- If packets are received out of order, the last segment isn't acknowledged until the missing segments are received. Rather, the last segment that was received in order is acknowledged.
- If a sender doesn't receive an acknowledgement of a segment, it doesn't immediately resend the segment. (It could still be in transit). Instead, it waits until it receives *three* duplicate acknowledgements of the previously received segment. At that point it assumes the next segment was lost and proceeds with the retransmission. Since this is often before a time out on the segment has occurred, the policy is called *fast retransmission*.

- It uses a technique called *slow start*. In this procedure one or two segments are sent and the acknowledgements are awaited. If they return quickly, the sender increases the block size (that is, the number of segments it sends simultaneously). It will continue to increase the block size until one of two things happens: either there is a time out because an acknowledgement isn't received (such as when a segment has been discarded) or the total data sent is one-half the receiver's advertised window value.

Let's consider some examples. Suppose the sender has a hundred or so segments of data to transmit. We show the interaction in Figure 11. However, rather than complicate things by using the actual acknowledgement values, we simply use consecutive integers that correspond to the number of blocks of data.

Step one starts after the three-way handshake has been completed and the client is ready to up load its information. The client sends a frame and awaits an acknowledgement. The acknowledgement returns quickly as Ack1. In step 2, because the client's TCP stack is using slow start and it has received one acknowledgement, it now increases the send group to two segments and sends segment 2 and segment 3. Again, the server responds quickly with Ack3. In step 3, it increases the outgoing group size to 4 because it received acknowledgements for two more segments. It sends all four segments. Once again, the server quickly responds with Ack7, assuring the client that it received all four segments.

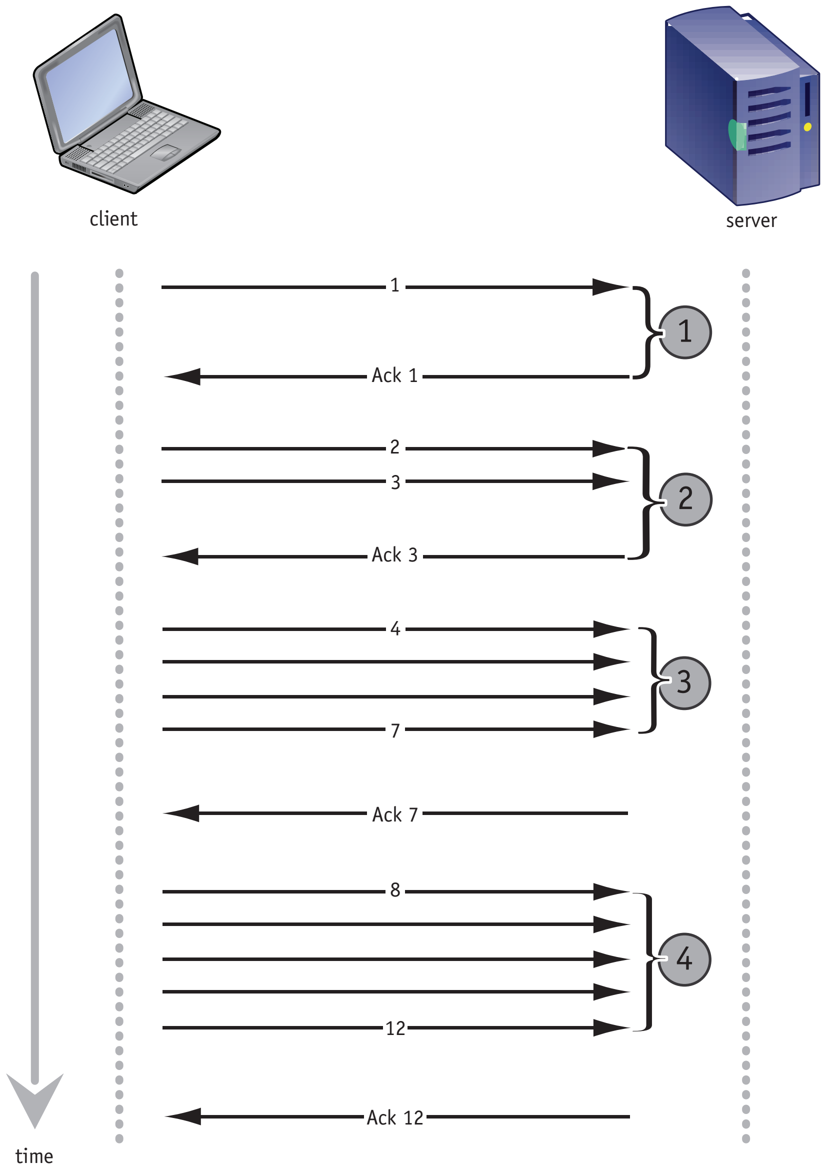


Figure 11 TCP Operation

At this point we make an assumption. Let us suppose that by sending the four segments, the receiver's incoming buffer became more than half full. That is, assume that the client has sent enough information that the next window advertisement from the server will be less than half what it started out as. In this case, the client is not allowed to increase its send group by 4, to total 8 out-going segments. That would overfill the receiver's buffer. So, in this new phase of the operation, the client increases the group size by one segment at a time. Therefore, it increases the number of outgoing blocks to five. If that succeeds, it will increase to six, and so forth. Of course, this entire scenario is subject to whether the server is able to completely empty its incoming buffer upon each transmission. We have assumed that is the case. Later, we'll consider an example where that isn't true.

This example points out several important features of applications that are using TCP. First, they will appear to adapt to the speed of the network. If the response of the server and the speed of the network are both good, TCP sends more and more data until either the link or the recipient has been saturated with information. Second, TCP operation is more complex than is generally thought. It is important to keep in mind that this is a very simplified explanation of how a very complex algorithm works.

Let's consider a less than ideal situation. This one may surprise you. In Figure 12 we see the client beginning to transfer data to the server in the same manner as before. However, this time, we'll assume a single segment is lost.

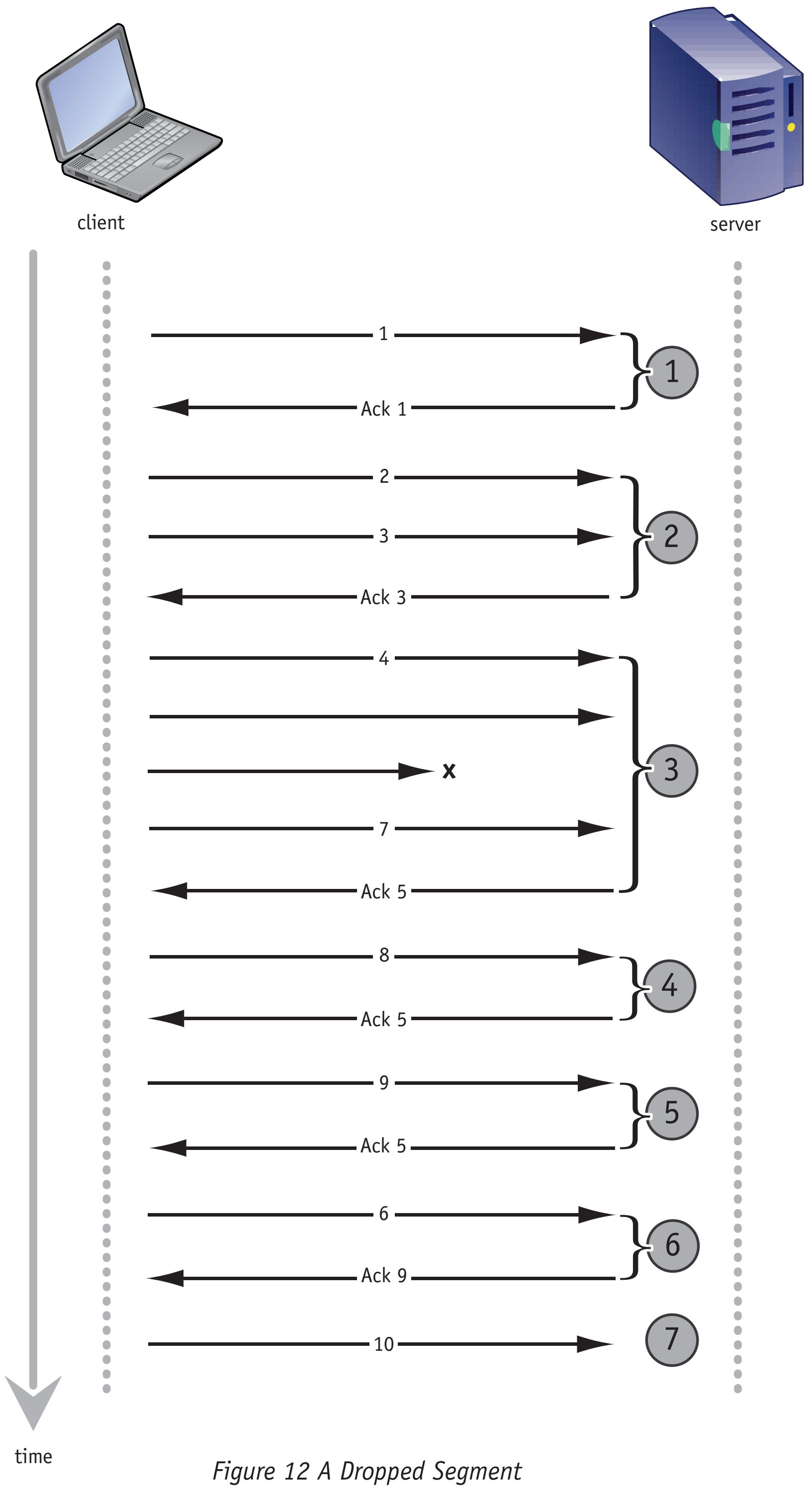


Figure 12 A Dropped Segment

In step 1, the client sends segment 1 and the server quickly responds with Ack1. In step 2, using slow start, the client sends two blocks of data and the server quickly responds with Ack3. In step three, again because of slow start, the client sends four segments, segments 4, 5, 6 and 7. But segment 6 is discarded by the network. When the server receives segments 4, 5 and 7, it doesn't acknowledge receipt of 7. Instead, it acknowledges segment 5, the last one that was successfully received *in order*. In step four, the client has received Ack5 but isn't sure if six was lost or is still wandering around the network waiting to arrive. So, the client sends just segment 8. Since, segment 6 isn't wandering around but has been lost, the server quickly acknowledges receipt of data but it acknowledges segment 5 again with Ack5. In step 5, the client still isn't permitted to assume that segment 6 is lost so it again sends one segment, segment 9. The server sends Ack5 again. Finally, because the client has received *three duplicate acknowledgements* (steps 3, 4 and 5), it is permitted to assume that segment 6 has been lost and it retransmits that segment. Now that the server has segments 4 though 9, it sends Ack9. As illustrated in the final step, the client continues as it begins and sends segment 10. It will continue to implement slow start with sending succeeding segments.

This illustration showed some rather remarkable facts. First, because a single frame was dropped, the sending TCP entity reacted very strongly by cutting back to sending one frame at a time until the problem was resolved by the retransmission. Keep in mind that if the segments each contained 1000 bytes, for example, this reaction may have been caused by a single bit error!

This clearly demonstrates that TCP does not behave well in error prone environments such as those created by wireless or poorly cabled networks. When comparing the first and second examples, it was clear that a single bit error could reduce the throughput by thousands of bytes in the time interval illustrated.

Our third example involves a clean network but a severely overwhelmed server. It's illustrated in Figure 13.

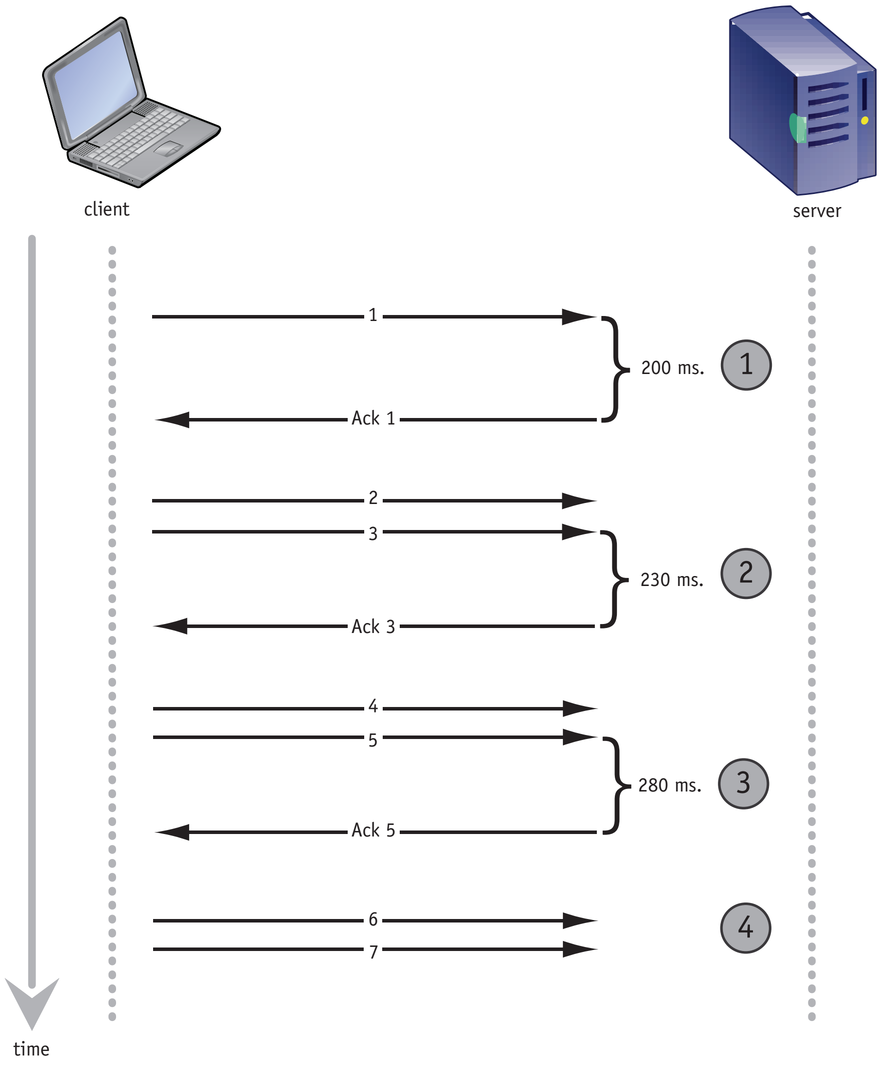


Figure 13 Overwhelmed Server

The client begins by sending a single segment in step 1. In order to comply with delayed acknowledgement, the server waits 200 ms. to see if more data will arrive. It does not. So, it sends Ack1. In step 2, the client increases the outgoing buffer to two segments and sends segments two and three. The server's reaction is important to understand. If the server were quickly processing the incoming data, it would respond on or before the 200 ms. wait time. Instead, the server uses 230 ms. to acknowledge that it received the two segments. Then it sends Ack3. In step three, the client's calculation of round trip time (RTT) tells TCP that this situation involves either a slow network or slow server. One of the two is not operating optimally. So, rather than increase its outgoing number of segments as it did in the other examples, the client sends two segments again, segments four and five. In step 3, the server is still operating sub optimally, so it takes it 280 ms. to determine that it received two more segments. Then the server sends Ack5. In the final step illustrated, step 4, the client still realizes the RTT is high, so it proceeds to send only two segments again. After these segments are delivered, if the server would happen to increase its performance (maybe it was busy with another large computational task), the RTT would be a lower value. The client's calculation would indicate that it can increase the number of simultaneous segments it is sending.

This example points out an additional important point. A subtle point is illustrated by the fact that the client never aggressively increased the number of segments being simultaneously sent. This indicated slowness somewhere. The turn-around time at the server was high and indicated inability to keep up with the rate at which it was receiving data. It was obvious that the server was the issue.

We have seen from these three examples, that TCP is a complicated process. However, it has certain core features that help us to isolate where problems are located. We should point out here, that in a real world analysis, the block numbers would be replaced by sequence and acknowledgement numbers that are often eight or more characters in length, rather than the single digit block numbers used in these examples. This means that the analysis of a capture file that is being analyzed may take considerable time. This could be the main reason that network engineers tend to avoid this deep inspection to search for problems. Yet, such inspection is analogous to the medical doctor who orders an MRI. It can be very revealing. With practice and good network troubleshooting tools, you can become very proficient in such analysis.

Throughout our discussion, we have illustrated one-way transfers of data. TCP is a full-duplex protocol. That means it is designed to allow for simultaneous flows in both directions. In practice, this isn't done very often. We think the half-duplex flows that have been illustrated not only simplify the illustrations but are more likely to be what you will encounter.

Closing the Connection

After the data relevant to the session has been exchanged, TCP indicates that it is finished by using a method that is often called the four-way handshake. Each end must close the connection independently. So, in our illustration, we'll suppose the client initiates the sequence because it is responsible for starting the session. The client will send a segment that usually doesn't contain data, but has a single bit in the TCP header set to one. That bit is called the FIN bit. The server will acknowledge that segment. Then the server will send a segment to the client with the FIN bit set to one. The client will acknowledge that segment. This is illustrated in Figure 14.

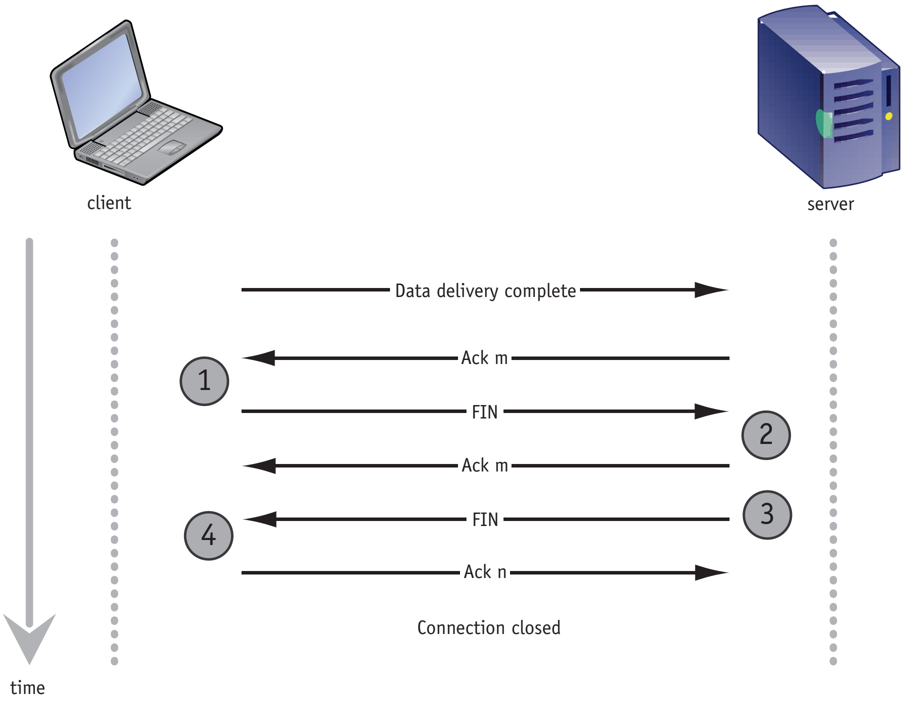


Figure 14 Closing the Connection

It is important for applications to properly close a connection. At both ends, this action tells the operating system that the resources that had been used such as input buffers, timers, and sequence values are no longer needed. The operating system is then free to use the resources for other activities.

In some of the early implementations of TCP applications, the connection was closed by issuing a segment with the RESET bit set to one. While the other TCP stack might see this as an indication the session was over, it might also correctly interpret it as an indication that the timers and sequence numbers were confused and needed to be corrected. Issuing resets occasionally causes servers to retain resources for the TCP session long after they are no longer being used by the applications. Of course, nearly the same thing might be expected if the FIN segment were to be lost. But in this case, the other end would not acknowledge receipt of the FIN segment and it would eventually be retransmitted. So, the four-way handshake is the proper way for a session to be closed.

Understanding How Applications Fail

Failures with DNS

In order to understand how DNS can fail we need to look more closely at how it works. When a client wants to make a request of a server, it usually has the name of the server such as www.psu.edu or www.flukenetworks.com. DNS is a distributed database that stores the names and the corresponding IP addresses. The database in DNS is a hierarchical tree which denotes two things. First, every node (DNS server) is above or below another DNS server. Second, there is a single path between any two servers. The nodes close to the root of the tree are called root servers. They keep track of who knows all the names in their own branch which is below them. This can be seen in Figure 15. For example, the .com server knows who is responsible for a name like flukenetworks.com because the suffix is .com. Likewise, the node flukenetworks.com knows who can resolve names containing support.flukenetworks.com.

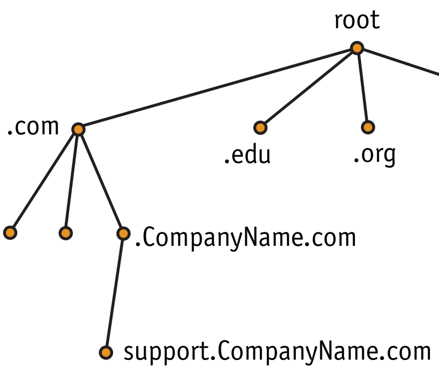


Figure 15 The DNS Structure

When a client application needs the address of a server, its DNS software called the *resolver* creates a query and forwards it to the local DNS server. The local DNS server is normally listed in its IP configuration. If the local DNS server knows the address corresponding to the name, the server returns it and we say the name has been *resolved*. If the local DNS server does not know the name, it has two choices: forward the query up the tree to the next DNS server or ask each server successively going up the tree to resolve the name on its behalf. Either way this takes time and slows the connection attempt. When the query reaches the root, the root will know how to resolve the name or will know which other root server can resolve the name. If the name can't be resolved, the response to the query will indicate the failure. At that point, the client's attempt to connect to the server will fail.

One common cause of failures in DNS is network configuration errors. Another common mistake that causes failures is mistyping the name. For example, typing www.flukenetwork.com will cause a DNS failure because the "s" is missing.

Failures with ARP

Remember that ARP is the protocol that is used by a station that needs to communicate with an IP address but doesn't know the local hardware address. Probably the two most common failures with ARP are duplicate IP addresses and incorrect configuration of the default gateway. If a user or other administrator of a computer assigns an address to their NIC that is already in use by someone else on the network, ARP tables in the network may have conflicting information in them. While each TCP/IP stack is permitted to handle ARP entry caching in their own manner, some use only entries they query for, while others use anything they hear from sending the query/response combination.

ARP can lead to problems caused by a device using the wrong subnet mask. Suppose a client and server are both on the same network 111.111.0.0 as in Figure16. If the mask is supposed to be 255.255.0.0 but the client is incorrectly configured to use 255.255.255.0, a problem is created. When the one phone wants to send a VoIP frames to the second, it will ARP for the router's address because it believes the second phone to be on a separate network. The router will respond with its own MAC address and the frames will be sent to the router. The router will relay the frame to the second phone. Computing cycles on the router are being used but the router should not have been involved in the exchange.

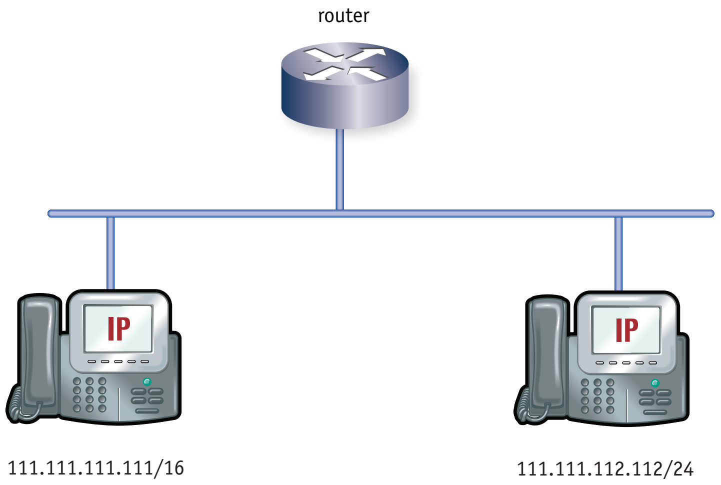


Figure 16 Unnecessary Routing

Proxy ARP is a protocol used to make devices that are on a separate LAN appear to be on the same LAN. It has many applications including configuring serial links, placing multiple IP addresses on a single interface of a server, and configuring firewalls. For example, a firewall could be configured to proxy ARP for two servers that were behind it. While the addresses of the servers would make it appear they were on the network, they are actually protected by the firewall. This is illustrated in Figure 17.

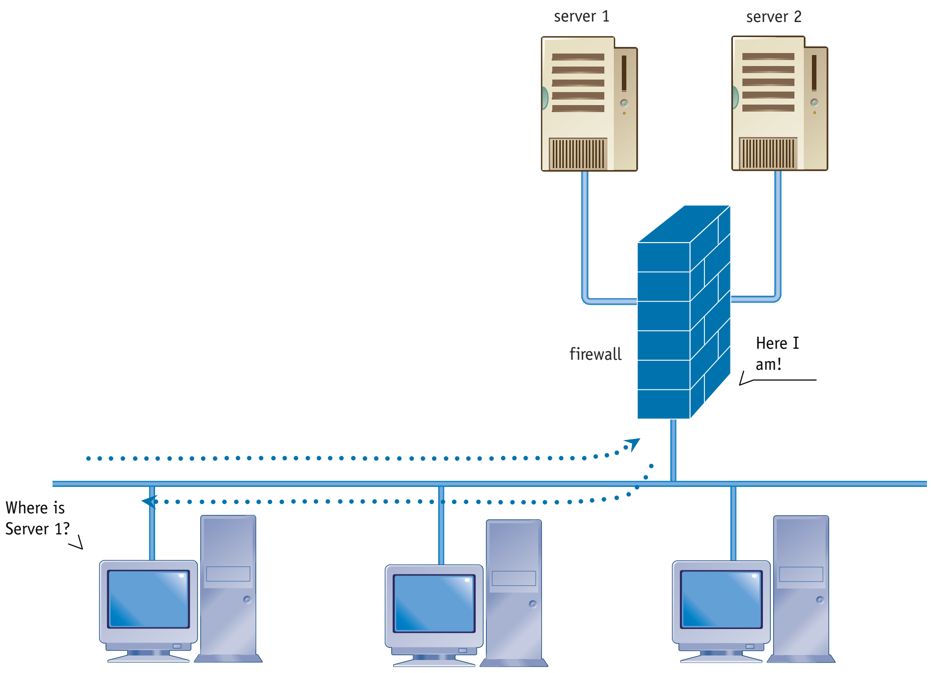


Figure 17 Proxy ARP

In this case when a station wants to communicate with Server 1, it will send an ARP broadcast. The firewall will respond with a unicast response. This creates the impression in the station that the firewall is, in fact, Server 1. Network administrators like this technique because it completely hides the fact that the two servers are on a separate, protected network.

But proxy ARP is prone to being misconfigured. Especially in the case of servers that need multiple IP addresses on a physical NIC and in cases where it is used in mobile IP applications.

On some occasions configuration mistakes can lead to incorrect ARP responses. Consider the example in Figure 18. Suppose the server has been accidentally configured with the incorrect address for the router as shown. If a packet arrives on the WAN interface of the router and is destined to the server, the router will send an ARP query for the server's address to get the correct response and the data frame will be sent correctly to the server. However, when the server decides to respond, it will decide the response needs to be routed and it will ARP for the router. But the ARP request will be for 192.168.0.2 and upon receipt of the response, the servers response will go to the wrong device. In all likelihood, the response will die at that point.

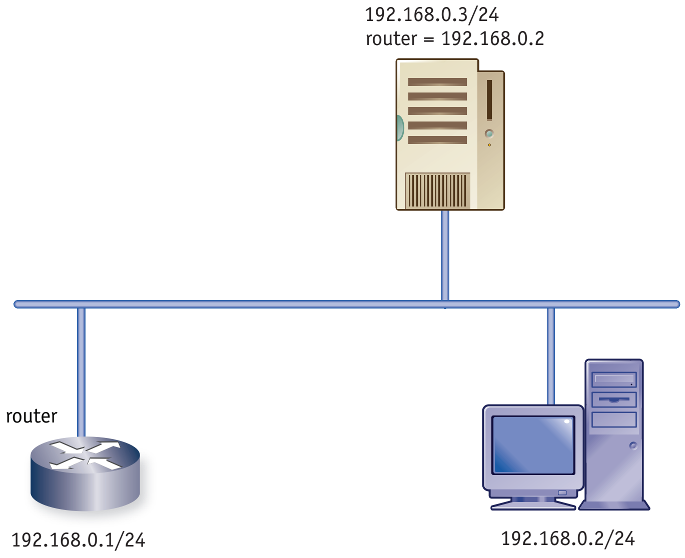


Figure 18 No Return Packet

Failures with Routing

Unfortunately, issues with poor or incorrect routes are common. Most often they are a result of administrative errors in configuring the network. Occasionally, the routing protocol being used leads to the problem. It may seem surprising that many low cost routers and most servers that act as routers use the RIP (routing information protocol). That protocol is over thirty years old. Routers and servers that use the protocol choose the destination path based on the number of hops (links between routers). However, no provision is made to assess the speed or throughput of a link. Consequently, links might be in a path that is very slow when an alternative path is not being used. The best way to spot a slow link is probably to use a tool like trace route. In Figure 8 (page 25) you can see that link 8 had a higher than normal RTT with 88 ms. While that may be an aberration, repeated use of trace route might confirm that it is always the slow link.

One factor that can be important is the network router discarding packets. Among routers, there is no provision for retransmission. So, the client or the server must determine that the routers have discarded packets. When they do, they will generally retransmit the packets. But you will recall that we pointed out that this retransmission can happen significantly later and cause TCP to slow down its rate of delivery by a large factor. So, why would routers discard packets? There are two main reasons. First, a packet arrives at a router and it is corrupted. That is, it has been damaged and the router detects that some information it contains is incorrect. It won't try to determine the correct information. That would use valuable processor cycles. It discards the packet. In certain instances, it will send a packet called an ICMP (Internet Control Message Protocol) packet back to the source indicating the discard. But the packet is lost. The second thing that can happen is that the router simply becomes too congested (that is, busy). Nearly all routers will reach a threshold where they begin to randomly discard traffic in order to do as much correct routing as they can. Again, the impact on TCP applications in the client or the server can be devastating.

Printing problems are often associated with routing errors. For example, if a client is logged into a server that is on the other side of a WAN link, but chooses to print to a local printer, the print file may go across the WAN link to the print spooler (queue). When the printer is available, the file will return across the WAN link to reach the printer. This is a remarkably common occurrence and usually creates slow printing because WAN links tend to have limited bandwidth.

Another common problem results when devices on a network with a multi-homed router are misconfigured. Look at Figure 19. The network administrator originally had the phones with addresses ending .1 through .126 on one network. When the administrator ran out of addresses he decided to add the range from .130 through .190 by giving the router a second address 192.168.0.129 with a 26 bit mask. Apparently the users at .5 and at .83 found out that the user at .133 had a 26 bit mask. They proceeded to dig out the manual from a pile of papers on their desk and look up how to change the phone configuration. They changed their configuration to be consistent with the phone at .133. This will introduce a series of possible problems. If the phone is smart enough to discover the misconfiguration, it may do what a Windows® client would do. In the case .83, the phone would indicate that the configured default router (.1) is not on the local network.

However, devices such as phones have a very limited operating systems. It is much more likely that the phone will allow the configuration to be activated. In this case, traffic between .5 and .83 will be routed. If the router is not overworked, this misconfiguration could be hidden for years. However, when the router becomes overly congested with bursty traffic, it could begin to drop packets in calls between these two stations.

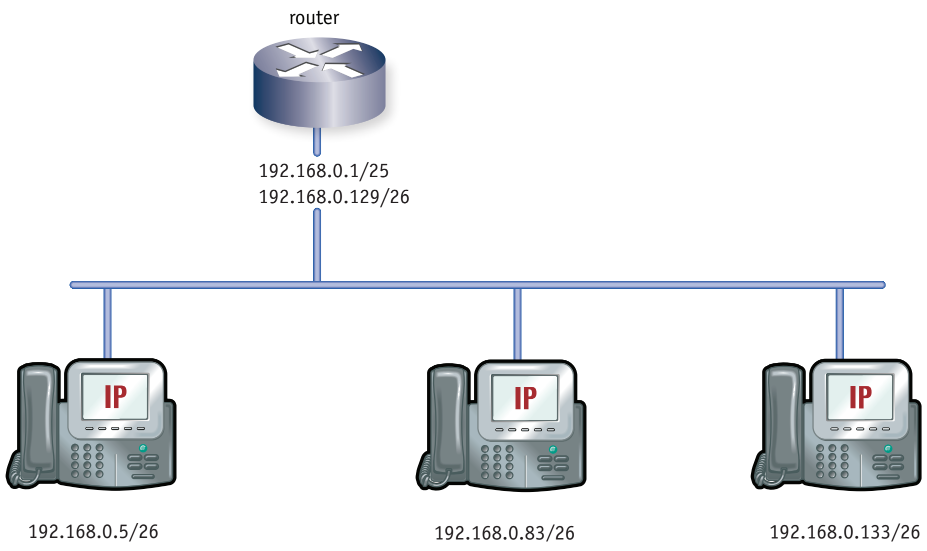


Figure 19 Incorrect Configuration

Packet loss along the route can be hidden. For example, suppose a pair of layer two switches are connected directly by a fiber or twisted pair connection. Because no other devices except the switches are normally monitoring such a link, bit errors can occur that cause the link to corrupt data frames. This problem will be dealt with by the end stations. If the applications that are losing the frames are TCP based, the effect will be significant. If the administrator of the network is using the link for VoIP transport, minor frame loss will not likely be evident in the quality of the calls. This can cause the administrator to search elsewhere for the cause and miss the actual source of the degradation in performance. Other causes of packet loss can be an over subscribed quality of service technique, bad NIC cards in routers, or intermediate wireless links (such as wireless bridges).

Problems in Establishing a Connection

Applications communicate between ports. In fact, in TCP and UDP discussions, we say the TCP *session* is defined or uniquely characterized by these four parameters: the sending IP address, the sending TCP port number, the receiving IP address, and the receiving TCP port. Programmers refer to the set of four parameters along with the protocol as a *socket*. These four values

(IP address A, port x) <--> (IP address B, port y)

uniquely identify the logical communications channel between the client and the server.

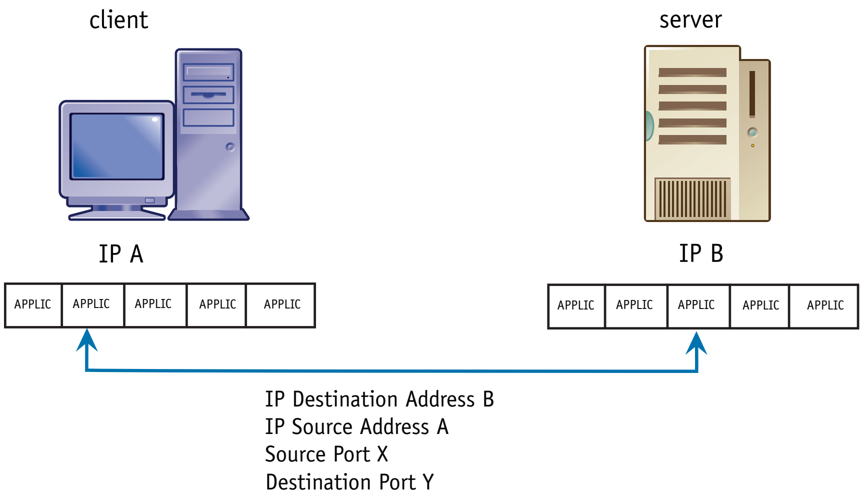


Figure 20 Defining a Session

Application Troubleshooting Resource Center:
www.flukenetworks.com/appts

For example, in the three-way handshake, the client sends the SYN packet from IP address A listing port x as the source port. In the packet the client also lists the destination as IP address B (server) and port y (the application). As we described previously, the operating system in each device needs to allocate memory for the operation of the session being established. These four values are the tags used to identify the exchange. Note that a single server, such as an email server, can use port 25 as part of a set of thousands of different sessions, so long as the client port or client address changes in each instance.

Occasionally, a client application will attempt to connect to a port that is not available on a server. We say the *port is not open*. If this happens, most TCP stacks will tell the source of this occurrence by sending an ICMP packet indicating this. However, most client devices will ignore the report and the user will not make the connection. If someone or some device is attempting to discover which ports are active on the server by *scanning* ports, you may see a sudden increase in the amount of ICMP traffic on your network. Scanning is a process in which a SYN packet is sent to each possible port number in succession. For example, a scanning utility might send successive SYN packets to 191.168.0.5: 1025, 191.168.0.5: 1026, 191.168.0.5: 1027, and so forth. Here we have used the common notation to show the port number behind the IP address separated by a colon.

There are generally two sources for scanning activities: good guys and bad guys. Among the good guys are network troubleshooting tools with functions built in that will scan the ports of a host in order to see if the host has the correct ports open.

Most servers have several open ports which represent services it will provide. For example, a server with ports 25 and 80 open would be available to provide email and web page serving. If the server actively announces these services, we say it advertises the services. So, often a network support tool will scan the network to see which devices have open ports and then send a SYN to verify the status of the port. Among the bad guys are hackers that are attempting to do reconnaissance on your network. They also want to know which devices are servers and which ports are open. But it isn't likely that they have honorable purposes.

Slow Responses from Servers

In the client-server relationship, when a client sends a request to a server, it is usually represented by a packet that has a unique format which is determined by the application program interface. For example, with HTTP, when a client requests a home page or part of a page from a web server, an HTTP GET packet is sent to the server. It contains a significant number of parameters such as the version of HTTP being used, the date and time, the file requested and other values. The GET is sent only after the client and session server has been established by the three-way handshake.

How quickly the service request is fulfilled may depend on several factors. The server may be slow to respond because it needs to compute something or search for the file. In the case of HTTP, it may even send a response that indicates acknowledgement of the request without supplying the file to fulfill the request. It's like saying, "I'm working on it." The request may get lost or be discarded by the network. The same thing can happen to the response. Finding the cause of server slowness is beyond the scope of this document. However, later we will consider how to spot signs that a server is over worked.

Closing the Connection

As we pointed out before, when we considered the operation of TCP, the session is ended with the four-way handshake. The client sends a FIN and the server acknowledges it. Then the server sends a FIN and the client acknowledges that FIN. Occasionally, one of the four frames of data is lost or discarded by the network. If the FIN is lost, the corresponding acknowledgement isn't received and eventually the station will send it again. If the acknowledgement is lost, the sender likewise eventually decides it must be sent again. It might seem that this would not affect the application performance since it appears to happen after the data has been transferred. But that is incorrect. As an example, consider when a web page is requested. It will typically contain eight to twenty components such as a logo, a banner, a copyright notice, a list of options to click on and other information. HTTP generally transfers the page in individual TCP sessions. So, if the session to retrieve the banner doesn't close properly, it may prevent the retrieval of the list of items that the user can click on. It will appear as if the application is responding slowly.

Finally, as we mentioned in our previous discussion of closing the connection, some poorly constructed applications and some older versions of protocols, use the RESET packet to close a session.

This may force the operating system to keep memory resources and times active when they are no longer needed. Keeping the resources viable will take additional computational cycles from the server and decrease its overall performance.

Troubleshooting Applications

Often troubleshooting is challenging and a “best” method isn’t obvious. Some technicians use the few tools they are familiar with to try to solve every problem. The adage, “When the only tool you have is a hammer, everything begins to look like a nail, “ comes to mind. In network troubleshooting, technicians who know the physical level sometimes attempt to solve a problem with cable testers and meters. Technicians who have an RF communications background gravitate towards spectrum analyzers.

On the other hand, team members working on a problem may each have a different view of the same problem. The systems analyst will think of CPU performance, insufficient memory, and fragmented disk space as the cause of the problem. The network architect might see it as routing issues. Or, the manager of network infrastructure might decide to test the WAN links for throughput or replace copper links with fiber connections.

Good troubleshooting requires having a broad base of experience, a solid knowledge of the technology involved, and the availability of the proper tools.

Show devices with IPv6 address prefix: all subnets

Name	IPv6 Address	Mfg-MAC Address
PC0017.testnetwork.com	2001:c001:c0de:248:21a:a0ff:fe92:f2a1	Dell-92f2a1
PC0019.testnetwork.com	2001:c001:c0de:248:214:22ff:fee0:c1a0	Dell-e0c1a0
PC0025.testnetwork.com	2001:c001:c0de:248:21a:a0ff:fe8d:292	Dell-8d0292
PC0029.testnetwork.com	2001:c001:c0de:248:21a:a0ff:fe92:f313	Dell-92f313
PC0037.testnetwork.com	2001:c001:c0de:248:21a:a0ff:fe92:edf8	Dell-92edf8
PC0054.testnetwork.com	fe80::71f6:a451:46f1:b08c	Compal-225f46
PC0076.testnetwork.com	2001:c001:c0de:248:208:74ff:fed4:d213	Dell-d4d213
PC0082.testnetwork.com	2001:c001:c0de:248:20b:dbff:fe95:4878	Dell-954878
PC0092.testnetwork.com	2001:c001:c0de:248:21a:a0ff:fe9f:8512	Dell-9f8512
PC0100.testnetwork.com	2001:c001:c0de:248:206:5bff:fe3c:30bb	Dell-3c30bb
PC0162.testnetwork.com	fe80::b871:c2fb:326c:4f79	Radsys-3cbf6c
PC0232.testnetwork.com	fe80::1eb:3b1e:5a0e:4f93	Netgea-6ebaaa
PC0247.testnetwork.com	2001:c001:c0de:248:211:43ff:feb9:bde8	Dell-b9bde8
PC0254.testnetwork.com	2001:c001:c0de:248:21c:23ff:fe17:cc36	Dell-17cc36
PC0281.testnetwork.com	2001:c001:c0de:248:2e0:4dff:fe02:b12b	Intlni-02b12b
PC0291.testnetwork.com	2001:c001:c0de:248:212:3fff:fe5d:cc33	Dell-5dcc33
PC0296.testnetwork.com	2001:c001:c0de:248:212:3fff:fe5d:c790	Dell-5dc790

OptiView® Analyzers: Portable and Rack Models.

Fluke Networks' OptiView Series III Network Analyzers are available in two form factors to give you a clear view of your entire enterprise – providing visibility into every piece of hardware, every application, and every connection on your network.

Choose the Integrated Network Analyzer for portable, all-in-one analysis or the Workgroup Analyzer for permanent or semi-permanent deployment that acts as a "virtual network engineer 24/7" in the core or at remote sites. Or, use them together to create a powerful combination – a troubleshooting tool for the access layer and an analyzer watching the core, remote site or critical network point. Network professionals can conduct all the necessary tests at the remote site without ever leaving the headquarters site.

Baselining

One of the techniques that is used the least is baselining the network. Yet, it is one of the most useful techniques. Baselining is recording data based on the "normal" performance of the network. By knowing how the network performs when users are satisfied, you have a point of comparison when problems are reported. Have the level of broadcasts gone up? Are there new protocols on the network? Is utilization higher? Is the ERP application response time slower? Questions like these are impossible to answer without the corresponding data gathered under normal operating circumstances. This information is rarely provided by network equipment manufacturers. So, network managers need to find time in their schedules to record it when they aren't under pressure.

Methodology

Selecting the right tools can be more challenging than it appears. The more theoretically oriented engineer often wants a protocol analyzer as their only tool. While it allows them to do a deep analysis of the network in some circumstances, finding a failing fiber optic cable or a network interface card that is out of specification, may be nearly impossible. Yet, the engineer that wants to avoid theory altogether might simply swap out components until the problem disappears. This can take too much time and lead to many discarded, expensive but perfectly functional network components. Here are some points that may be helpful as you approach a troubleshooting problem:

- Analyze the network as a whole.
- Follow a logical sequence of steps.
- Zero-in on the root source of the problem and make one adjustment or change one part to eliminate the problem.
- Don't focus on completely understanding the problem until after the network is functioning properly.
- Provide feedback and training to the user. It's good diplomacy and may eliminate user generated problems in the future.

Five Key Steps to Successful Application Troubleshooting

We will discuss each of these steps in succession:

1) Determine the domain of the problem and exonerate the network.

2) Conduct an Application flow analysis.

3) Fix the problem.

4) Validate the problem.

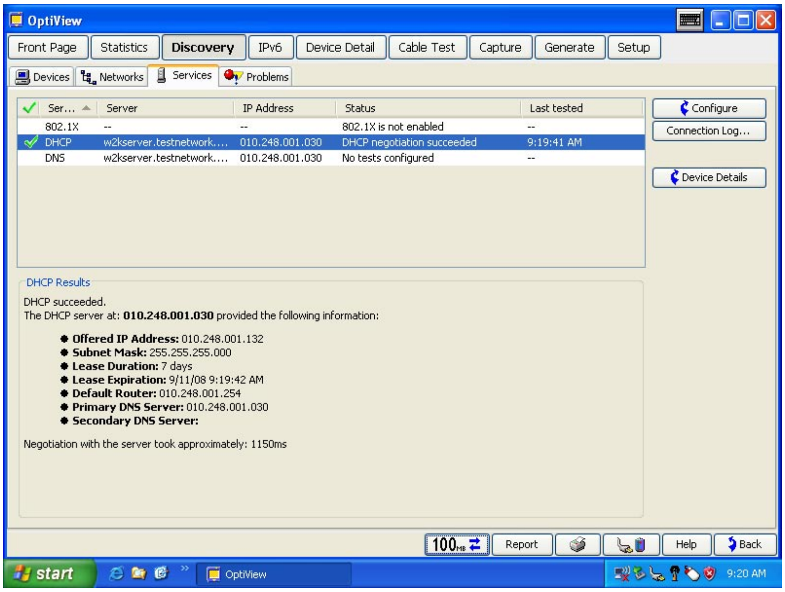
5) Document the problem.

Determine the domain of the problem and exonerate the network.

This process involves three steps: (1) Validating network services; (2) Validating connectivity to the server; and (3) Determining the network path.

Validating Network Services. To validate network service, we need to do two things. First, we must be sure that clients have or are getting an IP address, subnet mask, default router address, and DNS server address. On a Windows® client, you can use the command line tool *ipconfig* to do this. If there isn't an IP address assigned to the client, it could be because there is a failure in the connection to the DHCP server. From the command line, you can issue the command i*pconfig/release* followed by *ipconfig/renew*. this will cause the client to initiate the four step process to obtain the configuration parameters from the DHCP server.

Using Fluke Networks OptiView, it's much easier to test DHCP and you get significantly more information. By placing the OptiView at the point of connection used by the client and starting it, the OptiView automatically does a Discovery process to find devices on the network. That process includes obtaining an address for the analyzer from DHCP as shown in Figure 21.



Ser...	Server	IP Address	Status	Last tested
802.1X	--	--	802.1X is not enabled	--
✓ DHCP	w2kserver.testnetwork....	010.248.001.030	DHCP negotiation succeeded	9:19:41 AM
DNS	w2kserver.testnetwork....	010.248.001.030	No tests configured	--

DHCP Results

DHCP succeeded.
The DHCP server at: **010.248.001.030** provided the following information:

- **Offered IP Address:** 010.248.001.132
- **Subnet Mask:** 255.255.255.000
- **Lease Duration:** 7 days
- **Lease Expiration:** 9/11/08 9:19:42 AM
- **Default Router:** 010.248.001.254
- **Primary DNS Server:** 010.248.001.030
- **Secondary DNS Server:**

Negotiation with the server took approximately: 1150ms

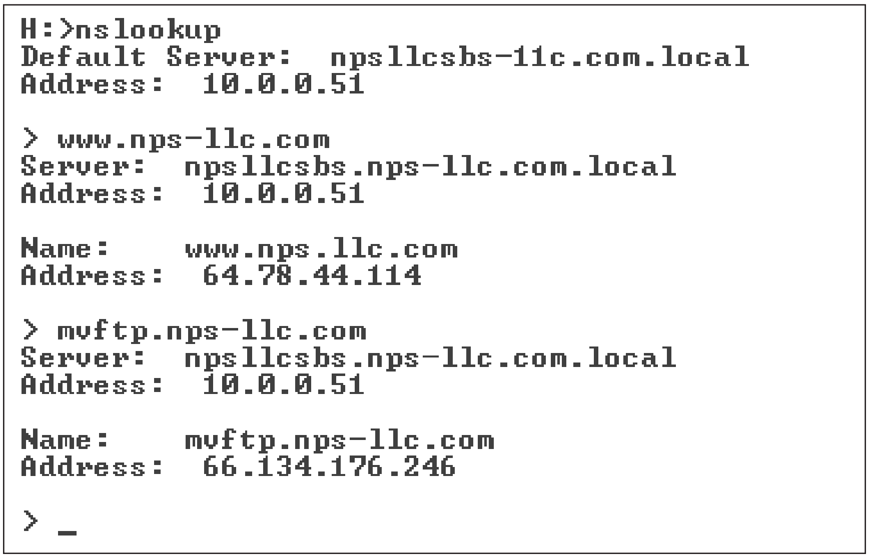
Figure 21 DHCP Response using OptiView

You can see that the test was successful and that the OptiView received an IP address, mask, default router, DNS server, and a seven day lease. You can also see that the process took a little over one second, a fact that isn't available from the command line tools.

The second thing we must test is DNS. To see whether a client can get names resolved we can again use two approaches. From the command line we can use *nslookup*.

The nslookup is a tool that is sometimes used to troubleshoot domain name servers. Its use is somewhat controversial because it is often used by hackers. It may not provide consistent results because name servers are often restricted from responding to nslookup queries. Yet, in competent hands that have good intentions, nslookup is helpful.

It can be used to do a general query in a domain and it can be used to specify a particular type of query. For example, if you type nslookup and hit enter you may receive a result such as is shown in Figure 22.

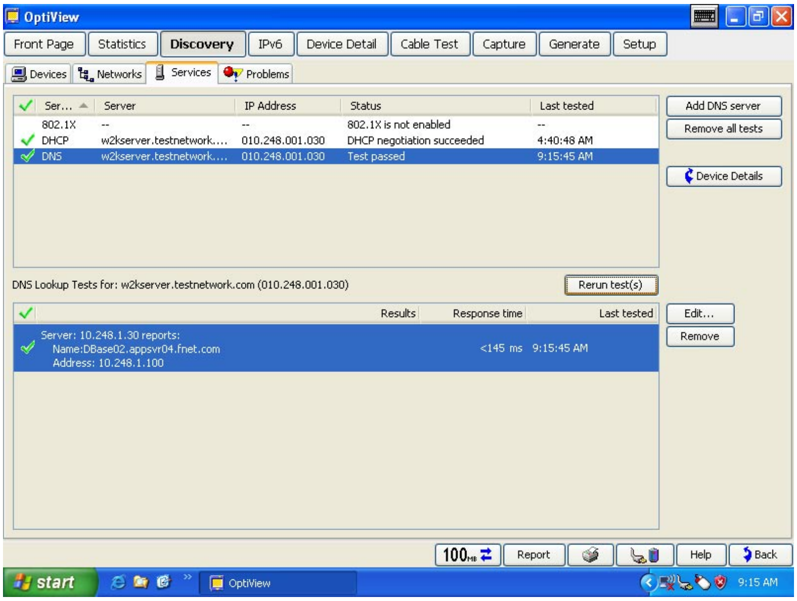


```
H:>nslookup
Default Server:  npsllcsbs-llc.com.local
Address:  10.0.0.51

> www.nps-llc.com
Server:  npsllcsbs.nps-llc.com.local
Address:  10.0.0.51

Name:    www.nps.llc.com
Address:  64.78.44.114

> mvftp.nps-llc.com
Server:  npsllcsbs.nps-llc.com.local
Address:  10.0.0.51

Name:    mvftp.nps-llc.com
Address:  66.134.176.246

> _
```

Figure 22 The nslookup Tool

The nslookup is a tool that tells us about servers, domains, and the addresses of the servers. Often, there are subcommands that allow querying about mail servers or exchange servers. Searching the Internet for descriptions will provide a great deal of information.

An easier approach is to use the same screen that we used to test DHCP. In figure 23 we can see that the OptiView has been configured to resolve the name DBase02.appsvr04.fnet.com. The test was successful and the IP address was returned in 140 ms. By clicking on the *edit* button on the right any specific name can be queried. By clicking on the *Add DNS Server* button, any specific server can be tested.



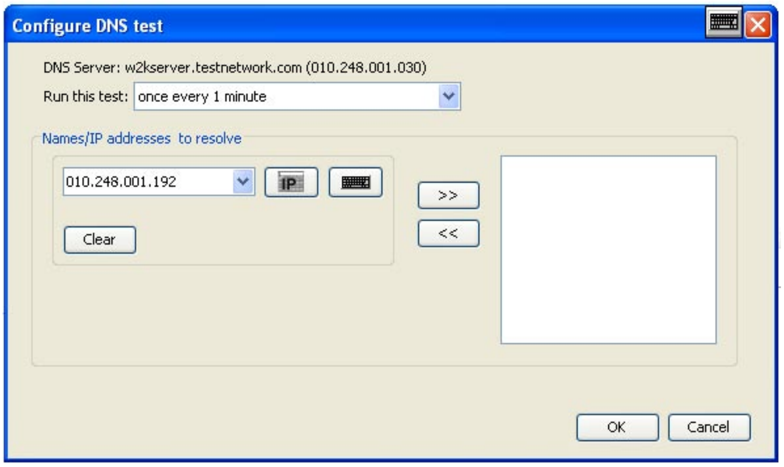
✓	Ser...	Server	IP Address	Status	Last tested
	802.1X	--	--	802.1X is not enabled	--
✓	DHCP	w2kserver.testnetwork....	010.248.001.030	DHCP negotiation succeeded	4:40:48 AM
✓	DNS	w2kserver.testnetwork....	010.248.001.030	Test passed	9:15:45 AM

DNS Lookup Tests for: w2kserver.testnetwork.com (010.248.001.030)

✓	Results	Response time	Last tested
✓	Server: 10.248.1.30 reports: Name:DBase02.appsvr04.fnet.com Address: 10.248.1.100	<145 ms	9:15:45 AM

Figure 23 DNS Test using OptiView

A test which we use less frequently is the *reverse lookup*. By submitting an IP address, the DNS server should return the corresponding server that uses that name. While the command line tool nslookup can be used, you can add a reverse name lookup to the OptiView by just entering the IP address in the screen that pops up when you select a DNS Server and click *Add*. This is shown in Figure 24.



DNS Server: w2kserver.testnetwork.com (010.248.001.030)

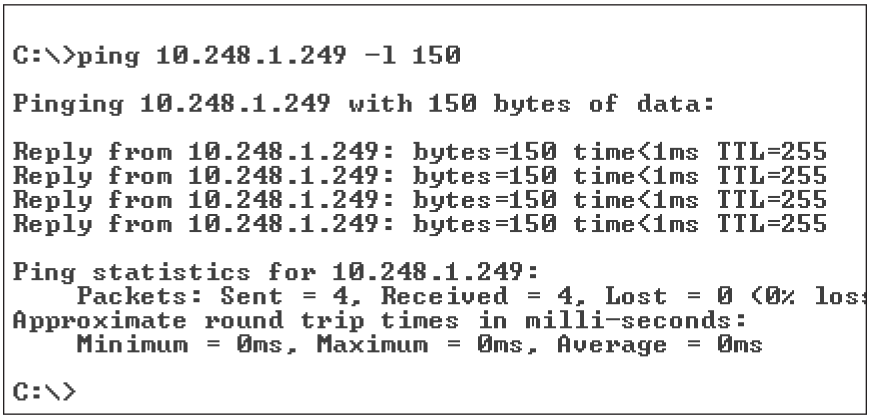
Run this test: once every 1 minute

010.248.001.192

Figure 24 Adding a Reverse using OptiView

Device Connectivity. The *ping* command is so widely used, that it has become part of our language. In networking, it has a very specific meaning. It is a small application that sends a query packet to a target device to see if the device will respond. For security reasons, the device is sometimes not allowed to respond. However, it remains the most widely used method of testing connectivity.

From a command prompt, you can execute the command ping x.x.x.x. This will cause your protocol stack to send a packet called an *ICMP echo request* to the address you listed as x.x.x.x. Some devices do not respond to pings and some firewalls block echo requests. But if your echo request survives the journey and reaches the target, the target device is likely to respond with another ICMP packet called an *ICMP echo response*. Network administrators use ping packets for many purposes. You do this by adding a *switch* which is similar to a subcommand. Among other things you can: see if a packet containing M bytes will get to the target (ping with – l M switch), ping continuously until interrupted (-t switch) and ping allowing up to N hops (ping – i N). You can see the first of these illustrated in Figure 25. The continuous ping is usually stopped with CNTL-C. While there are slight variations of the ping command across different operating systems, most support the usage described here.

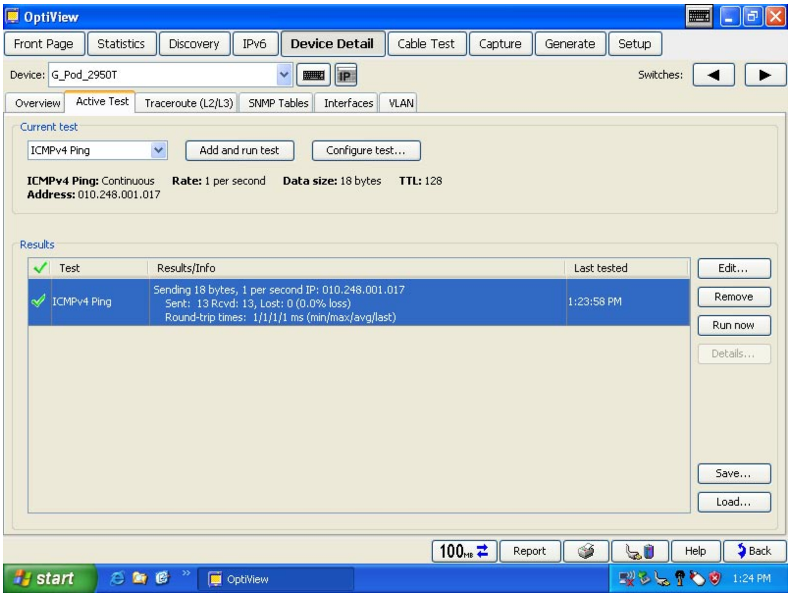


```
C:\>ping 10.248.1.249 -l 150

Pinging 10.248.1.249 with 150 bytes of data:

Reply from 10.248.1.249: bytes=150 time<1ms TTL=255
Reply from 10.248.1.249: bytes=150 time<1ms TTL=255
Reply from 10.248.1.249: bytes=150 time<1ms TTL=255
Reply from 10.248.1.249: bytes=150 time<1ms TTL=255

Ping statistics for 10.248.1.249:
    Packets: Sent = 4, Received = 4, Lost = 0 (0% los
Approximate round trip times in milli-seconds:
    Minimum = 0ms, Maximum = 0ms, Average = 0ms

C:\>
```

Figure 25 Ping with Designated Payload

So, suppose we believe that we have a route that is dropping packets that are over 1000 bytes because the MTU isn't the standard value 1460. We can send a ping that specifies a payload of 900 bytes followed by a ping that specifies a payload of 1100 bytes. This will give us an indication. Keep in mind that this depends on the fact that nothing in the path is blocking ICMP packets for security reasons.

Ping connectivity tests are easy to configure and run in the OptiView. In Figure 26 you can see the screen that appears when you select a device on the Discovery screen and click on *Device Detail*. From here you can run ping tests and other tests that we will discuss later.



Device: G_Pod_2950T

Current test: ICMPv4 Ping

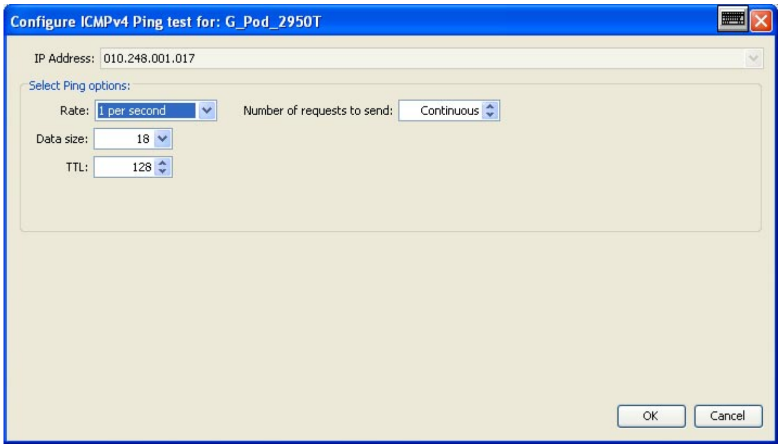
ICMPv4 Ping: Continuous **Rate:** 1 per second **Data size:** 18 bytes **TTL:** 128
Address: 010.248.001.017

Results

✓	Test	Results/Info	Last tested
✓	ICMPv4 Ping	Sending 18 bytes, 1 per second IP: 010.248.001.017 Sent: 13 Rcvd: 13, Lost: 0 (0.0% loss) Round-trip times: 1/1/1/1 ms (min/max/avg/last)	1:23:58 PM

Figure 26 Using OptiView

If you want to change the payload, run the test more or less frequently, or limit the hop count, you can click on *configure* test.



Configure ICMPv4 Ping test for: G_Pod_2950T

IP Address: 010.248.001.017

Select Ping options:

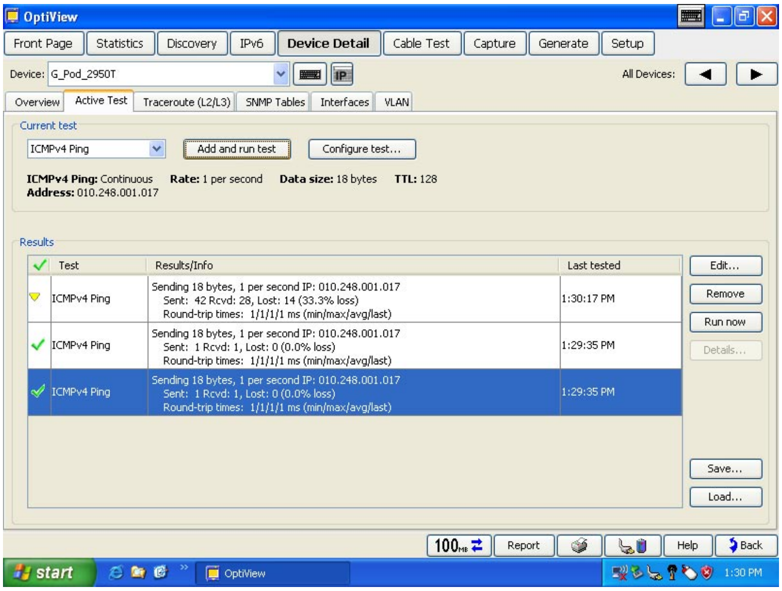
Rate: 1 per second Number of requests to send: Continuous

Data size: 18

TTL: 128

Figure 27 Ping options

The result will be a report like the one in Figure 28 that tells us to how fast the response came back and whether any of the packets or responses were lost.



Device: G_Pod_2950T

ICMPv4 Ping

ICMPv4 Ping: Continuous **Rate:** 1 per second **Data size:** 18 bytes **TTL:** 128
Address: 010.248.001.017

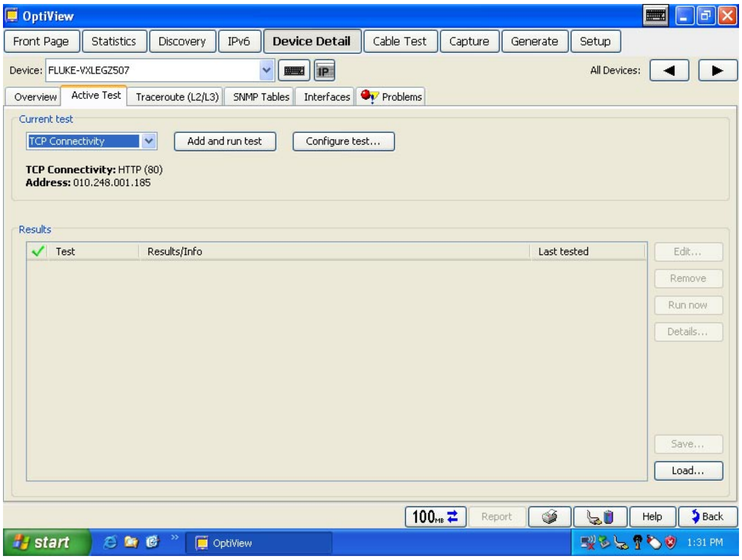
	Test	Results/Info	Last tested
▽	ICMPv4 Ping	Sending 18 bytes, 1 per second IP: 010.248.001.017 Sent: 42 Rcvd: 28, Lost: 14 (33.3% loss) Round-trip times: 1/1/1/1 ms (min/max/avg/last)	1:30:17 PM
✓	ICMPv4 Ping	Sending 18 bytes, 1 per second IP: 010.248.001.017 Sent: 1 Rcvd: 1, Lost: 0 (0.0% loss) Round-trip times: 1/1/1/1 ms (min/max/avg/last)	1:29:35 PM
✓	ICMPv4 Ping	Sending 18 bytes, 1 per second IP: 010.248.001.017 Sent: 1 Rcvd: 1, Lost: 0 (0.0% loss) Round-trip times: 1/1/1/1 ms (min/max/avg/last)	1:29:35 PM

Figure 28 The Ping Report

Validating Connectivity to the Application Server

There are three tasks to be accomplished here. First, we should see if we can connect to the application of interest. Second, we should measure the response time of the application. Third, we should check the vital statistics of the server.

Application Connectivity. As we discussed earlier, applications communicate through ports. In particular, server applications use well-known ports. Once the server has been selected, we can use the same screen that we used to generate a ping test. The drop down menu allows us to choose the TCP connectivity test. This is shown in Figure 29.



Device: FLUKE-VXLEGZ507

Current test: TCP Connectivity

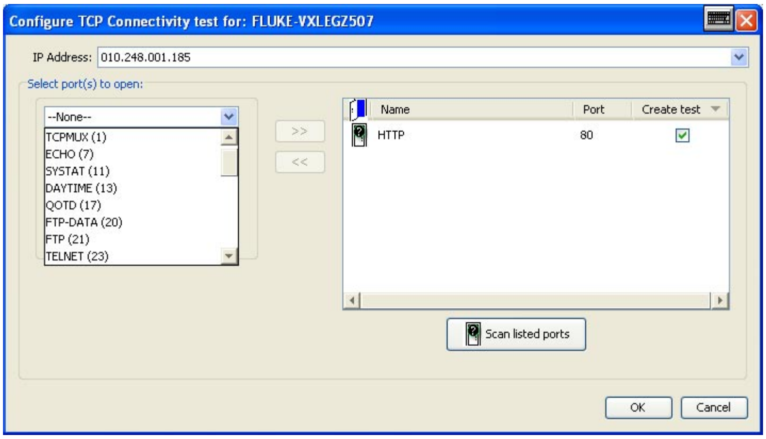
TCP Connectivity: HTTP (80)
Address: 010.248.001.185

Results

Test	Results/Info	Last tested

Figure 29 The TCP Connectivity Test

By clicking on the button labeled *Configure Test*, you can easily select nearly every well known application process. This is shown in Figure 30.



IP Address: 010.248.001.185

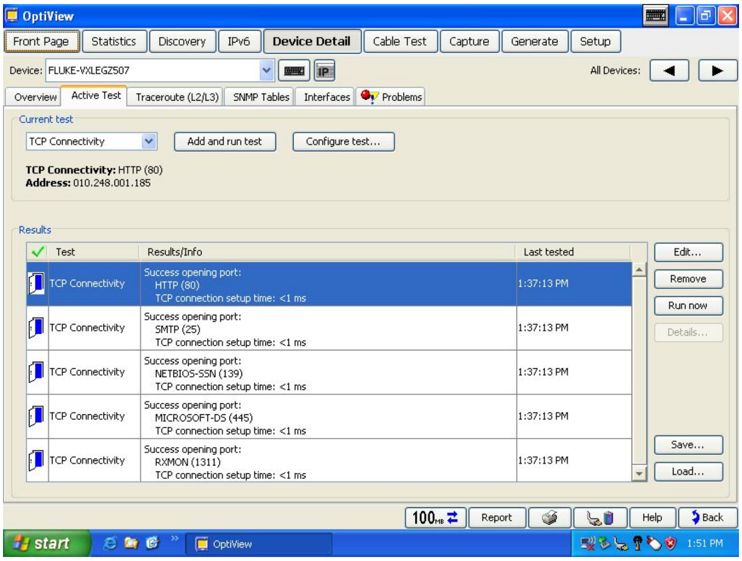
Select port(s) to open:

--None--
TCPMUX (1)
ECHO (7)
SYSTAT (11)
DAYTIME (13)
QOTD (17)
FTP-DATA (20)
FTP (21)
TELNET (23)

Name	Port	Create test
HTTP	80	✓

Figure 30 TCP Port Selection

We can attempt to open individual or multiple well known or registered ports, or any user defined ports which may be necessary for custom home-grown applications. The results of one such connectivity test is shown in Figure 31.



TCP Connectivity: HTTP (80)
Address: 010.248.001.185

Test	Results/Info	Last tested
TCP Connectivity	Success opening port: HTTP (80) TCP connection setup time: <1 ms	1:37:13 PM
TCP Connectivity	Success opening port: SMTP (25) TCP connection setup time: <1 ms	1:37:13 PM
TCP Connectivity	Success opening port: NETBIOS-SSN (139) TCP connection setup time: <1 ms	1:37:13 PM
TCP Connectivity	Success opening port: MICROSOFT-DS (445) TCP connection setup time: <1 ms	1:37:13 PM
TCP Connectivity	Success opening port: RXMON (1311) TCP connection setup time: <1 ms	1:37:13 PM

Figure 31 Application Connectivity Test Result

Application Test. In order to see if the application is responding adequately, we can use an Application Flow Analysis. After capturing data during server transactions, the protocol decode overview screen shown in Figure 32 appears. It shows that HTTP protocol was used to conduct transactions between the client and several servers. It also shows how many transactions (TCP sessions) were opened, used and then closed. If we click on the server that used HTTP, we get a list of servers that were involved in the transactions using HTTP.

Each time we click on a server, we get a table of statistics about the interaction with that server. This is shown in Figure 33. Finally, if we click on the hyperlink that says how many transactions there were, we get a list of the transactions, a Bounce Chart and a table of Transport Statistics about those transactions. From the transport statistics we can see what the transaction was about and how quickly the server responded. This is shown in Figure 34.

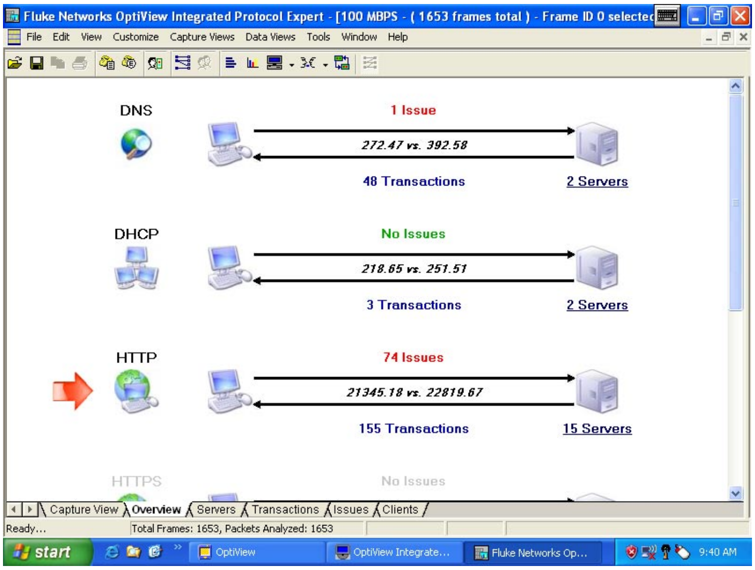
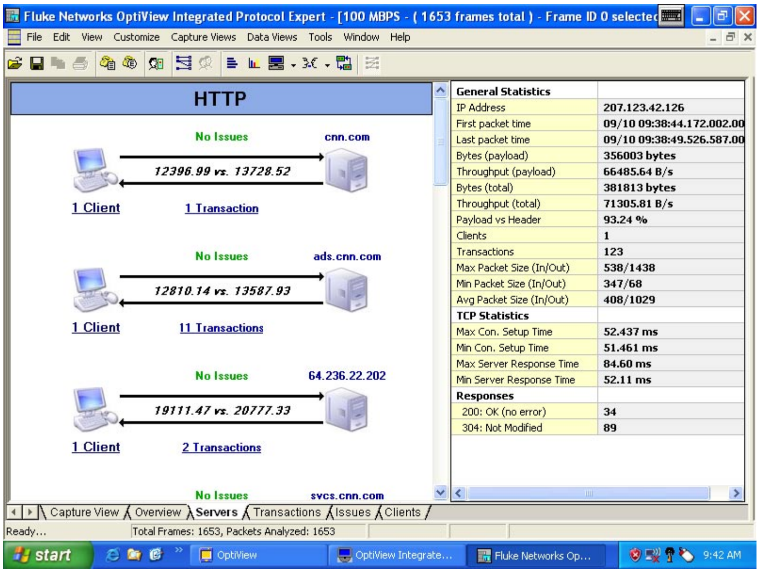
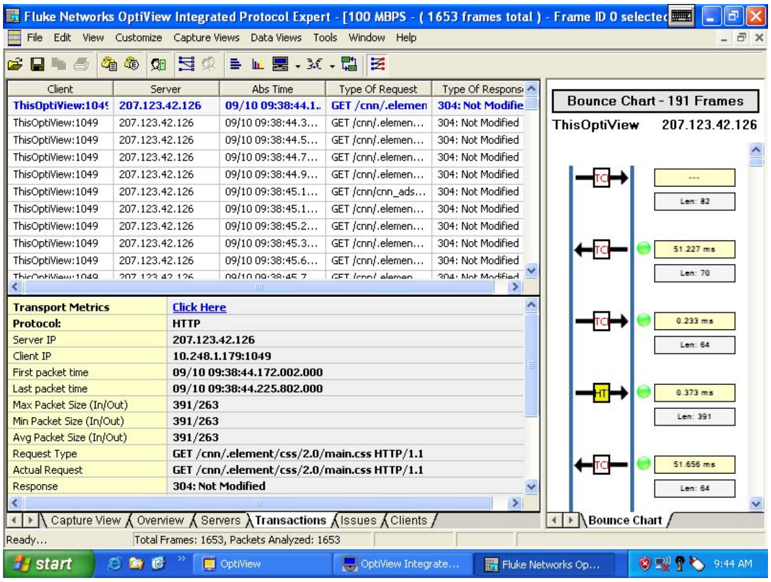

Fluke Networks OptiView Integrated Protocol Expert - [100 MBPS - (1653 frames total) - Frame ID 0 selected
File Edit View Customize Capture Views Data Views Tools Window Help
DNS
1 Issue
272.47 vs. 392.58
48 Transactions
2 Servers
DHCP
No Issues
218.65 vs. 251.51
3 Transactions
2 Servers
HTTP
74 Issues
21345.18 vs. 22819.67
155 Transactions
15 Servers
HTTPS
No Issues
Capture View Overview Servers Transactions Issues Clients
Ready...
Total Frames: 1653, Packets Analyzed: 1653
start
OptiView
OptiView Integrate...
Fluke Networks Op...
9:40 AM

Figure 32 Server Overview Screen



General Statistics	
IP Address	207.123.42.126
First packet time	09/10 09:38:44.172.002.00
Last packet time	09/10 09:38:49.526.587.00
Bytes (payload)	356003 bytes
Throughput (payload)	66485.64 B/s
Bytes (total)	381813 bytes
Throughput (total)	71305.81 B/s
Payload vs Header	93.24 %
Clients	1
Transactions	123
Max Packet Size (In/Out)	538/1438
Min Packet Size (In/Out)	347/68
Avg Packet Size (In/Out)	408/1029
TCP Statistics	
Max Con. Setup Time	52.437 ms
Min Con. Setup Time	51.461 ms
Max Server Response Time	84.60 ms
Min Server Response Time	52.11 ms
Responses	
200: OK (no error)	34
304: Not Modified	89

Figure 33 HTTP Servers


Fluke Networks OptiView Integrated Protocol Expert - [100 MBPS - (1653 frames total) - Frame ID 0 selected

Client	Server	Abs Time	Type Of Request	Type Of Response
ThisOptiView:1049	**207.123.42.126**	**09/10 09:38:44.1..**	**GET /cnn/.elemen**	**304: Not Modified**
ThisOptiView:1049	207.123.42.126	09/10 09:38:44.3...	GET /cnn/.elemen...	304: Not Modified
ThisOptiView:1049	207.123.42.126	09/10 09:38:44.5...	GET /cnn/.elemen...	304: Not Modified
ThisOptiView:1049	207.123.42.126	09/10 09:38:44.7...	GET /cnn/.elemen...	304: Not Modified
ThisOptiView:1049	207.123.42.126	09/10 09:38:44.9...	GET /cnn/.elemen...	304: Not Modified
ThisOptiView:1049	207.123.42.126	09/10 09:38:45.1...	GET /cnn/cnn_ads...	304: Not Modified
ThisOptiView:1049	207.123.42.126	09/10 09:38:45.1...	GET /cnn/.elemen...	304: Not Modified
ThisOptiView:1049	207.123.42.126	09/10 09:38:45.2...	GET /cnn/.elemen...	304: Not Modified
ThisOptiView:1049	207.123.42.126	09/10 09:38:45.3...	GET /cnn/.elemen...	304: Not Modified
ThisOptiView:1049	207.123.42.126	09/10 09:38:45.6...	GET /cnn/.elemen...	304: Not Modified
ThisOptiView:1049	207.123.42.126	09/10 09:38:45.7	GET /cnn/.elemen	304: Not Modified

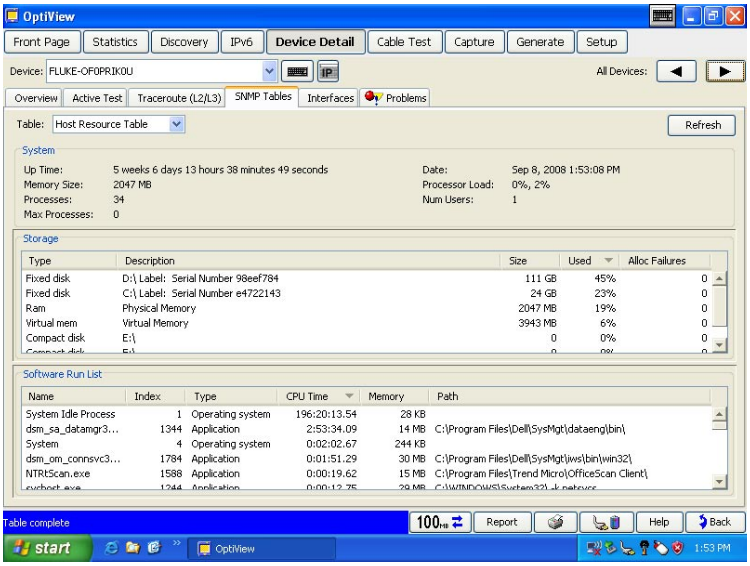
Transport Metrics	Click Here
Protocol:	**HTTP**
Server IP	**207.123.42.126**
Client IP	**10.248.1.179:1049**
First packet time	**09/10 09:38:44.172.002.000**
Last packet time	**09/10 09:38:44.225.802.000**
Max Packet Size (In/Out)	**391/263**
Min Packet Size (In/Out)	**391/263**
Avg Packet Size (In/Out)	**391/263**
Request Type	**GET /cnn/.element/css/2.0/main.css HTTP/1.1**
Actual Request	**GET /cnn/.element/css/2.0/main.css HTTP/1.1**
Response	**304: Not Modified**

Ready... Total Frames: 1653, Packets Analyzed: 1653

Figure 34 Individual Transactions

OptiView also indicates when there is a server that isn't responding well. This makes the analysis much easier.

Server Statistics. Once you have isolated the server that appears to be running too slowly, you can obtain server statistics though a series of SNMP queries sent by the OptiView analyzer. The result is a screen like the one shown in Figure 35 which indicates processor load, number of users, memory and disk utilization and a list of processes running on the server.



Device: FLUKE-OF0PRIK0U

All Devices:

Table: Host Resource Table

System

Up Time:	5 weeks 6 days 13 hours 38 minutes 49 seconds	Date:	Sep 8, 2008 1:53:08 PM
Memory Size:	2047 MB	Processor Load:	0%, 2%
Processes:	34	Num Users:	1
Max Processes:	0		

Storage

Type	Description	Size	Used	Alloc Failures
Fixed disk	D:\ Label: Serial Number 98eef784	111 GB	45%	0
Fixed disk	C:\ Label: Serial Number e4722143	24 GB	23%	0
Ram	Physical Memory	2047 MB	19%	0
Virtual mem	Virtual Memory	3943 MB	6%	0
Compact disk	E:\	0	0%	0
Compact disk	F:\	0	0%	0

Software Run List

Name	Index	Type	CPU Time	Memory	Path
System Idle Process	1	Operating system	196:20:13.54	28 KB	
dsm_sa_datamgr3...	1344	Application	2:53:34.09	14 MB	C:\Program Files\Dell\SysMgt\dataeng\bin\
System	4	Operating system	0:02:02.67	244 KB	
dsm_om_connsvc3...	1784	Application	0:01:51.29	30 MB	C:\Program Files\Dell\SysMgt\iws\bin\win32\
NTRtScan.exe	1588	Application	0:00:19.62	15 MB	C:\Program Files\Trend Micro\OfficeScan Client\
svchost.exe	1244	Application	0:00:12.75	29 MB	C:\WINDOWS\System32\ -k netsvcs

Table complete

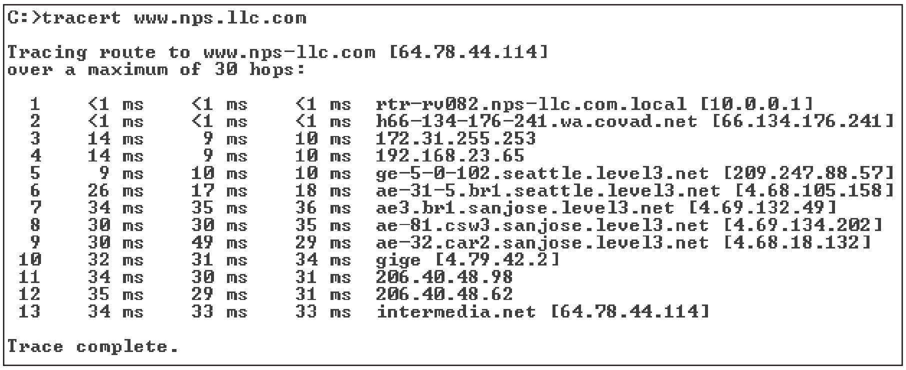
Figure 35 Server Statistics

This will allow you to do an analysis on the server system operation before determining that the slowness must be within the application.

Determine the Network Path

We must be sure there is a path to the server that is running our application. So, we need to make sure that both the layer two switch path and the layer three router path is functioning correctly. As we saw in the overview section, an incorrect path can lead to increased server response time. This can result form a variety of causes. The network routing topology may be incorrect. A link may be down causing a back-up route to be used. Or, a service provider may route traffic over unexpected paths. We can determine the layer two and layer three routes within our own network with a combination of a trace route procedure and SNMP queries to switches. We can also use trace route outside our network to discover the router path to the server.

From the command line of the operating system trace route (layer 3 only) is a utility that is actually an extension of the ping utility. In Windows,® if you type the command tracert x.x.x.x, a series of ping packets are sent to the target. In the first packet, the hop count (TTL) is set to zero. Therefore, the first routing device that receives the packet is required to discard it and report its action to the source. Then the utility increases the hop count in the packet to 1 and sends the ping. This packet passes thought the first router. But that router is required to decrease the hop count by one. So, when it arrives at the second router, the hop count has been reduced to zero and the second router drops the packet. However, like the first router, it reports to the source that it dropped the packet. You can see what is happening. Each successive ping packet goes one more hop. Also, each time a packet is discarded, the source gets a report telling it who dropped the packet. By using a timer on each ping, the source can build a report like the one in Figure 36.

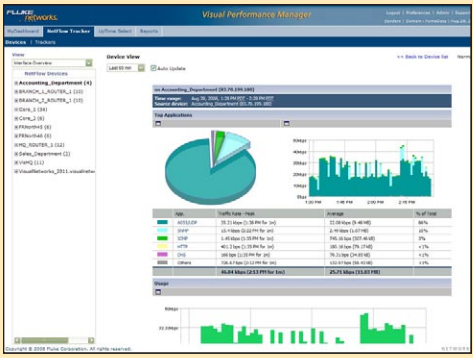


```
C:>tracert www.nps.llc.com

Tracing route to www.nps-llc.com [64.78.44.114]
over a maximum of 30 hops:

  1    <1 ms    <1 ms    <1 ms  rtr-rv082.nps-llc.com.local [10.0.0.1]
  2    <1 ms    <1 ms    <1 ms  h66-134-176-241.wa.covad.net [66.134.176.241]
  3    14 ms     9 ms    10 ms  172.31.255.253
  4    14 ms     9 ms    10 ms  192.168.23.65
  5     9 ms    10 ms    10 ms  ge-5-0-102.seattle.level3.net [209.247.88.57]
  6    26 ms    17 ms    18 ms  ae-31-5.br1.seattle.level3.net [4.68.105.158]
  7    34 ms    35 ms    36 ms  ae3.br1.sanjose.level3.net [4.69.132.49]
  8    30 ms    30 ms    35 ms  ae-81.csw3.sanjose.level3.net [4.69.134.202]
  9    30 ms    49 ms    29 ms  ae-32.car2.sanjose.level3.net [4.68.18.132]
 10    32 ms    31 ms    34 ms  gige [4.79.42.2]
 11    34 ms    30 ms    31 ms  206.40.48.98
 12    35 ms    29 ms    31 ms  206.40.48.62
 13    34 ms    33 ms    33 ms  intermedia.net [64.78.44.114]

Trace complete.
```

Figure 36 Trace Route

This creates a listing of the path that was followed from the source to the target x.x.x.x. The trace route tool also gives us important

Leveraging Netflow Data



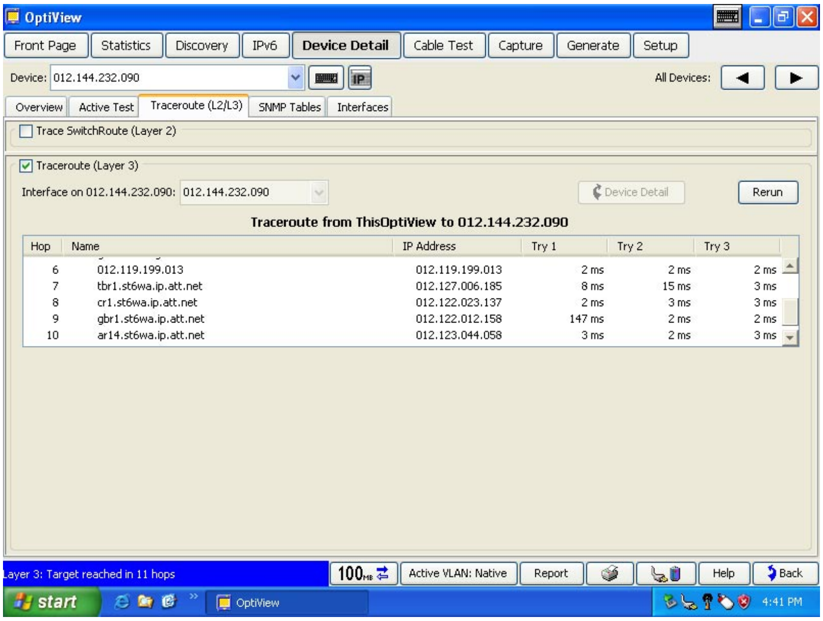
NetFlow tracker harnesses flow information from Cisco® IOS NetFlow, and flow standards from several other vendors, to give users detailed network traffic information from data already in your infrastructure devices. NetFlow Tracker provides information on all network conversations passing through the interfaces of supported routers and layer 3 switches, regardless of network design and can create unique databases that collect, store, and present valuable usage-based network data reports. This reporting provides you the visibility into every single conversation flow, to the per-minute level, up to the last two minutes, on every router, on every interface, right across your network. NetFlow Tracker provides complete coverage leveraging an existing data source already embedded within the network and usually already paid for (Cisco® Hardware and Cisco® IOS software or other NetFlow and IPFIX enabled networking devices from world leading manufacturers). When troubleshooting problems, the NetFlow Tracker Server can be easily accessed via a web browser on the OptiView Integrated Network Analyzer in order to obtain valuable data on router interface, WAN link performance and usage.

Use NetFlow Tracker to provide answers to many critical questions about the network, such as:

- Exactly what makes up my traffic over the WAN?
- Who are the users?
- What applications are they using?
- Who and what are consuming the bandwidth?
- How is quality of service working?
- Are network usage policies being followed?

information about how long it takes the ping to travel across each link.

Most technicians either don't know or don't remember the switches used with trace route. So, OptiView makes trace route an easy-to-use tool. In addition, it determines the layer two and layer three routes on a single screen and allows you to move on to router and switch queries and more. Figure 37 shows a trace route analysis from OptiView.



Traceroute from ThisOptiView to 012.144.232.090

Hop	Name	IP Address	Try 1	Try 2	Try 3
6	012.119.199.013	012.119.199.013	2 ms	2 ms	2 ms
7	tbr1.st6wa.ip.att.net	012.127.006.185	8 ms	15 ms	3 ms
8	cr1.st6wa.ip.att.net	012.122.023.137	2 ms	3 ms	3 ms
9	gbr1.st6wa.ip.att.net	012.122.012.158	147 ms	2 ms	2 ms
10	ar14.st6wa.ip.att.net	012.123.044.058	3 ms	2 ms	3 ms

Figure 37 OptiView Trace Route

When you find a link that is slow or appears to have a problem on one of its attached links, you can query the device to look at the link from the perspective of the switch or router. Later, we will consider a case study in which solving the problem uses such queries.

In some cases, the path between the client and the server will include WAN links. Since these are often provided by a carrier and have a usage charge associated with them, network architects tend to buy the minimum bandwidth they think will meet their needs. However, that can lead to problems if the links create a bottleneck for traffic moving between a client and a server. Later, we will look at a case study in which the analysis of the WAN link is used to solve a problem.

Application Flow Analysis

If your analysis of the path to the server doesn't unravel a problem with slow application response and the server statistics seem to indicate the server is fast enough, you must now suspect the application. This part of your analysis may involve capturing traffic going to and from the server. So, we'll approach it from that stand point. In a later case study, we'll show that you can do some analysis without capturing this traffic.

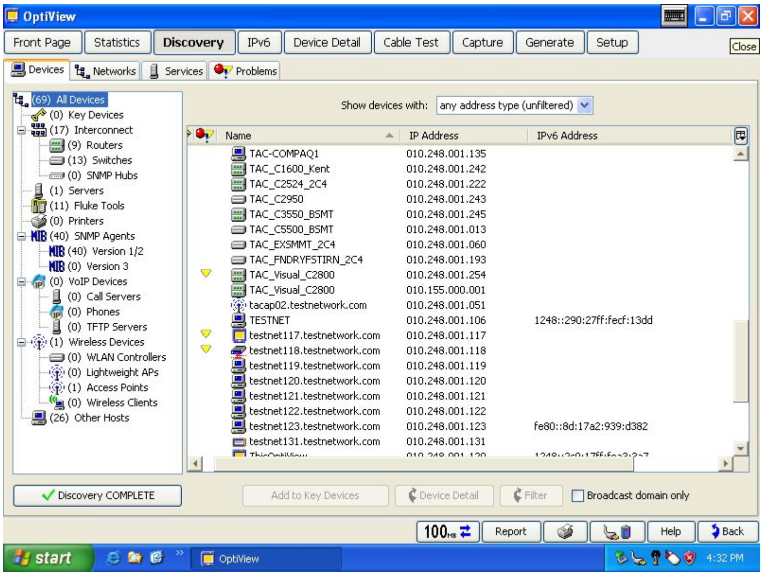
Capturing the traffic that is flowing between the client and the server is often called passive testing. This means that the OptiView must be placed at a point in the path between the client and the server where it can make a copy of every passing packet. One method of doing this is to insert a hub (Ethernet repeater) into the connection. But this is becoming a difficult technique to implement for several reasons. First, most hubs are 10Mbps half-duplex devices. Few network links operate at that speed.

Second, inserting the hub means disrupting the link to make the connection. And, third, performance is likely to be degraded by the fact that it is passing through this 10 Mbps bottleneck.

A better technique is to connect to a switch and then use port mirroring or port monitoring within the switch. Most Ethernet switches provide the capability to have the traffic passing in and out of one port, copied to a second port, where a device like the OptiView is attached. This is often referred to as a *mirror*, *monitor* or *span* port. If you are not the administrator of the switch, you need to have the person who does administer the switch set up his port.

Another technique is to tap one of the links over which the client-server traffic is passing. Passive taps for copper and fiber both exist and are easy to use. However, they involve interrupting the circuit in order to insert the tap. Once this is done, all traffic passing through the tap is copied to the tap port where the OptiView is attached.

Once you have connected to a point where the capture can be made, you need to establish a filter so that only appropriate traffic will be captured. You can build this filter based on the protocol the server is using, the address of the server or nearly any other characteristic of the traffic passing into and out of the server application. One easy method to accomplish this is to use the Discovery Screen shown in Figure 38.



Show devices with: any address type (unfiltered)

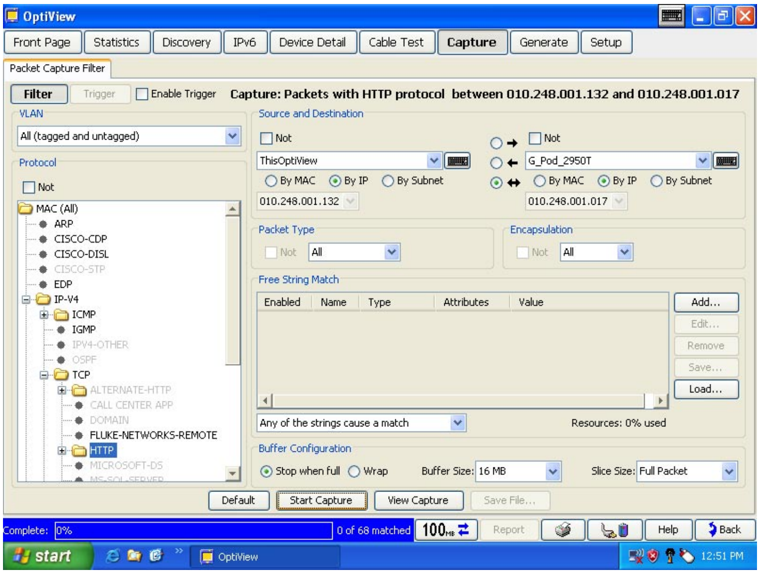
(69) All Devices
(0) Key Devices
(17) Interconnect
(9) Routers
(13) Switches
(0) SNMP Hubs
(1) Servers
(11) Fluke Tools
(0) Printers
(40) SNMP Agents
(40) Version 1/2
(0) Version 3
(0) VoIP Devices
(0) Call Servers
(0) Phones
(0) TFTP Servers
(1) Wireless Devices
(0) WLAN Controllers
(0) Lightweight APs
(1) Access Points
(0) Wireless Clients
(26) Other Hosts

Name	IP Address	IPv6 Address
TAC-COMPAQ1	010.248.001.135	
TAC_C1600_Kent	010.248.001.242	
TAC_C2524_2C4	010.248.001.222	
TAC_C2950	010.248.001.243	
TAC_C3550_BSMT	010.248.001.245	
TAC_C5500_BSMT	010.248.001.013	
TAC_EXSMMT_2C4	010.248.001.060	
TAC_FNDRYFSTIRN_2C4	010.248.001.193	
TAC_Visual_C2800	010.248.001.254	
TAC_Visual_C2800	010.155.000.001	
tacap02.testnetwork.com	010.248.001.051	
TESTNET	010.248.001.106	1248::290:27ff:fecf:13dd
testnet117.testnetwork.com	010.248.001.117	
testnet118.testnetwork.com	010.248.001.118	
testnet119.testnetwork.com	010.248.001.119	
testnet120.testnetwork.com	010.248.001.120	
testnet121.testnetwork.com	010.248.001.121	
testnet122.testnetwork.com	010.248.001.122	
testnet123.testnetwork.com	010.248.001.123	fe80::8d:17a2:939:d382
testnet131.testnetwork.com	010.248.001.131	

Discovery COMPLETE

Figure 38 The Discovery Screen

If you highlight the device and select the Filter button, you will be provided with a screen that looks like the one in Figure 39.



Capture: Packets with HTTP protocol between 010.248.001.132 and 010.248.001.017

VLAN: All (tagged and untagged)

Protocol: MAC (All) › IP-V4 › TCP › HTTP

Source and Destination: ThisOptiView, By IP, 010.248.001.132 ↔ G_Pod_2950T, By IP, 010.248.001.017

Packet Type: All

Encapsulation: All

Free String Match: Any of the strings cause a match

Resources: 0% used

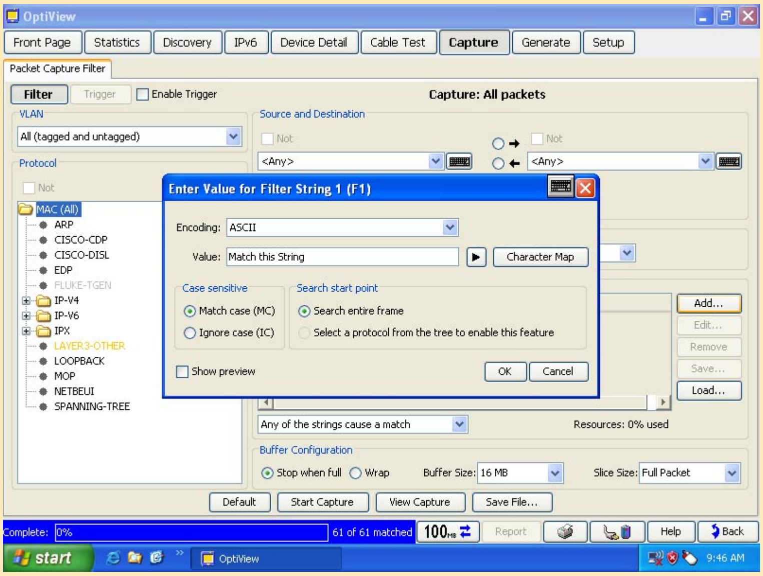
Buffer Configuration: Stop when full; Buffer Size: 16 MB; Slice Size: Full Packet

Complete: 0%   0 of 68 matched

Figure 39 Setting Up a Filter

Troubleshooting Intermittent Problems

When intermittent problems occur, most IT professionals turn to packet capture to diagnose the problem. The OptiView analyzers make it easy to troubleshoot intermittent problems with advanced triggering and filtering to capture the traffic before, after or around the event occurrence and ensures the event is captured the first time to avoid doing random traffic captures that may not contain anything of interest. To capture a specific event the analyzer must inspect the contents of each packet to see if it matches a pattern or error message string which is indicative of the event occurring. The string matching is performed in hardware in real-time with line-rate gigabit capture to ensure all the relevant packets are captured – after all, you can't analyze packets that were never captured. A total of eight sets of triggers or filters can be defined to trigger a capture unattended for later analysis, allowing analysis when you have time, not when the event occurred.



VLAN: All (tagged and untagged)

<Any>

<Any>

Enter Value for Filter String 1 (F1)

Encoding: ASCII

Value: Match this String

Case sensitive: [✓ Match case (MC)] Ignore case (IC)

Search start point: [✓ Search entire frame]

Any of the strings cause a match

Resources: 0% used

[✓ Stop when full] Wrap

Buffer Size: 16 MB

Slice Size: Full Packet

Complete: 0%

61 of 61 matched

100MB

Free String Match

Application Troubleshooting Resource Center:
www.flukenetworks.com/appts

Using OptiView® Protocol Expert

The Integrated Protocol Expert (iPE), which is built into the portable OptiView analyzer, is a full featured, easy to use, protocol analyzer. The screen similar to the one in Figure 40 will appear.

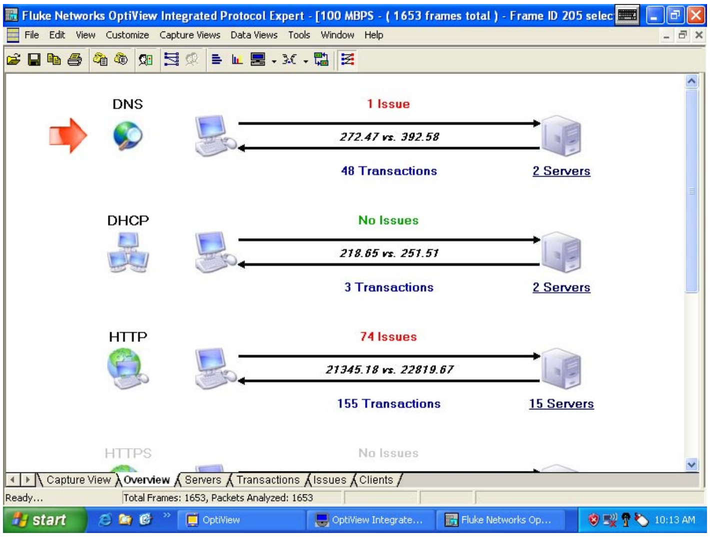
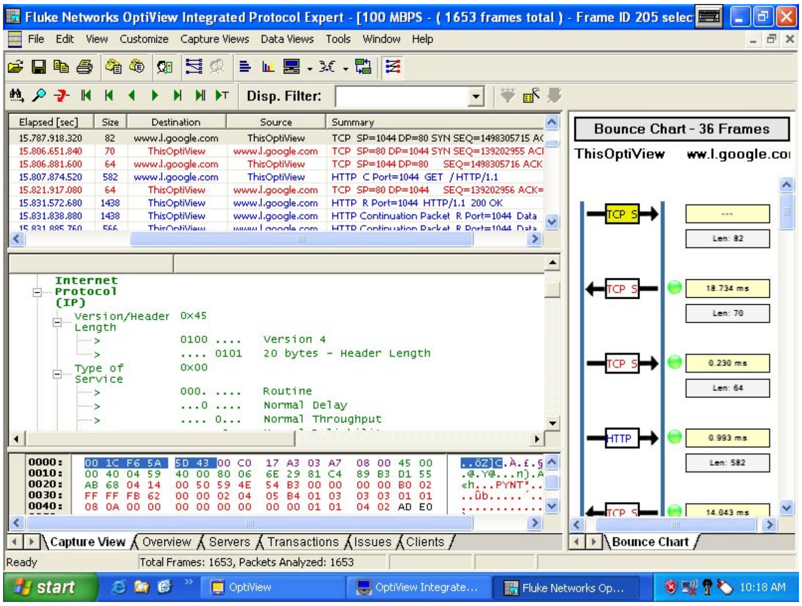


Figure 40 The Initial Screen in iPE

This is the overview of protocols and servers seen in the captured traffic. It shows the protocols used between the client and several servers and indicates whether issues in performance were detected. In the illustration you can see that DNS appeared to function well. However, HTTP issues are evident. By clicking on the *Capture* tab, you get a screen like the one in Figure 41



```
Elapsed [sec]    Size  Destination       Source            Summary
15.787.918.320   82    www.l.google.com  ThisOptiView      TCP  SP=1044 DP=80 SYN SEQ=1498305715 AC
15.806.651.840   70    ThisOptiView      www.l.google.com  TCP  SP=80 DP=1044 SYN SEQ=139202955 ACI
15.806.881.600   64    www.l.google.com  ThisOptiView      TCP  SP=1044 DP=80     SEQ=1498305716 ACK
15.807.874.520   582   www.l.google.com  ThisOptiView      HTTP  C Port=1044  GET  / HTTP/1.1
15.821.917.080   64    ThisOptiView      www.l.google.com  TCP  SP=80 DP=1044     SEQ=139202956 ACK=
15.831.572.680   1438  ThisOptiView      www.l.google.com  HTTP  R Port=1044  HTTP/1.1  200 OK
15.831.838.880   1438  ThisOptiView      www.l.google.com  HTTP Continuation Packet  R Port=1044  Data
15.831.885.760   566   ThisOptiView      www.l.google.com  HTTP Continuation Packet  R Port=1044  Data
```

```
Internet
Protocol
(IP)
   Version/Header  0x45
   Length
      >            0100 ....   Version 4
      >            .... 0101   20 bytes - Header Length
   Type of         0x00
   Service
      >            000. ....   Routine
      >            ...0 ....   Normal Delay
      >            .... 0...   Normal Throughput
```

```
0000:  00 1C F6 5A  5D 43 00 C0  17 A3 03 A7  08 00 45 00   ..öZ]C.À.£.§
0010:  00 40 04 59  40 00 80 06  6E 29 81 C4  89 B3 D1 55   .@.Y@...n).Ä
0020:  AB 68 04 14  00 50 59 4E  54 B3 00 00  00 00 B0 02   «h...PYNT³..
0030:  FF FF FB 62  00 00 02 04  05 B4 01 03  03 03 01 01   ..ûb.....´..
0040:  08 0A 00 00  00 00 00 00  00 00 01 01  04 02 AD E0   ............
```

```
Ready    Total Frames: 1653, Packets Analyzed: 1653
```

Figure 41 The Capture Tab

This shows the details of the packets on the left side of the screen and a bounce chart on the right side of the screen. The *bounce chart* shows the interaction and the time and size of packets in the interaction. The packet detail on the left is broken into three sections. The top of the screen shows a listing of the frames as they were captured. The middle section shows the detailed decode of the frame highlighted and the bottom section shows both the ASCII value and the hex value of each byte in order.

From a portion of the bounce chart shown in Figure 42, you can see the TCP connection being made between port 1080 on the client and port 80 (HTTP) on the server. It takes 73.193 ms.

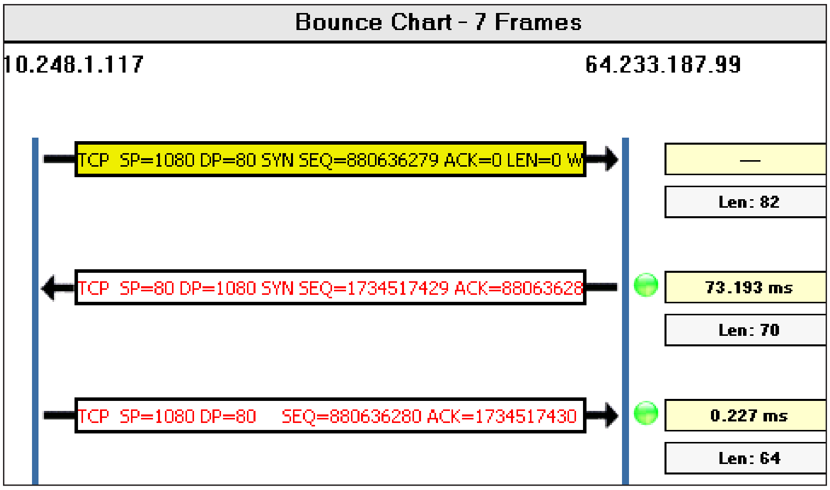

Bounce Chart - 7 Frames
10.248.1.117
64.233.187.99
TCP SP=1080 DP=80 SYN SEQ=880636279 ACK=0 LEN=0 W
—
Len: 82
TCP SP=80 DP=1080 SYN SEQ=1734517429 ACK=88063628
73.193 ms
Len: 70
TCP SP=1080 DP=80 SEQ=880636280 ACK=1734517430
0.227 ms
Len: 64

Figure 42 The TCP Connection

The first line shows the SYN being sent from the client to the server. The response is the SYN/ACK (only the SYN is indicated). We know this is the response because the pair of port numbers is the same as on the first line. On the third line, the port numbers are also the same pair, so this must represent the ACK that confirms the SYN/ACK. Notice that the client's reaction time between line two and line three is only a fraction of the server's reaction time between line one and line two.

Additional information about the connections to this server can be obtained by clicking on the server from the Overview tab. This is shown in Figure 43.

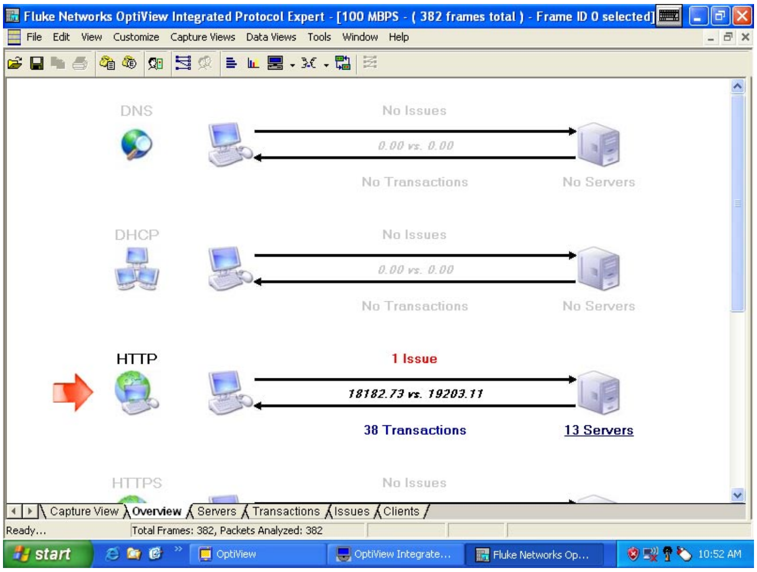
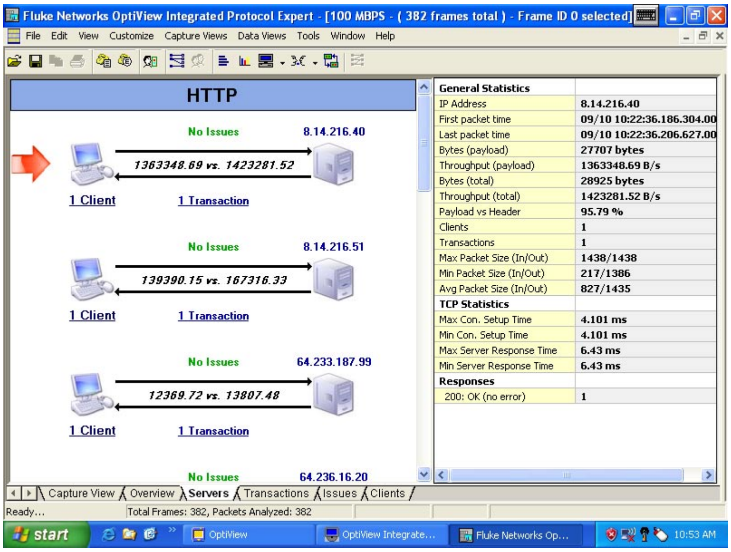


Figure 43 Server Information

The screen that appears will look like Figure 44. It contains a wealth of information about the exchange between the client and the server you picked. Here you can look for evidence of poor server performance, small payload sizes and other issues.

Reporting and documenting

Fluke Networks' OptiView® Reporter works with the OptiView Integrated Network Analyzers and Workgroup Analyzers to provide documenting and trending capabilities from one central location. The OptiView hardware agents automatically perform detailed device, VLAN and network discovery, problem identification, and infrastructure device analysis. This information, in turn, is automatically imported into OptiView Reporter for reporting, trending, and event notification.

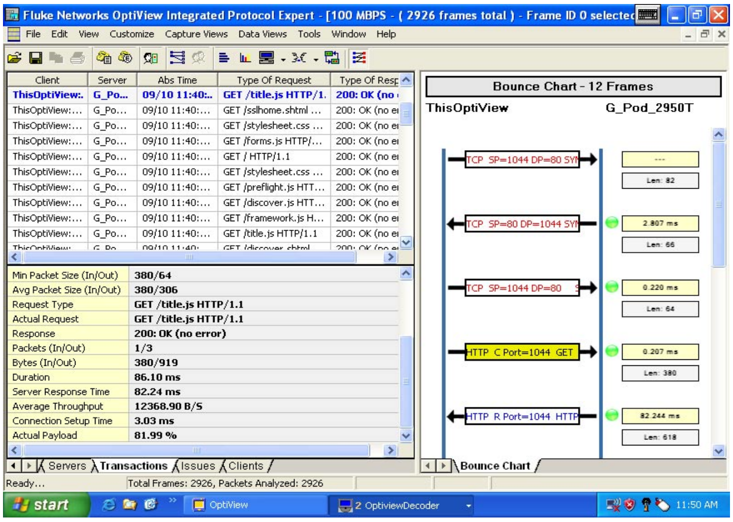


General Statistics	
IP Address	8.14.216.40
First packet time	09/10 10:22:36.186.304.00
Last packet time	09/10 10:22:36.206.627.00
Bytes (payload)	27707 bytes
Throughput (payload)	1363348.69 B/s
Bytes (total)	28925 bytes
Throughput (total)	1423281.52 B/s
Payload vs Header	95.79 %
Clients	1
Transactions	1
Max Packet Size (In/Out)	1438/1438
Min Packet Size (In/Out)	217/1386
Avg Packet Size (In/Out)	827/1435
TCP Statistics	
Max Con. Setup Time	4.101 ms
Min Con. Setup Time	4.101 ms
Max Server Response Time	6.43 ms
Min Server Response Time	6.43 ms
Responses	
200: OK (no error)	1

Ready... Total Frames: 382, Packets Analyzed: 382

Figure 44 Server Details

Starting from the Overview screen, if you click on the Transactions tab, you can see all the transactions for a particular server. This is shown in Figure 45.



Client	Server	Abs Time	Type Of Request	Type Of Resp
ThisOptiView:.	G_Po...	09/10 11:40:..	GET /title.js HTTP/1.	200: OK (no
ThisOptiView:...	G_Po...	09/10 11:40:...	GET /sslhome.shtml ...	200: OK (no e
ThisOptiView:...	G_Po...	09/10 11:40:...	GET /stylesheet.css ...	200: OK (no e
ThisOptiView:...	G_Po...	09/10 11:40:...	GET /forms.js HTTP/...	200: OK (no e
ThisOptiView:...	G_Po...	09/10 11:40:...	GET / HTTP/1.1	200: OK (no e
ThisOptiView:...	G_Po...	09/10 11:40:...	GET /stylesheet.css ...	200: OK (no e
ThisOptiView:...	G_Po...	09/10 11:40:...	GET /preflight.js HTT...	200: OK (no e
ThisOptiView:...	G_Po...	09/10 11:40:...	GET /discover.js HTT...	200: OK (no e
ThisOptiView:...	G_Po...	09/10 11:40:...	GET /framework.js H...	200: OK (no e
ThisOptiView:...	G_Po...	09/10 11:40:...	GET /title.js HTTP/1.1	200: OK (no e
ThisOptiView:	G_Po	09/10 11:40:	GET /discover.shtml	200: OK (no e

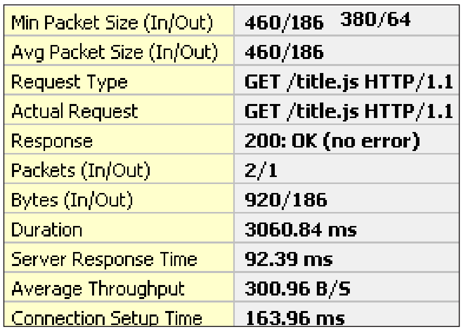
Min Packet Size (In/Out)	380/64
Avg Packet Size (In/Out)	380/306
Request Type	GET /title.js HTTP/1.1
Actual Request	GET /title.js HTTP/1.1
Response	200: OK (no error)
Packets (In/Out)	1/3
Bytes (In/Out)	380/919
Duration	86.10 ms
Server Response Time	82.24 ms
Average Throughput	12368.90 B/S
Connection Setup Time	3.03 ms
Actual Payload	81.99 %

Ready... Total Frames: 2926, Packets Analyzed: 2926

Figure 45 The Transactions Tab

In Figure 45, you can see almost the entire transaction. On the bounce chart you see the three-way handshake on the first three lines. Line four shows the request for the file stylesheet.css. Line five shows the server returning the requested file.

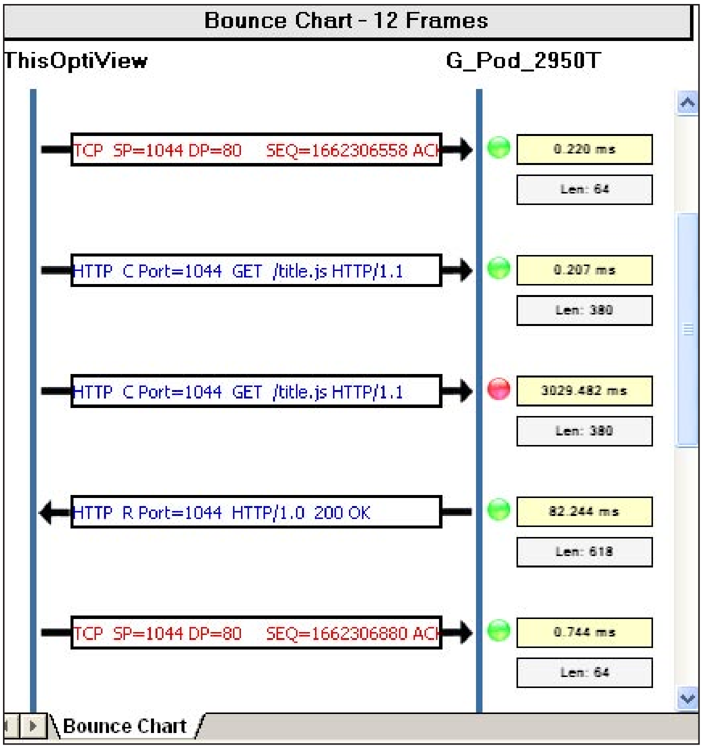
In the next image, Figure 46, the statistics table shows a problem. Only one medium sized packet was received by the client. However, the session used to retrieve that file required over three seconds.



Min Packet Size (In/Out)	460/186 380/64
Avg Packet Size (In/Out)	460/186
Request Type	GET /title.js HTTP/1.1
Actual Request	GET /title.js HTTP/1.1
Response	200: OK (no error)
Packets (In/Out)	2/1
Bytes (In/Out)	920/186
Duration	3060.84 ms
Server Response Time	92.39 ms
Average Throughput	300.96 B/S
Connection Setup Time	163.96 ms

Figure 46 Slow Server Response

The bounce chart for that transaction is shown in Figure 47. After the three-way handshake to open the session, you can see the request for the file green.gif. When the server doesn't respond in three seconds, you can see the second request for the file. One of two things occurred, either the first request never arrived or the server application was very slow to respond. Since the response to the second request was rather quick, you can suspect that the first request was lost or not processed for some reason.



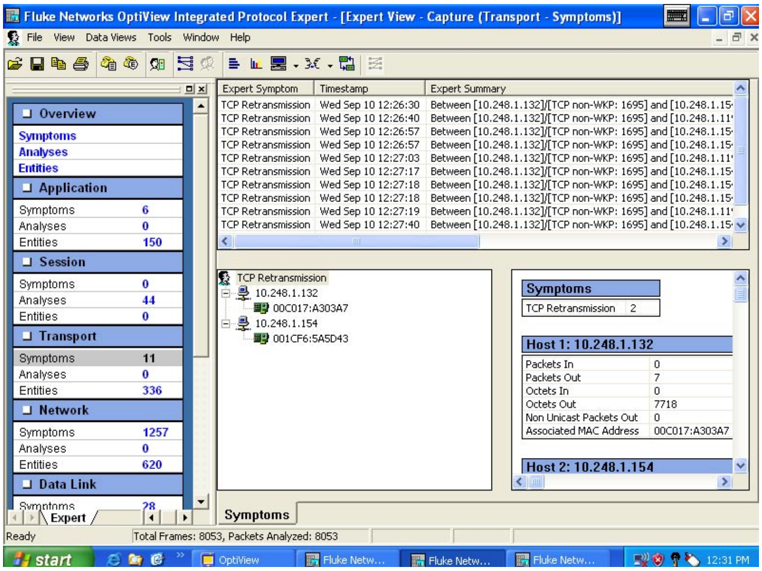
Bounce Chart - 12 Frames

ThisOptiView G_Pod_2950T

```
TCP  SP=1044 DP=80    SEQ=1662306558 ACK     0.220 ms   Len: 64
HTTP  C Port=1044  GET  /title.js HTTP/1.1   0.207 ms   Len: 380
HTTP  C Port=1044  GET  /title.js HTTP/1.1   3029.482 ms   Len: 380
HTTP  R Port=1044  HTTP/1.0  200 OK          82.244 ms   Len: 618
TCP  SP=1044 DP=80    SEQ=1662306880 ACK     0.744 ms   Len: 64
```

Figure 47 The Bounce Chart for Slow Server

From the Overview screen, you can click on the Issues Tab. That will display all of the issues that OptiView found in the connection. In Figure 48 you see a list of retransmissions that were discovered. As we discussed in the section on TCP operation, retransmissions will cause the TCP timing algorithm to assume the link is not robust. As a result it will slow down the rate at which it offers packets to the connection.


Fluke Networks OptiView Integrated Protocol Expert - [Expert View - Capture (Transport - Symptoms)]
File View Data Views Tools Window Help
Expert
Symptoms
start
OptiView
Fluke Netw...
Fluke Netw...
Fluke Netw...
12:31 PM

Overview	
Symptoms	
Analyses	
Entities	
Application	
Symptoms	6
Analyses	0
Entities	150
Session	
Symptoms	0
Analyses	44
Entities	0
Transport	
Symptoms	11
Analyses	0
Entities	336
Network	
Symptoms	1257
Analyses	0
Entities	620
Data Link	
Symptoms	28

Expert Symptom	Timestamp	Expert Summary
TCP Retransmission	Wed Sep 10 12:26:30	Between [10.248.1.132]/[TCP non-WKP: 1695] and [10.248.1.15
TCP Retransmission	Wed Sep 10 12:26:40	Between [10.248.1.132]/[TCP non-WKP: 1695] and [10.248.1.11
TCP Retransmission	Wed Sep 10 12:26:57	Between [10.248.1.132]/[TCP non-WKP: 1695] and [10.248.1.15
TCP Retransmission	Wed Sep 10 12:26:57	Between [10.248.1.132]/[TCP non-WKP: 1695] and [10.248.1.15
TCP Retransmission	Wed Sep 10 12:27:03	Between [10.248.1.132]/[TCP non-WKP: 1695] and [10.248.1.11
TCP Retransmission	Wed Sep 10 12:27:17	Between [10.248.1.132]/[TCP non-WKP: 1695] and [10.248.1.15
TCP Retransmission	Wed Sep 10 12:27:18	Between [10.248.1.132]/[TCP non-WKP: 1695] and [10.248.1.15
TCP Retransmission	Wed Sep 10 12:27:18	Between [10.248.1.132]/[TCP non-WKP: 1695] and [10.248.1.15
TCP Retransmission	Wed Sep 10 12:27:19	Between [10.248.1.132]/[TCP non-WKP: 1695] and [10.248.1.11
TCP Retransmission	Wed Sep 10 12:27:40	Between [10.248.1.132]/[TCP non-WKP: 1695] and [10.248.1.15

```
TCP Retransmission
  10.248.1.132
    00C017:A303A7
  10.248.1.154
    001CF6:5A5D43
```

Symptoms	
TCP Retransmission	2

Host 1: 10.248.1.132	
Packets In	0
Packets Out	7
Octets In	0
Octets Out	7718
Non Unicast Packets Out	0
Associated MAC Address	00C017:A303A7

Host 2: 10.248.1.154

Ready Total Frames: 8053, Packets Analyzed: 8053

Figure 48 The Issues Tab

Fix the Problem

The third step in our five step process is to fix the problem. Because there can be a great number of causes, we can only assume you've found it by this time. However, you should know what single or what few items combined to cause the inadequate performance of the application. Sometimes this step will be easy. However, it can be both time consuming and expensive. For example, it could be as easy as changing the configuration of a router. On the other hand, you might need a larger server or a faster WAN link. However, it could also be the application itself – as we pointed out earlier, TCP routinely operates with 50% or more overhead, so imagine what is happening when an application uses minimum length frames of 64 bytes. The MAC frame headers, IP and TCP headers and frame check sequences can occupy

46 bytes of the frame, leaving only 18 bytes available to transfer data. Badly written applications could require thousands of frames exchanged to complete a single simple transaction.

Validate the Fix

If the problem affected one particular user, spend some time to be sure the user sees a difference after the problem has been fixed. If the entire network was down, thoroughly check that all parts are now up and functioning well. If connectivity to a server application was the issue, don't just check connectivity to the server. Check that the application can be reached by making sure a complete transaction can occur. If a particular link was slow, run a series of ping tests over time and check back to make sure that all of the response times are good. Taking a little extra care after a fix is made can lead to work time saved because the problem won't occur again.

Document the Fix

It's a very good idea to keep written records of what you fix and to have the records well organized. This is important to other individuals who may need to fix a similar problem if you aren't available to consult. Also, records are necessary for documenting the need to acquire larger servers or test equipment. Documentation should include screen shots, notes stored in text documents, spread sheets with incidents and time on task and a host of other kinds of documents. What is important is to create a record for your future referral and for others who may have need of such history.

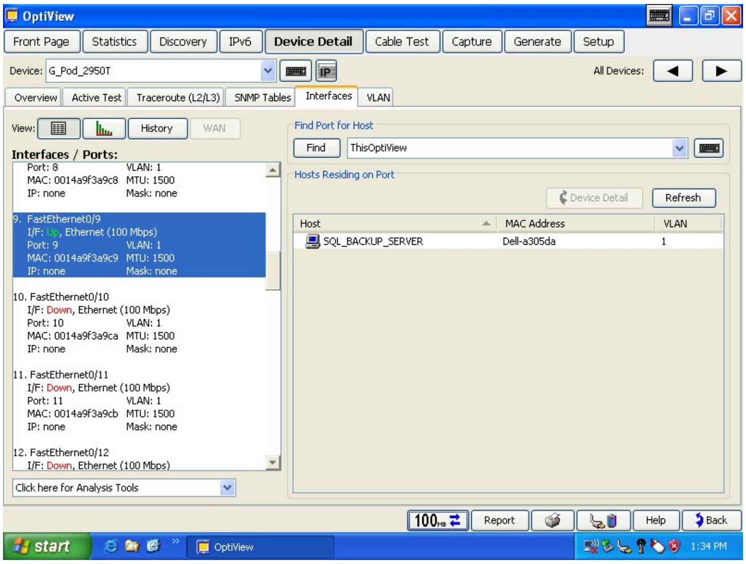
Case Studies

By looking at examples of troubleshooting, we can understand the methods we're outlining in this document. Consequently, we will consider two case studies.

Case Study 1: Obtaining Switch Statistics

Sometimes obtaining the information you need can involve moving from one building or site to another. Travel time is lost time. So, in this illustration we see how OptiView can be used to learn about a link to a server and the configuration of the network at that point.

Let's say we have a server that is connected to a switch in some part of the network and users are complaining that the server is responding slowly. Using the unique layer 2 and layer 3 trace route techniques described before, we find the path to the server and the layer 2 switch to which the server is connected. By highlighting that switch and clicking on Device Detail and the Interfaces tab, we can get the screen shown in Figure 49.



Device: G_Pod_2950T

ThisOptiView

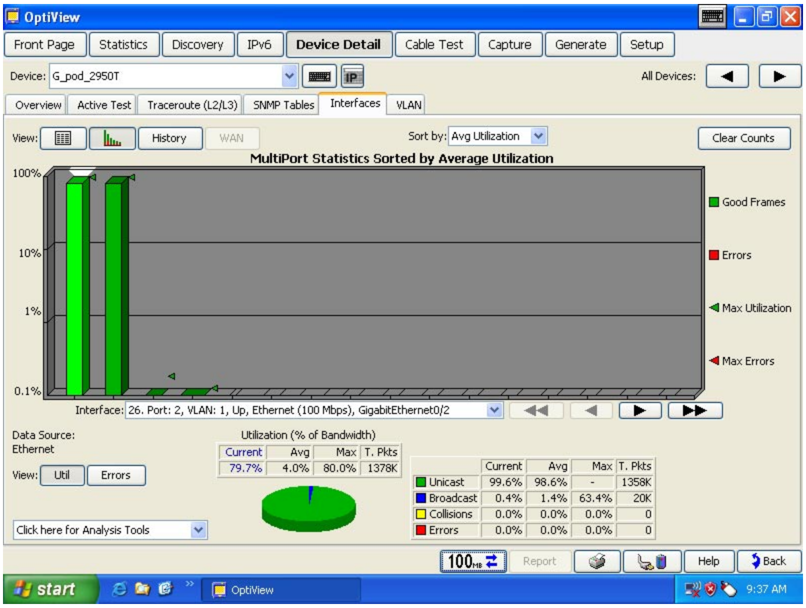
Interfaces / Ports:

```
   Port: 8              VLAN: 1
   MAC: 0014a9f3a9c8    MTU: 1500
   IP: none             Mask: none

9.  FastEthernet0/9
   I/F: Up, Ethernet (100 Mbps)
   Port: 9              VLAN: 1
   MAC: 0014a9f3a9c9    MTU: 1500
   IP: none             Mask: none

10. FastEthernet0/10
   I/F: Down, Ethernet (100 Mbps)
   Port: 10             VLAN: 1
   MAC: 0014a9f3a9ca    MTU: 1500
   IP: none             Mask: none

11. FastEthernet0/11
   I/F: Down, Ethernet (100 Mbps)
   Port: 11             VLAN: 1
   MAC: 0014a9f3a9cb    MTU: 1500
   IP: none             Mask: none

12. FastEthernet0/12
   I/F: Down, Ethernet (100 Mbps)
```

Host	MAC Address	VLAN
SQL_BACKUP_SERVER	Dell-a305da	1

Figure 49 The Switch Interface Screen

From this screen, we can determine which port is connected to the server, in this case port 9. Then by clicking on the bar graph button next to *View*, we can query the switch to get the statistics on that link to the server. This is shown in Figure 50.



Interface: 26. Port: 2, VLAN: 1, Up, Ethernet (100 Mbps), GigabitEthernet0/2

Data Source: Ethernet

View: Util Errors

Utilization (% of Bandwidth)

Current	Avg	Max	T. Pkts
79.7%	4.0%	80.0%	1378K

	Current	Avg	Max	T. Pkts
🟩 Unicast	99.6%	98.6%	-	1358K
🟦 Broadcast	0.4%	1.4%	63.4%	20K
🟨 Collisions	0.0%	0.0%	0.0%	0
🟥 Errors	0.0%	0.0%	0.0%	0

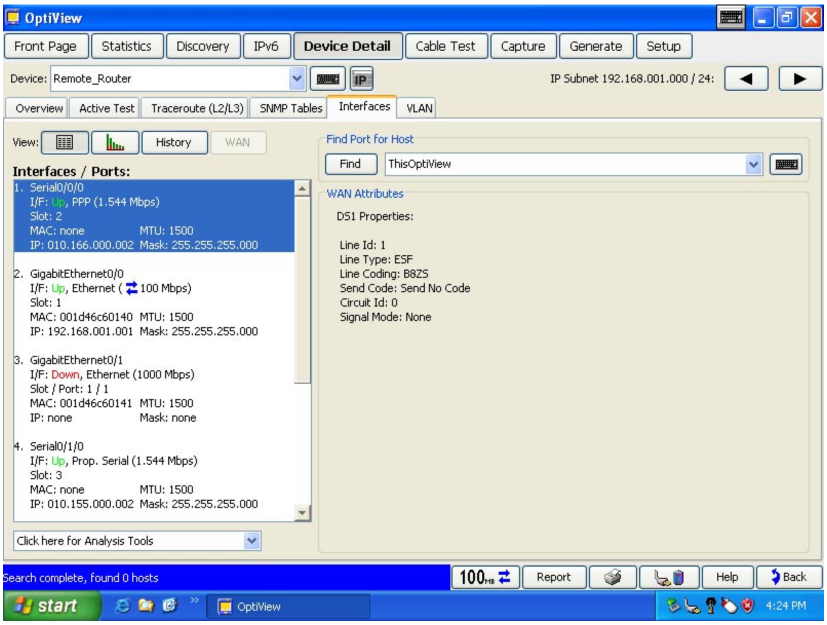
Figure 50 Link Statistics

This allows us to see the utilization and error levels on the link and even access an RMON history study of the link, if it is enabled on the switch. Sometimes servers are browsing on a link. If that is the case, there will be an abnormally high level of broadcast traffic. Or, you might find that there are errors caused by a physical link problem such as a defective NIC or bad copper cable. Being able to see the link without being physically at the switch provides a considerable savings in time.

Case Study 2: Investigating WAN Link Performance

In this case study, we'll see how the OptiView can be used to analyze a WAN link that appears to be a bottleneck in the path to the server. In this scenario, access and performance across a WAN link has been slow. The service provider is saying that the company needs an additional T-1 circuit.

In Figure 51 below, we begin with the Device Detail Interface screen that is similar to the one in Case Study 1. This time we are querying a router.



Device: Remote_Router

IP Subnet 192.168.001.000 / 24:

Interfaces / Ports:

```
1. Serial0/0/0
   I/F: Up, PPP (1.544 Mbps)
   Slot: 2
   MAC: none            MTU: 1500
   IP: 010.166.000.002  Mask: 255.255.255.000

2. GigabitEthernet0/0
   I/F: Up, Ethernet ( 100 Mbps)
   Slot: 1
   MAC: 001d46c60140    MTU: 1500
   IP: 192.168.001.001  Mask: 255.255.255.000

3. GigabitEthernet0/1
   I/F: Down, Ethernet (1000 Mbps)
   Slot / Port: 1 / 1
   MAC: 001d46c60141    MTU: 1500
   IP: none             Mask: none

4. Serial0/1/0
   I/F: Up, Prop. Serial (1.544 Mbps)
   Slot: 3
   MAC: none            MTU: 1500
   IP: 010.155.000.002  Mask: 255.255.255.000
```

ThisOptiView

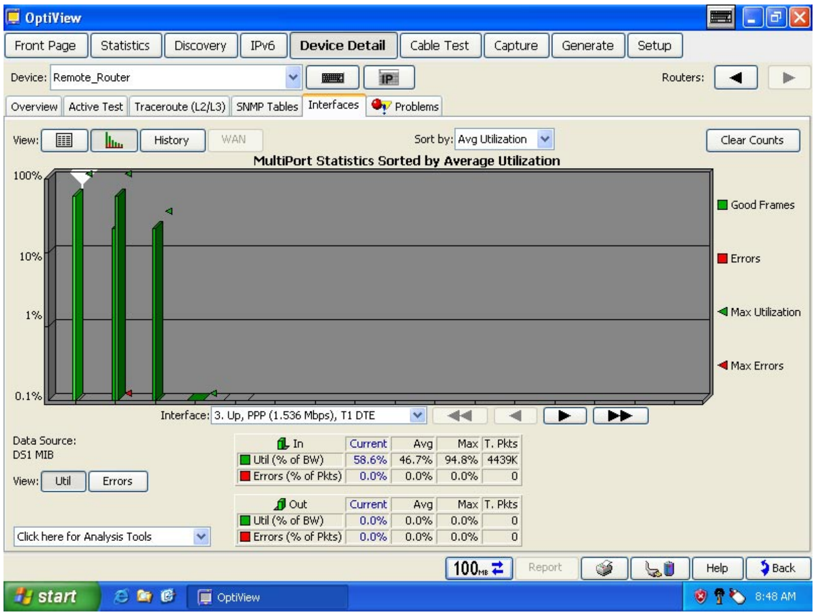
WAN Attributes

```
DS1 Properties:

 Line Id: 1
 Line Type: ESF
 Line Coding: B8ZS
 Send Code: Send No Code
 Circuit Id: 0
 Signal Mode: None
```

Search complete, found 0 hosts

Figure 51 Interfaces on the Router

By highlighting the interface with the suspected WAN link, we can see some of the interface statistics and see that it is, in fact at T-1 circuit running at 1.544 Mbps and that the MTU is 1500 bytes. This means that IP packets with the standard maximum size of 1500 bytes will not be fragmented in order to traverse the link. By clicking on the bar graph button next to *View*. the screen in Figure 52 is produced.



In	Current	Avg	Max	T. Pkts
🟩 Util (% of BW)	58.6%	46.7%	94.8%	4439K
🟥 Errors (% of Pkts)	0.0%	0.0%	0.0%	0

Out	Current	Avg	Max	T. Pkts
🟩 Util (% of BW)	0.0%	0.0%	0.0%	0
🟥 Errors (% of Pkts)	0.0%	0.0%	0.0%	0

Figure 52 Utilization of the WAN Link

This screen makes it obvious that the utilization of the WAN link is only slightly under 50%. This screen is periodically updated, so you could study it over a period of time to be sure about your conclusion. As we mentioned before, it would be wise to move on to an analysis of the server or the application running on the server.

Summary

In this document, we have provided information on how network applications work. We have discussed the typical sequence for a client to use when it makes a connection to a server and asks for a service to be provided. As part of that analysis we considered the DNS lookup, the role of ARP, how the TCP connection is opened, how the request is sent and the response is received, and how the connection is closed. We considered the major categories of applications such as web applications, database applications and transaction processing systems. During that explanation we noted that 90% of the applications are based on the TCP protocol.

In analyzing how the client interacts with the server, we considered the route to the server and how it can hide problems that affect performance. We also did a brief tutorial on the operation of TCP since its operation has such a significant influence on how well the application appears to perform. We discovered that when packets are lost in the network, the impact on TCP performance is significant. This led us to the conclusion that the quality of the network is very important.

In considering TCP operation, we described the three-way handshake that is used to create the connection, the method of assuring reliability of the transfer of data and the four-way handshake that closes the connection.

Application Troubleshooting Resource Center:
www.flukenetworks.com/appts

We moved on to a discussion of a Five Step Process to troubleshoot applications. We stress that a logical approach would include: (1) determining the domain of the problem; (2) performing an application flow analysis; (3) fixing the problem; (4) validating that the problem is fixed; and (5) documenting that was fixed and how it was fixed.

Using the techniques suggested in this paper should reduce finger pointing between your application developers and your network staff. And, it will increase your success in solving network problems, reducing costs to your company.

For more information on Network Troubleshooting and Application Diagnostics visit the Application Troubleshooting Resource Center at **www.flukenetworks.com/appts** to request a free 5-day network analyzer trial, sign-up for upcoming events and download technical content.

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