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# THESIS

**FEASIBILITY STUDY OF VOIP INTEGRATION INTO THE MYSEA ENVIRONMENT**

by

Lily Tse

September 2005

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# FEASIBILITY STUDY OF VOIP INTEGRATION INTO THE MYSEA ENVIRONMENT

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# ABSTRACT

Voice over Internet Protocol (VoIP) is becoming popular due to its low cost and the management advantages it offers over traditional PSTN phone systems. VoIP is widely implemented with H.323 and Session Initiation Protocol (SIP) standards. However, both protocols are poorly designed for networks with common security solutions such as firewalls and Network Address Translation (NAT).

This project is a feasibility study of SIP-based VoIP integration into the Monterey Security Architecture (MYSEA), a multilevel secure environment that uses NAT as a security mechanism. A gathering of comparative studies on VoIP protocols was performed to guide the selection of SIP as the test protocol. A set of experiments was devised and conducted using SIP-based softphones for this study. The insights gained from the experiments provide useful insights to the MYSEA project concerning VoIP security.

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# ACRONYMS AND ABBREVIATIONS

| | |
|---|---|
| AP | Access Point |
| ASN.1 | Abstract Syntax Notation One |
| ATA | Analog Telephone Adapter |
| | |
| BES | Back End Service |
| | |
| CODEC | Coder-Decoder |
| | |
| DNAT | Destination Network Address Translation |
| DoD | Department of Defense |
| DoS | Denial of Service |
| | |
| GSM | Global System for Mobile Communication |
| | |
| IANA | Internet Assigned Numbers Authority |
| IETF | Internet Engineering Task Force |
| IP | Internet Protocol |
| IPsec | Internet Protocol Security |
| IPv4 | Internet Protocol version 4 |
| IPv6 | Internet Protocol version 6 |
| ITU-T | International Telecommunications Union – Telecommunications Standards Committee |
| | |
| LAN | Local Area Network |
| | |
| MEGACO | Media Gateway Control |
| MIME | Multipurpose Internet Mail Extensions |
| MGCP | Media Gateway Control Protocol |
| MCU | Multipoint Control Unit |
| MYSEA | Monterey Security Architecture |
| | |
| NAT | Network Address Translation |
| NIST | National Institute of Standards and Technology |
| | |
| PBX | Public Branch Exchange |
| PDA | Personal Digital Assistant |
| PSTN | Public Switched Telephone Network |
| | |
| QoS | Quality of Service |
| | |
| RTCP | Real-time Control Protocol |
| RTP | Real-time Transport Protocol |

| | |
|---|---|
| SDP | Session Description Protocol |
| SIP | Session Initiation Protocol |
| S/MIME | Secure/ Multipurpose Internet Mail Extensions |
| SNAT | Source Network Address Translation |
| SS7 | Signaling System 7 |
| TCP | Transmission Control Protocol |
| UDP | User Datagram Protocol |
| VoIP | Voice over Internet Protocol |
| WAN | Wide Area Network |
| WEP | Wired Equivalent Privacy |
| WLAN | Wireless Local Area Network |

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# I. INTRODUCTION

## A. MOTIVATION

Voice over Internet Protocol (VoIP) is becoming a popular technology. Currently, three million households use VoIP. It is estimated that the number will increase to twenty-seven million by the end of 2009 [1]. Furthermore, companies and public-sector organizations are expected to invest over nine hundred million dollars into the VoIP technology in 2005, an increase of more than two hundred million dollars compared to last year [2]. This indicates that VoIP continues to grow in popularity and remains a promising telecommunication technology that will gradually replace traditional PSTN phone systems.

Integration of VoIP capabilities into the existing MYSEA architecture is highly desirable for both economical and management reasons. Deploying VoIP greatly reduces the cost of making long distance calls from the MLS LAN to an external network such as the Internet. Management of telephone systems in the MLS LAN will also be simplified with the use of VoIP as rewiring phones for nomadic users is no longer needed. Thus, extending MYSEA to include VoIP is beneficial.

## B. PURPOSE OF STUDY

This study is a preliminary step in integrating VoIP capabilities into the existing MYSEA architecture. The objective is to determine the feasibility of this integration. The current MYSEA environment has a number of NAT components that may make the integration of VoIP into the MLS environment difficult if not impossible. A set of experiments were devised and conducted to test if VoIP works with the NAT components in MYSEA.

## C. ORGANIZATION OF PAPER

This paper is organized into five chapters and six appendices. A brief introduction is provided in Chapter I. Chapter II provides the background information related to this research study. A technical comparison between two popular VoIP protocols, H.323 and SIP, is presented in Chapter III. One of the two protocols will be selected for testing purposes. Chapter IV describes the test plan used to confirm the feasibility of integrating VoIP capabilities into MYSEA. The problems encountered and results from each test are also discussed in this chapter. The last chapter or Chapter V talks about future work and conclusion.

Six appendices are also included in this paper. Appendix A has surveys of hard and softphones. Appendices B through F contain descriptions, instructions, and results of tests described in Chapter IV.

# II. BACKGROUND

This chapter presents background information pertaining to this thesis study. It includes a high-level technical overview of the current VoIP technology. Finally, the MYSEA architecture is described in the last section.

## A. INTRODUCTION

The traditional phone system has been evolving since the first voice transmission in 1876 using a ring-down circuit. Today, phone systems no longer run on an analog network, instead they use a digital network known as the Public Switched Telephone Network (PSTN). The PSTN greatly reduces the amount of noise inflicted by analog voice amplification during transmission and provides number of services, such as call waiting, call forwarding, three-way calling, call blocking, etc. Despite the benefits obtained from the PSTN, there are drawbacks to the system that motivated the VoIP solution.

### 1. Advantages of VoIP

VoIP has many advantages over the traditional PSTN phone systems. These include the efficient use of bandwidth, reduction or possible elimination of long distance and phone charges, convergence of the voice and data networks, and advanced features. Some of them will be discussed further below.

#### *a. Efficient Use of Bandwidth*

Bandwidth is a key performance measure of a network. It defines how many bits can be transmitted every second, which means the more bandwidth available, the more data can be sent in a given period of time.

PSTN phone system requires a minimum of 64-kbps of dedicated circuit between the two calling devices. The circuit is reserved for the entire duration of the call regardless of whether or not any data is in transmission. Hence, bandwidth is unnecessarily wasted. On the other hand, VoIP uses IP networks that have the flexibility to allocate bandwidth as needed and reserve the unallocated bandwidth for other data. Thus, the use of network bandwidth in VoIP is more efficient.

***b. Reduction or Possibly Elimination of Long Distance and Phone Charges***

The cost of a long distance call generally depends on two factors: duration and destination of call. Charges can accumulate when an enterprise or individual frequently makes this type of call. VoIP service providers have monthly flat-rate plans that offer unlimited or fixed-number of minutes to make calls, including long distance calls. These plans are much more economical than the traditional charge-by-minute service. Thus will greatly reduce or possibly eliminate the phone and long distance charges for individuals and enterprises that make frequent long distance phone calls.

***c. Convergence of Voice and Data Networks***

Traditionally, a voice network only transmits voice and a data network only carries data. This is no longer true. Data makes up major traffic on voice networks. Unlike data networks, voice networks are not efficient in carrying data due to its inflexible bandwidth allocation and limited bandwidth. Therefore, most enterprises maintain both networks.

However, in many cases, management and maintenance of two different networks has proved to be cumbersome and costly for enterprises. Upgrading the voice network equipment such as the Public Branch Exchanges (PBX) telephones burdens enterprise budgets. If VoIP is deployed, the voice network will no longer be needed and will leave the enterprise with only the data network.

***d. Advanced Features***

VoIP provides all the services of a traditional phone system including speed-dial, call waiting, busy signaling, caller-ID, etc. In addition, VoIP can interoperate with services traditional phone systems lack such as video-conferencing, instant messaging, email, click-to-dial, and directory service.

**2. Disadvantages of VoIP**

While more home users and businesses are in the transition to use VoIP, the technology does have shortcomings. These will be discussed in detail below.

***a. Quality of Voice***

Voice data traveling across an IP network is highly susceptible to delay and loss due to routing and network latency. Voice in analog form has to be converted and compressed into digital packets before transmission over an IP network. A compression method that aggressively minimizes the size of voice packets will deteriorate the quality of voice. As a result, the quality of voice using VoIP may be worse than that obtained from PSTN due to delay, loss, and compression of the information.

#### *b. Security*

In many cases, maintaining separate voice and data networks can be difficult and costly. Convergence of both networks simplifies management and greatly reduces cost. However, convergence leads to security problems. Voice will be vulnerable to the same attacks as other data traveling across an IP network. Attacks include interception, modification, spoofing, man-in-the-middle attacks and denial of service.

#### *c. Availability*

Making a VoIP call requires a connection to an IP network through properly configured network devices that are dependent on a stable electrical power supply. Power outage and connection problems will prevent an individual from making or receiving a VoIP call.

#### *d. 911*

Currently, none of the VoIP protocols provide information regarding the caller's physical location to the emergency operator. When the caller dials 911, there is no guarantee the call will be routed to the nearest 911 police station. At the same time, the 911 operator has no way of identifying the location of the caller.

## B. VOIP OVERVIEW

VoIP uses IP or a packet-switched network as the data transmission vehicle. A VoIP system digitizes voice using an audio codec, divides the digitized voice into packets, and sends the packets over an IP network to the destination. All packets are routed without a guarantee that they will travel the same path. Unlike a PSTN call, no dedicated circuit is ever created for a VoIP call.

The exact process required to set up a VoIP call is dependent on the VoIP protocol. Two types of protocols are necessary to complete a VoIP call: signaling and media transport. A signaling protocol has the responsibility of establishing a session between the call participants. A media transport protocol specifies the rules and formats of the actual voice packets. Currently, the Real Time Protocol is commonly used as the media transport protocol in VoIP. However, there is a wider variety of signaling protocols. VoIP protocols will be further discussed later in this chapter.

For PSTN systems, a phone number consisting of digits is used to locate a phone. A phone number in VoIP can be a regular PSTN phone number, an address, or an alias. The "phone number" ultimately is translated to a 32-bit or 128-bit IP address depending on whether IPv4 or IPv6 is used. Every VoIP signaling protocol must provide address resolution capability.

There are four general VoIP communication modes. They are Phone-to-Phone, Phone-to-PC, PC-to-Phone, and PC-to-PC.[1] Voice transmission is carried by both PSTN and IP networks under the first three modes. A VoIP service provider that interconnects the PSTN and VoIP networks is needed for the first three modes when a call originates from a PSTN network and arrives at a VoIP network or vice versa. Voice travels exclusively across the IP network in the fourth mode.

### 1. VoIP Phone Overview

VoIP phones generally fall into three categories: PSTN phones, hardphones, and softphones. A PSTN phone is the type of phone almost every household has and is connected to a phone jack using a telephone cable. Technically, a PSTN phone in itself is not a VoIP device, but it can be used to make VoIP calls with the use of a phone adapter known as the Analog Telephone Adapter (ATA) that converts voice from analog to digital form.

A hardphone, also commonly known as an IP phone, can look identical to a PSTN phone. It is an independent device that understands VoIP protocols. Unlike a PSTN phone, a hardphone does not require an external device such as an ATA to make VoIP calls. All it needs is an Internet connection. Table 1 lists various types of hardphones and their corresponding descriptions from [3].

[1] Phone refers to a traditional phone and PC refers to a personal computer.

| Hardphone Type | Description/Characteristics |
|---|---|
| Ethernet | Has an Ethernet port<br>Connects directly to the IP network |
| Cordless | Has IP interface on base stations |
| WLAN or WiFi | Has built-in WiFi transceivers<br>Connects to a WiFi base station |
| WLAN/WiFi and GSM | Same as WLAN/WiFi phones but can also transfer calls to GSM network |
| Voice and Video | Supports for both voice and video |

Table 1. Hardphones

A softphone is a VoIP phone in the form of software. A softphone runs on a computing device such as a desktop, laptop, or PDA, and is typically Operating System dependent. This type of phone needs audio support such as speakers and microphones for communication purposes. Appendix A has a survey of hard and softphones.

## C. VOIP SERVICES

The VoIP technology is made up of four distinct services: signaling, encoding, transport, and gateway control [4]. A signaling VoIP protocol establishes and manages a connection between the endpoints when a call is made. Signaling protocols are discussed in Section D. When the conversation takes place, voice has to be encoded before it is transmitted over the IP network. The encoded voice packets will then be transported via the IP network to the destination. A gateway may be needed to convert voice into another format suitable for the receiving network. For example, a gateway will convert voice from digital PSTN to digital IP form when voice packets come into an IP network from a PSTN network.

### 1. Encoding

Voice, in its native form, cannot be transmitted over an IP network. A voice codec is used to convert voice from analog to digital data or digital to analog data, compress voice to optimize bandwidth usage, and packetize the voice data in preparation for transmission. A codec determines bandwidth usage and quality of voice. Higher quality voice transmission usually requires more bandwidth. The tradeoff between the two factors is critical when deciding what codec to use in VoIP applications. Various codecs exist to support VoIP. However, the three most commonly used codecs are G.711, G.723.1, and G.729A. More information on the above codec specifications can be found on the ITU-T website.

### 2. Transport

Media transport protocols such as Real Time Protocol (RTP) deliver the encoded voice packets over an IP network. RTP is a standard developed by Internet Engineering Task Force (IETF) to transport real-time audio and video data. RTP does not guarantee reliable transmission of packets. It usually runs on top of UDP due to the delay-intolerance of voice conversation and uses a dynamically assigned UDP port in the range 1024 – 65535.

The RTP Control Protocol (RTCP) is the control counterpart of RTP. RTCP, also developed by IETF, is not required to be used with RTP. However, RTCP can be used to monitor transmission performance. End users in a VoIP session can send transmission statistics in RTCP packets upon receiving packets. This information is useful to determine network and delivery performances and keep track of retransmission needs. Similar to RTP, RTCP uses a dynamically assigned UDP port.

### 3. Gateway Control

A gateway connects the PSTN and VoIP networks. Voice packets arriving at its IP interface will be converted by the gateway from a format understandable by IP to one that is understandable by PSTN and vice versa. A gateway is necessary for communications between Phone-to-Phone, Phone-to-PC and PC-to-Phone.

## D. VOIP PROTOCOLS

Table 2 lists some well-known VoIP signaling protocols. A VoIP signaling protocol defines the formats of VoIP messages and rules for message exchange necessary to establish a VoIP call.  The signaling protocol is responsible for setting up a VoIP call, which includes tasks like locating users and negotiating session parameters between the two end devices.  A media gateway control protocol control communication amongst the gateways in an IP networks.

| **Protocol** | **Organization** | **Type** |
|---|---|---|
| H.323 | ITU-T | Signaling |
| Session Initiation Protocol (SIP) | IETF | Signaling |
| MGCP | ITU-T | Signaling |
| Megaco/H.248 | ITU-T/IETF | Signaling |

Table 2. VoIP Protocols

### 1. H.323

H.323 is an open standard developed by ITU-T in 1996.  H.323 was originally designed for multimedia conferencing and was later extended to support VoIP.  H.323 is a suite of protocols that provide services such as end-to-end multipoint conferencing, audio and video codecs, management and accounting, and security.  Since 1996, the protocol has undergone a series of changes, with the latest version (H.323v4) providing many enhanced feature and services.  Refer to Chapter III for more details.

### 2. Session Initiation Protocol

The Session Initiation Protocol is developed by IETF. It is an application protocol designed to establish a two-way communication session.  SIP is gaining popularity in the VoIP market despite the fact that it is a fairly young protocol developed in 1998.  SIP is generally more scalable, simple, and extensible than H.323.  Some believe that SIP will eventually become the official VoIP signaling protocol standard.  Refer to Chapter III for more details.

### 3. MGCP

Media Gateway Control Protocol (MGCP), developed by IETF, controls communication among VoIP gateways in an IP network.  Two components exist in the MGCP architecture: call agents, also known as media gateway controller, and gateways.

MGCP is a master-slave protocol in which the master call agent sends signaling, control, and processing commands to the gateway. The gateway acts as a slave and executes the commands sent by the call agent. MGCP does not replace SIP or H.323. Rather, the protocol is used to manage signaling and control activities for VoIP network gateways such as H.323, SIP, and SS7 signaling.

#### 4. Megaco/H.248

Megaco/H.248, developed jointly by ITU-T and IETF, has the same architecture as MGCP. However, Megaco/H.248 offers several advantages over MGCP such as the support of multimedia and multipoint conferencing enhanced services, improved syntax for more efficient semantic message processing, TCP and UDP transport options, support for both text and binary encoding, and formalized extension process for enhanced functionality [5].

### E. VOIP CHALLENGES

#### 1. Quality of Service

Quality of Service (QoS) is not a major concern in PSTN systems because a fixed amount of bandwidth is dedicated to a call and transmission of voice follows the same circuit for the duration of the call. On the other hand, when a VoIP call is made, digitized voice will be transmitted using an IP network that has no fixed-bandwidth allocation mechanism. Thus, it is subject to jitter, latency, and packet loss problems, of which VoIP is intolerant.

There are many good reasons to deploy VoIP, however, it makes no sense to use VoIP if the quality of a VoIP call is lower than a traditional PSTN phone call. QoS must be addressed to an acceptable level such that end users can carry on a smooth conversation with minimal interruptions

#### 2. Security

Security is another important aspect of VoIP. Convergence of voice and data networks means that both voice and data have to be protected. Eavesdropping a pure PSTN phone conversation is more difficult than a VoIP conversation, because interception of a regular PSTN call requires physical access to the phone lines or compromise of the corporate PBX. On the other hand, a VoIP conversation can be intercepted by an adversary anywhere along the path where the digitized voice packets travel. The security problem is intensified when sensitive personal information such as social security and credit card numbers are given out over a VoIP call.

The transmission of voice over an IP network is subject to security risks. It is important to ensure the confidentiality, integrity, and authenticity of a VoIP conversation and availability of resources when a VoIP call needs to be placed. In summary, the conversation and the VoIP network resources are the two main assets that require protection. Security mechanisms must be used to prevent both internal and external eavesdropping, spoofing, replay, and denial of service attacks. Many security mechanisms such as encryption, firewalls, and Network Address Translation (NAT) exist to address these threats. However, almost every one of them raises problems or affects the overall performance of a VoIP in some ways.

Encryption effectively protects the confidentiality and possibly authenticity of VoIP packets by making it impossible for people other than the intended recipient to read the packets. However, encrypting every packet, at the sending end and decrypting it at intermediate nodes and at the receiving end could cause an immense amount of delay, thus lowering the QoS. The size of an encrypted packet is often bigger than the plaintext packet, thus requiring more bandwidth and leading to a possibility of packet drop. This is a typical tradeoff between security and performance.

A firewall is often the first layer of defense in securing a network. It sits between the internal and external network, inspects every incoming and outgoing packet, and blocks those packets that it thinks is malicious. Firewalls usually inspect packets by examining certain fields, such as IP addresses, ports, and protocol type, in the packet headers. However, some VoIP protocols such as H.323 use dynamic ports to send or receive messages. A stateless firewall that only looks at header information to determine packet admissibility might drop some of the messages. To ensure the admission of those messages, the stateless firewall would have to open many ports and leave itself in a vulnerable statue. A stateful firewall, one that stores information about a session along with previous packet transactions, can inspect a packet's application layer data and can manage the dynamic port problem. However, a stateful firewall introduces latency due to the extensive packet inspection. As a result, network performance may not be optimal.

Network Address Translation (NAT) is a method of mapping a group of private IP addresses to a group of public network IP addresses. NAT conserves IP addresses by sharing a limited number of public IP addresses among many internal hosts. A public IP address can be mapped to multiple internal hosts. Furthermore, NAT hides internal IP addresses from the outside world so that adversaries outside the network cannot directly attack internal hosts. VoIP signaling and media transport protocols often use different ports. Furthermore, RTP and RTCP use random ports to exchange data, thus complicating the NAT process. When NAT receives the actual digitized voice packets, it has no knowledge of where to send it. As a result, the packets may get dropped.

Similar to the use of encryption, the use of firewalls and NAT will also affect QoS because every packet coming in will have to be processed to determine admission. The uses of encryption, firewall, and NAT are just three examples of defense mechanisms against possible VoIP threats. However, these defenses often cause problems in the operation of VoIP processing and performance.

**F. NAT**

Network Address Translation (NAT) is primarily used for two purposes: public IP address conservation and security. The advantage of using NAT is that any number of internal hosts using un-routable private IP addresses can be connected to another network, such as the Internet, using a small number of public IP addresses. When an internal host wants to communicate with an external host, NAT maps the private IP address of the internal host to one of its un-used public IP addresses. The process of NAT rewriting the private IP address with a public IP address in the source IP address field of all packets initiated from a local host is known as Source Network Address Translation (SNAT). Destination Network Address Translation (DNAT) refers to the process of NAT rewriting the public IP address with a private IP address in the destination IP address field of all packets received at its public interface. Note that a NAT device must have routing capabilities to route packets in and out of two different networks. SNAT and DNAT allow an internal host to communicate with an external host without ever exposing its internal IP address. This mechanism of hiding internal IP address provides another layer of protection for system security.

### 1. netfilter/iptables

Any Linux system can be turned into a NAT device using *netfilter* and *iptables*. These two open source kernel modules are included in most Linux distributions that provide networking functions including NAT, routing, and firewall. Furthermore, *iptables* has a powerful connection tracking mechanism that allows it to associate packets with their corresponding sessions. This mechanism is essential in stateful firewalls. More information about *netfilter* and *iptables* can be found at [6]. The next section describes a component in the MYSEA architecture that uses *netfilter* to do NAT.

*netfilter/iptables* consults the *nat* table to determine if the IP address and/or port of a packet needs to be rewritten and how those fields should be rewritten. The *nat* table contains three chains of rules. Two of them are PREROUTING and POSTROUTING. The PREROUTING chain is referenced when a DNAT decision has to be made while the POSTROUTING chain is used to make SNAT decisions. The following are two sample NAT rules:

```
iptables -t nat -A POSTROUTING -o eth0 -j SNAT --to 192.168.0.1

iptables -t nat -A PREROUTING -ieth0 -j DNAT --to 192.168.1.10
```

The first rule instructs the NAT device to modify the source IP address of all outgoing packets to 192.168.0.1 after the packet is processed by the routing logic. The second rule tells the NAT device to rewrite the destination IP address of all incoming packets to 192.168.1.10 before routing logic takes place.

### 2. SIP with NAT

SIP-based VoIP calls rely on two protocols: SDP for negotiating of session parameters and RTP for transporting of voice data. Two endpoints setup a VoIP connection with exchange of the INVITE and 200 OK messages. Each message has SDP information embedded in the payload specifying the IP address the endpoint expects to receive RTP voice packets. Before sending out either the INVITE or 200 OK packet during the call setup, an endpoint located behind a NAT device writes its private IP address as part of the SDP data. The NAT device, a layer three device, performs SNAT to the message it receives from the endpoint and then sends the packet to the destination address. Since SDP is a layer five protocol, the NAT device is unable to examine and rewrite the private IP address embedded in the SDP section of the message. When the other endpoint wants to send RTP packets, it will send it to the private IP address indicated in the SDP portion of the received message and hence the RTP packets will get dropped.

## G. MYSEA OVERVIEW

The Monterey Security Architecture (MYSEA) is a multi-level distributed operating environment designed to allow secure access to information at different classifications. MYSEA consists of a combination of commercial-off-the-shelf (COTS) and high assurance components. The COTS components are used to perform common user tasks whereas the high assurance applications are used to enforce security policies. Such a design is especially advantageous to organizations such as the DoD that invest heavily on COTS products but has a need to manage information with different sensitivities [7].

Figure 1 is an illustration of the MYSEA network architecture. Communication between an untrusted client on the MLS LAN and another client is mediated by a MLS server running XTS-400. The MLS server enforces security policies and provides a number of security-related services. Each MLS LAN client communicates with the MYSEA server via an inline Trusted Path Extension (TPE). The TPE establishes an encrypted trusted path and negotiates session level information with the MLS server on behalf of the client. Each TPE is also a NAT device that hides the internal IP address of the clients. Currently every TPE on the MYSEA testbed has a unique private IP address whereas every client uses the same private IP addresses. Refer to [8] for more detail discussion of MYSEA.

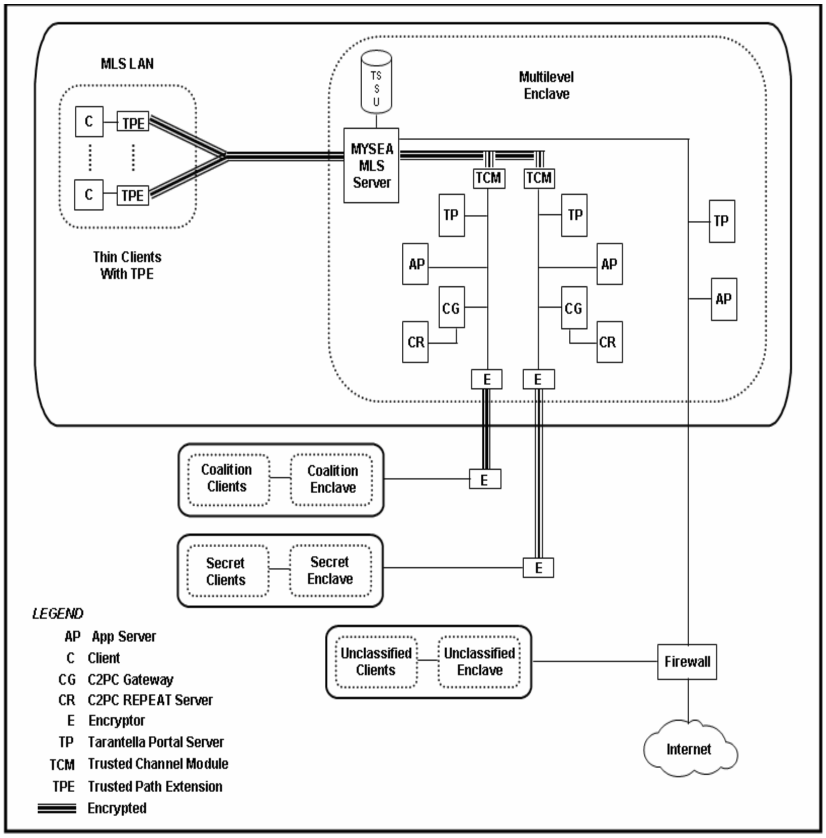


Figure 1. MYSEA Network Architecture [From Ref. 8]

## H. SUMMARY

This chapter presents background information that is relevant to this project. To prepare for testing, a VoIP protocol must be selected as the test protocol. The next chapter compares two popular VoIP protocols, namely H.323 and SIP, and ends with a protocol selection for testing.

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# III. TECHNICAL COMPARISON BETWEEN H.323 AND SIP

The purpose of this chapter is to compare two dominant VoIP signaling protocols: H.323 and Session Initiation Protocol (SIP). At the end of the study, one of the two protocols will be selected for testing. Selection is based on the protocols' simplicity and flexibility, extensibility, scalability, and security. This chapter consists of two sections. The first section describes the protocols whereas the second section is a summary of two SIP and H.323 comparisons presented in [9] and [10].

## A. H.323 AND SIP OVERVIEW

H.323 and SIP are the front-runners in the VoIP industry. H.323 came about in 1996, two years before the birth of SIP in July 1998. H.323 is still widely used in enterprises and continues to be improved, while SIP is gaining popularity and undergoes more development. The following subsections present a high level overview of the two protocols.

### 1. Background

H.323v1 was developed by ITU-U as a "standard for real-time video-conferencing over non-guaranteed quality of service LANs" [9]. The standard has undergone several revisions. The latest version, H.323v4, defines basic call control and signaling for multimedia applications. The protocol is specifically designed to support multimedia and voice applications. It was extended to support VoIP. The ITU-U initially concentrated on developing multimedia functionalities, supplementary services, and internetworking capabilities into H.323. As those capabilities become standardized, ITU-U works to address the protocol's security, QoS, and mobility issues.

SIP, IETF's standard for establishing VoIP connections, was standardized in 1999 and revised in 2002. SIP is an application layer protocol designed to setup, modify, and tear down generic sessions. Other fundamental services it provides include user location, session invitation and session negotiation. As with all IEFT protocols, SIP was not developed to support a particular type of application. Rather, SIP is designed to work with any application that may need its services. The IEFT initial focus was on standardizing the protocol to support session initiation. Currently, a large amount of effort is placed on defining specific applications, such as internetworking with legacy networks and providing supplementary services [9].

### 2. Architecture

Both H.323 and SIP have both peer-to-peer and client-server architectures. H.323 specifies a complete framework that defines the protocols and the message flows for multimedia communications. The standard covers all phases of a VoIP call including set-up, call control, and media transport. Other issues critical to the quality of a VoIP call such as QoS, security, and mobility are also addressed in the standard.

H.323 is actually a suite of protocols that can be broken down into six classes: call control and signaling, audio processing, video processing, data conferencing, media transportation, security, and supplementary services. Information regarding the different protocols used in H.323 can be found in [11].

Figure 2 depicts the H.323 protocol stack. The lighter colored blocks represent optional components whereas the darker colored blocks represent mandatory components necessary to complete a VoIP call. It is important to note that both H.323 and SIP rely on the support of several common protocols such as TCP/UDP, IP, RTP and RTCP and audio processing services. In summary, only H.245, H.225.0/Q.931, and H.225.0/RAS are essential to achieve the signaling part H.323 VoIP call.

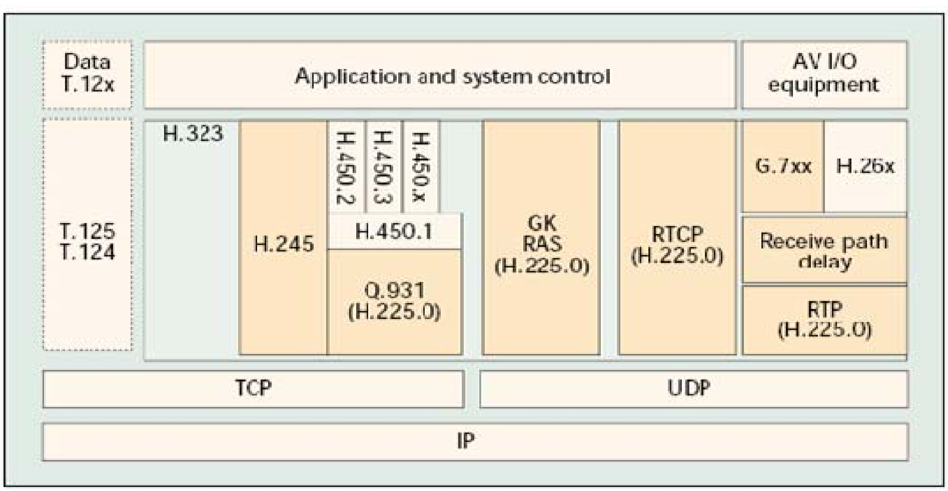


Figure 2. H.323 Protocol Stack [From Ref. 9]

SIP by itself only defines setup and teardown of sessions. Advance signaling features are specified as SIP extensions. Furthermore, QoS and mobility are not addressed by SIP but can be supported by other protocols. SIP depends on the Session Description Protocol (SDP) to describe parameters for multimedia session between two endpoints. SDP is a text-based media-description format that is carried in SIP messages. Figure 3 depicts the SIP protocol stack. Again, the darker component is essential to the signaling part a SIP VoIP call.

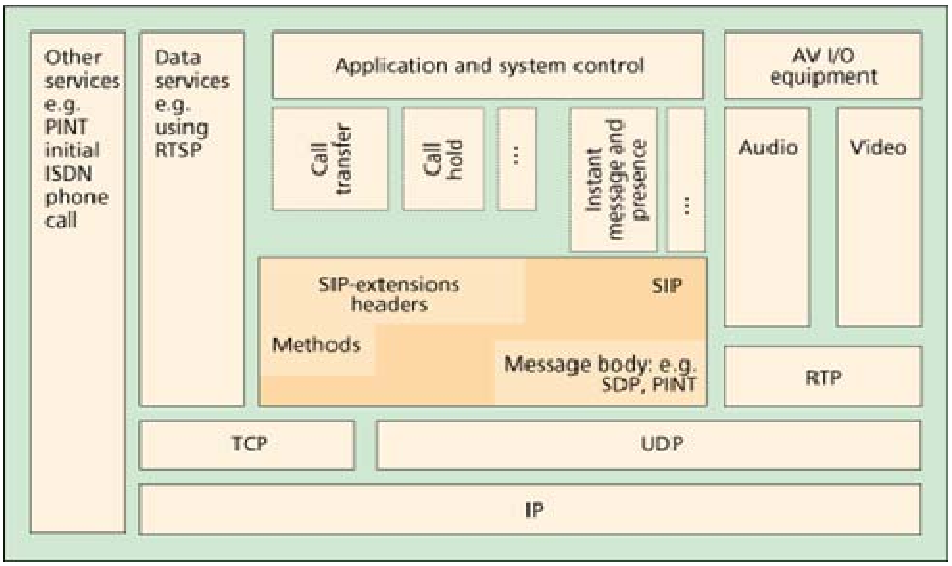


Figure 3. SIP Protocol Stack [From Ref. 9]

### 3. Components

H.323 and SIP divide functions to components in a similar fashion. Basic call function controls are assigned to the terminals whereas services requiring network support are assigned to network servers. Table 3 lists SIP and H.323 network components by types.

| | Client | Servers in the network | | |
|---|---|---|---|---|
| SIP (IETF) | Terminal | Proxy server, registrar | Conference server[1] | Gateway[1,2] |
| H.323 (ITU-T) | Terminal | Gatekeeper | MCU | Gateway |

[1] Not standardized up to now. [2] Addressed in several drafts, e.g., SIP-T
MCU: Multipoint control unit. Gateway: maintains transition to traditional telephony.

Table 3. SIP and H.323 components [From Ref. 9]

An H.323 network consists of terminals and a gateway. A gatekeeper, Multipoint Control Unit (MCU), and Back End Service (BES) may also be deployed as part of the network. A terminal is any VoIP-enabled device. The gateway provides translation services for terminals that use different communication protocols including non-VoIP protocols such as PSTN. A gatekeeper performs address translation, controls accesses of terminals, manages bandwidth and makes routing decisions. Although a gatekeeper is optional, it is often an important component in an H.323 network because of the services it provides. A H.323 network can have a MCU that facilitates communication among multiple endpoints. A Back End Service usually exists to support the gatekeeper by maintaining information about endpoints such as the endpoints' permissions, configurations, and services [5].

A SIP network has endpoints, a proxy server or redirect server, location server, and a registrar. The registrar authenticates users and stores location information from users. A proxy server, which can be integrated with the registrar, resolves addresses and forwards messages on behalf of the endpoint to another proxy server or the destination endpoint during call setup and teardown. The redirect server performs tasks similar to those of a proxy sever but instead of forwarding the message, the redirect server sends the resolved address back to the endpoint and lets the endpoint communicate directly with the other endpoint. The location server supports the registrar by maintaining location information of endpoints [5].

### 4. Call Setup

Call setup refers to the actions necessary to establish a connection between two endpoints. This process must be completed before the endpoints can exchange actual voice data. The following subsections describe simple H.323 and SIP call setups. The scenarios assume that Alice initiates a non-local call to Bob.

#### *a. H.323*

H.225.0/RAS, Q.931 in H.225.0, and H.245 are necessary to establish an H.323 VoIP connection. These protocols provide functions necessary for call registration, call setup, and capability exchange. Figure 4 illustrates a simple H.323 call setup.

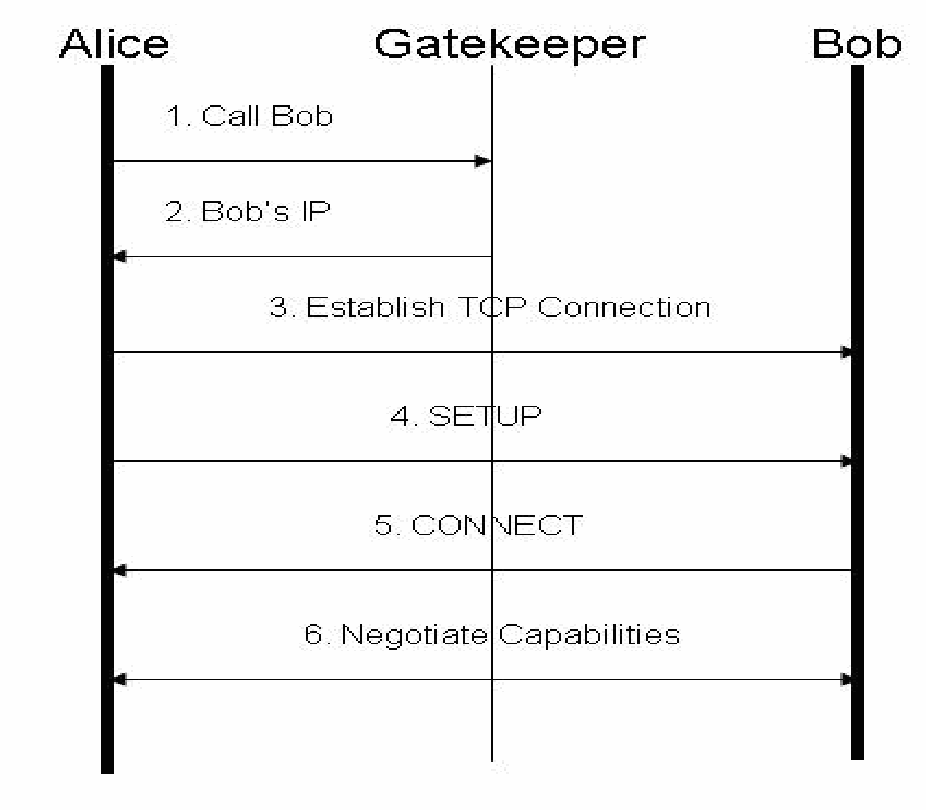


Figure 4. Simple H.323 Call Setup

When Alice dials Bob's phone number, (Step 1) Alice's terminal sends a Registration Admission Request to the gatekeeper using H.225.0/RAS. The gatekeeper registers Alice into the system, admits and grants resources to Alice and finds Bob's IP address. Next, (Step 2) the gatekeeper sends the IP address to Alice. Alice then establishes a TCP connection with Bob at the IP address she received (Step 3). Alice sends a SETUP message to Bob using Q.931/H.255.0, an ISDN-connection control protocol (Step 4). Bob sends Alice back a CONNECT (Step 5) message to Alice using the same protocol indicating acceptance to the connection. Finally, Alice and Bob negotiate terminal capabilities using H.245 (Step 6). Then H.245 will open logical channels for both endpoints to start the conversation.

***b. SIP***

Figure 5 illustrates a simple SIP call setup. In this example, an integration of the registrar into the proxy server is assumed. Before Alice calls Bob, Alice's terminal must register itself with the registrar (Step 1). This step is similar to the first step in the

H.323 simple call setup. After the registration is completed, Alice may call Bob by sending the proxy server an INVITE Bob message (Step 2). The proxy server looks up Bob's IP address and forwards the invitation to Bob (Step 3). An OK response will be received by the proxy server from Bob indicating acceptance to the call (Step 4) and the response will in turn be forwarded to Alice (Step 5). Throughout this process, session parameters and terminal capabilities are transparently exchanged inside the INVITE and OK messages from both parties using SDP or some other methods. From now on, the two parties may communicate in a peer-to-peer fashion.

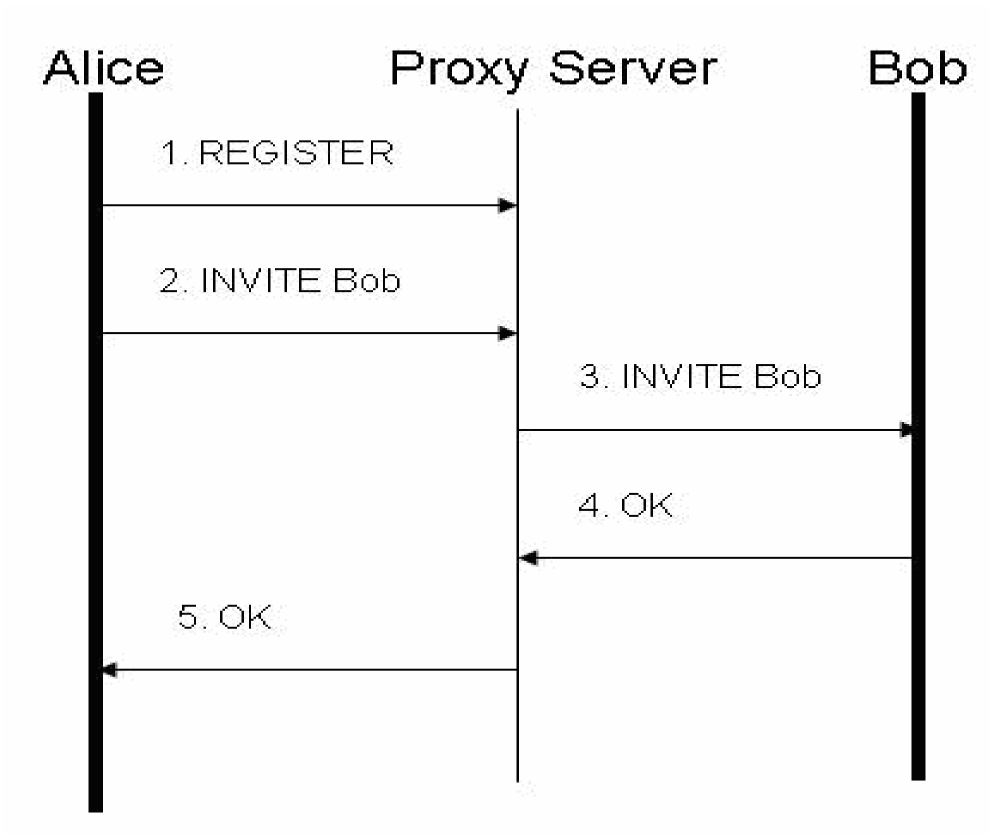


Figure 5. Simple SIP Call Setup

## 5. Services

H.323 and SIP both provide basic call controls as well as advanced features. Table 4 lists the features common to both protocols.

| Feature | H.323 | SIP |
|---|---|---|
| Call Setup | Yes | Yes |
| Call Teardown | Yes | Yes |
| Call Waiting | Yes | Yes |
| Call Hold | Yes | Yes |
| Call Transfer | Yes | Yes |
| Call Forwarding | Yes | Yes |
| Call Return | Yes | Yes |
| Call Identification | Yes | Yes |
| Call Park | Yes | Yes |
| Capabilities Exchange | Yes | Yes |

Table 4. Basic Call Control Features

## B. H.323 AND SIP COMPARISON

H.323 and SIP are different in many ways despite the fact that they provide similar call control services and are widely used in VoIP applications. One of their fundamental differences lies in their original intents and designs. H.323 was designed with a focus on multimedia and voice communications whereas the SIP design focused on providing only session initiation services. H.323 uses a top-down approach to specify a complete framework for providing multimedia and voice services. It is telecommunication-oriented as it uses existing multimedia protocols in the ITU H-series to support provide various services. SIP, on the other hand, uses a bottom-up approach. Its modular design allows it to work with a wide range of applications. SIP takes on an Internet-oriented design by adopting a number of features from HTTP and SMTP, two of the most successful Internet protocols [9, 10].

Their implementation approaches lead to differences in the simplicity and flexibility, extensibility, scalability, and security of the protocols. The following subsections examine these differences.

### 1. Simplicity and Flexibility

Simplicity and flexibility are two important measurements to determine the quality of the protocols' design. This subsection compares H.323 and SIP based on these two aspects. Protocol specification, message encoding, and protocol interactions will be closely examined.

#### *a. Protocol Specification*

SIP is simpler in nature than H.323. According to [9], a SIP-based VoIP implementation can be done with four headers (To, From, Call-ID, and Cseq) and three types of requests (INVITE, ACK, and BYE).

H.323, on the other hand, consists of numerous protocols such as H.225.0 for call signaling, H.245 for call control, H.332 for conferences, H.450.1 to H.450.9 for supplementary services, H.235 for security and encryption, etc. Many services require a number of H.323 protocols to interact with each other. This further intensifies H.323's complexity problem [10].

#### *b. Message Encoding*

H.323 messages are encoded by the ASN.1, an international standard used to specify data used in communication protocols. Since ASN.1 messages exist in binary form, a special tool is needed to parse the messages. SIP adopts the HTTP tradition by using text-based messages. Hence, SIP messages are generally easy to parse, generate, and debug [10].

#### *c. Protocol Interactions*

H.323 is complicated because of the many of protocols it encompasses. A number of protocols are often required for a single service in H.323. For example, connection establishment, as illustrated in Figure 3, requires Q.931/H.255.0, H.225.0/RAS, and H.245. On the other hand, SIP uses a single INVITE request to establish the connection even though it depends on SDP to negotiate session parameters. H.323 also allows those three protocols to be used in different orders to establish a connection. Thus, network devices such as firewalls, endpoints, gatekeepers, and gateways must support and understand all three connection establishment methods [12].

#### *d. Applications*

SIP is designed to create, manage, and tear down generic sessions. As a result, SIP has the flexibility to work with a wide range of applications. Voice and multimedia are just two applications of SIP. Others include voice-enriched e-commerce, web page click-to-dial, Instant Message, and IP Centrex services. H.323 was designed to focus on a specific type of communication, namely voice and multimedia conferencing. Thus, its applications are not as wide as SIP's. ITU-U is currently working toward providing non-VoIP services in H.323 [9].

### 2. Extensibility

Extensibility defines how easy it is to add new features to the existing protocols. This aspect of the protocols will be evaluated based on extensible mechanisms, backward-compatibility, and interoperability.

#### *a. Extensible Mechanisms*

Both H.323 and SIP have certain extensible mechanisms. H.323 has nonstandardParm fields in its ASN.1 messages. Each nonstandardParm field is identified by a unique vendor code and the information contained in this field is only meaningful to the specific vendor. These fields allow vendors to add extensions [10].

SIP adopts the HTTP's use of hierarchical numeric warning codes. Its warning codes are represented by three digit numbers where the first digits identify which of the six categories the codes belong to.[2] The list of warning codes can be easily extended under this hierarchical structure system. SIP features are also extensible. New SIP features can be officially added by registering the names of the features with the Internet Assigned Numbers Authority (IANA). New features will not cause confusion on the server side because SIP specifies that a server shall ignore the request of a feature in a SIP header if the server does not understand or support the feature.

---

[2] SIP warning codes are divided into six categories: provisional, redirection, request and server failure, busy everywhere responses, successful and bad requests.

*b. Backward-Compatibility*

H.323 supports full backward compatibility among different implementation versions. In other words, obsolete features have to be carried over from one version to the next. Hence, the code can become complicated. However, compatibility between two different versions of H.323 implementations is guaranteed. On the other hand, a new version of the SIP implementation does not need to support obsolete features from past versions. When an obsolete feature is requested from a server, the server, by default, ignores the request, informs the requestor of the unsupported feature and lets the requestor decide what to do. This way, a SIP implementation is typically cleaner while still maintaining some compatibility. Unlike H.323, different versions of SIP implementation may suffer from compatibility problems [10].

*c. Interoperability*

H.323 has higher interoperability than SIP. H.323 has well-defined implementation guidelines available to help improve interoperability among different H.323 vendors. Also, the H.32x family specifies standards to guarantee interoperability among circuit-switched networks such as ISDN, B-ISDN and GSTN. SIP is also highly interoperable with other protocols due to its flexibility and modular design. For example, SIP can be used in conjunction with H.323 where SIP provides the location service and H.323 performs the rest of the communication services. However, SIP is loosely defined and open to various interpretations. This may lead to potential interoperability issues. There is a growing effort focused on addressing interoperability issues in SIP [9].

**3. Scalability**

The scalability of the two protocols, or the ability to support small or large volume of data or users, is compared in below. The design, server components, and conference mechanisms of the protocols will be evaluated for scalability.

*a. Protocol Design*

H.323 was originally designed for local area networks. Addressing in a wide area network (WAN) and user location were not initial concerns for H.323. As networks employing H.323 have grown in size, H.323 has been augmented to address these issues. However, H.323 still has a scalability problem because its loop detection algorithm using path values does not work well. SIP, on the other hand, was designed to support WAN addressing and user location. It uses a loop detection mechanism that is similar to the one employed by Border Gateway Protocol (BGP). Thus unlike H.323, the SIP loop detection algorithm scales well [10].

#### *b. Servers*

Scalability generally decreases if servers have to maintain state for all calls. Servers used in both protocols can be stateful or stateless. Endpoints in both protocols need to keep states in stateless call implementations. Endpoints as well as servers need to maintain states in stateful call implementation. The drawback of maintaining states is the large amount of memory and processing that is required. Most current H.323 gatekeeper implementations are designed to be call stateful whereas most SIP proxy implementations are designed to be call-stateless [12]. Therefore, SIP scales better in large networks.

#### *c. Conferencing*

H.323 relies on the Multipoint Control Unit (MCU) to manage signaling in multiparty conferences regardless of the number of participants. However, MCU can be a bottleneck in large conferences. SIP is more scalable with regard to conferencing because it employs a distributed control scheme [12].

### 4. Security

This section compares the security provided by H.323 and SIP based on [5]. More specifically, it examines the protection of authenticity, confidentiality, and integrity for both signaling and media data provided by H.323 and SIP.

#### *a. H.323 Security - H.235*

H.323's relies on H.235 to specify security standards. However, H.235 "does not mandate particular [security] features" [13]. To address interoperability among different H.235 vendors, H.235 defines security profiles corresponding different security levels. Table 5 summaries the different H.235 security profiles or annexes described in [5].[3]

[3] H.235 Annex A (H.235 ASN.1), Annex B (H.323 Specific Topics), and Annex C (H.334 Specific Topics) are not listed in Table 4.

#### *b. SIP Security*

The SIP standard specifies several security features including HTTP Digest Authentication and S/MIME. The protocol does recommend other best security practices to address authentication, confidentiality, and integrity for both signaling and media data. Table 6 lists the existing SIP security features presented in [5].

#### *c. Security Comparison*

H.235, the security protocol for H.323, and SIP both have recommendations to protect the authenticity, confidentiality, and integrity for both signaling and media data. At the same time, ITU-T and IETF are making serious efforts to address security problems by continuously devising new security recommendations. Even though H.235 has effective security measures, H.323 does not mandate vendors to implement any of the H.235 security measures. According to an online website, not many H.323 products have support for H.235 and those that have "only use H.235 (baseline security) for the communication between gatekeeper and gateway and not for communication with the endpoint" [13]. SIP, on the contrary, has security mechanisms specified in the protocol implementation and is inherently more secure than H.323.

| Annex | Name | Description | Protection |
|---|---|---|---|
| **D** | Baseline Security Profile | Protect messages with shared secrets and hashed values. | Authentication, Integrity |
| **E** | Signatures Security Profile | Protect messages with certificates and digital signatures. | Authentication, Integrity |
| | Voice Encryption Option | Encrypt voice streams. Can add on top of Annex D or E. | Confidentiality |
| **F** | Hybrid Security Profile | Establish shared secret using certificates and digital signatures. Use shared secret to protect messages. | Authentication, Integrity |
| **G** (draft) | SRTP & MIKEY Usage | Use MIKEY to manage keys in SRTP. | Authentication, Confidentiality, Replay |
| **H** (draft) | RAS Key Management | Protect keys established during gatekeeper discovery with PINs or passwords. | Confidentiality |
| **I** | H.235 Annex D for Direct Routed Scenarios | Enhances security provided by Annex D and F with option to use in direct routed calls. | Authentication, Integrity |
| **J** | N/A | Describes security for simple endpoint types. | Authentication, Integrity |

Table 5. H.235 Security Profiles

| Name | Description | Protection |
|---|---|---|
| **HTTP Digest** | Challenge and response scheme to protect signaling information. Endpoint sends to server an MD5 hash of the username, password, and nonce provided by server. | Authentication |
| **S/MIME** | Use S/MIME to provide protection to signaling information. | Authentication, Confidentiality, Integrity |
| **RTP and SRTP** | Encrypt RTP data defined in RFC 1889 or use SRTP to protect media data. | Confidentiality |
| **SDP** | Uses SDP to carry session keys for media data. | Confidentiality |
| **TLS** | SIP requires proxy and redirect servers and registrars and recommends endpoints to use TLS to protect signaling data. | Confidentiality, Integrity, Replay |
| **IPsec** | Use IPsec to provide security for signaling data at the network layer | Authentication, Confidentiality, Integrity |
| **AIB** (draft) | Use a digitally-signed message or message fragment to provide authentication | Authentication |
| **Authenticated Identity Management** (draft) | Recommends on how to correctly verify end users | Authentication |
| **S/MIME AES Requirement** (draft) | Use AES instead of DES or 3DES as a minimum requirement for encryption implementations of S/MIME in SIP | Confidentiality |
| **Security Mechanism Agreement** (draft) | Provide a method to negotiate which security mechanism to use between two endpoints | N/A |
| **End-to-Middle, Middle-to-Middle, Middle-to-End Security** (draft) | Describes security in different modes of communication (i.e. end-to-end, middle-to-middle, middle-to-end) | N/A |

Table 6. SIP Security Features

### 5. Conclusion

Both protocols have strengths as well as weaknesses. SIP is more flexible and light-weight but less well-defined compared to H.323. H.323 has a detailed specification and offers higher interoperability but supports fewer applications. Nevertheless, H.323 and SIP are widely used in VoIP applications and both are undergoing more development to address their weaknesses. Neither of the two will become obsolete. Thus, interoperability between them will become necessary.

Nevertheless, SIP is simpler, more flexible, extensible, scalable, and can be more secure than H.323 based on the above comparison. SIP, the younger protocol of the two, is showing the potential to become a highly successful Internet protocol. Products based on SIP are becoming increasingly available for these reasons. For example, Microsoft has shifted H.323-based NetMeeting implementation to a SIP-based implementation in Windows XP. Furthermore, Microsoft also incorporates a SIP-like protocol stack in its .Net framework that can be used on desktops and mobile devices such as PDAs and smart phones [14].[4] Further research on this young protocol is highly valuable to the community, as the number of applications supported by SIP is expected to grow. For this work, SIP has been selected for use in the experiments described in Chapter IV.

## C. SUMMARY

H.323 and SIP provide similar VoIP services using different approaches. Thus they differ in simplicity and flexibility, extensibility, scalability, and security. Both protocols have advantages as well as disadvantages and they continue to be improved. For this project, SIP is chosen as the test protocol for research purposes.

[4] NetMeeting is standard video-conferencing program included in Windows 2000 and XP.

# IV. TESTING

This chapter describes the test methodology and test plan to verify the feasibility of SIP-based VoIP communications in different network architectures that included Network Address Translation (NAT) devices. An overview of the five tests and a brief summary of the findings are also presented. The testing described in this chapter is a preliminary step in integrating VoIP capabilities into the existing MYSEA architecture.

## A. TEST METHODOLOGY

Testing is conducted on a dedicated testbed using an incremental approach. After each test, the result is thoroughly analyzed before proceeding to a more complicated test. The incremental testing approach is preferred in this study because it allows easy identification and debugging of problems that emerged during the tests.

A number of free tools are deployed in the testbed. In particular, SJPhone, a softphone developed by SJ Labs [15], is used to make and receive SIP-based VoIP calls. Ethereal [16], an open source packet capture tool, is used to capture packet exchanges during each VoIP session for post-testing analysis. *netfilter* and *iptables* [6], modules in Linux Operating System kernel, provide Network Address Translation and routing functions in the testbed. Finally, ZoneAlarm [17], a free software-based firewall, is used to block certain traffic during the tests.

A number of systems are used to model the different components that make up the MYSEA environment. For example, Windows laptops with SJPhone installed are in the testbed to represent the untrusted clients that sit behind the TPEs. Linux systems are used to perform NAT and/or routing functionalities. They simulate the TPEs in MYSEA.

This project focuses on testing the feasibility of making VoIP calls from the MLS LAN. Therefore, VoIP calls are always initiated from the clients located behind TPEs on the MLS LAN to the clients located on simulated single level networks. In terms of MYSEA, this is equivalent to allowing calls to be initiated from clients on the MLS LAN to clients on external networks.

## B. TEST DESCRIPTION

The main objective of the tests described in the following subsections is to verify that VoIP conversations can be carried out in each network configuration. Procedures and results pertaining to each test can be found in Appendices B, C, D, E, F, and G. Note that private IP addresses assigned to network devices were used for demonstration purposes only. However, public devices such as public NATs and routers should use public IP addresses in practical scenarios.

### 1. Test 1: No NAT VoIP Configuration

The objective of this experiment is to observe the behavior of SJPhone in the simplest possible setup. The testbed consists of two directly connected VoIP-enabled clients as shown in Figure 6. In this scenario, Client B initiates a VoIP call to Client A. Test procedures and results are included in Appendix B.

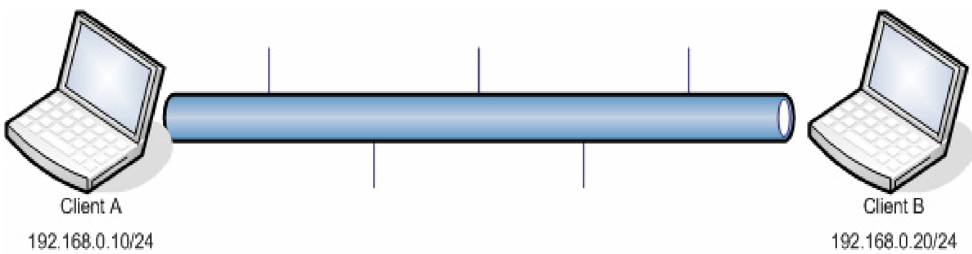


Figure 6. Test 1: Physical and Logical Network Topology

### 2. Test 2: Single NAT VoIP Configuration

The goal of this experiment is to confirm the feasibility of SIP-based VoIP calls when a NAT system is present. The testbed for this experiment consists of two VoIP-enabled clients and one NAT device as illustrated in Figure 7. The combination of the NAT device and Client B simulates the TPE-client pair in the MYSEA architecture. Client A is aware of Client B in this setup. Client B, on the other hand, is hidden behind a NAT device and is not visible to Client A. All packets exchanged between the two clients must traverse the NAT device that is configured with Source NAT and Destination NAT.

Two similar tests are conducted with this NAT configuration. The first test has a physical and logical network topology depicted in Figure 7. Even though the second test uses the same physical network topology as the first test, it has a logical topology depicted in Figure 8. Since the NAT device is not configured to drop packets destined for private IP addresses, Client A can send RTP packets directly to the private IP address of Client B. This is exactly what Client A does based on the packet captures provided in Appendix C. In non-experimental scenarios, a firewall at the client is not necessary because packets destined for a private IP address will eventually be dropped as they traverse the networks. For demonstration purposes, a firewall is introduced at Client A to block packets initiated by Client A and destined for Client B. More information including the test procedures and results for both tests can be found in Appendix C.

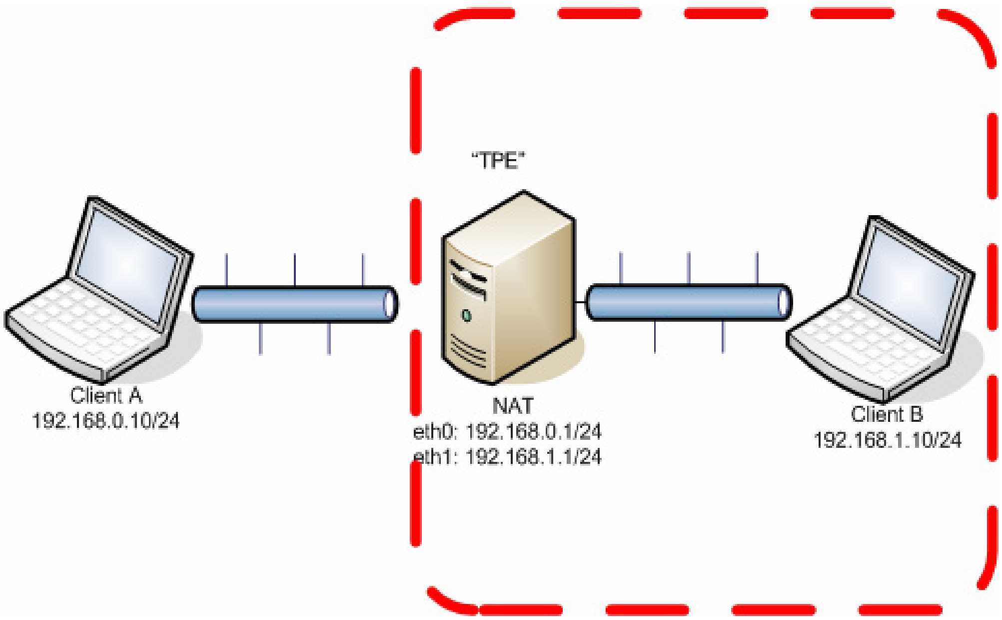


Figure 7. Test 2: Physical Network Topology (with and without firewall)

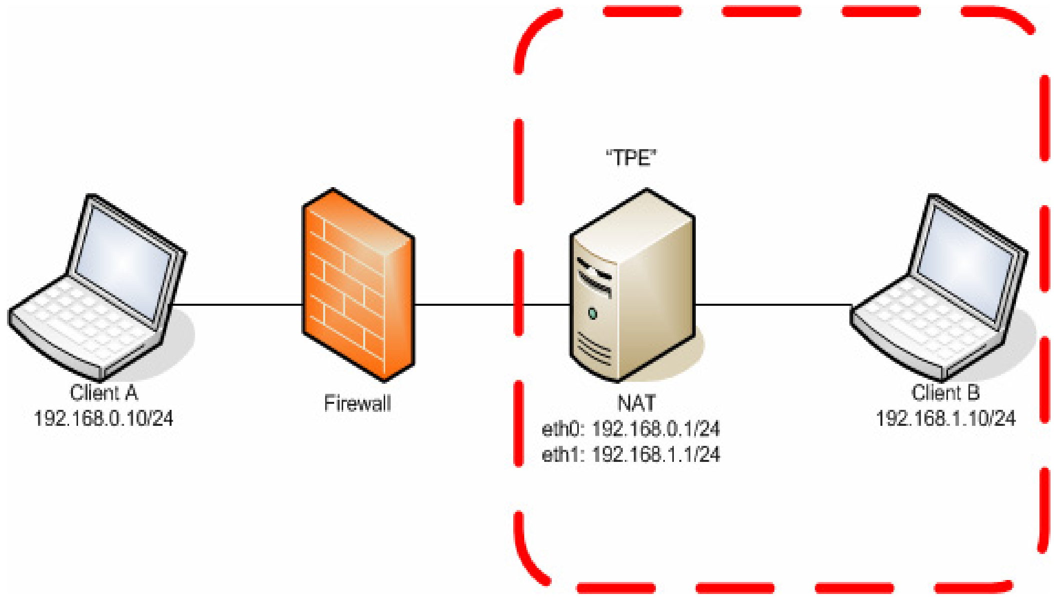


Figure 8. Test 2: Logical Network Topology (with firewall)

### 3. Test 3: Double NAT VoIP Configuration

The goal of this experiment is to confirm the feasibility of a SIP-based VoIP call using SJPhone when two NAT systems are present. In this test, a VoIP session between two clients have to traverse two different NAT devices as depicted in Figure 9 and Figure 10. Similar to the previous test, Client B and NAT 2 represent the client-TPE pair in the MYSEA architecture. NAT 1 simulates the NAT device located between the MYSEA network and the Internet whereas Client A acts as a VoIP-enabled client in the Internet. Both NAT devices are configured to perform Source NAT and Destination NAT. Test procedures and results are included in Appendix D.

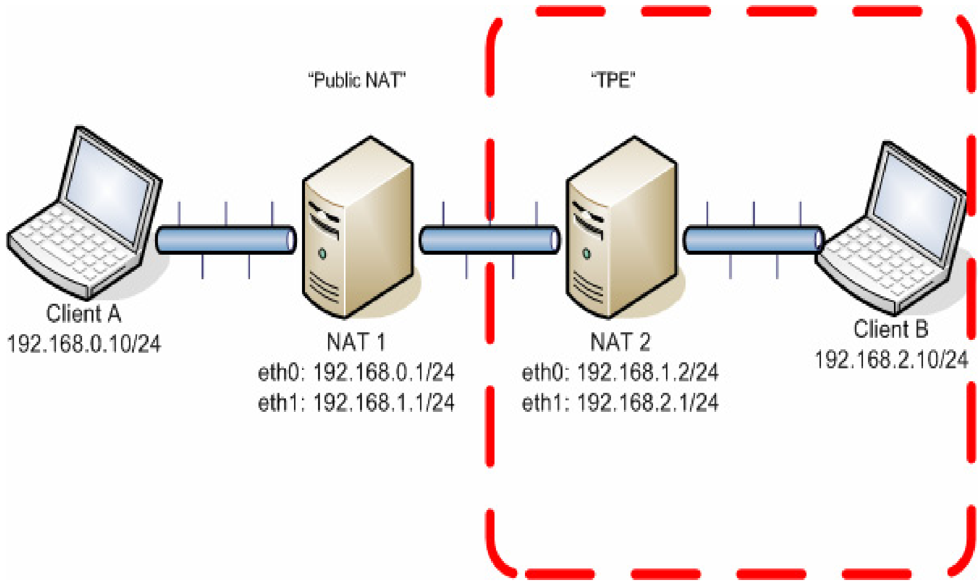


Figure 9. Test 3: Physical Network Topology

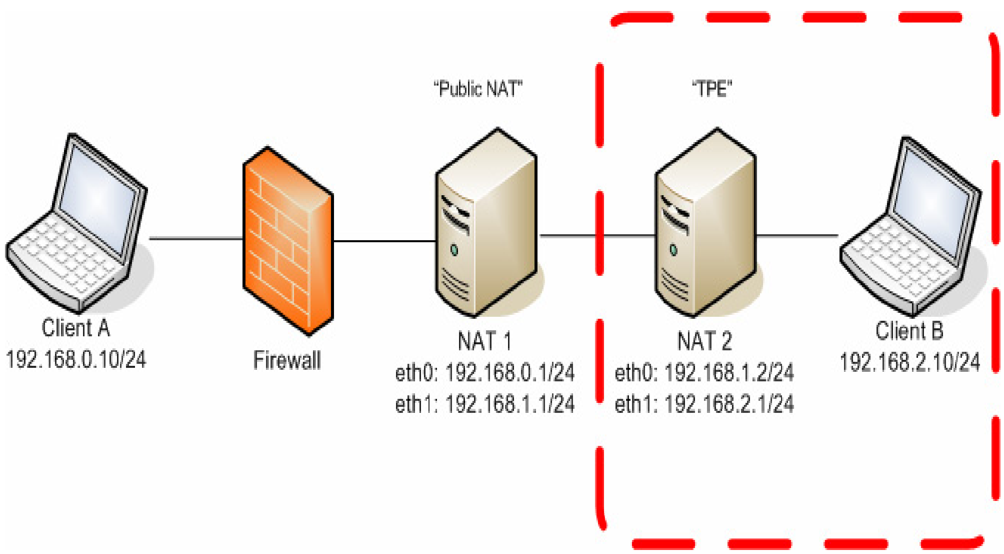


Figure 10. Test 3: Logical Network Topology

### 4. Test 4: Extended Double NAT VoIP Configuration

The purpose of this experiment is to confirm that two different VoIP sessions can take place at different times when the sessions have to traverse a common public NAT device. This and the next setup only work when the public NAT device implements a connection tracking mechanism that is similar to what *iptables* provides. The test is setup so that the public NAT device is not explicitly instructed to forward packets to any specific network. In other words, the public NAT only performs Source NAT but not

Destination NAT. A firewall is installed on Client A to prevent Client A from sending RTP packets directly to the private address of Client B. The firewall is necessary in this demonstration because NAT 1 and NAT 2 are not configured to drop packets destined for private IP addresses.

This test consists of three VoIP-enabled clients and three NAT devices as depicted in Figures 11 and 12. Two client-TPE pairs are simulated in this setup, Client B and NAT 2 being one pair and Client A and NAT 3 being the second pair. NAT 1 resembles the public NAT that is located between the MLS LAN and the Internet. NAT 1 is only configured with a SNAT rule whereas NAT 2 and NAT 3 are configured with both SNAT and DNAT rules. The test proceeds as follows: Client B initiates a call to Client A, terminates the call, and then Client C initiates a call to Client A. Procedures and results can be found in Appendix D.

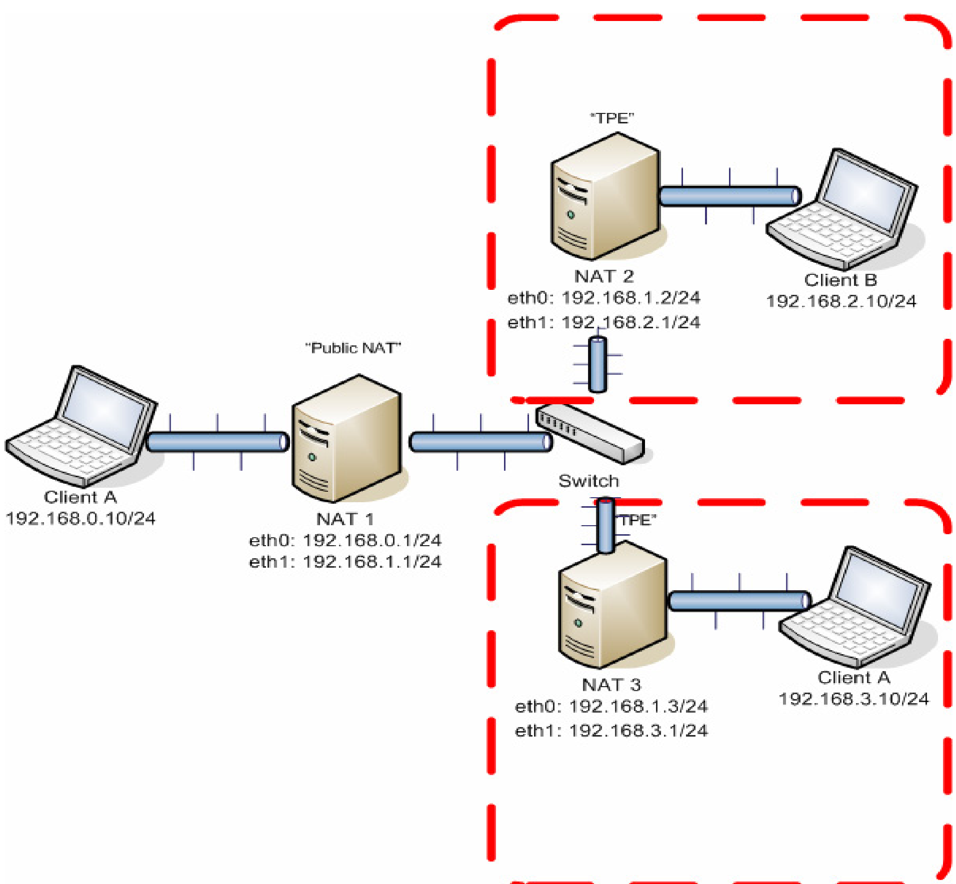


Figure 11. Test 4: Physical Network Topology

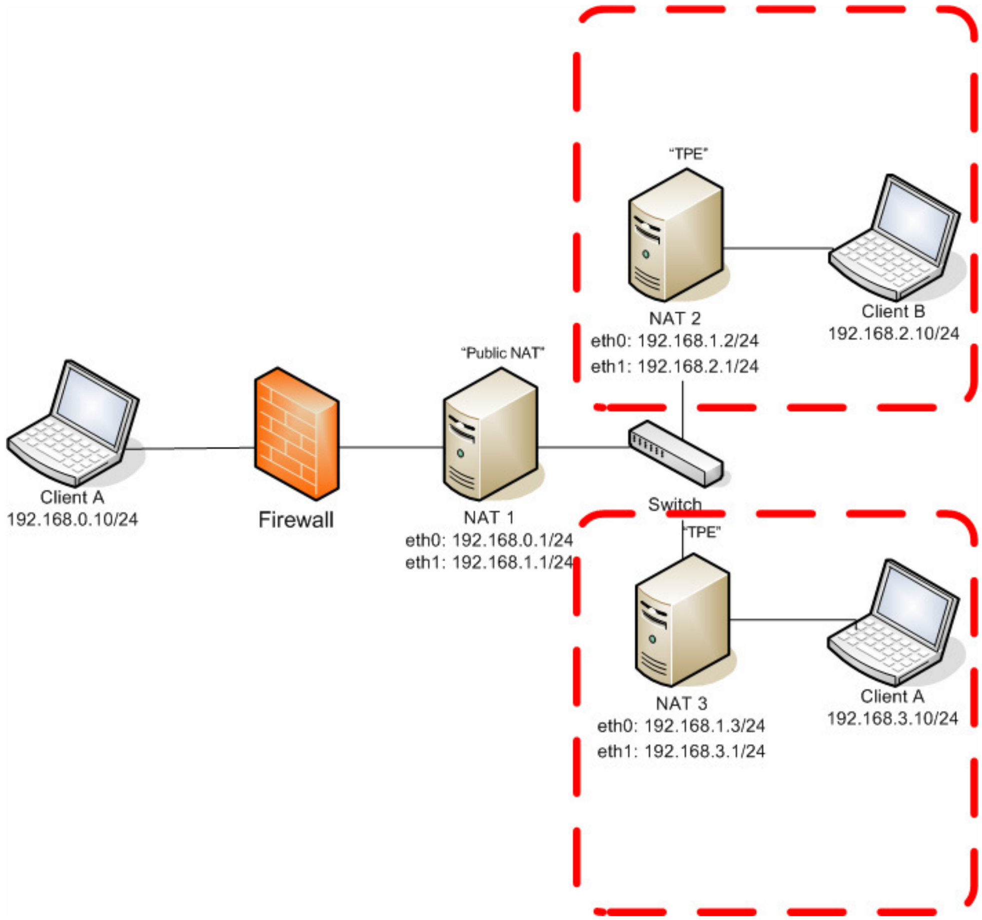


Figure 12. Test 4: Logical Network Topology

### 5. Test 5: Extended Double NAT VoIP Configuration with Simultaneous VoIP Sessions

The purpose of this test is to confirm the feasibility of two simultaneous VoIP sessions between two pairs of clients. The setup of this test, shown in Figures 13 and 14, closely resembles a simplified version of the MYSEA architecture. The IP addresses used for different components in this demonstration are the same as the ones used in the MYSEA testbed. This test consists of four VoIP-enabled clients, three NAT devices, and a router. The router is introduced here as a preparation for the next test. Refer to the next section for a description of the router. Similar to the previous test, two pairs of client-TPEs are simulated using Client C, NAT 2, Client D, and NAT 3. Furthermore, NAT 1 is not configured with a DNAT rule and two firewalls are installed on Client A and Client B to block RTP packets destined to Client C and Client, respectively. In this scenario, Client C calls Client A and Client D calls Client B at the same time.

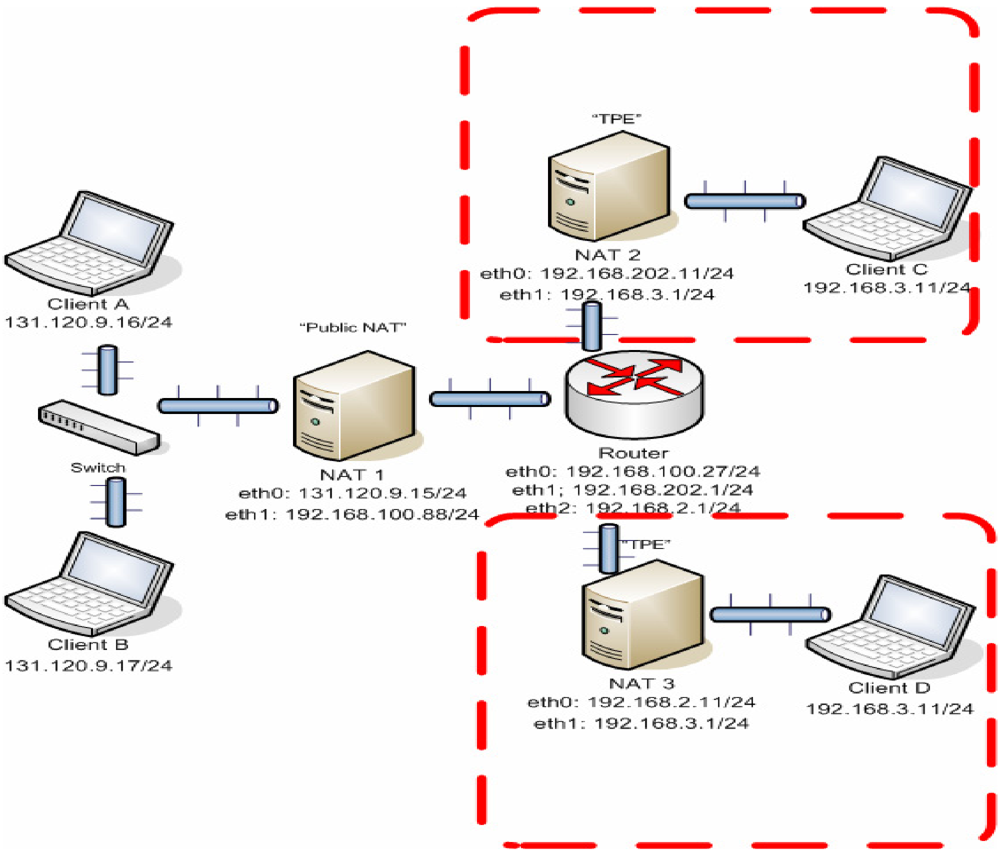


Figure 13. Test 5: Physical Network Topology

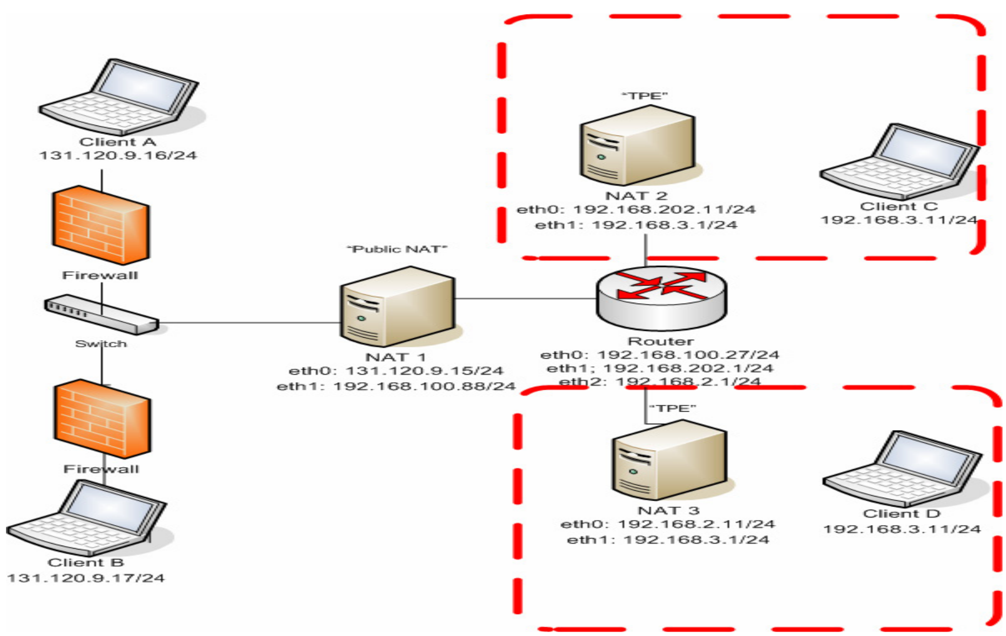


Figure 14. Test 5: Logical Network Topology

### 6. Test 6: MYSEA Configuration

The setup of this test, illustrated in figures 15 and 16, is an extension to the last one with the addition of a MLS server. The objective of this test is to verify that the MLS Server could support two simultaneous VoIP sessions between two pairs of clients. Since the MLS server does not perform routing in the testbed, a router is introduced to perform that function. The MLS server simply forwards the received packets to the correct network interfaces according to the network configurations. In this test, unexpected routing problems were encountered on the MLS server. Refer to the next section for a discussion of this test.

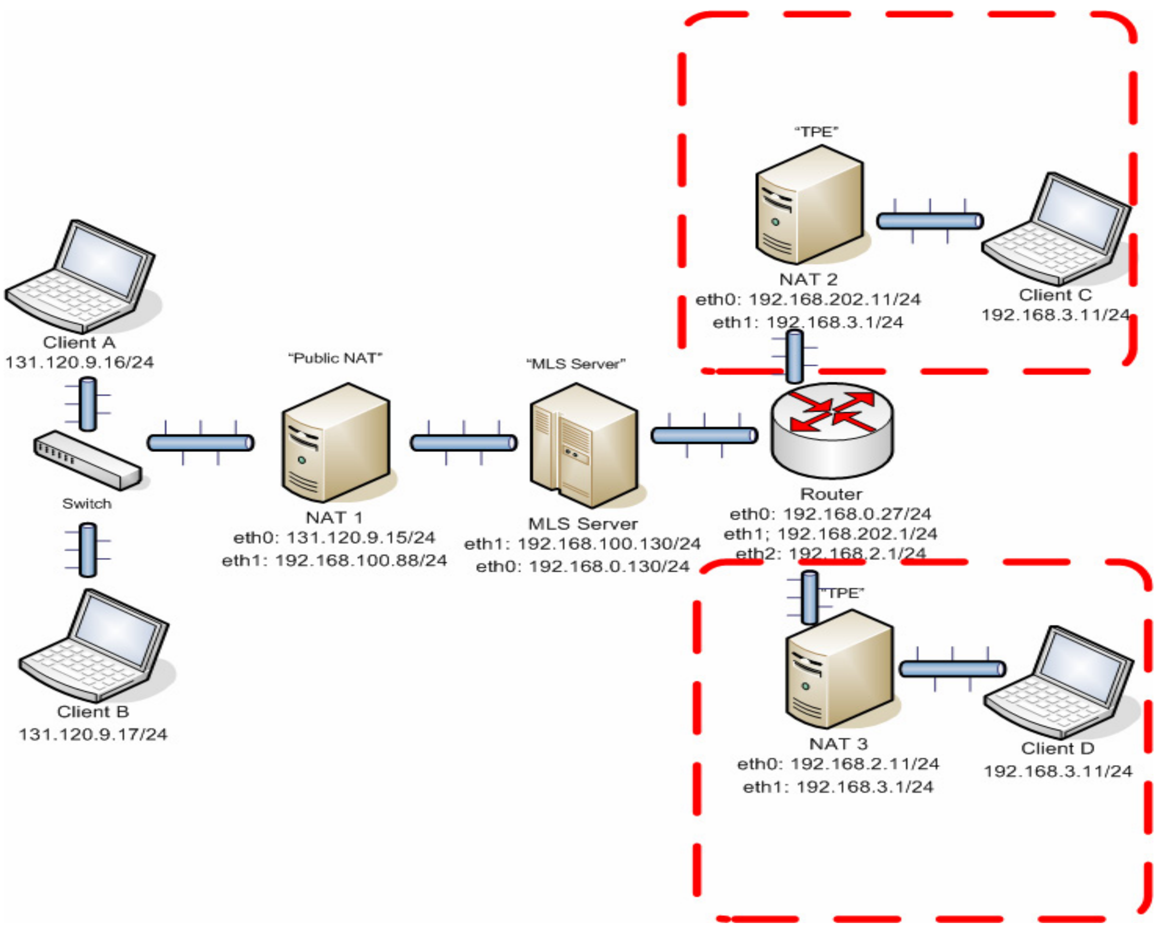


Figure 15. Test 6: Physical Network Topology

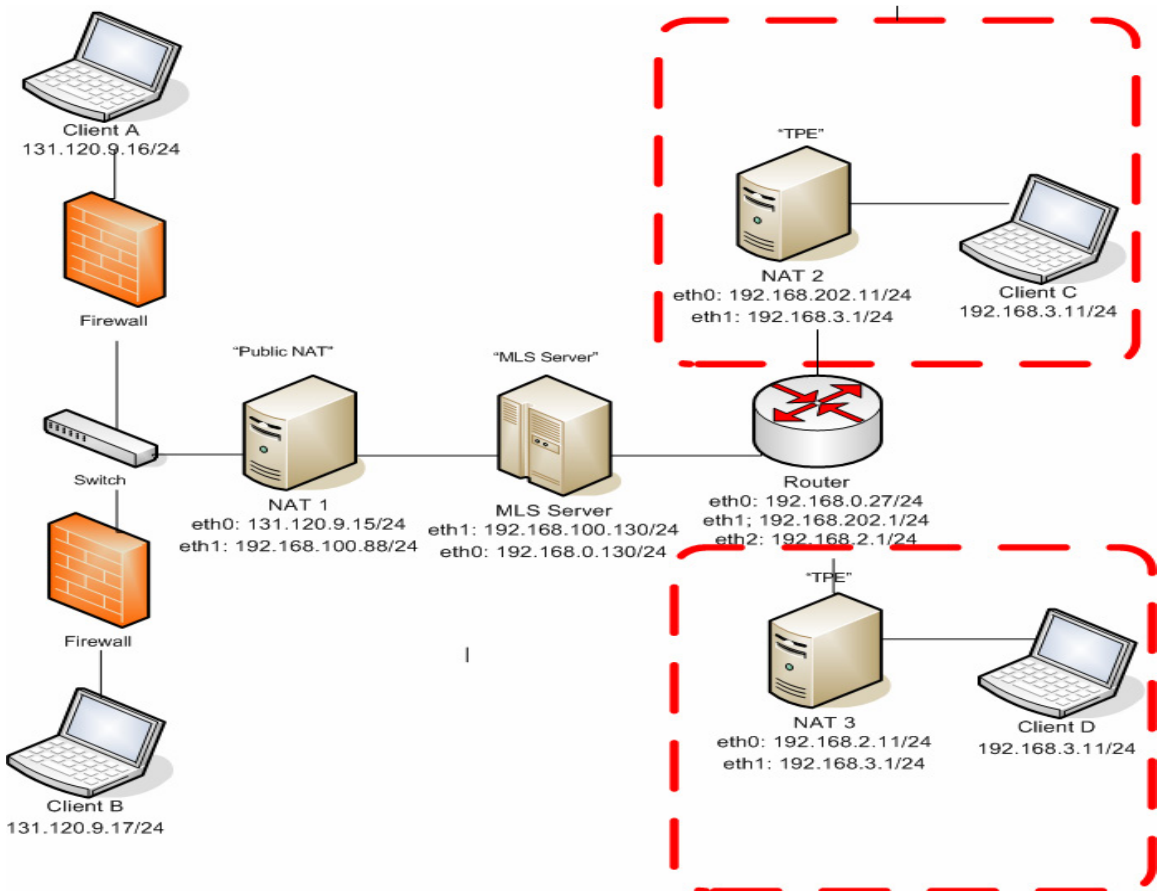


Figure 16. Test 6: Logical Network Topology

## C. PROBLEMS ENCOUNTERED

Setting up and running the tests was fairly straightforward with the exception of the MYSEA Configuration. The major problem encountered that ultimately led to the failure of the MYSEA test was configuring the MLS server to perform proper routing. The MLS server, currently running XTS-400, has routing limitation such that only one static route can be configured for each network interface. An attempt to add a second route to the network interface X resulted in the following error message: "Route /dev/etherX already exists". Since every packet has to traverse the server, this limitation prevents an incoming or outgoing packet arriving at the MLS server from routing to the other side of the network. According to the customer service response to our inquiry,, "there is a known restriction on [the XTS-400] configuration tool tcpip_edit (not the network stack) that there can only be one route per interface device..." [18]. Four test scenarios were conducted to ensure that the MLS server, in fact, has routing limitation. Refer to Appendix G for details. In conclusion, the MYSEA Configuration test was not conducted successfully.

## D. TEST RESULT

All the tests described in the previous section generate positive results, i.e. the results indicate that VoIP communications are possible in five of the six scenarios. However, it is important to recognize that the last test was unsuccessful not because of VoIP limitation or the network topology. Instead, the last test failed was due to configuration difficulties in the MLS server.

Several interesting findings are discovered from analyzing the packet captures. First, SJPhone will attempt to send RTP packets to the IP address indicated in the SDP data. If that fails, then SJPhone will resort to send subsequent RTP packets to the IP address it received RTP packets from. Second, the client that initiates the VoIP call is always the first to send out a RTP packet to the other party. Third, *iptables* has a connection tracking mechanism that allows it to associate incoming packets with previous outgoing packets and determine which VoIP session the incoming packets belong to. Since *iptables* creates an entry in its connection tracking table for the first RTP packet sent, subsequent incoming packets can be correctly forwarded to the correct next hop based on the information stored in the table. This mechanism plays a significant role in the success of tests 4 and 5 where the public NAT was not configured with a DNAT rule to forward incoming packets to any particular IP addresses. Refer to Appendices B through G for more details on the findings.

## E. SUMMARY

The purposes and configurations of the experiments designed for this feasibility study are described in this Chapter. The results of the experiments are also briefly discussed. The results are optimistic and they indicate that the integration of VoIP into MYSEA is possible.

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# V. FUTURE WORK AND CONCLUSIONS

## A. FUTURE WORK

The test results described in Chapter IV and various appendices suggest that VoIP capabilities can be potentially integrated into the existing MYSEA architecture with little effort. However, further research in the following areas is required.

### 1. Routing in MLS Server

Every MLS client communication is mediated by the MLS server that runs on XTS-400, a high-assurance Unix-like system. Unfortunately, the MLS server has routing limitation such that only one static route can be configured for each network interface. The ability to configure more than one route for each interface in the MYSEA server is necessary for VoIP packets to route between clients on the MLS network and on external networks. Therefore, further study of the routing configurations or capabilities is required to allow proper routing of VoIP packets.

### 2. VoIP Conversations Initiated from the Internet

This research study is primarily concerned with testing the scenarios in which VoIP conversations are initiated from the MLS LAN. The ability for externally initiated VoIP conversation is also desirable. Currently, a client on an external network only knows the IP address of the public NAT device and there exists no way of distinguishing calls intended for different clients on the MLS LAN. In order for an internal client to receive an external call, three features may have to be implemented. First, each internal client must own a unique SIP address that is publicly known. This allows an external client to direct call to a specific internal client. Second, a server must exist to translate a SIP address to the corresponding internal client IP address. Third, softphones with reconfigurable RTP port, such as the SipXphone, are needed for each internal client. Each client needs to use a different RTP port for sending and receiving RTP packets and the public NAT must be configured to perform port forwarding. This allows NAT to forward RTP packets to the correct client according to the destination port. Research on how to implement this scheme is highly recommended.

## B. CONCLUSIONS

The tests conducted in this research study were generally successful. Furthermore, the test results indicate that VoIP conversations, at least in the scenario we studied, between internal and external clients are possible even when various NAT devices are present. It is important to recognize that the success of Test 4 and Test 5 was dependent on the connection tracking mechanism in *iptables*. NAT devices without connection tracking mechanisms were not tested in this project. Thus, it is unknown whether the tests will work if those devices are used instead. In conclusion, VoIP capabilities may be integrated into the existing MYSEA architecture.

# APPENDIX A. A SURVEY OF VOIP HARDPHONES AND SOFTPHONES

## A. HARDPHONES

The following table is a survey of some VoIP hardphones. Each hardphone is listed with information including its manufacture (Brand/Company), phone type (category and Sub-Category), the VoIP protocol (VoIP Protocol), and Wi-Fi protocol (Wi-Fi Protocol) it supports.

| Name | Brand/Company | Category | Sub-Category | VoIP Protocol | Wi-Fi Protocol |
|---|---|---|---|---|---|
| ACT P202 | Advantage Century Telecomm Phones | WLAN/WiFi Phone | Mobile IP Phone | SIP | 802.11b |
| Senao SI-7800H | Senao | WLAN/WiFi Phone | Mobile IP Phone | SIP | 802.11b |
| F1000 | UTStarcom | WLAN/WiFi Phone | Mobile IP Phone | SIP | 802.11b |
| WLAN600 | BCM | WLAN/WiFi Phone | Cordless Handset | SIP | 802.11b |
| Cisco 7920 | Cisco | WLAN/WiFi Phone | Cordless Handset | Skinny | 802.11b |
| WirelessIP5000 | Hitachi Cable | WLAN/WiFi Phone | Cordless Handset | SIP | 802.11b |
| ZyXEL P2000W | ZyXEL | WLAN/WiFi Phone | Cordless Handset | SIP | 802.11b |
| Vesta 100 | Arkon Networks | WLAN/WiFi Phone | Cordless Handset | N/A | 802.11b/g |
| Motorola CN620 | Motorola | GSM/WiFi Phone | Handset | SIP | 802.11a/b/g |
| Nostrand 200 | Arkon Networks | WLAN/WiFi Phone | Handset | SIP | 802.11b/g |
| Stonehenge WP150 | Moimstone | WLAN/WiFi Phone | Wireless Phone | SIP | 802.11b/g |

Table 7. Summary of Hardphones

## B. SOFTPHONES

The following table is a survey of some VoIP softphones. Each softphone is listed with information including what Operating System(s) and VoIP protocol(s) (VoIP Protocol) it supports and whether it is a commercial or an open source product.

| Name | Pocket PC 2003 | MAC OS X | Linux | Win2K | WinXP | VoIP Protocol | Commercial | Open Source |
|---|---|---|---|---|---|---|---|---|
| X-Lite/eyeBeam | Y | Y | | Y | Y | SIP | Y | |
| X-Pro | Y | Y | | Y | Y | SIP | Y | |
| eyeP Phone Lite | | | | Y | Y | SIP | | Y |
| SIPPS | | | | | | SIP | Y | |
| Ubiquity User Agent | | | | Y | Y | SIP | | Y |
| EZ-Phone | | | | Y | Y | SIP | Y | |
| MySIP | | | | Y | Y | SIP | Y | |
| SJPhone | Y | Y | Y | Y | Y | SIP and H.323 | Y | |
| Linphone | | | Y | | | SIP | | Y |
| Kphone | | | Y | | | N/A | | Y |
| Vovida | | | Y | | | SIP | | Y |
| Siphon | | | Y | | | SIP | | Y |
| sipXphone | | | Y | Y | Y | SIP | | Y |
| Shtoom | | Y | | Y | Y | SIP | | Y |
| Cornfed SIP-UA | | | Y | | | SIP | | Y |
| SFLphone | | Y | | Y | Y | SIP | | Y |
| Asterisk | | | Y | | | SIP and H.323 | | Y |
| Empower Pro Internet Phone | Y | Y | | | | N/A | Y | |
| Skype | | Y | Y | Y | Y | SIP | Y | |

Table 8. Summary of Softphones

# APPENDIX B. TEST 1: NO NAT VOIP DEMONSTRATION USING SJPHONE

The instructions in this appendix describe how to setup and demonstrate a SIP-based VoIP communication between two directly connected SIP-enabled clients using SJPhone. Figure 6 illustrates the physical network as well as the logical topology for this demonstration. A VoIP session is initiated from Client B to Client A. Packet captures from both clients are included at the end of this appendix along with an analysis.

A. Network Topology

Refer to Figure 6 for the physical and logical network topology.

B. Equipment Requirements

B.1. Clients A and B

B.1.1. Windows XP Operating System

B.1.2. Sound card

B.1.3. SJPhone v.1.60

B.1.4. Ethereal

B.2. Additional Equipment

B.2.1. Cross-over cable to implement the network architecture Figure 6

B.2.2. Microphones as audio input devices for clients A and B

C. Installation and Configuration

C.1. Client A

IP Address: 192.168.0.10

Subnet Mask: 255.255.255.0

Default Gateway: 192.168.0.20

C.2. Client B

IP Address: 192.168.0.20

Subnet Mask: 255.255.255.0

Default Gateway: 192.168.0.10

C.3. SJPhone Installation and Configuration

C.3.1. Client A and Client B

C.3.1.1. Download the Windows version of SJPhone v.1.60 from SJ Labs

C.3.1.2. Install SJPhone v.1.60

C.3.1.3. Launch SJPhone

C.3.1.4. Right-click on SJPhone

C.3.1.5. Go to Services

C.3.1.6. Select PC-to-PC (SIP)

C.4. Ethereal Installation and Configuration

C.4.1. Clients A and B

C.4.1.1. Download the latest Windows version of Ethereal

C.4.1.2. Install Ethereal

D. Preparation and Testing

D.1. Adjust volume on both clients accordingly

D.2. Plug microphones into both clients

D.3. On Client A,

D.3.1. Launch Ethereal

D.3.2. Go to the Capture menu

D.3.3. Go to Interfaces

D.3.4. Click on Capture 192.168.0.10

D.4. On Client B,

D.4.1. Launch Ethereal

D.4.2. Go to the Capture menu

D.4.3. Go to Interfaces

D.4.4. Click on Capture 192.168.0.20

D.4.5. Call Client A by dialing 192.168.0.10 in SJPhone

D.5. On Client A,

D.5.1. Select Accept in the pop-up dialog box when SJPhone rings

D.5.2. Clients A and B may engage in a VoIP conversation at this point

D.5.3. Click on the Hang-Up bottom on either SJPhone to terminate the call when finished

D.6. On Clients A and B,

D.6.1. Stop packet captures by selecting Stop on Ethereal

E. Packet Captures

E.1. Client A

Figure 17 is a snapshot of the packets captured on Client A.

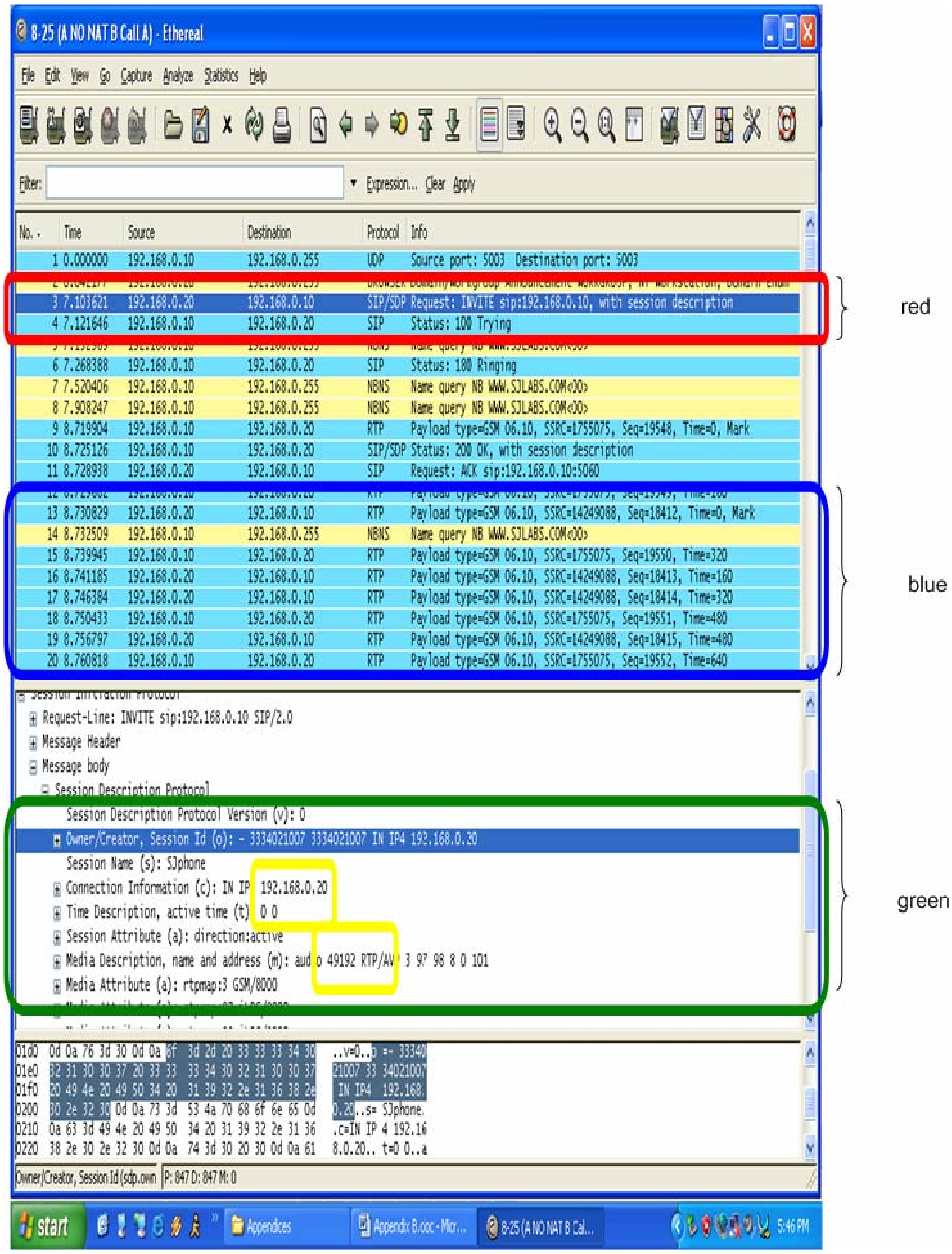


Figure 17. Test 1: Packet Capture on Client A

E.2. Client B

Figure 18 is a snapshot of the packets captured on Client B.

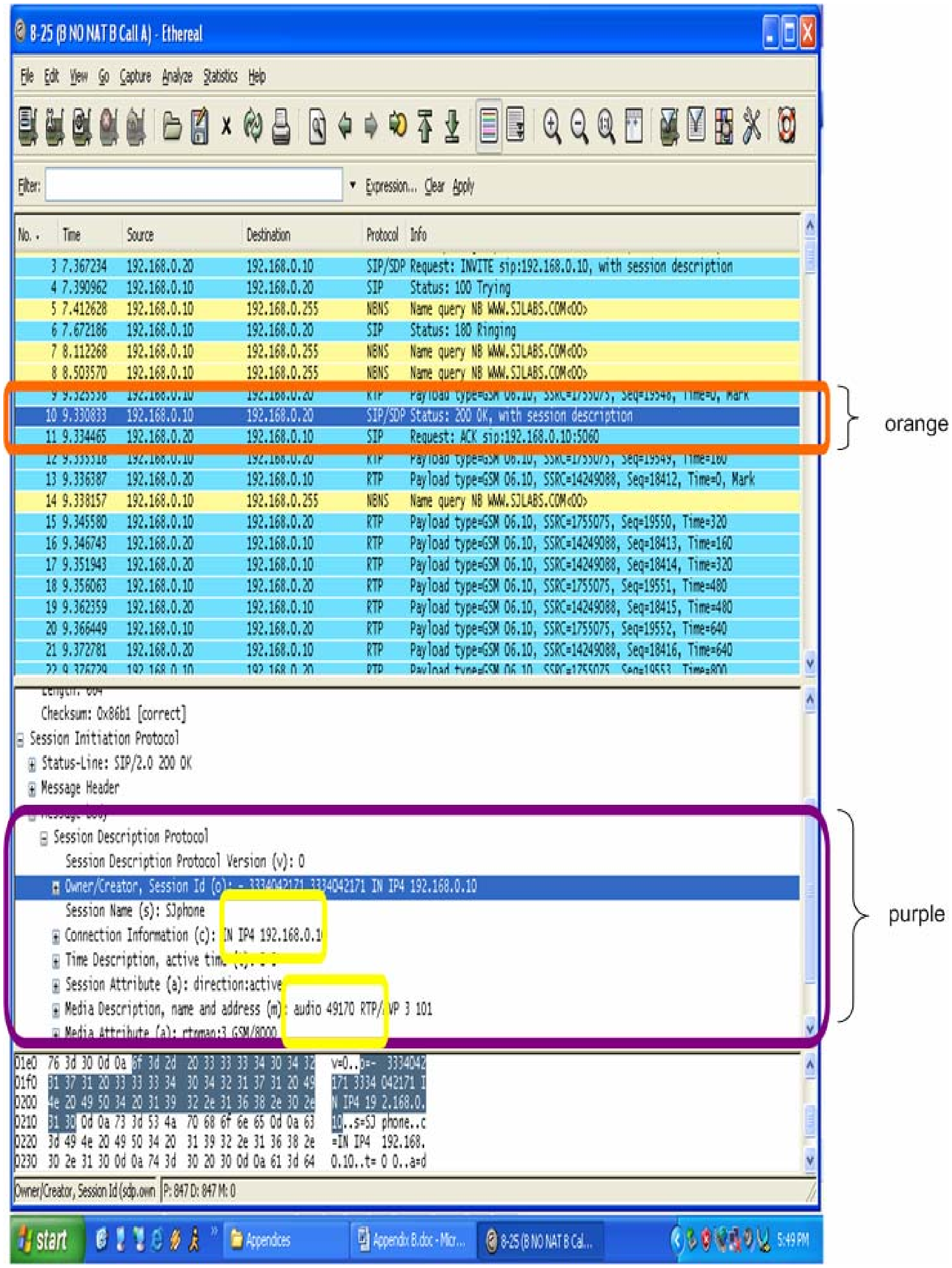


Figure 18. Test 1: Packet Capture on Client B

### E.3. Analysis

The packet captures indicate that as soon as Client B initiated a call to Client A, Client B sent out an “INVITE” message from 192.168.0.20:5060 to Client A at 192.168.0.10:5060 (red outline in Figure 17). The “INVITE” message had embedded SDP information to inform Client A that Client B will send and receive RTP voice packets at 192.168.0.20 on port 49192 (green outline in Figure 17). Client A acknowledged the invitation by sending Client B a “200 OK” message (orange outline in Figure 18) with embedded SDP information indicating that it will send and receive RTP voice packets at 192.168.0.l0 on port 49170 (purple outline in Figure 18). Subsequent voice exchanges between the two clients were achieved via 192.168.0.20: 49192 on Client B and 192.16.0.10: 49170 on Client A (blue outline in Figure 17).

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# APPENDIX C. TEST 2: SINGLE NAT VOIP DEMONSTRATION USING SJPHONE

The instructions contained in this appendix describe how to setup and demonstrate a SIP-based VoIP communication between two SIP-enabled clients via a Network Address Translation (NAT) device. In this setup, Clients A and B belong to different networks and Client B is located behind a NAT device. The NAT device is configured to act as a router and modify the destination or source IP address of all packets that traverse it. In this scenario, Client B initiates a VoIP call to Client A.

The demonstration consists of two parts. They are very similar in nature except that a firewall is introduced in the second part. Packet captures and an analysis are included for each part.

## A. Without Firewall

### A.1. Network Topology

Refer to Figure 7 for the physical and logical network topology.

### A.2. Equipment Requirements

#### A.2.1. Client A and Client B

A.2.1.1. Windows XP Operating System

A.2.1.2. Sound Card

A.2.1.3. SJPhone v.1.60

A.2.1.4. Ethereal

A.2.1.5. ZoneAlarm (Client A only)

#### A.2.2. NAT

A.2.2.1. Linux Operating System (Fedora Core 4)

A.2.2.2. netfilter and iptables

A.2.2.3. Ethereal

A.2.2.4. Two network cards

A.2.3. Additional Equipment

A.2.3.1. Cross-over cables to implement the network architecture illustrated in Figure 7

A.2.3.2. Microphones as audio input devices for Client A and Client B

A.2.4. Installation and Configuration

A.2.4.1. Client A

IP Address: 192.168.0.10
Subnet Mask: 255.255.255.0
Default Gateway: 192.168.0.1

A.2.4.2. Client B

IP Address: 192.168.0.20
Subnet Mask: 255.255.255.0
Default Gateway: 192.168.1.1

A.2.4.3. NAT

A.2.4.3.1. Configure eth0 by editing /etc/sysconfig/network-scripts/ifcfg-eth0:

```
DEVICE=eth0
BOOTPROTO=NONE
IPADDR=192.168.0.1
NETMASK=255.255.255.0
```

A.2.4.3.2. Activate eth0 by running:

```
ifup eth0
```

A.2.4.3.3. Configure eth1 by editing /etc/sysconfig/network-scripts/ifcfg-eth1:

```
DEVICE=eth1

BOOTPROTO=NONE
```

IPADDR=192.168.1.1

NETMASK=255.255.255.0

A.2.4.3.4. Activate eth1 by running:

ifup eth1

A.2.4.3.5. Enable IP Forwarding by running:

echo 1 > /proc/sys/net/ipv4/ip_forward

A.2.4.3.6. Flush any existing firewall and NAT rules by running:

iptables -F

iptables -t nat -F

A.2.4.3.7. Configure NAT rules by running:

iptables -t nat -A POSTROUTING -o eth0 -j SNAT --to 192.168.0.1

iptables -t nat -A PREROUTING -i eth0 -j DNAT --to 192.168.1.10

A.2.4.4. SJPhone Installation and Configuration

A.2.4.4.1. Client A and Client B

A.2.4.4.1.1. Download the Windows version of SJPhone v.160 from SJLabs

A.2.4.4.1.2. Install SJPhone v.160

A.2.4.4.1.3. Launch SJPhone

A.2.4.4.1.4. Right-click on SJPhone

A.2.4.4.1.5. Go to Services

A.2.4.4.1.6. Select PC-to-PC (SIP)

A.2.4.5. Ethereal Installation and Configuration

A.2.4.5.1. Client A and Client B

A.2.4.5.1.1. Download the latest Windows version of Ethereal

A.2.4.5.1.2. Install Ethereal

A.2.4.5.2. NAT

A.2.4.5.2.1. Install Ethereal if it is not already installed:

A.2.4.5.2.1.1. Go to the Desktop menu

A.2.4.5.2.1.2. Go to System Settings

A.2.4.5.2.1.3. Go to Add/Remove Applications

A.2.4.5.2.1.4. Click on Details under System Tools

A.2.4.5.2.1.5. Find and then check ethereal-gnome

A.2.4.5.2.1.6. Click on Close

A.2.4.5.2.1.7. Click on Update

A.2.4.5.2.1.8. Put in the correct Fedora Core 4 CDs when prompted

A.3. Preparation and Testing

A.3.1. Adjust volume on both clients accordingly

A.3.2. Plug microphones into both clients

A.3.3. On client A,

A.3.3.1. Launch Ethereal

A.3.3.2. Go to the Capture menu

A.3.3.3. Go to Interfaces

A.3.3.4. Click on Capture 192.168.0.10

A.3.4. On client B,

A.3.4.1. Launch Ethereal

A.3.4.2. Go to the Capture menu

A.3.4.3. Go to Interfaces

A.3.4.4. Click on Capture 192.168.1.10

A.3.5. On NAT,

A.3.5.1. Launch one instance of Ethereal

A.3.5.2. Go to the Capture menu

A.3.5.3. Go to Interfaces

A.3.5.4. Click on Capture Eth0

A.3.5.5. Launch another instance of Ethereal

A.3.5.6. Go to the Capture menu

A.3.5.7. Go to Interfaces

A.3.5.8. Click on Capture Eth1

A.3.6. On Client B,

A.3.6.1. Call A by dialing 192.168.0.10 in SJPhone

A.3.7. On Client A,

A.3.7.1. Select Accept in the pop-up dialog box when SJPhone rings

A.3.8. Clients A and B may engage in a VoIP conversation at this point

A.3.9. Click on the Hang-Up bottom on either SJPhone to terminate the call when finished

A.3.10. On Client A, Client B, and NAT,

A.3.10.1. Stop packet captures by selecting Stop on Ethereal

## A.4. Packet Captures

### A.4.1. Client A

Figure 19 is a snapshot of the packets captured on Client A.

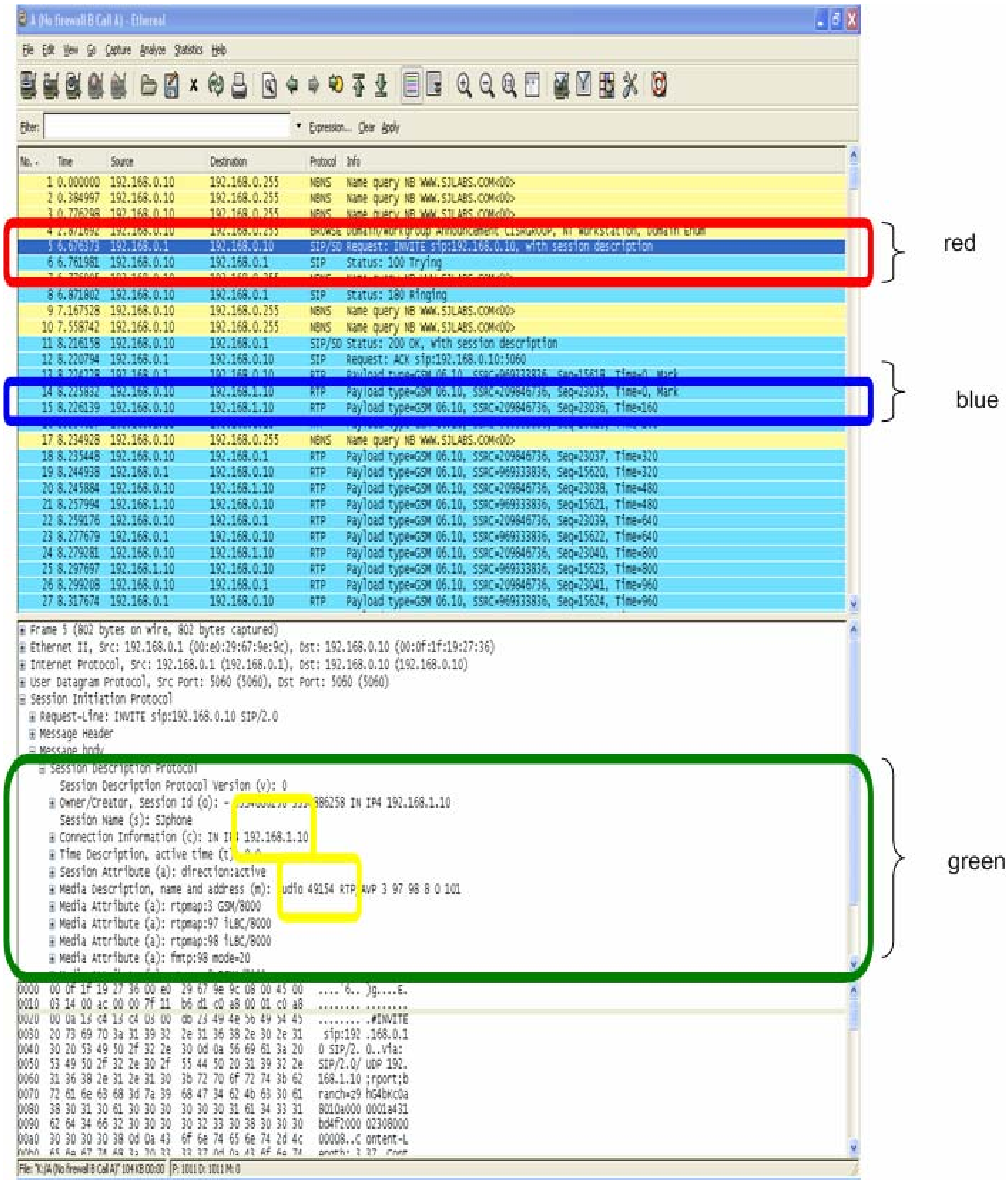



Figure 19. Test 2: Packet Capture on Client A (without firewall)

A.4.2. Client B

Figure 20 is a snapshot of the packets captured on Client B.

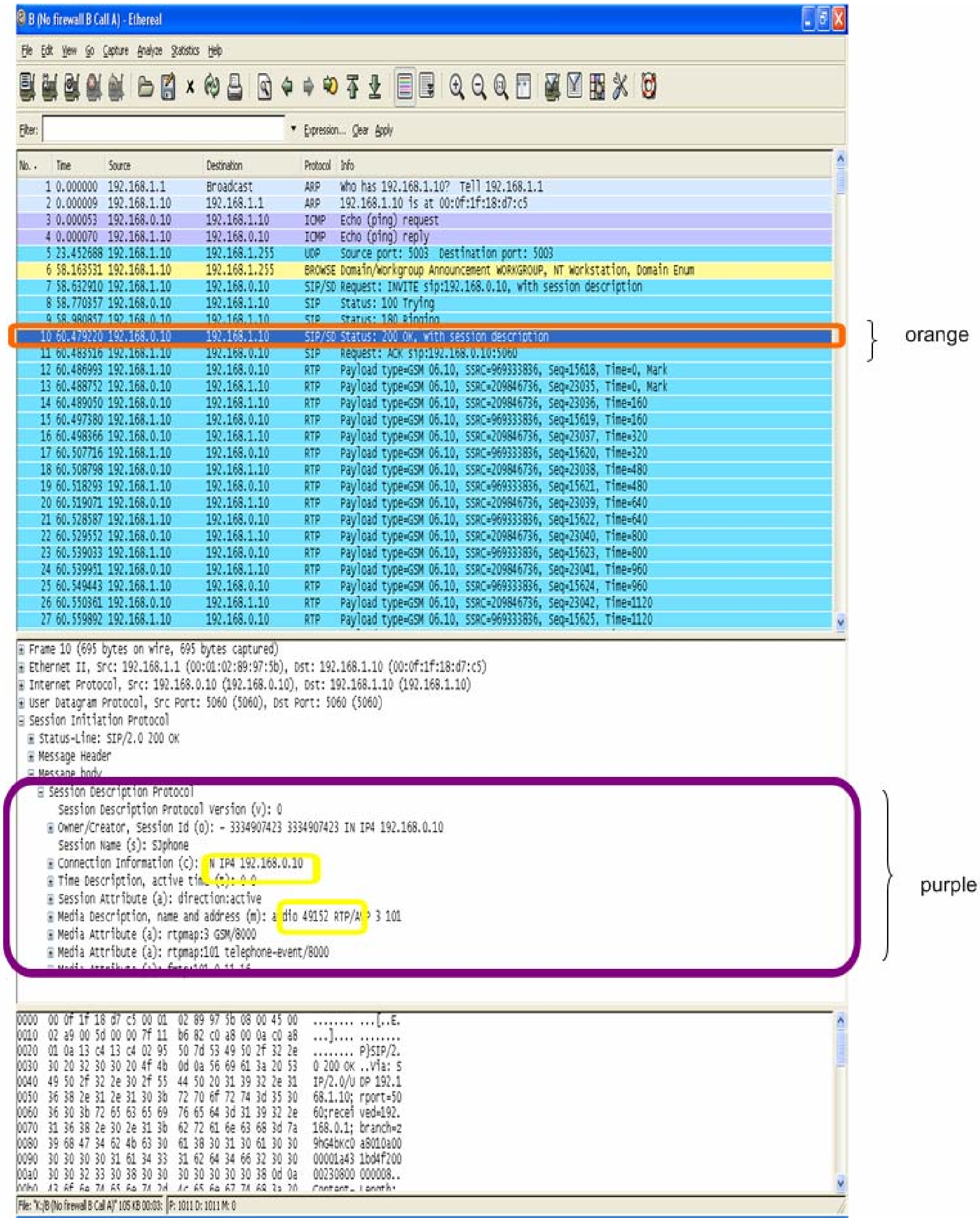



Figure 20. Test 2: Packet Capture on Client B (without firewall)

A.4.3. NAT eth0

Figure 21 is a snapshot of the packets captured on the first interface (eth0) of the NAT device.

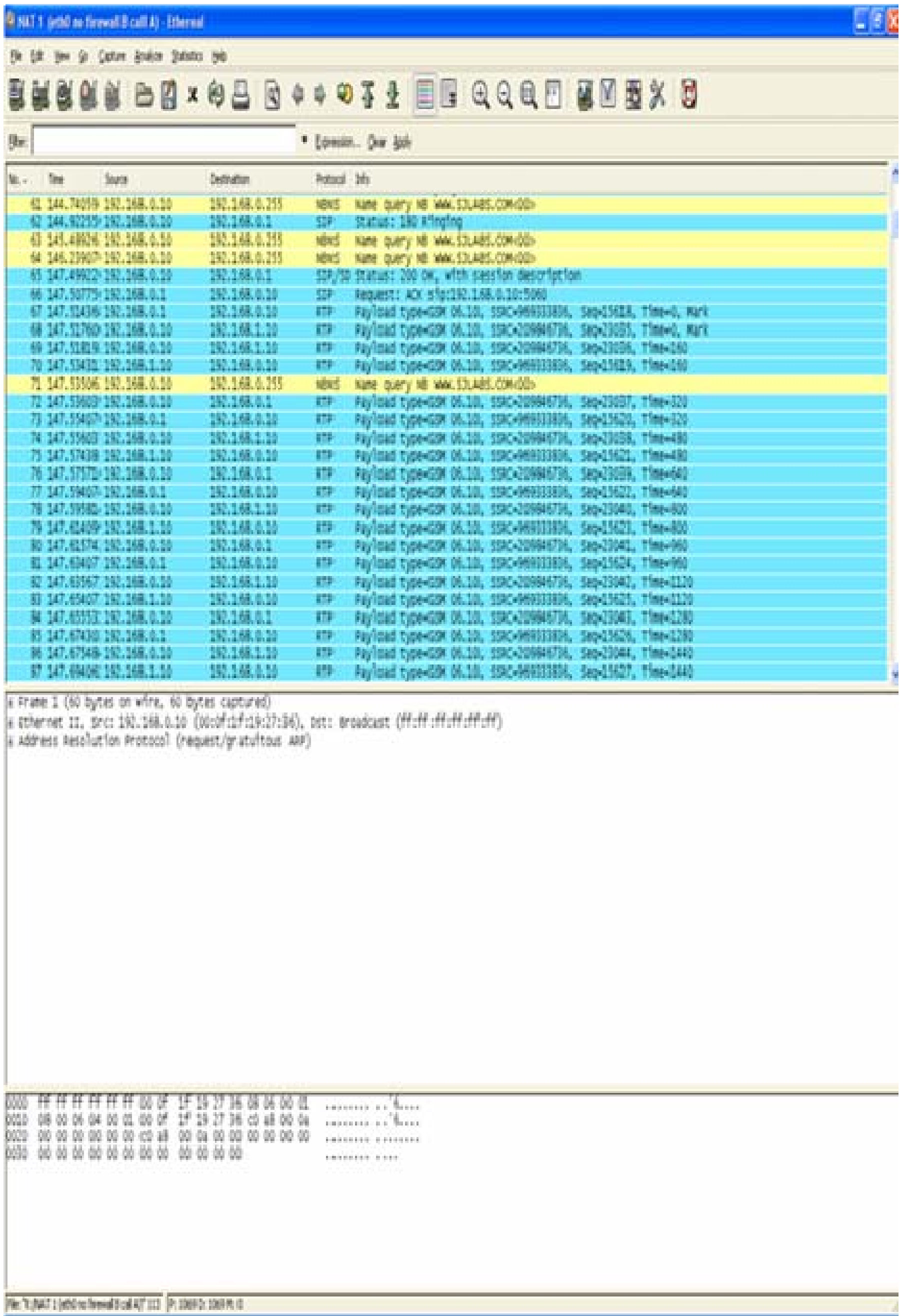


Figure 21. Test 2: Packet Capture on eth0 of NAT (without firewall)

A.4.4. NAT eth1

Figure 22 is a snapshot of the packets captured on the second interface (eth1) of the NAT device.

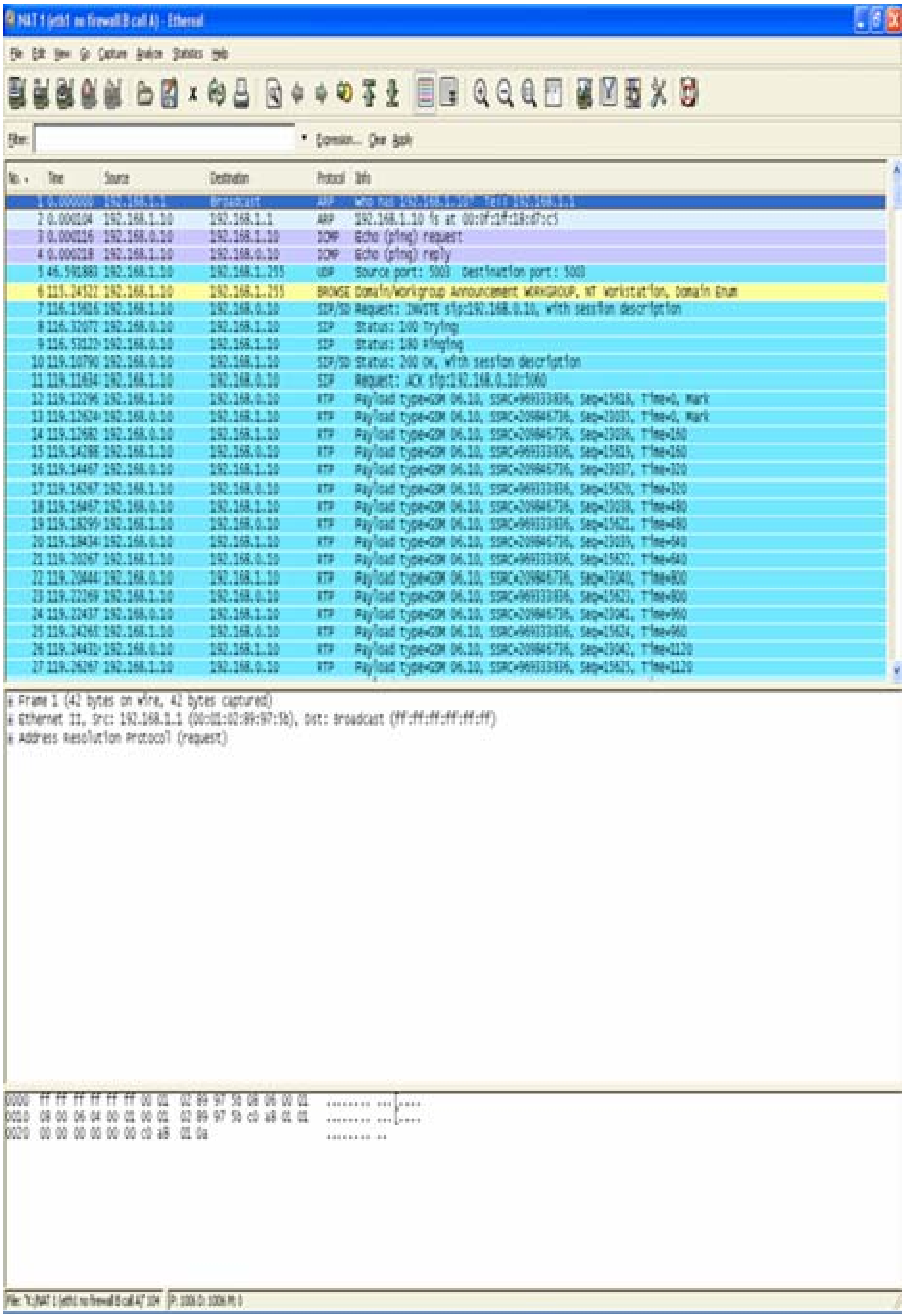


Figure 22. Test 2: Packet Capture on eth1 of NAT (without firewall)

A.4.5. Analysis

Since all packets exchanged between Clients A and B are processed by the NAT rules, understanding those rules is essential when analyzing the traffic flow captured by Ethereal. The SNAT rule "iptables -t nat -A POSTROUTING -o eth0 -j SNAT --to 192.168.0.1" instructed the NAT device to modify the source IP address of all outgoing packets to 192.168.0.1 before routing them. Thus, all the packets received by Client A appeared to come from the NAT device (see packet capture for Client A). The DNAT rule "iptables -t nat -A PREROUTING -i eth0 -j DNAT --to 192.168.1.10" instructed the NAT device to change the destination IP address of all incoming packets to 192.168.1.10 before routing those packets. This operation allowed packets sent to the NAT to be routed to Client B.

The process of establishing a connection between Clients A and B in this test was very similar to the one described in Appendix B. First, Client B sent out an "INVITE" message from 192.168.1.10:5060 to 192.168.0.10:5060 on Client A (red outline in Figure 19). The "INVITE" message had embedded SDP information to inform Client A that Client B will send and receive RTP voice packets at 192.168.1.10 on port 49154 (green outline in Figure 19). Client A acknowledged the invitation by sending Client A an "200 OK" message with embedded SDP information indicating that it will send and receive RTP packets at 192.168.0.l0 on port 49152 (orange outline in Figure 20).

Subsequent voice communication between the two clients was sent to the IP addresses and ports specified in SDP. In other words, Client B sent RTP packets directly to Client A at 192.168.0.10 and Client A sent RTP packets directly to Client B at 192.168.1.10 (blue outline in Figure 19). The former was legitimate because Client A was publicly reachable. But the latter was only possible in our setup since neither Client A nor the NAT device was configured to drop packets destined for private IP addresses. In this case, the NAT device simply forwarded the RTP packets to client B (see Figures 21 and 22). To simulate a more realistic network configuration, a firewall was needed to drop packets sent by Client A and destined for Client B.

B. With Firewall

B.1. Network Topology

Refer to Figure 7 and Figure 8 for the physical and logical network topology.

B.2. Preparation and Testing (in addition to all steps described in Section A)

B.2.1. On Client A,

B.2.1.1. Download the ZoneAlarm from Zone Labs

B.2.1.2. Install ZoneAlarm

B.2.1.3. When ZoneAlarm is being run for the first time, it will ask the user to choose between Basic ZoneAlarm or the trial version of ZoneAlarm Pro, select the trial version of ZoneAlarm

B.2.1.4. When asked to select a security level for the detected network, select Allow into Trusted Zone

B.2.1.5. Configure firewall rule in ZoneAlarm:

B.2.1.5.1. Go to Firewall menu on the left panel

B.2.1.5.2. Click on the Expert tab

B.2.1.5.3. Click on Add

B.2.1.5.4. Type in a name for the firewall rule in the Name textbox

B.2.1.5.5. Under Action, select Block

B.2.1.5.6. Under Destination,

B.2.1.5.6.1. Select Modify

B.2.1.5.6.2. Select Add Location

B.2.1.5.6.3. Select IP Address

B.2.1.5.6.4. Type in a description in the Description textbox

B.2.1.5.6.5. Type 192.168.1.10 in the IP Address textbox

B.2.1.5.6.6. Click OK

B.2.1.5.6.7. Click OK

B.2.1.5.6.8. Click Apply

B.2.2. Run test as described in Section A

### B.3. Packet Captures

#### B.3.1. Client A

Figure 23 is a snapshot of the packets captured on Client A.


Figure 23. Test 2: Packet Capture on Client A (with firewall)

## B.3.2. Client B

Figure 24 is a snapshot of the packets captured on Client B.


Figure 24. Test 2: Packet Capture on Client B (with firewall)

B.3.3. NAT eth0

Figure 25 is a snapshot of the packets captured on the first interface (eth0) of the NAT device.

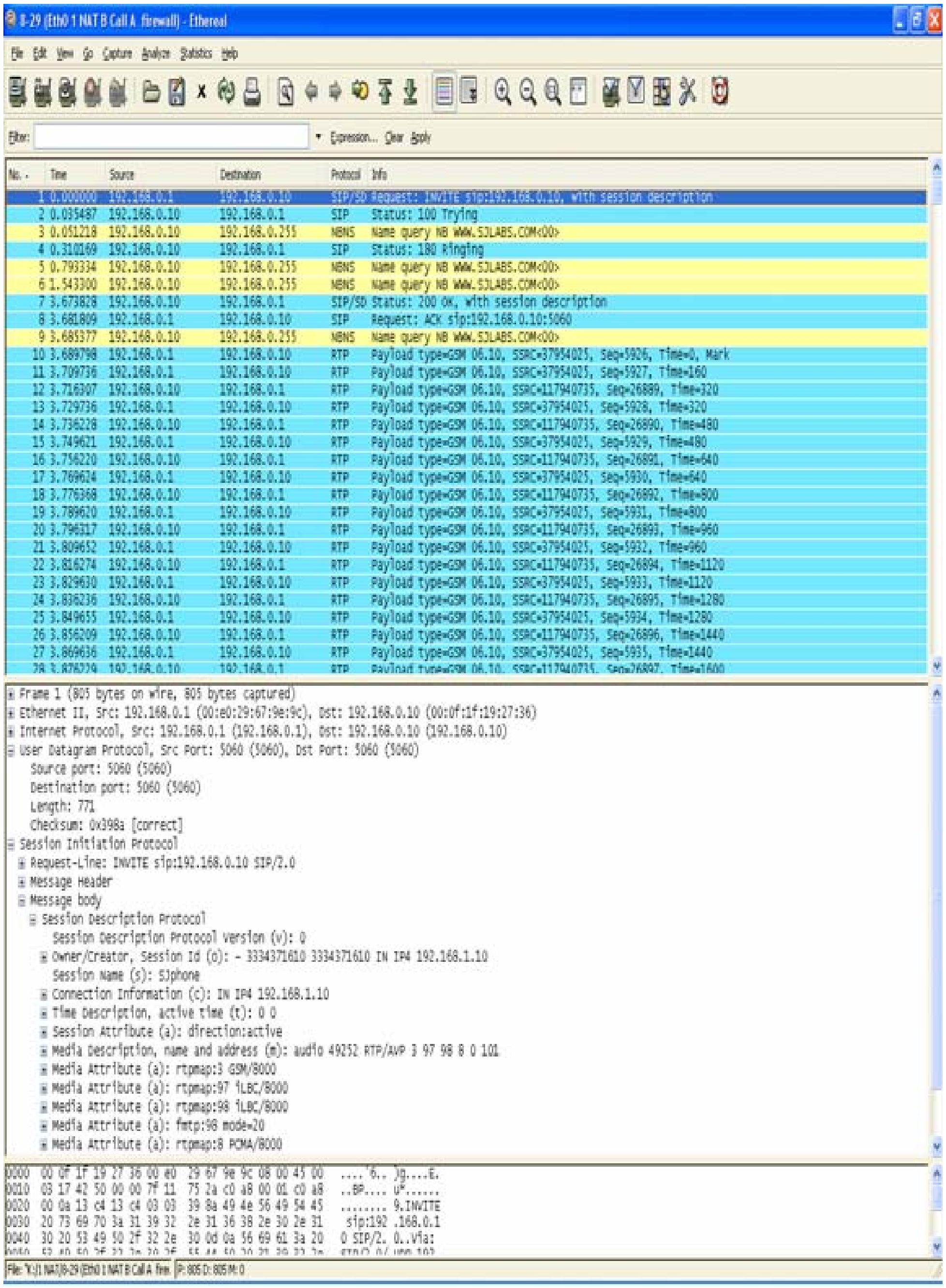


Figure 25. Test 2: Packet Capture on eth0 of NAT (with firewall)

B.3.4. NAT eth1

Figure 26 is a snapshot of the packets captured on the second interface (eth1) of the NAT device.

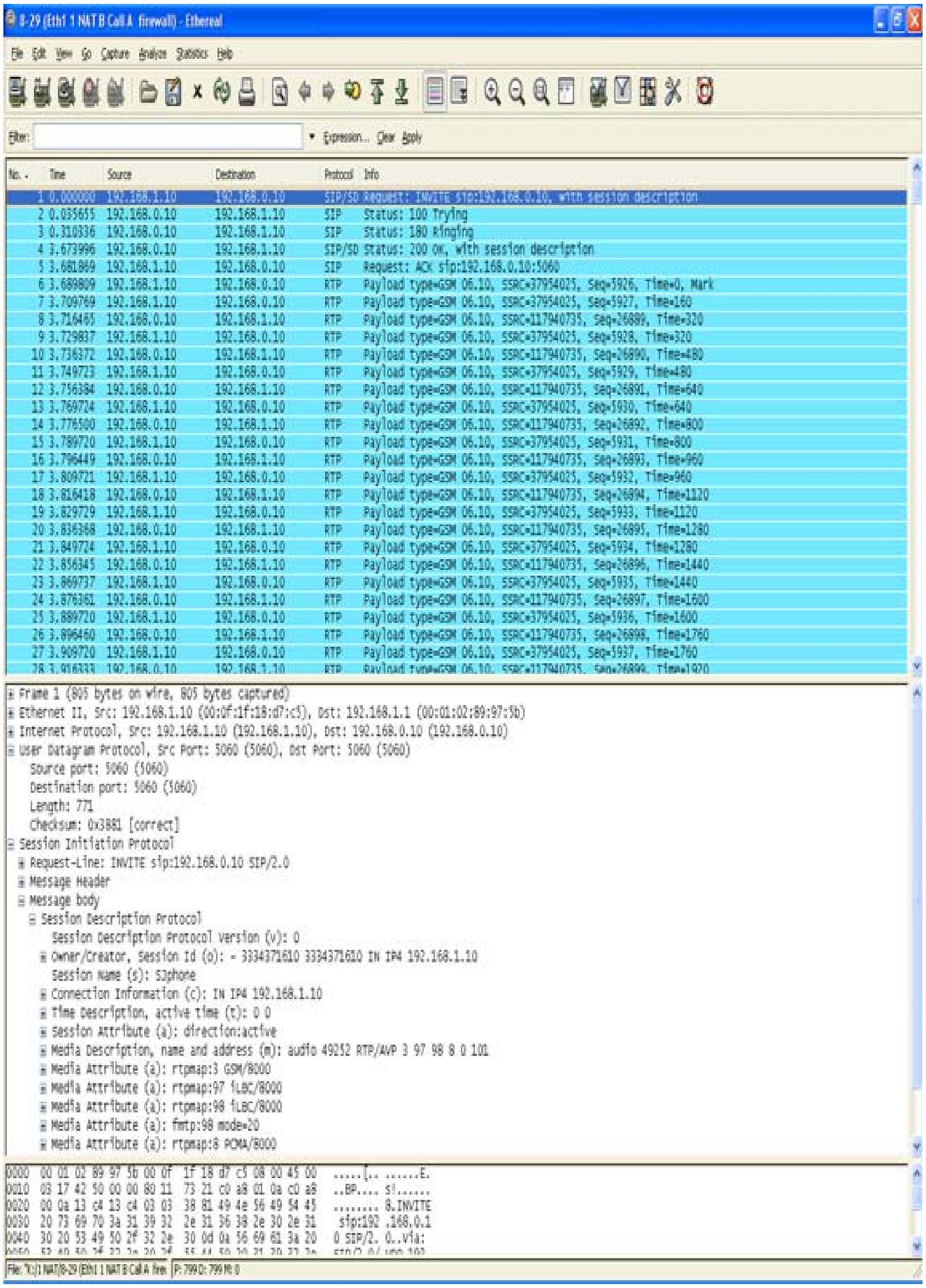



Figure 26. Test 2: Packet Capture on eth1 of NAT (with firewall)

### B.3.5. Analysis

The signaling part of the call was processed as usual with the exchanges of "INVITE" (red outline in Figure 23) and "200 OK" (orange outline in Figure 24) messages between the two clients. Client B still told Client A to send RTP media packets to its private IP address or 192.168.1.10 (green outline in Figure 23). However, Client A can no longer send RTP packets directly to Client B at its private address because ZoneAlarm was configured to block those packets. The ZoneAlarm log files were examined to confirm that packets destined for 192.168.1.10 were, in fact, dropped. The packet captures on Client A indicate that when Client A failed to send RTP packets to the IP address of Client B, it tried to send subsequent RTP packets to the IP address from which it received RTP packets (due to the SNAT rule). In this case, Client A sent the RTP packets to the public IP address NAT or 192.168.0.1 (blue outline in Figure 23). When NAT received the packets, it modified the destination address in the packet header according to the configured DNAT rule. In other words, NAT changed the destination address from its own public IP address (192.168.0.1) to the private IP address of Client B (192.168.1.10) before forwarding the packets (dark red outline in Figure 24). Figures 25 and 26 confirm that DNAT and SNAT were done correctly.

# APPENDIX D. TEST 3: DOUBLE NAT VOIP DEMONSTRATION USING SJPHONE

The instructions contained in this appendix describe how to setup and demonstrate a SIP-based VoIP communication between two SIP-enabled clients via two Network Address Translation (NAT) devices. In this setup, Client B is located behind two NATs. Each NAT is configured to act as a router and modifies the destination or source IP address of all packets that traverses it. Packet captures from both clients are included at the end of this appendix along with an analysis.

A. Network Topology

Refer to Figures 9 and Figure 10 for the physical and logical network topology.

B. Equipment Requirements

B.1. Clients A and B

B.1.1. Windows XP Operating System

B.1.2. Sound card

B.1.3. SJPhone v.1.60

B.1.4. Ethereal

B.1.5. ZoneAlarm (Client A only)

B.2. NAT 1 and NAT 2

B.2.1. Linux Operating System (Fedora Core 4)

B.2.2. netfilter and iptables

B.2.3. Ethereal

B.2.4. Two network cards

B.3. Additional equipment

B.3.1. Cross-over cables to implement the network architecture illustrated in Figure 9

B.3.2. Microphones as audio input devices for clients A and B

C. Installation and Configuration

C.1. Client A

IP Address: 192.168.0.10

Subnet Mask: 255.255.255.0

Default Gateway: 192.168.0.1

C.2. Client B

IP Address: 192.168.2.10

Subnet Mask: 255.255.255.0

Default Gateway: 192.168.2.1

C.3. NAT 1

C.3.1. Configure eth0 by editing /etc/sysconfig/network-scripts/ifcfg-eth0:

```
DEVICE=eth0
BOOTPROTO=NONE
IPADDR=192.168.0.1
NETMASK=255.255.255.0
```

C.3.2. Activate eth0 by running:

```
ifup eth0
```

C.3.3. Configure eth1 by editing /etc/sysconfig/network-scripts/ifcfg-eth1:

```
DEVICE=eth1
BOOTPROTO=NONE
IPADDR=192.168.1.1
NETMASK=255.255.255.0
```

C.3.4. Activate eth1 by running:

```
ifup eth1
```

C.3.5. Enable IP Forwarding by running:

```
echo 1 > /proc/sys/net/ipv4/ip_forward
```

C.3.6. Flush any existing firewall and NAT rules by running:

```
iptables -F
iptables -t nat -F
```

C.3.7. Configure NAT rules by running:

```
iptables -t nat -A POSTROUTING -o eth0 -j SNAT --to 192.168.0.1
iptables -t nat -A PREROUTING -i eth0 -j DNAT --to 192.168.1.2
```

C.4. NAT 2

C.4.1. Configure eth0 by editing /etc/sysconfig/network-scripts/ifcfg-eth0:

DEVICE=eth0

BOOTPROTO=NONE

IPADDR=192.168.1.2

NETMASK=255.255.255.0

GATEWAY=192.168.1.1

C.4.2. Activate eth0 by running:

ifup eth0

C.4.3. Configure eth1 by editing /etc/sysconfig/network-scripts/ifcfg-eth1:

DEVICE=eth1

BOOTPROTO=NONE

IPADDR=192.168.2.1

NETMASK=255.255.255.0

C.4.4. Activate eth1 by running:

ifup eth1

C.4.5. Enable IP Forwarding by running:

echo 1 > /proc/sys/net/ipv4/ip_forward

C.4.6. Flush any existing firewall and NAT rules by running:

iptables -F

iptables -t nat -F

C.4.7. Configure NAT rules by running:

iptables -t nat -A POSTROUTING -o eth0 -j SNAT --to 192.168.1.2

iptables -t nat -A PREROUTING -i eth0 -j DNAT --to 192.168.2.10

C.5. SJPhone Installation and Configuration

C.5.1. Clients A and B

C.5.1.1. Download the Windows version of SJPhone v.1.60 from SJ Labs

C.5.1.2. Install SJPhone v.1.60

C.5.1.3. Launch SJPhone

C.5.1.4. Right-click on SJPhone

C.5.1.5. Right-click

C.5.1.6. Go to Services

C.5.1.7. Select PC-to-PC (SIP)

C.6. Ethereal Installation and Configuration

C.6.1. Clients A and B

C.6.1.1. Download the Windows version of Ethereal v.0.10.12

C.6.1.2. Install Ethereal v.0.10.12 by following on-screen instructions

C.6.2. NAT 1 and NAT 2

C.6.2.1.1. Install Ethereal if it is not already installed

C.6.2.1.1.1. Go to the Desktop menu

C.6.2.1.1.2. Go to System Settings

C.6.2.1.1.3. Go to Add/Remove Applications

C.6.2.1.1.4. Click on Details under System Tools

C.6.2.1.1.5. Find and then check ethereal-gnome

C.6.2.1.1.6. Click on Close

C.6.2.1.1.7. Click on Update

C.6.2.1.1.8. Put in the correct Fedora Core 4 CDs when prompted

C.7. ZoneAlarm Installation and Configuration

C.7.1. On client A,

C.7.1.1. Download the free ZoneAlarm from Zone Labs

C.7.1.2. Install ZoneAlarm by following on-screen instructions

C.7.1.3. When ZoneAlarm is being run for the first time, it will ask the user to choose between Basic ZoneAlarm or trial version of ZoneAlarm Pro, select the trial version of ZoneAlarm

C.7.1.4. Answer on-screen questions

C.7.1.5. When asked to select a security level for the detected network, select Allow into Trusted Zone

C.7.1.6. Configure firewall rule in ZoneAlarm:

C.7.1.6.1. Go to Firewall menu on the left panel

C.7.1.6.2. Click on the Expert tab

C.7.1.6.3. Click on Add

C.7.1.6.4. Type in a name for the firewall rule in the Name textbox

C.7.1.6.5. Under Action, select Block

C.7.1.6.6. Under Destination,

C.7.1.6.6.1. Select Modify

C.7.1.6.6.2. Select Add Location

C.7.1.6.6.3. Select IP Address

C.7.1.6.6.4. Type in a description in the Description textbox

C.7.1.6.6.5. Type 192.168.2.10 in the IP Address textbox

C.7.1.6.6.6. Click OK

C.7.1.6.6.7. Click OK

C.7.1.6.6.8. Click Apply

D. Preparation and Testing

D.1. Adjust volume on both clients accordingly

D.2. Plug microphones into both clients

D.3. On Client A,

D.3.1. Launch Ethereal

D.3.2. Go to the Capture menu

D.3.3. Go to Interfaces

D.3.4. Click on Capture 192.168.0.10

D.4. On Client B,

D.4.1. Launch Ethereal

D.4.2. Go to the Capture menu

D.4.3. Go to Interfaces

D.4.4. Click on Capture 192.168.2.10

D.5. On NAT 1 (Ethereal not installed),

D.5.1. Launch one terminal and run:

tcpdump -n -i eth0

D.5.2. Launch another terminal and run:

tcpdump -n -i eth1

D.6. On NAT 2,

D.6.1. Launch one instance of Ethereal

D.6.1.1. Go to the Capture menu

D.6.1.2. Go to Interfaces

D.6.1.3. Click on Capture Eth0

D.6.2. Launch another instance of Ethereal

D.6.2.1. Go to the Capture menu

D.6.2.2. Go to Interfaces

D.6.2.3. Click on Capture Eth1

D.7. On Client B,

D.7.1. Call A by dialing 192.168.0.10 in SJPhone

D.8. On Client A,

D.8.1. Select Accept in the pop-up dialog box when SJPhone rings

D.8.2. Clients A and B may engage in a VoIP conversation at this point.

D.8.3. Click on the Hang-Up bottom on either SJPhone to terminate call when finished

D.8.4. On NAT 1,

D.8.4.1. Stop tcpdump packet captures by pressing Control-C on the terminals

D.8.5. On Client A, Client B and NAT 2,

D.8.5.1. Stop packet captures by selecting Stop on Ethereal

E. Packet Captures

E.1. Client A

The following is a snapshot of the packets captured on Client A.

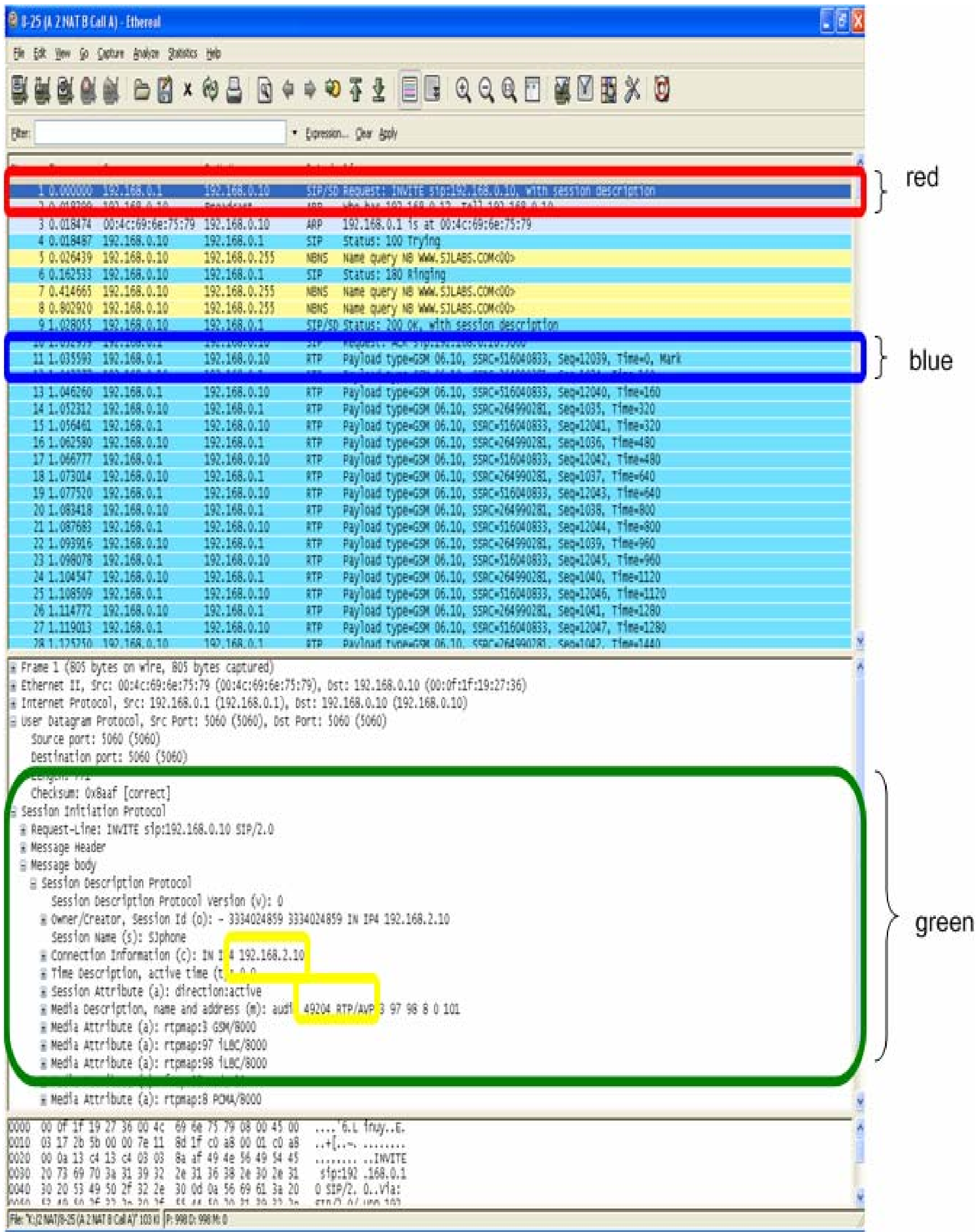



Figure 27. Test 3: Packet Capture on Client A

E.2. Client B

The following is a snapshot of the packets captured on Client B.

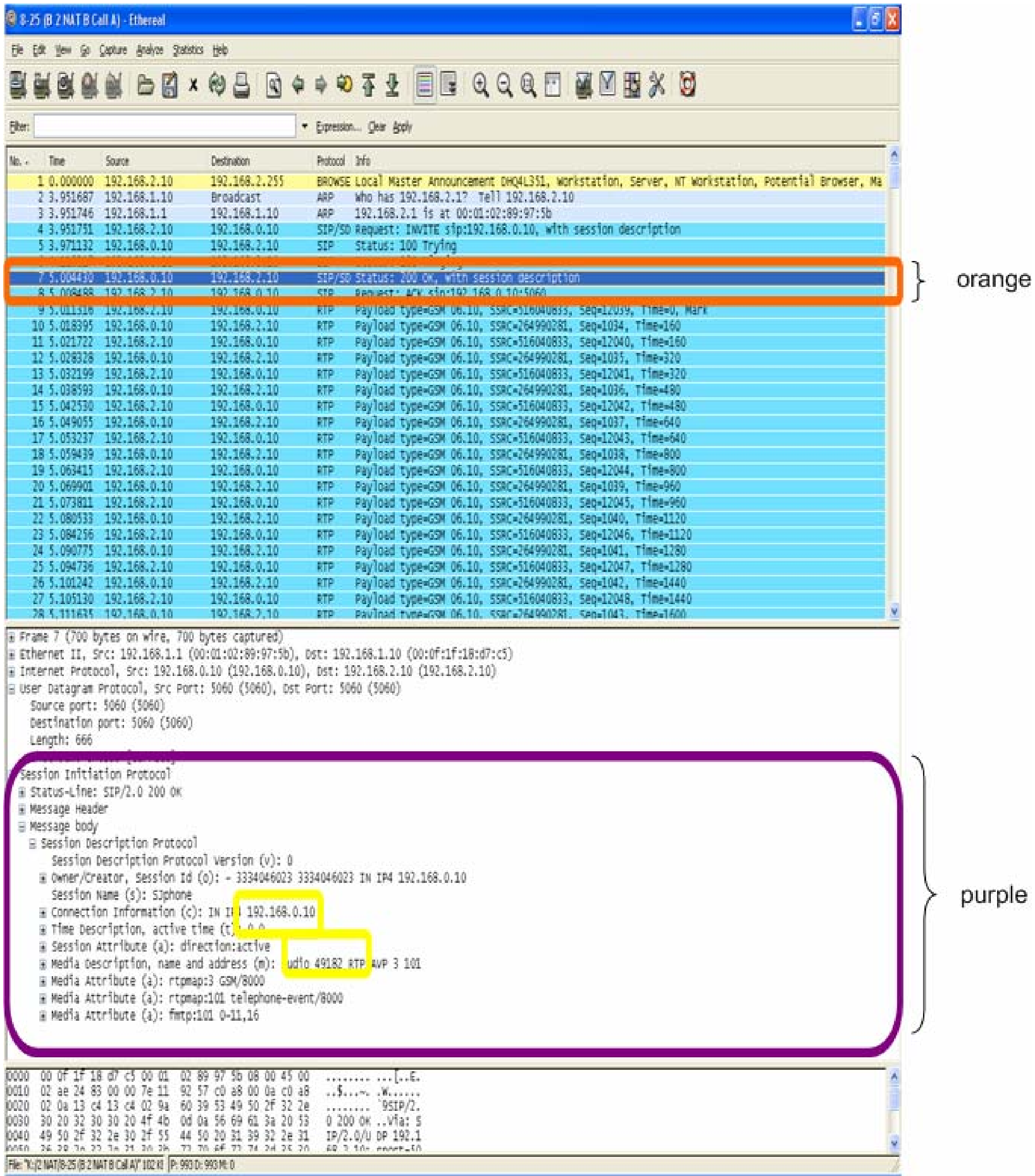
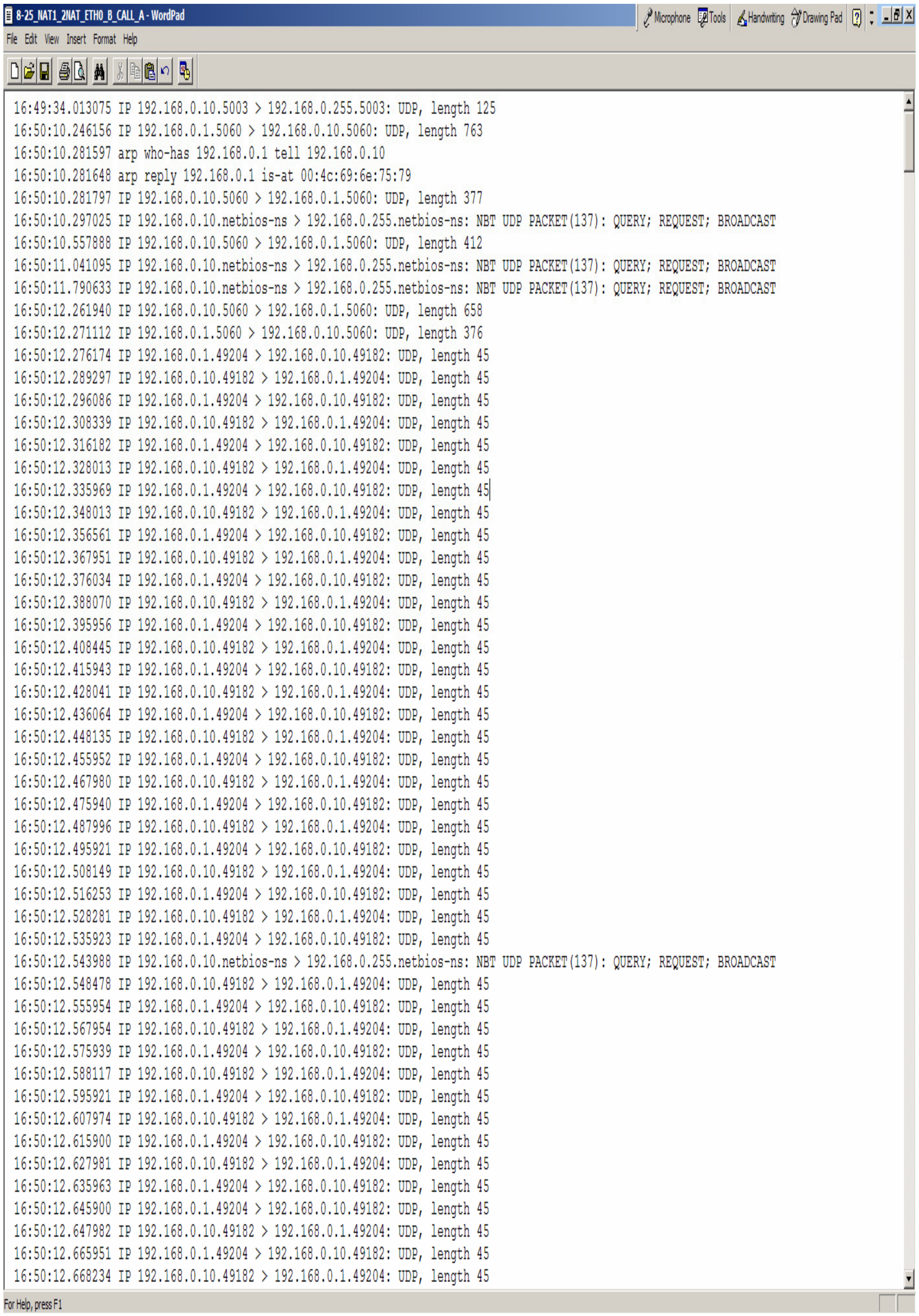



Figure 28. Test 3: Packet Capture on Client B

## E.3. NAT 1 Eth0

The following is a snapshot of the packets captured on the first interface (eth0) of NAT 1.

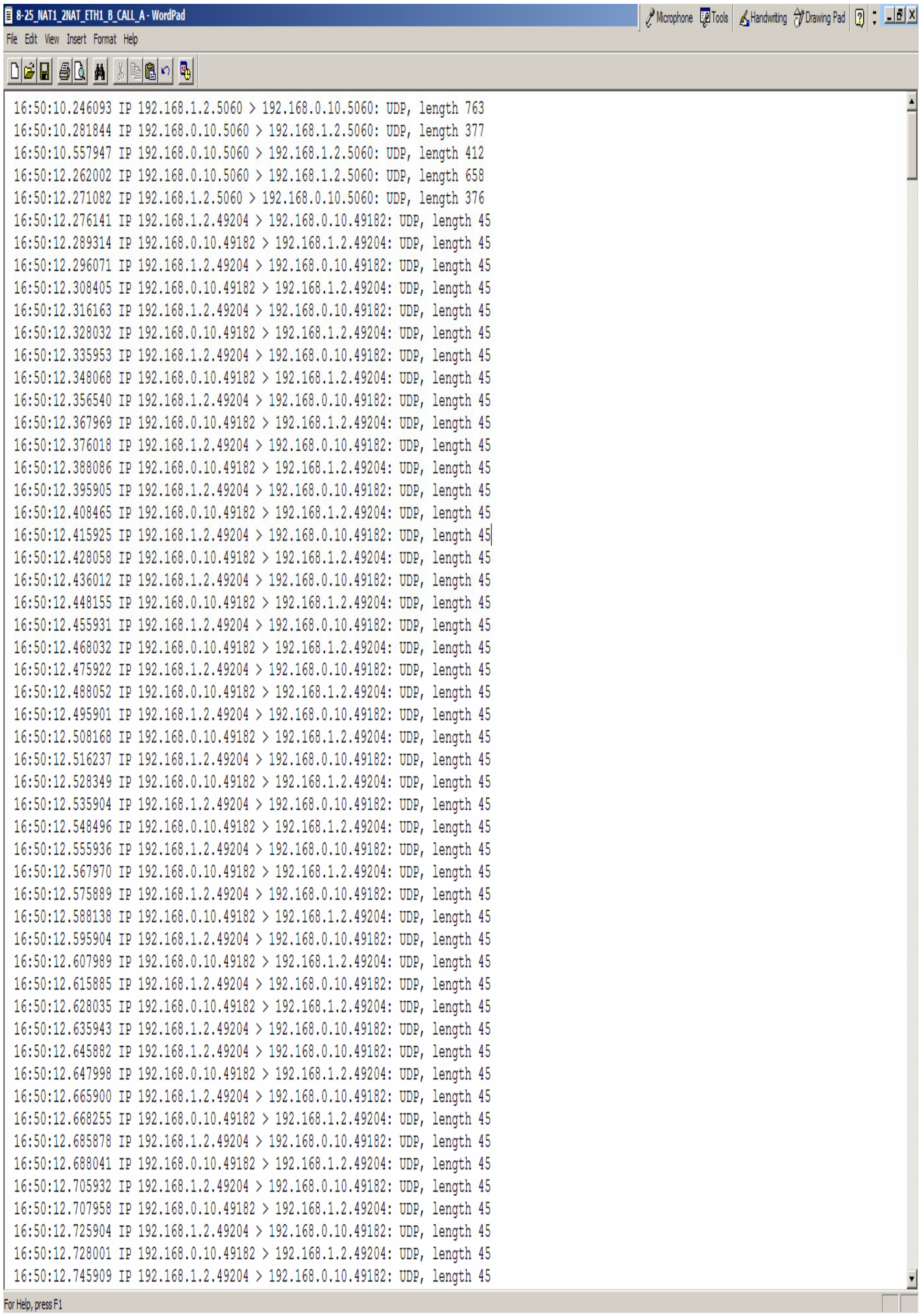


```
16:49:34.013075 IP 192.168.0.10.5003 > 192.168.0.255.5003: UDP, length 125
16:50:10.246156 IP 192.168.0.1.5060 > 192.168.0.10.5060: UDP, length 763
16:50:10.281597 arp who-has 192.168.0.1 tell 192.168.0.10
16:50:10.281648 arp reply 192.168.0.1 is-at 00:4c:69:6e:75:79
16:50:10.281797 IP 192.168.0.10.5060 > 192.168.0.1.5060: UDP, length 377
16:50:10.297025 IP 192.168.0.10.netbios-ns > 192.168.0.255.netbios-ns: NBT UDP PACKET(137): QUERY; REQUEST; BROADCAST
16:50:10.557888 IP 192.168.0.10.5060 > 192.168.0.1.5060: UDP, length 412
16:50:11.041095 IP 192.168.0.10.netbios-ns > 192.168.0.255.netbios-ns: NBT UDP PACKET(137): QUERY; REQUEST; BROADCAST
16:50:11.790633 IP 192.168.0.10.netbios-ns > 192.168.0.255.netbios-ns: NBT UDP PACKET(137): QUERY; REQUEST; BROADCAST
16:50:12.261940 IP 192.168.0.10.5060 > 192.168.0.1.5060: UDP, length 658
16:50:12.271112 IP 192.168.0.1.5060 > 192.168.0.10.5060: UDP, length 376
16:50:12.276174 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.289297 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.296086 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.308339 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.316182 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.328013 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.335969 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.348013 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.356561 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.367951 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.376034 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.388070 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.395956 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.408445 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.415943 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.428041 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.436064 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.448135 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.455952 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.467980 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.475940 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.487996 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.495921 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.508149 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.516253 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.528281 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.535923 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.543988 IP 192.168.0.10.netbios-ns > 192.168.0.255.netbios-ns: NBT UDP PACKET(137): QUERY; REQUEST; BROADCAST
16:50:12.548478 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.555954 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.567954 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.575939 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.588117 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.595921 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.607974 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.615900 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.627981 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.635963 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.645900 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.647982 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
16:50:12.665951 IP 192.168.0.1.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.668234 IP 192.168.0.10.49182 > 192.168.0.1.49204: UDP, length 45
```

Figure 29. Test 3: Packet Capture on eth0 of NAT 1

E.4. NAT 1 Eth1

The following is a snapshot of the packets captured on the second interface (eth1) of NAT 1.



8-25_NAT1_2NAT_ETH1_B_CALL_A - WordPad

File Edit View Insert Format Help

```
16:50:10.246093 IP 192.168.1.2.5060 > 192.168.0.10.5060: UDP, length 763
16:50:10.281844 IP 192.168.0.10.5060 > 192.168.1.2.5060: UDP, length 377
16:50:10.557947 IP 192.168.0.10.5060 > 192.168.1.2.5060: UDP, length 412
16:50:12.262002 IP 192.168.0.10.5060 > 192.168.1.2.5060: UDP, length 658
16:50:12.271082 IP 192.168.1.2.5060 > 192.168.0.10.5060: UDP, length 376
16:50:12.276141 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.289314 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.296071 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.308405 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.316163 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.328032 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.335953 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.348068 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.356540 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.367969 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.376018 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.388086 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.395905 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.408465 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.415925 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.428058 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.436012 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.448155 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.455931 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.468032 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.475922 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.488052 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.495901 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.508168 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.516237 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.528349 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.535904 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.548496 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.555936 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.567970 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.575889 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.588138 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.595904 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.607989 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.615885 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.628035 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.635943 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.645882 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.647998 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.665900 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.668255 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.685878 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.688041 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.705932 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.707958 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.725904 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
16:50:12.728001 IP 192.168.0.10.49182 > 192.168.1.2.49204: UDP, length 45
16:50:12.745909 IP 192.168.1.2.49204 > 192.168.0.10.49182: UDP, length 45
```

For Help, press F1

Figure 30. Test 3: Packet Capture on eth1 of NAT 1

## E.5. NAT 2 Eth0

The following is a snapshot of the packets captured on the first interface (eth0) of NAT 2.

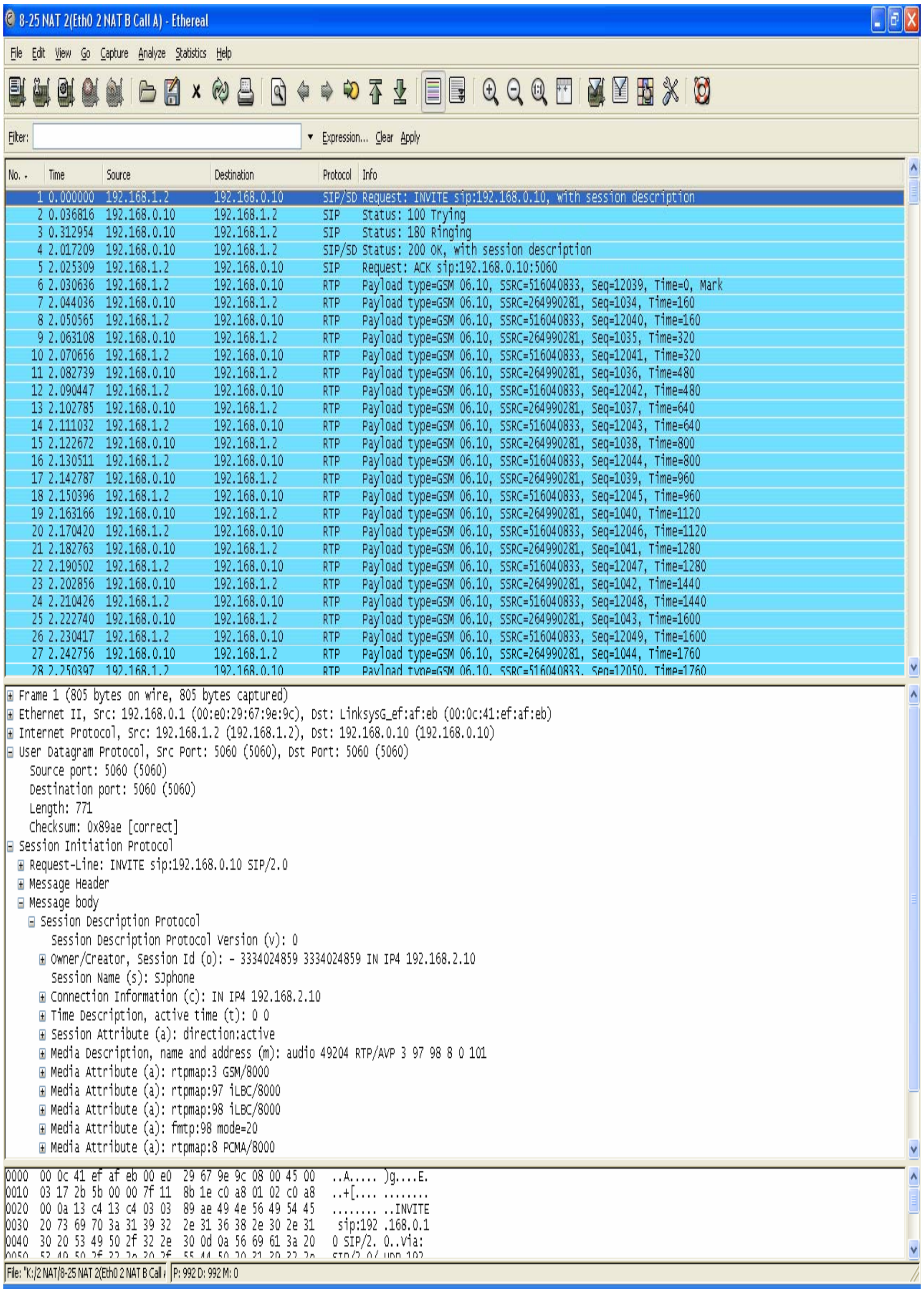


Figure 31. Test 3: Packet Capture on eth0 of NAT 2

E.6. NAT 2 Eth1

The following is a snapshot of the packets captured on the first interface (eth0) of NAT 2.

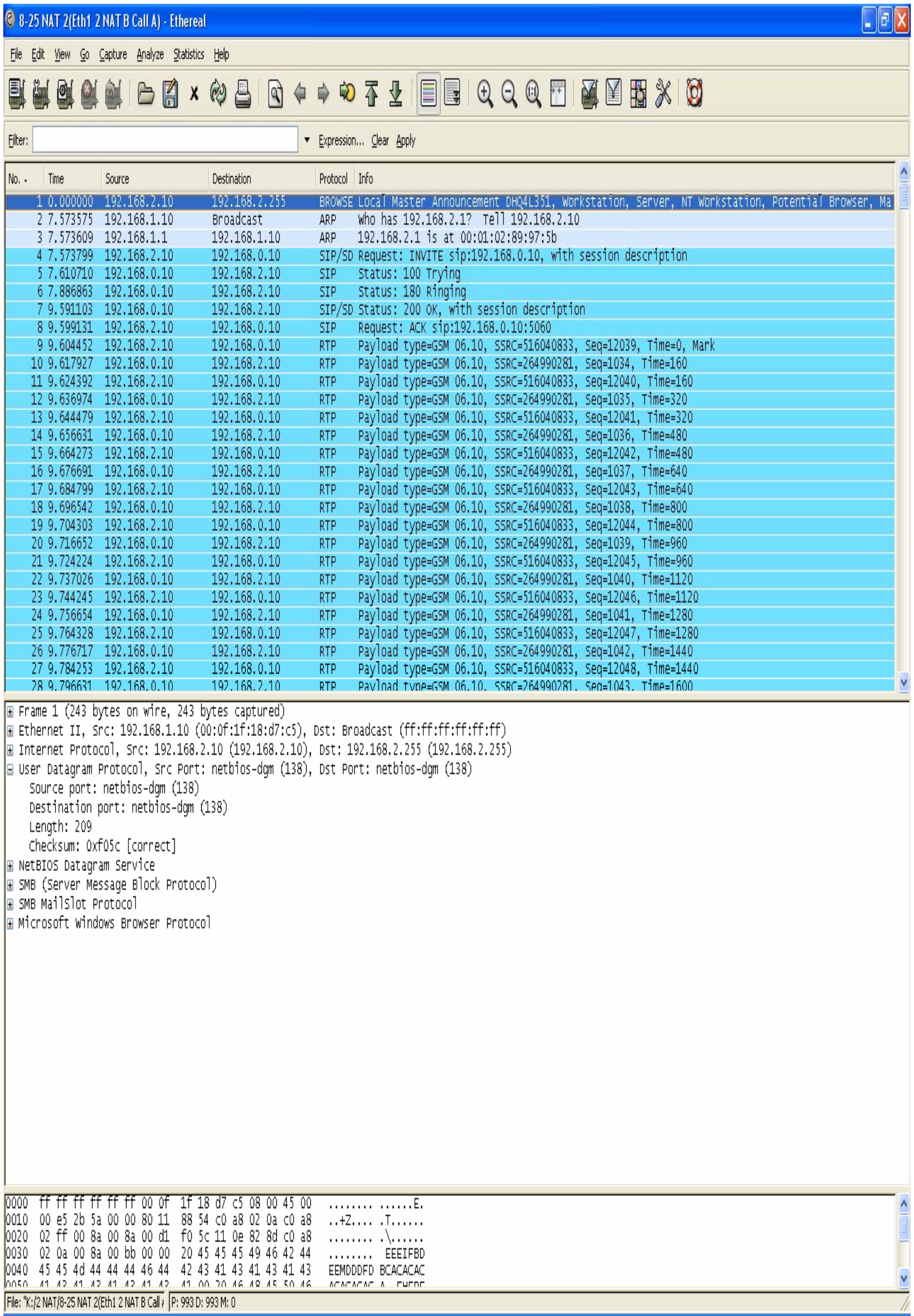



Figure 32. Test 3: Packet Capture on eth1 of NAT 2

### E.7. Analysis

This demonstration was very similar to the one described in Appendix C. It differed in that now Client B is located behind two NAT devices instead of one. In other words, two layers of network address translation occurred before any packet can be moved between Client A and Client B.

The “INVITE” message (red outline in Figure 27) indicated that Client B will be sending and receiving RTP packets at 192.168.2.10 on port 49204 (purple outline in Figure 28). Client A acknowledged the invitation by sending a “200 OK” message to Client B with embedded SDP information indicating that it would send and receive RTP packets at 192.168.2.l0 on port 49182 (green outline in Figure 27). Figures 27 and 29 show that Client A sends and receives RTP packet directly to/from the public IP address of NAT 1. As explained in Appendix C, SJPhone will first attempt to send RTP media packets to the IP address indicated in the SDP message (or the private IP address of Client B). Since the firewall installed on Client A was configured to drop packets destined to for Client B, none of the packets sent out by Client A ever reached Client B. Therefore, Client A resorted to sending subsequent RTP packets to the IP address from which it received Client B’s RTP media packets (blue outline in Figure 27). In this case, the packets were sent to the public IP address of the NAT 1 device (192.168.0.1). Figures 29 and 30 confirmed that the configured NAT operated correctly.

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# APPENDIX E. TEST 4: EXTENDED DOUBLE NAT VOIP DEMONSTRATION USING SJPHONE

The instructions contained in this appendix describe how to setup and demonstrate a SIP-based VoIP communication between two SIP-enabled clients via Network Address Translation (NAT) devices. In this setup, NAT 1 is no longer configured with a DNAT rule to rewrite the destination IP address of the packets that traverse it. Therefore, NAT 1 only performs SNAT. NAT 2 and NAT 3 are each configured with both SNAT and DNAT rules. The demonstration is conducted as follows: Client B initiates a call to Client A. Then Client C initiates a call to Client A after the VoIP session between Client B and A is terminated. Packet captures from all three clients and the NATs are included at the end of this appendix along with an analysis.

A. Network Topology

Refer to Figure 13 and Figure 14 for the physical and logical network topology.

B. Equipment Requirements

B.1. Clients A, B and C

B.1.1. Windows XP Operating System

B.1.2. Sound card

B.1.3. SJPhone v.1.60

B.1.4. Ethereal

B.1.5. ZoneAlarm (Client A only)

B.2. NAT 1, NAT 2 and NAT 3

B.2.1. Linux Operating System (Fedora Core 4)

B.2.2. netfilter and iptables

B.2.3. Ethereal

B.2.4. Two network cards

B.3. Additional Equipment

B.3.1. Cross-over cables and a switch or hub to implement the network architecture illustrated in Figure 13

B.3.2. Microphones as audio input devices for clients A, B, and C

C. Installation and Configuration

C.1. Client A

IP Address: 192.168.0.10

Subnet Mask: 255.255.255.0

Default Gateway: 192.168.0.1

C.2. Client B

IP Address: 192.168.2.10

Subnet Mask: 255.255.255.0

Default Gateway: 192.168.2.1

C.3. Client C

IP Address: 192.168.3.10

Subnet Mask: 255.255.255.0

Default Gateway: 192.168.3.1

C.4. NAT 1

C.4.1. Configure eth0 by editing /etc/sysconfig/network-scripts/ifcfg-eth0:

```
DEVICE=eth0
BOOTPROTO=NONE
IPADDR=192.168.0.1
NETMASK=255.255.255.0
```

C.4.2. Activate eth0 by running:

```
ifup eth0
```

C.4.3. Configure eth1 by editing /etc/sysconfig/network-scripts/ifcfg-eth1:

```
DEVICE=eth1
BOOTPROTO=NONE
IPADDR=192.168.1.1
NETMASK=255.255.255.0
```

C.4.4. Activate eth1 by running:

```
ifup eth1
```

C.4.5. Enable IP Forwarding by running:

```
echo 1 > /proc/sys/net/ipv4/ip_forward
```

C.4.6. Flush any existing firewall and NAT rules by running:

iptables -F

iptables -t nat -F

C.4.7. Configure NAT rule by running:

iptables -t nat -A POSTROUTING -o eth0 -j SNAT --to 192.168.0.1

## C.5. NAT 2

C.5.1. Configure eth0 by editing /etc/sysconfig/network-scripts/ifcfg-eth0:

DEVICE=eth0

BOOTPROTO=NONE

IPADDR=192.168.1.2

NETMASK=255.255.255.0

GATEWAY=192.168.1.1

C.5.2. Activate eth0 by running:

ifup eth0

C.5.3. Configure eth1 by editing /etc/sysconfig/network-scripts/ifcfg-eth1:

DEVICE=eth1

BOOTPROTO=NONE

IPADDR=192.168.2.1

NETMASK=255.255.255.0

C.5.4. Activate eth1 by running:

ifup eth1

C.5.5. Enable IP forwarding by running:

echo 1 > /proc/sys/net/ipv4/ip_forward

C.5.6. Flush any existing firewall and NAT rules by running:

iptables -F

iptables -t nat -F

C.5.7. Configure NAT rules by running:

iptables -t nat -A POSTROUTING -o eth0 -j SNAT --to 192.168.1.2

iptables -t nat -A PREROUTING -i eth0 -j DNAT --to 192.168.2.10

## C.6. NAT 3

C.6.1. Configure eth0 by editing /etc/sysconfig/network-scripts/ifcfg-eth0:

DEVICE=eth0

BOOTPROTO=NONE

IPADDR=192.168.1.3

NETMASK=255.255.255.0

GATEWAY=192.168.1.1

C.6.2. Activate eth0 by running:

ifup eth0

C.6.3. Configure eth1 by editing and saving /etc/sysconfig/network-scripts/ifcfg-eth1:

DEVICE=eth1

BOOTPROTO=NONE

IPADDR=192.168.3.1

NETMASK=255.255.255.0

C.6.4. Activate eth1 by running:

ifup eth1

C.6.5. Enable IP forwarding by running:

echo 1 > /proc/sys/net/ipv4/ip_forward

C.6.6. Flush any existing firewall and NAT rules by running:

iptables –F

iptables -t nat -F

C.6.7. Configure NAT rules by running:

iptables -t nat -A POSTROUTING -o eth0 -j SNAT --to 192.168.1.3

iptables -t nat -A PREROUTING -ieth0 -j DNAT --to 192.168.3.10

C.7. SJPhone Installation and Configuration

C.7.1. Clients A, B, and C

C.7.1.1. Download the Windows version of SJPhone v.1.60 from SJ Labs

C.7.1.2. Install SJPhone v.1.60

C.7.1.3. Launch SJPhone

C.7.1.3.1. On SJPhone,

C.7.1.3.1.1. Right-click

C.7.1.3.1.2. Go to Services

C.7.1.3.1.3. Select PC-to-PC (SIP)

C.8. Ethereal Installation and Configuration

C.8.1. Clients A, B, and C

C.8.1.1. Download the Windows version of Ethereal v.0.10.12

C.8.1.2. Install Ethereal v.0.10.12

C.8.2. NAT 2 and 3

C.8.2.1. Install Ethereal if it is not already installed

C.8.2.1.1. Go to the Desktop menu

C.8.2.1.2. Go to System Settings

C.8.2.1.3. Go to Add/Remove Applications

C.8.2.1.4. Click on Details under System Tools

C.8.2.1.5. Find and then check ethereal-gnome

C.8.2.1.6. Click on Close

C.8.2.1.7. Click on Update

C.8.2.1.8. Put in the correct Fedora Core 4 CDs when prompted

C.8.2.1.9.

C.9. ZoneAlarm Installation and Configuration

C.9.1. On client A,

C.9.1.1. Download the free ZoneAlarm from Zone Labs

C.9.1.2. Install ZoneAlarm by following on-screen instructions

C.9.1.3. When ZoneAlarm is being run for the first time, it will ask the user to choose between Basic ZoneAlarm or trial version of ZoneAlarm Pro, select the trial version of ZoneAlarm

C.9.1.4. Answer on-screen questions

C.9.1.5. When asked to select a security level for the detected network, select Allow into Trusted Zone

C.9.1.6. Configure firewall rule in ZoneAlarm:

C.9.1.6.1. Go to Firewall menu on the left panel

C.9.1.6.2. Click on the Expert tab

C.9.1.6.3. Click on Add

C.9.1.6.4. Type in a name for the firewall rule in the Name textbox

C.9.1.6.5. Under Action, select Block

C.9.1.6.6. Under Destination,

C.9.1.6.6.1. Select Modify

C.9.1.6.6.2. Select Add Location

C.9.1.6.6.3. Select IP Address

C.9.1.6.6.4. Type in a description in the Description textbox

C.9.1.6.6.5. Type 192.168.2.10 in the IP Address textbox

C.9.1.6.6.6. Click OK

C.9.1.6.7. Repeat steps C.9.1.6.1 to C.9.1.6.6.6 to create a rule to block 192.168.3.10

C.9.1.6.8. Click OK

C.9.1.6.9. Click Apply

D. Preparation and Testing

D.1. Adjust volume on both clients accordingly

D.2. Plug microphones into both clients

D.3. On Client A,

D.3.1. Launch Ethereal

D.3.2. Go to the Capture menu

D.3.3. Go to Interfaces

D.3.4. Click on Capture 192.168.0.10

D.4. On Client B,

D.4.1. Launch Ethereal

D.4.2. Go to the Capture menu

D.4.3. Go to Interfaces

D.4.4. Click on Capture 192.168.2.10

D.5. On NAT 1,

D.5.1. Launch one terminal and run:

```
tcpdump -n -i eth0
```

D.5.2. Launch another terminal and run:

```
tcpdump -n -i eth1
```

D.6. On NAT 2 and NAT 3,

D.6.1. Launch one instance of Ethereal

D.6.1.1. Go to the Capture menu

D.6.1.2. Go to Interfaces

D.6.1.3. Click on Capture Eth0

D.6.2. Launch another instance of Ethereal

D.6.2.1. Go to the Capture menu

D.6.2.2. Go to Interfaces

D.6.2.3. Click on Capture Eth1

D.7. On Client B,

D.7.1. Call A by dialing 192.168.0.10 in SJPhone

D.8. On Client A,

D.8.1. Select Accept in the pop-up dialog box when SJPhone rings

D.8.2. Clients A and B may engage in a VoIP conversation at this point.

D.8.3. Click on the Hang-Up bottom on either SJPhone to terminate call when finished

D.9. On Client A,

Stop tcpdump packet captures by pressing Control-C

D.10. On Client A, Client B, NAT 1, NAT 2 and NAT 3,

D.10.1. Stop packet captures by selecting Stop on Ethereal

D.10.2. Stop packet captures on NAT Box 1 by pressing Control-C

D.10.3. Repeat steps D.3 to D.8 for Clients A, Client C , NATs 1 and NAT 3

E. Packet Captures

E.1. Client B Calls A

E.1.1. Client A

The following is a snapshot of the packets captured on Client A when Client B calls Client A.

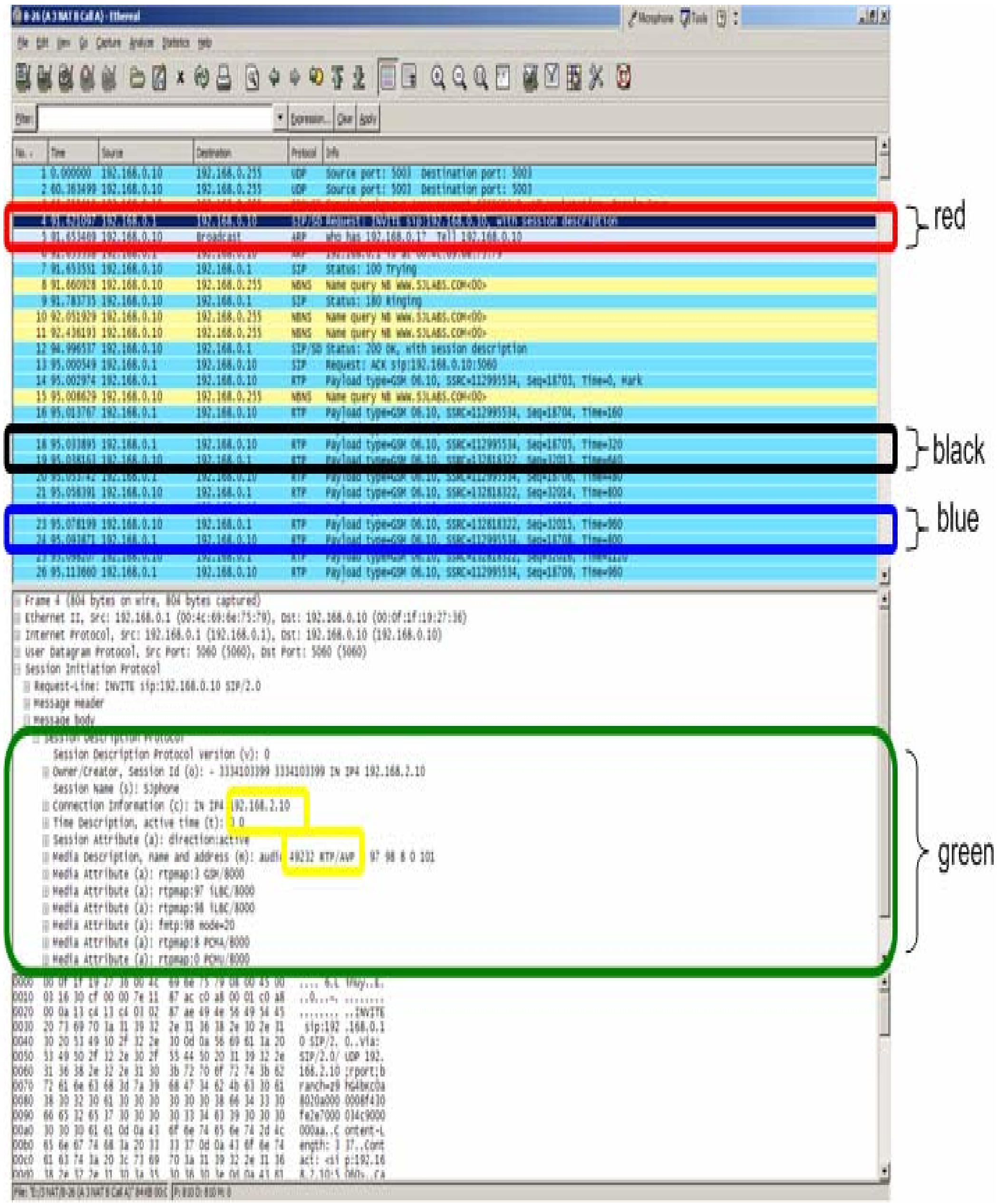



Figure 33. Test 4: Packet Capture on Client A (Client B Calls Client A)

E.1.2. Client B

The following is a snapshot of the packets captured on Client B when Client B calls Client A.

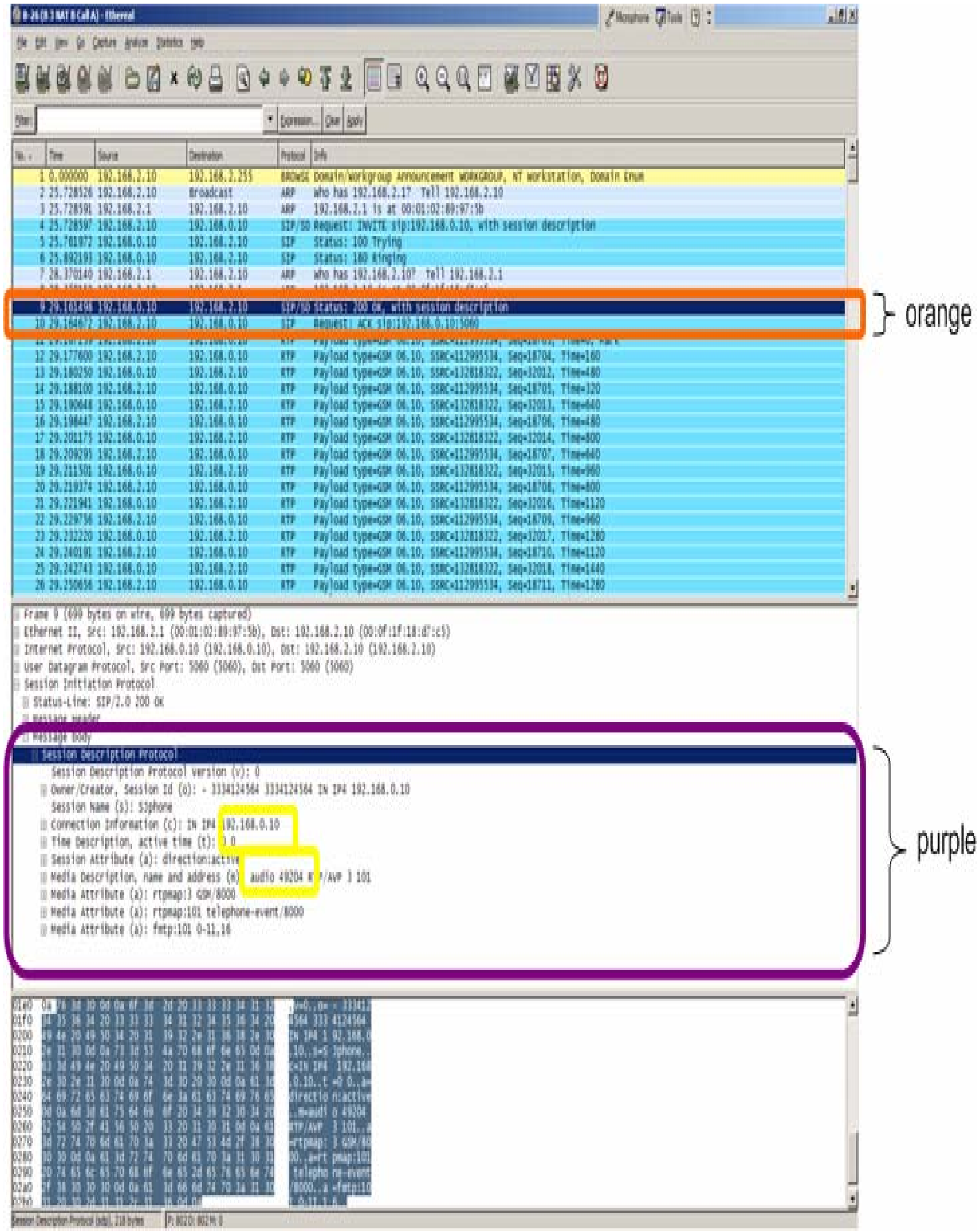
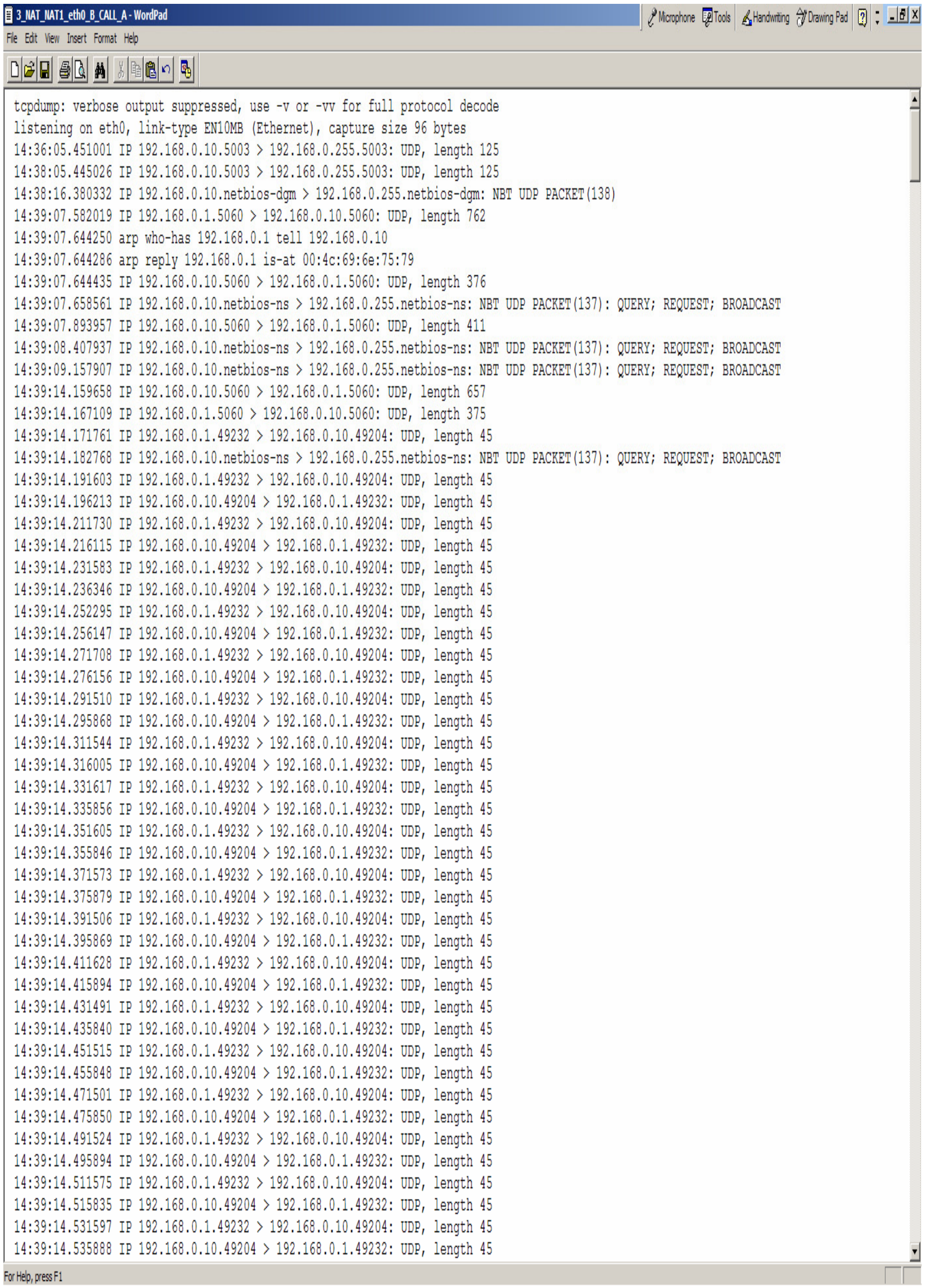



Figure 34. Test 4: Packet Capture on Client B (Client B Calls Client A)

### E.1.3. NAT 1 Eth0

The following is a snapshot of the packets captured on the first interface (eth0) of NAT 1 when Client B calls Client A.

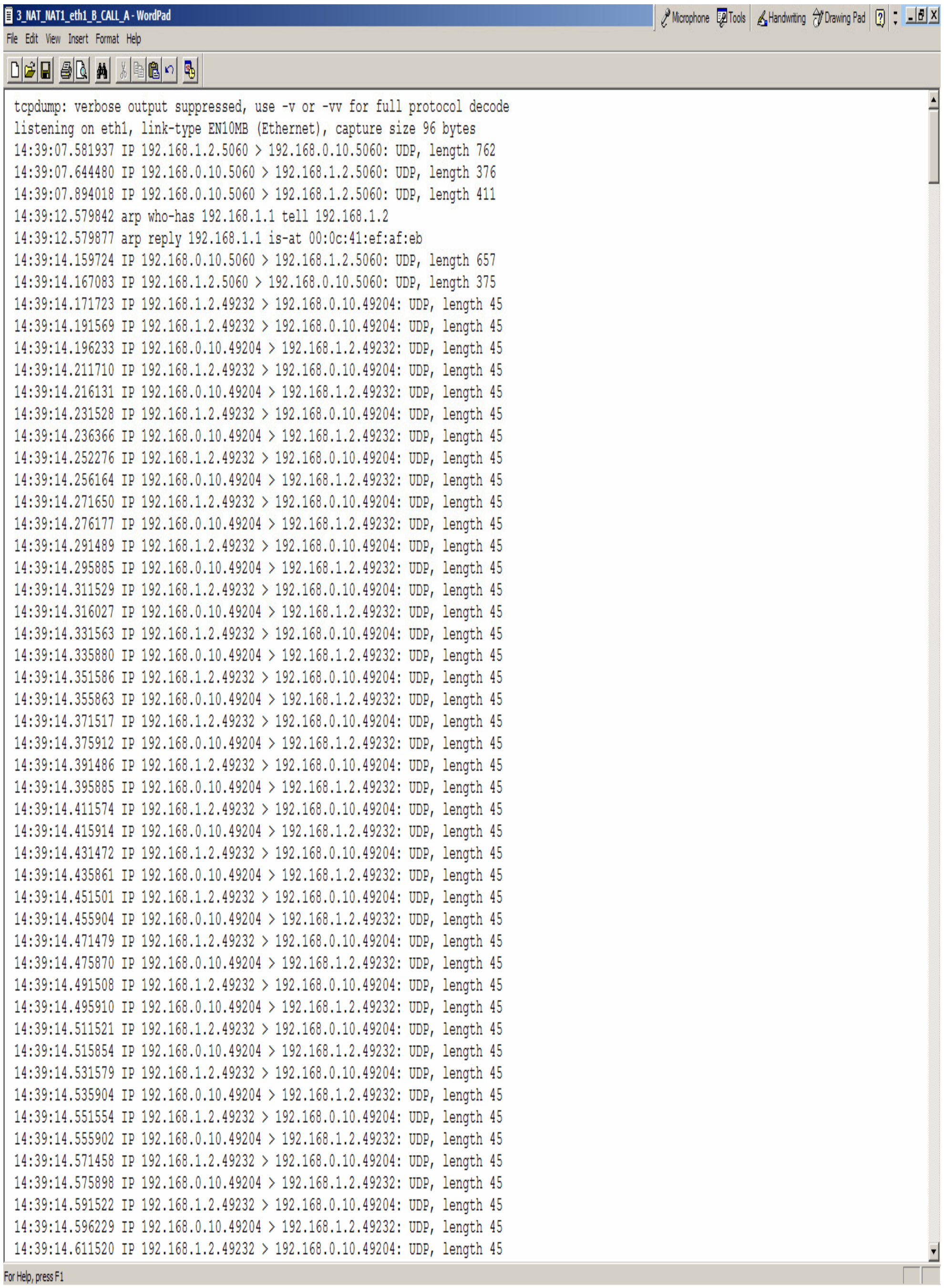


```
tcpdump: verbose output suppressed, use -v or -vv for full protocol decode
listening on eth0, link-type EN10MB (Ethernet), capture size 96 bytes
14:36:05.451001 IP 192.168.0.10.5003 > 192.168.0.255.5003: UDP, length 125
14:38:05.445026 IP 192.168.0.10.5003 > 192.168.0.255.5003: UDP, length 125
14:38:16.380332 IP 192.168.0.10.netbios-dgm > 192.168.0.255.netbios-dgm: NBT UDP PACKET(138)
14:39:07.582019 IP 192.168.0.1.5060 > 192.168.0.10.5060: UDP, length 762
14:39:07.644250 arp who-has 192.168.0.1 tell 192.168.0.10
14:39:07.644286 arp reply 192.168.0.1 is-at 00:4c:69:6e:75:79
14:39:07.644435 IP 192.168.0.10.5060 > 192.168.0.1.5060: UDP, length 376
14:39:07.658561 IP 192.168.0.10.netbios-ns > 192.168.0.255.netbios-ns: NBT UDP PACKET(137): QUERY; REQUEST; BROADCAST
14:39:07.893957 IP 192.168.0.10.5060 > 192.168.0.1.5060: UDP, length 411
14:39:08.407937 IP 192.168.0.10.netbios-ns > 192.168.0.255.netbios-ns: NBT UDP PACKET(137): QUERY; REQUEST; BROADCAST
14:39:09.157907 IP 192.168.0.10.netbios-ns > 192.168.0.255.netbios-ns: NBT UDP PACKET(137): QUERY; REQUEST; BROADCAST
14:39:14.159658 IP 192.168.0.10.5060 > 192.168.0.1.5060: UDP, length 657
14:39:14.167109 IP 192.168.0.1.5060 > 192.168.0.10.5060: UDP, length 375
14:39:14.171761 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.182768 IP 192.168.0.10.netbios-ns > 192.168.0.255.netbios-ns: NBT UDP PACKET(137): QUERY; REQUEST; BROADCAST
14:39:14.191603 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.196213 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.211730 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.216115 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.231583 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.236346 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.252295 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.256147 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.271708 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.276156 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.291510 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.295868 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.311544 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.316005 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.331617 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.335856 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.351605 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.355846 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.371573 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.375879 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.391506 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.395869 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.411628 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.415894 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.431491 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.435840 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.451515 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.455848 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.471501 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.475850 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.491524 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.495894 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.511575 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.515835 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
14:39:14.531597 IP 192.168.0.1.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.535888 IP 192.168.0.10.49204 > 192.168.0.1.49232: UDP, length 45
```

Test 4: Packet Capture on eth0 of NAT 1 (Client B Calls Client A)

E.1.4. NAT 1 Eth1

The following is a snapshot of the packets captured on the second interface (eth1) of NAT 1 when Client B calls Client A.




```
tcpdump: verbose output suppressed, use -v or -vv for full protocol decode
listening on eth1, link-type EN10MB (Ethernet), capture size 96 bytes
14:39:07.581937 IP 192.168.1.2.5060 > 192.168.0.10.5060: UDP, length 762
14:39:07.644480 IP 192.168.0.10.5060 > 192.168.1.2.5060: UDP, length 376
14:39:07.894018 IP 192.168.0.10.5060 > 192.168.1.2.5060: UDP, length 411
14:39:12.579842 arp who-has 192.168.1.1 tell 192.168.1.2
14:39:12.579877 arp reply 192.168.1.1 is-at 00:0c:41:ef:af:eb
14:39:14.159724 IP 192.168.0.10.5060 > 192.168.1.2.5060: UDP, length 657
14:39:14.167083 IP 192.168.1.2.5060 > 192.168.0.10.5060: UDP, length 375
14:39:14.171723 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.191569 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.196233 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.211710 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.216131 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.231528 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.236366 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.252276 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.256164 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.271650 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.276177 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.291489 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.295885 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.311529 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.316027 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.331563 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.335880 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.351586 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.355863 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.371517 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.375912 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.391486 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.395885 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.411574 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.415914 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.431472 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.435861 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.451501 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.455904 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.471479 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.475870 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.491508 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.495910 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.511521 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.515854 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.531579 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.535904 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.551554 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.555902 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.571458 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.575898 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.591522 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
14:39:14.596229 IP 192.168.0.10.49204 > 192.168.1.2.49232: UDP, length 45
14:39:14.611520 IP 192.168.1.2.49232 > 192.168.0.10.49204: UDP, length 45
```

Figure 35. Test 4: Packet Capture on eth1 of NAT 1 (Client B Calls Client A)

### E.1.5. NAT 2 Eth0

The following is a snapshot of the packets captured on the first interface (eth0) of NAT 2 when Client B calls Client A.

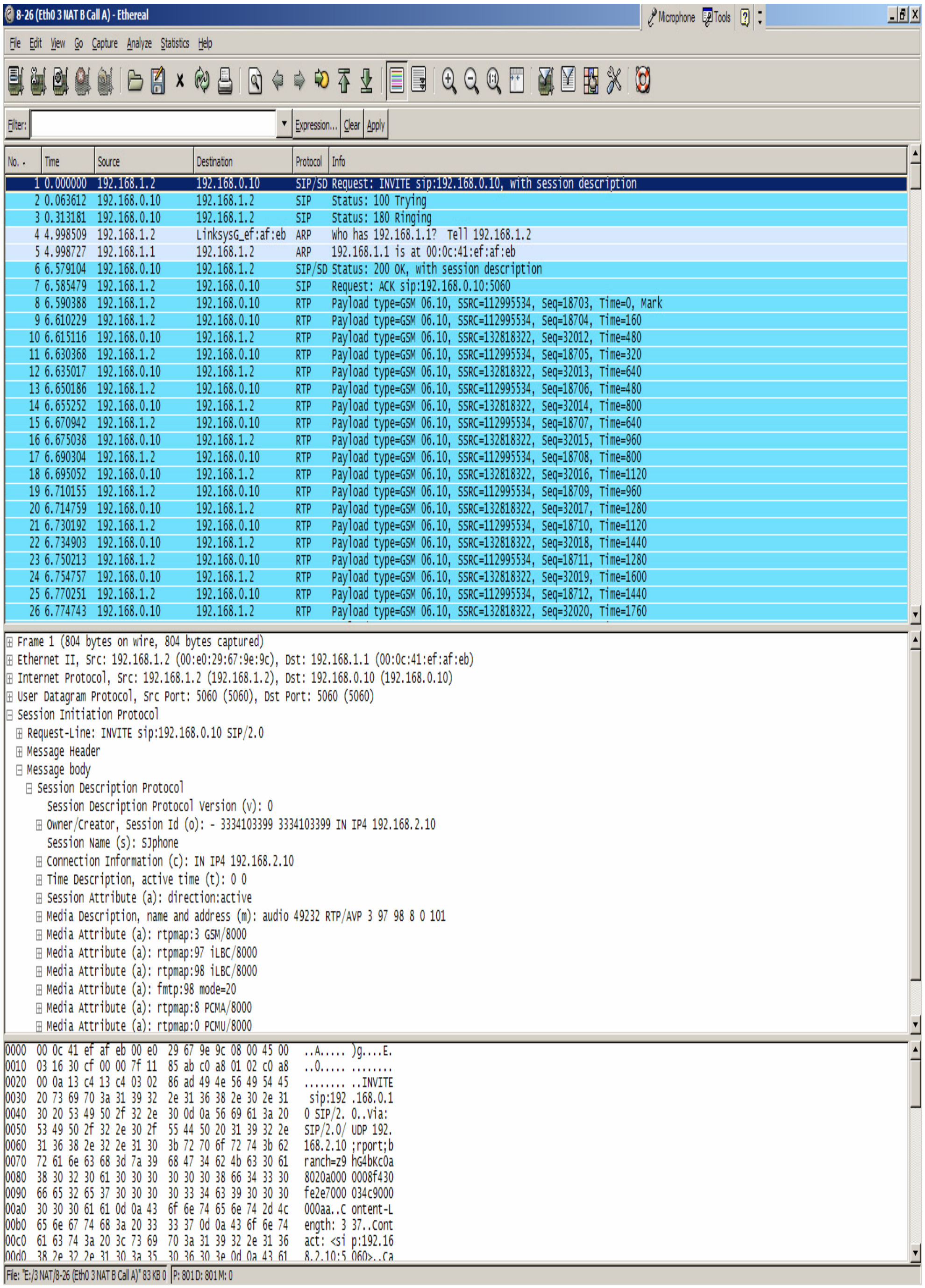


Figure 36. Test 4: Packet Capture on eth0 of NAT 2 (Client B Calls Client A)

E.1.6. NAT 2 Eth1

The following is a snapshot of the packets captured on the second interface (eth1) of NAT 2 when Client B calls Client A.

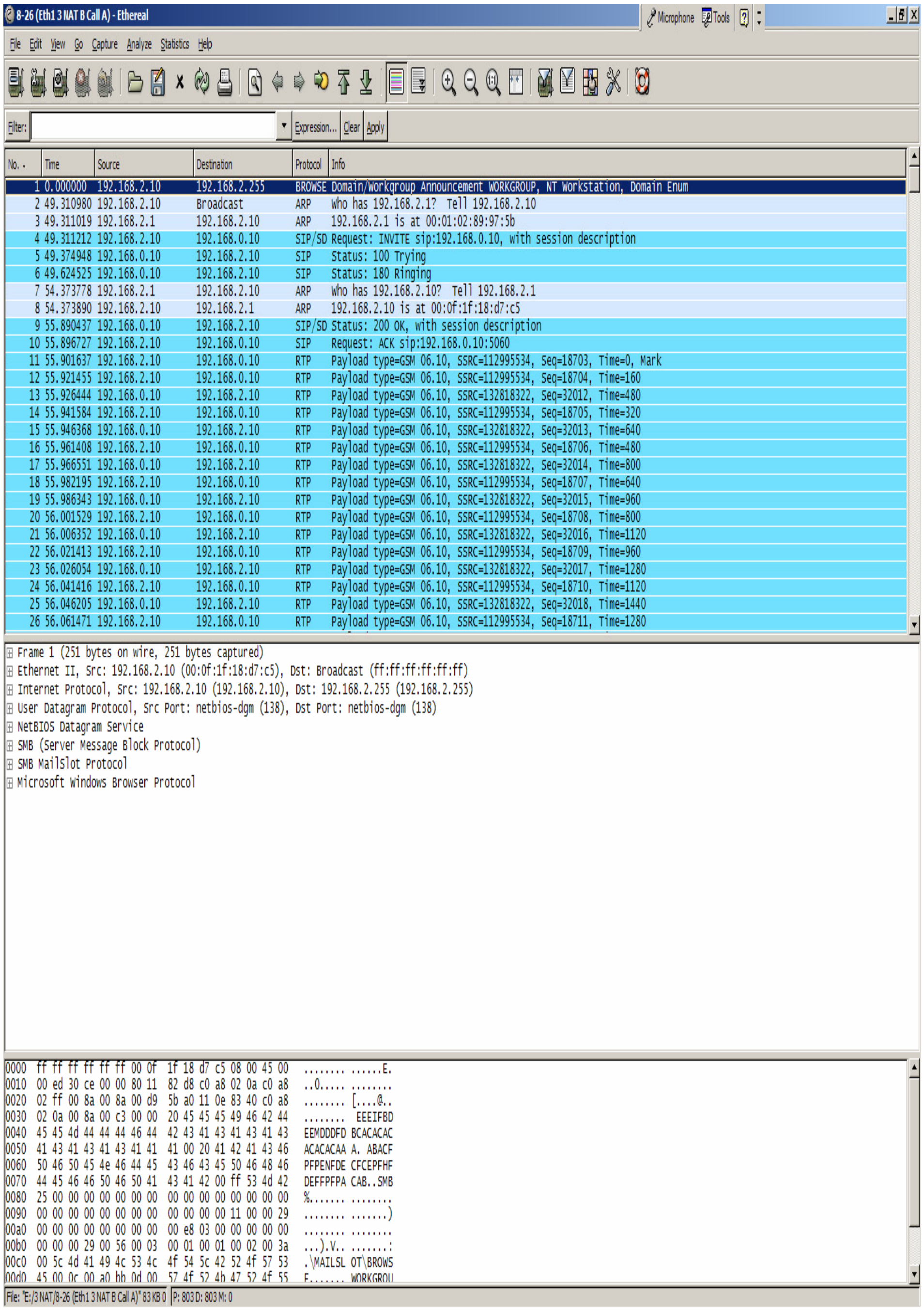


Figure 37. Test 4: Packet Capture on eth1 of NAT 2 (Client B Calls Client A)

### E.1.7. Analysis

As soon as Client B dialed the IP address of Client A, Client B sent out an "INVITE" message to Client A (red outline in Figure 33). The message had embedded SDP information to inform Client A that Client B would be sending and receiving RTP packets at 192.168.2.10 on port 49232 (purple outline in Figure 34). To acknowledge the invitation, Client A sent a "200 OK" packet to Client B with embedded SDP information indicating that it would send and receive RTP packets at 192.168.2.l0 on port 49204 (green outline in Figure 33). The packets captured on Client A indicate that Client A sent and received RTP packet directly to/from the public IP address of NAT 1. As explained in Appendix C, SJPhone will first attempt to send RTP media packets to the IP address indicated in the SDP message (or the private IP address of Client B). Since the firewall installed on Client A was configured to drop packets destined to the private IP address of Client B, none of the packets sent out by Client A could reach Client B. Therefore, Client A then sent subsequent RTP packets to the IP address in which received the RTP media packets (blue outline in Figure 34)

The packet captures indicate that the first RTP packet is sent by Client B (black outline in Figure 33). Even though NAT 1 was not explicitly configured to rewrite the destination IP address of incoming packets to 192.168.1.2 (public IP address of NAT 2), NAT 1 intelligently does this on its own. A reasonable explanation for this behavior is that *iptables* in NAT 1 maintained information for packets that are initiated from the local network [19]. In our scenario, the first RTP packet is processed according to the SNAT rule when it arrives at NAT 1. At the same time, NAT 1 created an entry in its connection tracking table to store essential information (such as source and destination IP addresses and ports) that would allow it to associate incoming packets with the session between Client A and Client B. Packets determined to belong to a certain packet previously received from NAT 2 (due to SNAT) had their destination IP address changed to the public IP address of NAT 2 (refer to Figures 35 through 38). This is evident from observing the packet captures on eth1 of NAT 2.

## E.2. Client C Calls Client A

### E.2.1. Client A

The following is a snapshot of the packets captured on Client A when Client C calls Client A.

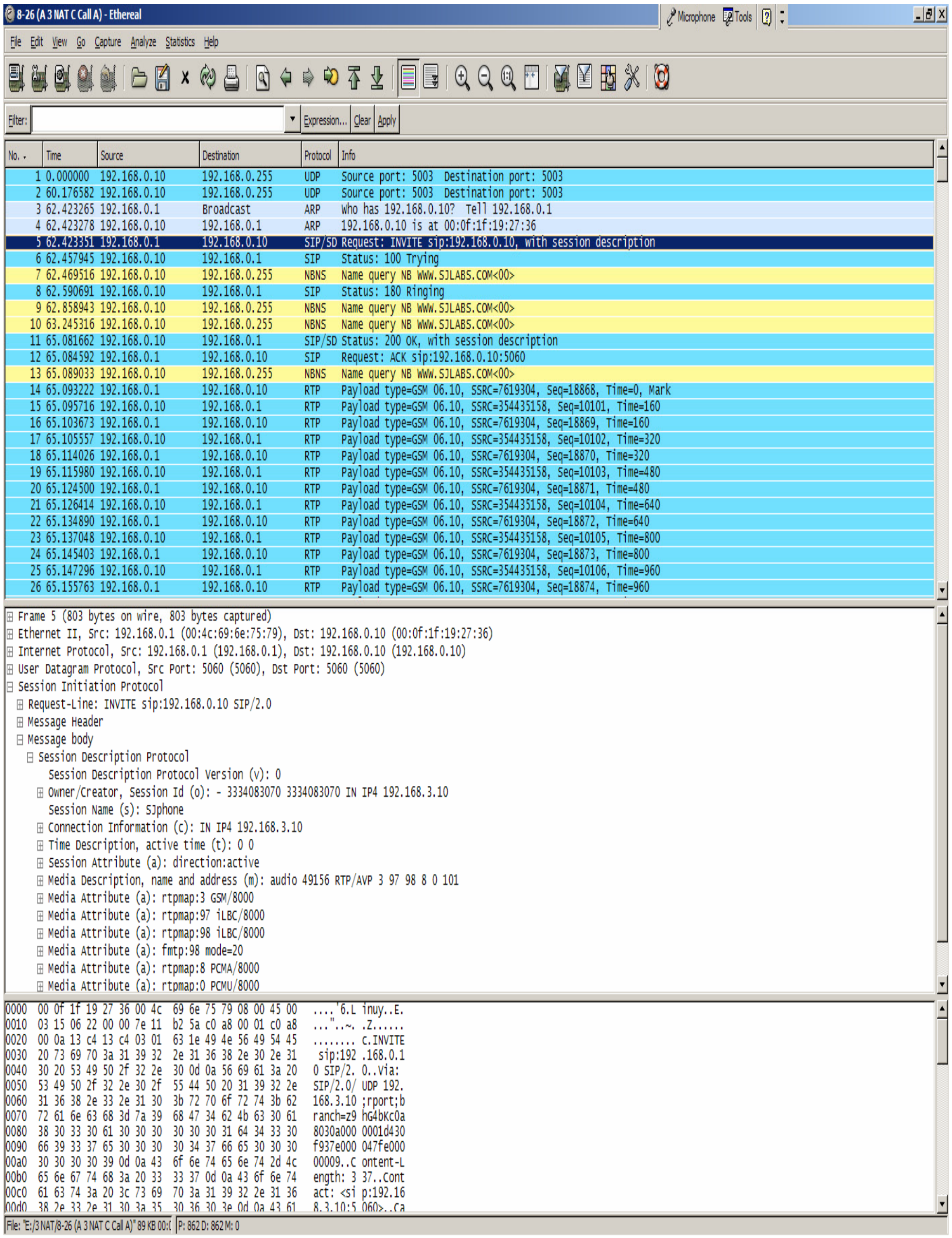



Figure 38. Test 4: Packet Capture on Client A (Client C Calls Client A)

E.2.2. Client C

The following is a snapshot of the packets captured on Client C when Client C calls Client A.

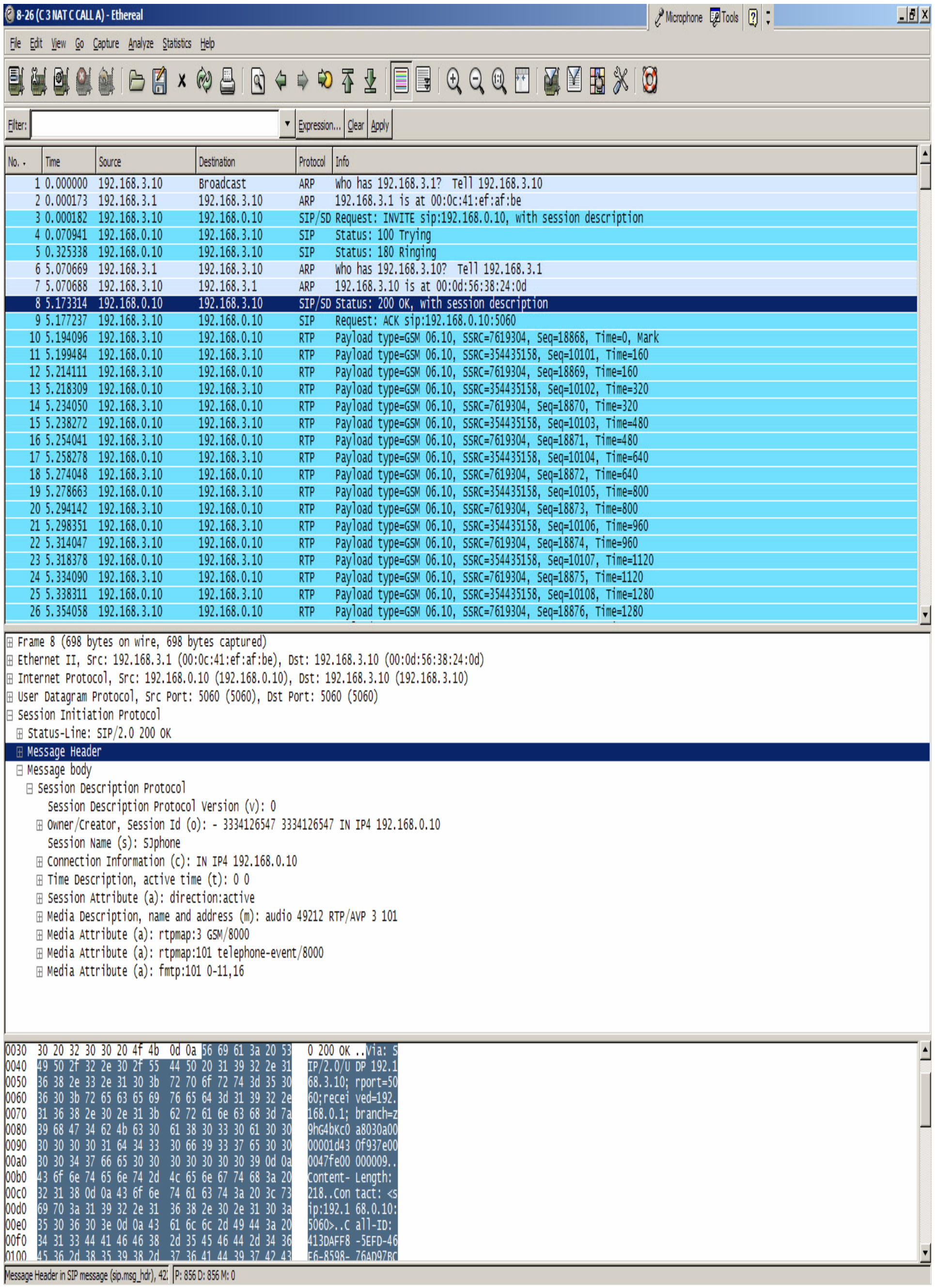


Figure 39. Test 4: Packet Capture on Client C (Client C Calls Client A)

### E.2.3. NAT 1 Eth0

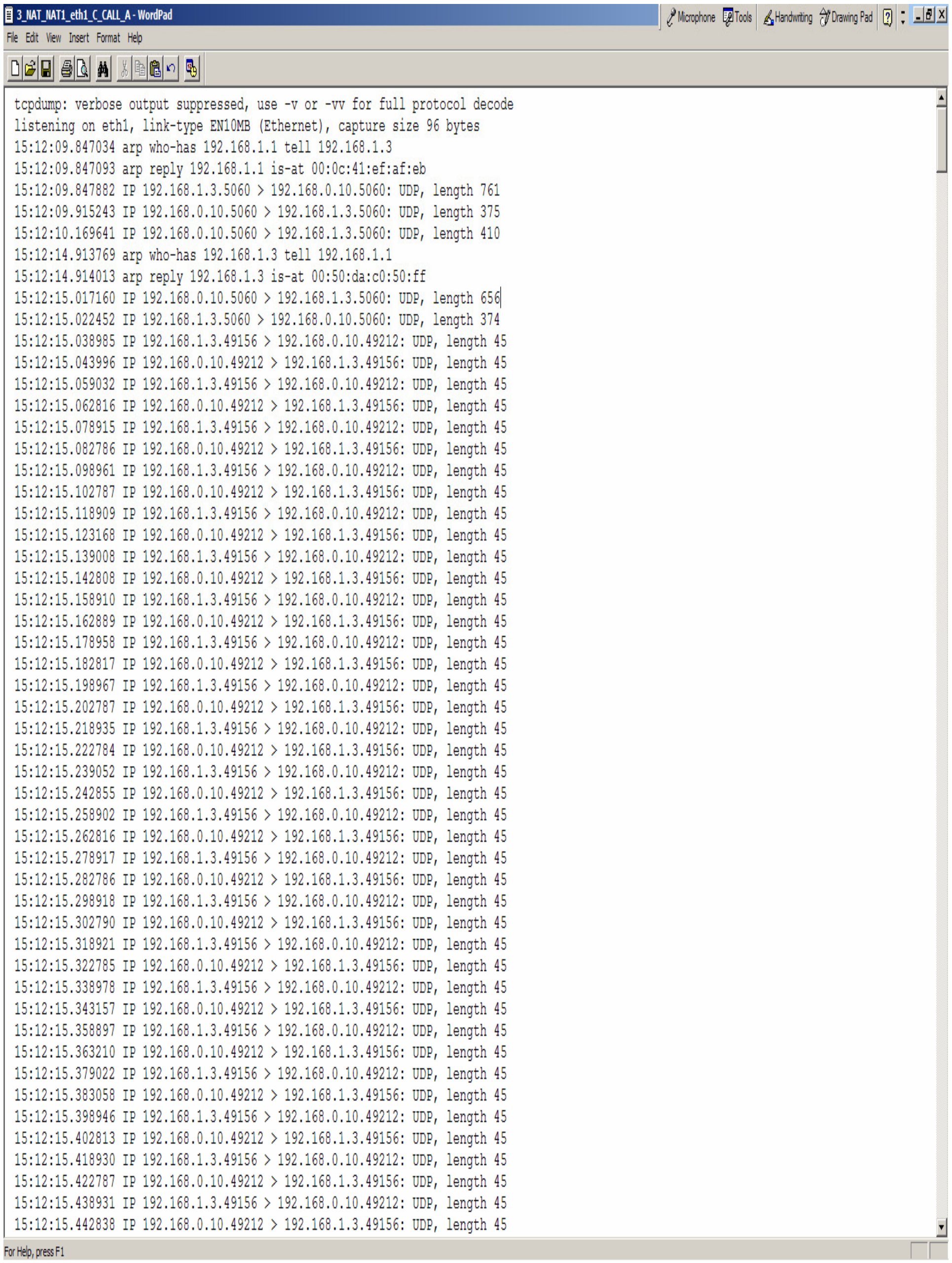
The following is a snapshot of the packets captured on the first interface (eth0) of NAT 1 when Client C calls Client A.

```
3_NAT_NAT1_eth0_C_CALL_A - WordPad
File Edit View Insert Format Help

tcpdump: verbose output suppressed, use -v or -vv for full protocol decode
listening on eth0, link-type EN10MB (Ethernet), capture size 96 bytes
15:12:05.343305 IP 192.168.0.10.5003 > 192.168.0.255.5003: UDP, length 125
15:12:09.848537 arp who-has 192.168.0.10 tell 192.168.0.1
15:12:09.848653 arp reply 192.168.0.10 is-at 00:0f:1f:19:27:36
15:12:09.848666 IP 192.168.0.1.5060 > 192.168.0.10.5060: UDP, length 761
15:12:09.915163 IP 192.168.0.10.5060 > 192.168.0.1.5060: UDP, length 375
15:12:09.937317 IP 192.168.0.10.netbios-ns > 192.168.0.255.netbios-ns: NBT UDP PACKET(137): QUERY; REQUEST; BROADCAST
15:12:10.169579 IP 192.168.0.10.5060 > 192.168.0.1.5060: UDP, length 410
15:12:10.683671 IP 192.168.0.10.netbios-ns > 192.168.0.255.netbios-ns: NBT UDP PACKET(137): QUERY; REQUEST; BROADCAST
15:12:11.433633 IP 192.168.0.10.netbios-ns > 192.168.0.255.netbios-ns: NBT UDP PACKET(137): QUERY; REQUEST; BROADCAST
15:12:15.017096 IP 192.168.0.10.5060 > 192.168.0.1.5060: UDP, length 656
15:12:15.022473 IP 192.168.0.1.5060 > 192.168.0.10.5060: UDP, length 374
15:12:15.031145 IP 192.168.0.10.netbios-ns > 192.168.0.255.netbios-ns: NBT UDP PACKET(137): QUERY; REQUEST; BROADCAST
15:12:15.039016 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.043942 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.059084 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.062793 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.078938 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.082769 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.099013 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.102766 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.118927 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.123153 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.139059 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.142788 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.158931 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.162873 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.178973 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.182766 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.198986 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.202767 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.218950 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.222769 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.239103 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.242836 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.258924 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.262800 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.278968 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.282766 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.298936 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.302774 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.318972 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.322765 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.338997 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.343141 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.358925 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.363191 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.379073 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.383038 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.398964 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.402797 IP 192.168.0.10.49212 > 192.168.0.1.49156: UDP, length 45
15:12:15.418980 IP 192.168.0.1.49156 > 192.168.0.10.49212: UDP, length 45

For Help, press F1
```

Figure 40. Test 4: Packet Capture on eth0 of NAT 1 (Client C Calls Client A)

### E.2.4. NAT 1 Eth1

The following is a snapshot of the packets captured on the second interface (eth1) of NAT 1 when Client C calls Client A.



```
tcpdump: verbose output suppressed, use -v or -vv for full protocol decode
listening on eth1, link-type EN10MB (Ethernet), capture size 96 bytes
15:12:09.847034 arp who-has 192.168.1.1 tell 192.168.1.3
15:12:09.847093 arp reply 192.168.1.1 is-at 00:0c:41:ef:af:eb
15:12:09.847882 IP 192.168.1.3.5060 > 192.168.0.10.5060: UDP, length 761
15:12:09.915243 IP 192.168.0.10.5060 > 192.168.1.3.5060: UDP, length 375
15:12:10.169641 IP 192.168.0.10.5060 > 192.168.1.3.5060: UDP, length 410
15:12:14.913769 arp who-has 192.168.1.3 tell 192.168.1.1
15:12:14.914013 arp reply 192.168.1.3 is-at 00:50:da:c0:50:ff
15:12:15.017160 IP 192.168.0.10.5060 > 192.168.1.3.5060: UDP, length 656
15:12:15.022452 IP 192.168.1.3.5060 > 192.168.0.10.5060: UDP, length 374
15:12:15.038985 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.043996 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.059032 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.062816 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.078915 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.082786 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.098961 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.102787 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.118909 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.123168 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.139008 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.142808 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.158910 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.162889 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.178958 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.182817 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.198967 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.202787 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.218935 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.222784 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.239052 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.242855 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.258902 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.262816 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.278917 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.282786 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.298918 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.302790 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.318921 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.322785 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.338978 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.343157 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.358897 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.363210 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.379022 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.383058 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.398946 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.402813 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.418930 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.422787 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
15:12:15.438931 IP 192.168.1.3.49156 > 192.168.0.10.49212: UDP, length 45
15:12:15.442838 IP 192.168.0.10.49212 > 192.168.1.3.49156: UDP, length 45
```

Figure 41. Test 4: Packet Capture on eth1 of NAT 1 (Client C Calls Client A)

### E.2.5. NAT 3 Eth0

The following is a snapshot of the packets captured on the first interface (eth0) of NAT 3 when Client C calls Client A.

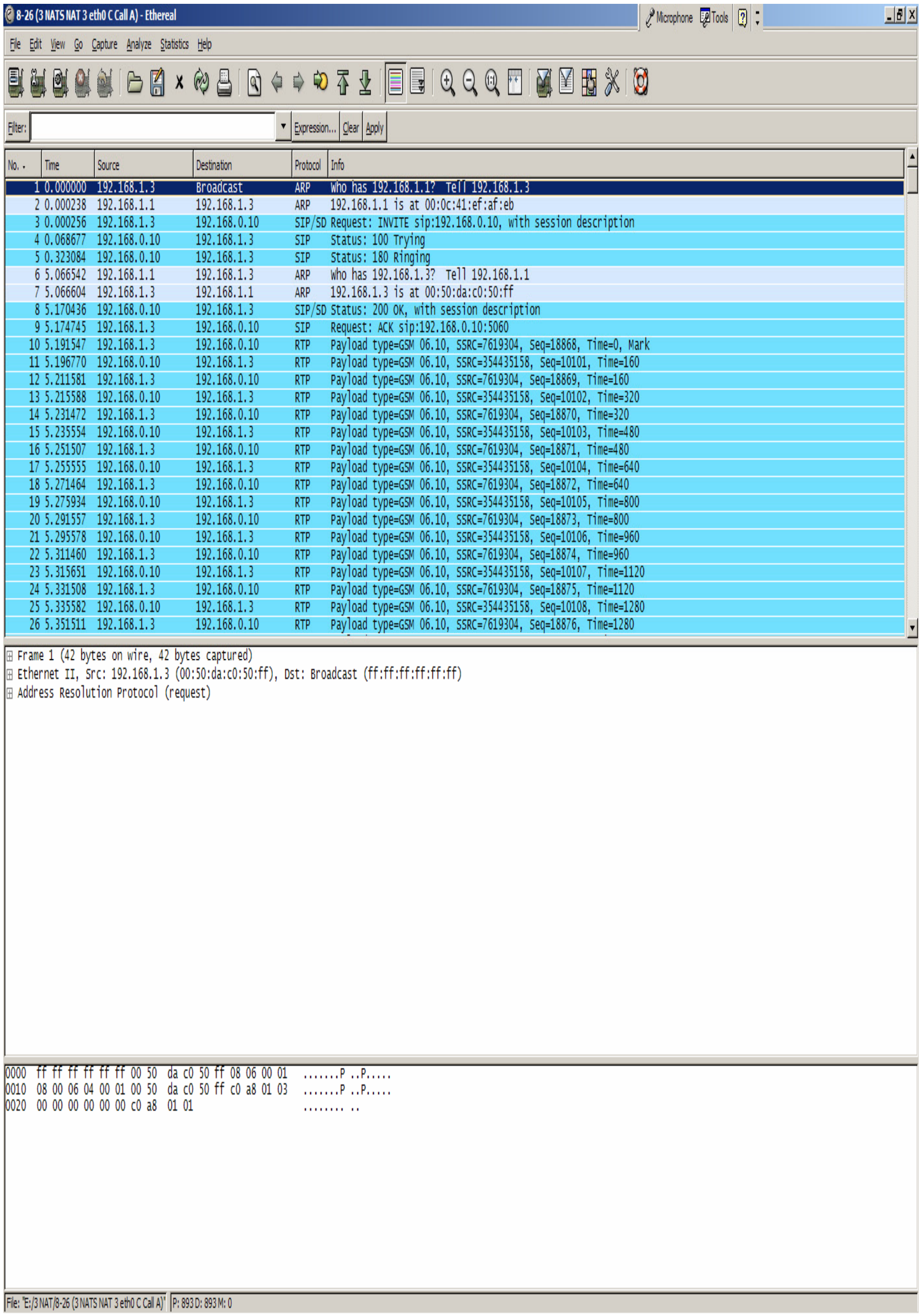


Figure 42. Test 4: Packet Capture on eth0 of NAT 3 (Client C Calls Client A)

E.2.6. NAT 3 Eth1

The following is a snapshot of the packets captured on the second interface (eth1) of NAT 3 when Client C calls Client A.

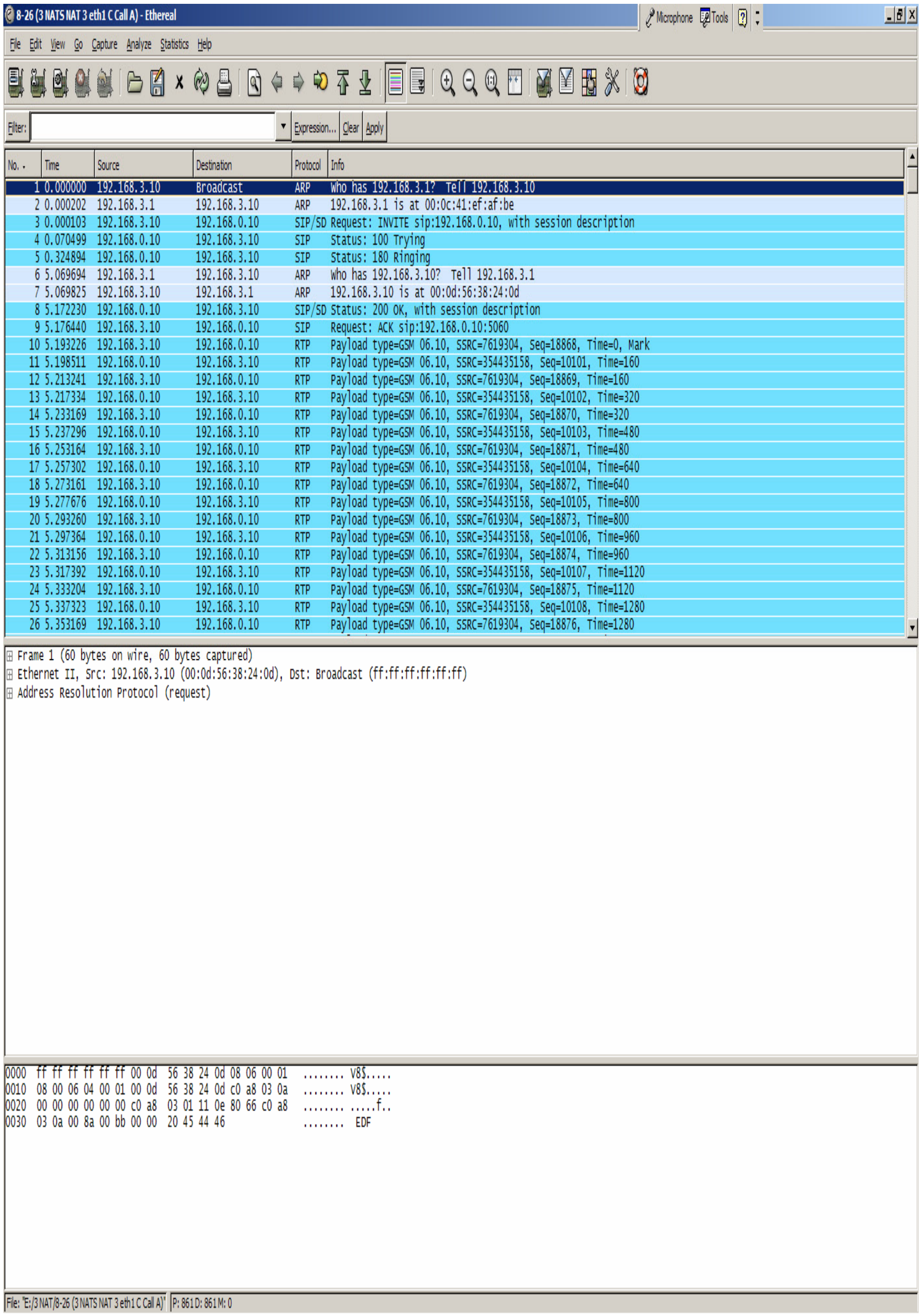


Figure 43. Test 4: Packet Capture on eth1 of NAT 3 (Client C Calls Client A)

### E.2.7. Analysis

The sequences of packet exchanges between Clients A and C are similar to that of Clients A and B described in the previous subsection. See Section E.1.7 for explanation.

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# APPENDIX F. TEST 5: EXTENDED DOUBLE NAT VOIP WITH SIMULTANEOUS VOIP SESSIONS DEMONSTRATION USING SJPHONE

The instructions contained in this appendix describe how to setup and demonstrate a SIP-based VoIP communication using the network topology illustrated in the Figure 13. In this test scenario, two VoIP communication sessions will take place simultaneously, i.e. one between Clients A and C and the other between Clients B and D. Similar to the previous test, the DNAT rule is purposely taken out from NAT 1. Packet captures from all four clients are included at the end of this appendix along with their corresponding analysis.

A. Network Topology

Refer to Figure 13 and Figure 14 for the physical and logical network topology.

B. Equipment Requirements

- B.1. Clients A, B, C, and D
  - B.1.1. Windows XP Operating System
  - B.1.2. Sound card
  - B.1.3. SJPhone v.1.60
  - B.1.4. Ethereal
  - B.1.5. ZoneAlarm (for Clients A and B only)
- B.2. NAT 1, NAT 2, NAT 3 and Router
  - B.2.1. Linux Operating System (Fedora Core 4)
  - B.2.2. netfilter and iptables
  - B.2.3. Ethereal
  - B.2.4. Two network cards (for NAT 1, NAT 2, and NAT 3)
  - B.2.5. Three network cards (for Router only)
- B.3. Additional Equipment
  - B.3.1. Cross-over cables and a switch or hub to implement the network architecture illustrated in Figure 13.
  - B.3.2. Microphones as audio input devices for clients A, B, C and D

C. Installation and Configuration

C.1. Client A

IP Address: 131.120.9.16

Subnet Mask: 255.255.255.0

Default Gateway: 131.120.9.15

C.2. Client B

IP Address: 131.120.9.17

Subnet Mask: 255.255.255.0

Default Gateway: 131.120.9.15

C.3. Clients C and D

IP Address: 192.168.3.11

Subnet Mask: 255.255.255.0

Default Gateway: 192.168.3.1

C.4. Router

C.4.1. Configure eth0 by editing /etc/sysconfig/network-scripts/ifcfg-eth0 to include the following:

```
DEVICE=eth0
BOOTPROTO=NONE
IPADDR=192.168.100.27
NETMASK=255.255.255.0
GATEWAY=192.168.100.88
```

C.4.2. Activate eth0 by running:

```
ifup eth0
```

C.4.3. Configure eth1 by editing /etc/sysconfig/network-scripts/ifcfg-eth1 to include the following:

```
DEVICE=eth1
BOOTPROTO=NONE
IPADDR=192.168.202.1
NETMASK=255.255.255.0
```

C.4.4. Activate eth1 by running:

```
ifup eth1
```

C.4.5. Configure eth2 by editing and saving /etc/sysconfig/network-scripts/ifcfg-eth2 to include the following:

```
DEVICE=eth2
BOOTPROTO=NONE
IPADDR=192.168.2.1
NETMASK=255.255.255.0
```

C.4.6. Activate eth2 by running:

```
ifup eth2
```

C.4.7. Enable IP Forwarding by running:

```
echo 1 > /proc/sys/net/ipv4/ip_forward
```

C.4.8. Flush any existing firewall and NAT rules by running:

```
iptables -F
iptables -t nat -F
```

C.5. NAT 1

C.5.1. Configure eth0 by editing /etc/sysconfig/network-scripts/ifcfg-eth0 to include the following:

```
DEVICE=eth0
BOOTPROTO=NONE
IPADDR=131.120.9.15
NETMASK=255.255.255.0
```

C.5.2. Activate eth0 by running:

```
ifup eth0
```

C.5.3. Configure eth1 by editing and saving /etc/sysconfig/network-scripts/ifcfg-eth1 to include the following:

```
DEVICE=eth1
BOOTPROTO=NONE
IPADDR=192.168.100.88
NETMASK=255.255.255.0
```

C.5.4. Activate eth1 by running:

```
ifup eth1
```

C.5.5. Configure static routes by running:

route add –net 192.168.202.0 netmask 255.255.255.0 gw 192.168.100.27 eth1
route add –net 192.168.2.0 netmask 255.255.255.0 gw 192.168.100.27 eth1

C.5.6. Enable IP Forwarding by running:

echo 1 > /proc/sys/net/ipv4/ip_forward

C.5.7. Flush any existing firewall and NAT rules by running:

iptables -F

iptables -t nat -F

C.5.8. Configure NAT rule by running:

iptables -t nat -A POSTROUTING -o eth0 -j SNAT --to 131.120.9.15

C.6. NAT 2

C.6.1. Configure eth0 by editing and saving /etc/sysconfig/network-scripts/ifcfg-eth0 to include the following:

DEVICE=eth0

BOOTPROTO=NONE

IPADDR=196.168.202.11

NETMASK=255.255.255.0

GATEWAY=198.168.202.1

C.6.2. Activate eth0 by running:

ifup eth0

C.6.3. Configure eth1 by editing and saving /etc/sysconfig/network-scripts/ifcfg-eth1 to include the following:

DEVICE=eth1

BOOTPROTO=NONE

IPADDR=192.168.3.1

NETMASK=255.255.255.0

C.6.4. Activate eth1 by running:

ifup eth1

C.6.5. Enable IP Forwarding by running:

echo 1 > /proc/sys/net/ipv4/ip_forward

C.6.6. Flush any existing firewall and NAT rules by running:

iptables -F

iptables -t nat -F

C.6.7. Configure NAT rules by running:

```
iptables -t nat -A POSTROUTING -o eth0 -j SNAT --to 192.168.202.11
iptables -t nat -A PREROUTING -i eth0 -j DNAT --to 192.168.3.11
```

C.7. NAT 3

C.7.1. Configure eth0 by editing /etc/sysconfig/network-scripts/ifcfg-eth0 to include the following:

```
DEVICE=eth0
BOOTPROTO=NONE
IPADDR=192.168.2.11
NETMASK=255.255.255.0
GATEWAY=192.168.2.1
```

C.7.2. Activate eth0 by running:

```
ifup eth0
```

C.7.3. Configure eth1 by editing /etc/sysconfig/network-scripts/ifcfg-eth1 to include the following:

```
DEVICE=eth1
BOOTPROTO=NONE
IPADDR=192.168.3.1
NETMASK=255.255.255.0
```

C.7.4. Activate eth1 by running:

```
ifup eth1
```

C.7.5. Enable IP Forwarding by running:

```
echo 1 > /proc/sys/net/ipv4/ip_forward
```

C.7.6. Flush any existing firewall and NAT rules by running:

```
iptables -F
iptables -t nat -F
```

C.7.7. Configure NAT rules by running:

```
iptables -t nat -A POSTROUTING -o eth0 -j SNAT --to 192.168.2.11
iptables -t nat -A PREROUTING -i eth0 -j DNAT --to 192.168.3.11
```

C.8. SJPhone Installation and Configuration

C.8.1. Clients A, B, C and D

C.8.1.1. Download the Windows version of SJPhone v.1.60 from SJ Labs

C.8.1.2. Install SJPhone v.1.60

C.8.1.3. Launch SJPhone

C.8.1.4. Right-click on SJPhone

C.8.1.5. Go to Services

C.8.1.6. Select PC-to-PC (SIP)

C.9. Ethereal Installation and Configuration

C.9.1. Clients A, B, C and D

C.9.1.1. Download the Windows version of Ethereal v.0.10.12

C.9.1.2. Install Ethereal v.0.10.12 by following on-screen instructions

C.9.2. Router, NAT 2 and NAT 3

C.9.2.1.1. Install Ethereal if it is not already installed

C.9.2.1.1.1. Go to the Desktop menu

C.9.2.1.1.2. Go to System Settings

C.9.2.1.1.3. Go to Add/Remove Applications

C.9.2.1.1.4. Click on Details under System Tools

C.9.2.1.1.5. Find and then check ethereal-gnome

C.9.2.1.1.6. Click on Close

C.9.2.1.1.7. Click on Update

C.9.2.1.1.8. Put in the correct Fedora Core 4 CDs when prompted

C.10. ZoneAlarm Installation and Configuration

C.10.1. On clients A and B,

C.10.1.1. Download the free ZoneAlarm from Zone Labs

C.10.1.2. Install ZoneAlarm by following on-screen instructions

C.10.1.3. When ZoneAlarm is being run for the first time, it will ask the user to choose between Basic ZoneAlarm or trial version of ZoneAlarm Pro, select the trial version of ZoneAlarm

C.10.1.4. Answer on-screen questions

C.10.1.5. Click to use Trial Version

C.10.1.6. When asked to select a security level for the detected network, select Allow into Trusted Zone

C.10.1.7. Allow all pop-ups for testing

C.10.1.8. Configure firewall rule in ZoneAlarm:

C.10.1.8.1. Go to Firewall menu on the left panel

C.10.1.8.2. Click on the Expert tab

C.10.1.8.3. Click on Add

C.10.1.8.4. Type in a name for the firewall rule in the Name textbox

C.10.1.8.5. Under Action, select Block

C.10.1.8.6. Under Destination,

C.10.1.8.6.1. Select Modify

C.10.1.8.6.2. Select Add Location

C.10.1.8.6.3. Select IP Address

C.10.1.8.6.4. Type in a description in the Description textbox

C.10.1.8.6.5. Type 192.168.3.11 in the IP Address textbox

C.10.1.8.6.6. Click OK

C.10.1.8.6.7. Click OK

C.10.1.8.6.8. Click Apply

D. Preparation and Testing

D.1. Adjust volume on both clients accordingly

D.2. Plug microphones into all clients

D.3. On Client A,

D.3.1. Launch Ethereal

D.3.2. Go to the Capture menu

D.3.3. Go to Interfaces

D.3.4. Click on Capture 131.120.9.16

D.4. On Client B,

D.4.1. Launch Ethereal

D.4.2. Go to the Capture menu

D.4.3. Go to Interfaces

D.4.4. Click on Capture 131.120.9.17

D.5. On Client C and Client D,

D.5.1. Launch Ethereal

D.5.2. Go to the Capture menu

D.5.3. Go to Interfaces

D.5.4. Click on Capture 192.168.3.11

D.6. On Router,

D.6.1. Launch one instance of Ethereal

D.6.1.1. Go to the Capture menu

D.6.1.2. Go to Interfaces

D.6.1.3. Click on Capture Eth0

D.6.2. Launch a second instance of Ethereal

D.6.2.1. Go to the Capture menu

D.6.2.2. Go to Interfaces

D.6.2.3. Click on Capture Eth1

D.6.3. Launch a third instance of Ethereal

D.6.3.1. Go to the Capture menu

D.6.3.2. Go to Interfaces

D.6.3.3. Click on Capture Eth2

D.7. On NAT 1, NAT 2 and NAT 3,

D.7.1. Launch one instance of Ethereal

D.7.1.1. Go to the Capture menu

D.7.1.2. Go to Interfaces

D.7.1.3. Click on Capture Eth0

D.7.2. Launch another instance of Ethereal

D.7.2.1. Go to the Capture menu

D.7.2.2. Go to Interfaces

D.7.2.3. Click on Capture Eth1

D.8. On Client C,

D.8.1. Call A by dialing 131.120.9.16 in SJPhone

D.9. On Client A,

D.9.1. Select Accept in the pop-up dialog box when SJPhone rings

D.9.2. Clients A and C may engage in a VoIP conversation at this point.

D.10. On Client D,

D.10.1. Call B by dialing 131.120.9.17 in SJPhone

D.11. On Client B,

D.11.1. Select Accept in the pop-up dialog box when SJPhone rings

D.12. Clients B and D may engage in a VoIP conversation at this point.

D.13. Click on the Hang-Up bottom on either SJPhone to terminate call when finished

D.14. On all clients, NATs and Router,

D.14.1. Stop packet captures selecting Stop on Ethereal

E. Packet Captures

E.1. Client A

The following is a snapshot of the packets captured on Client A when it receives a call from Client C.

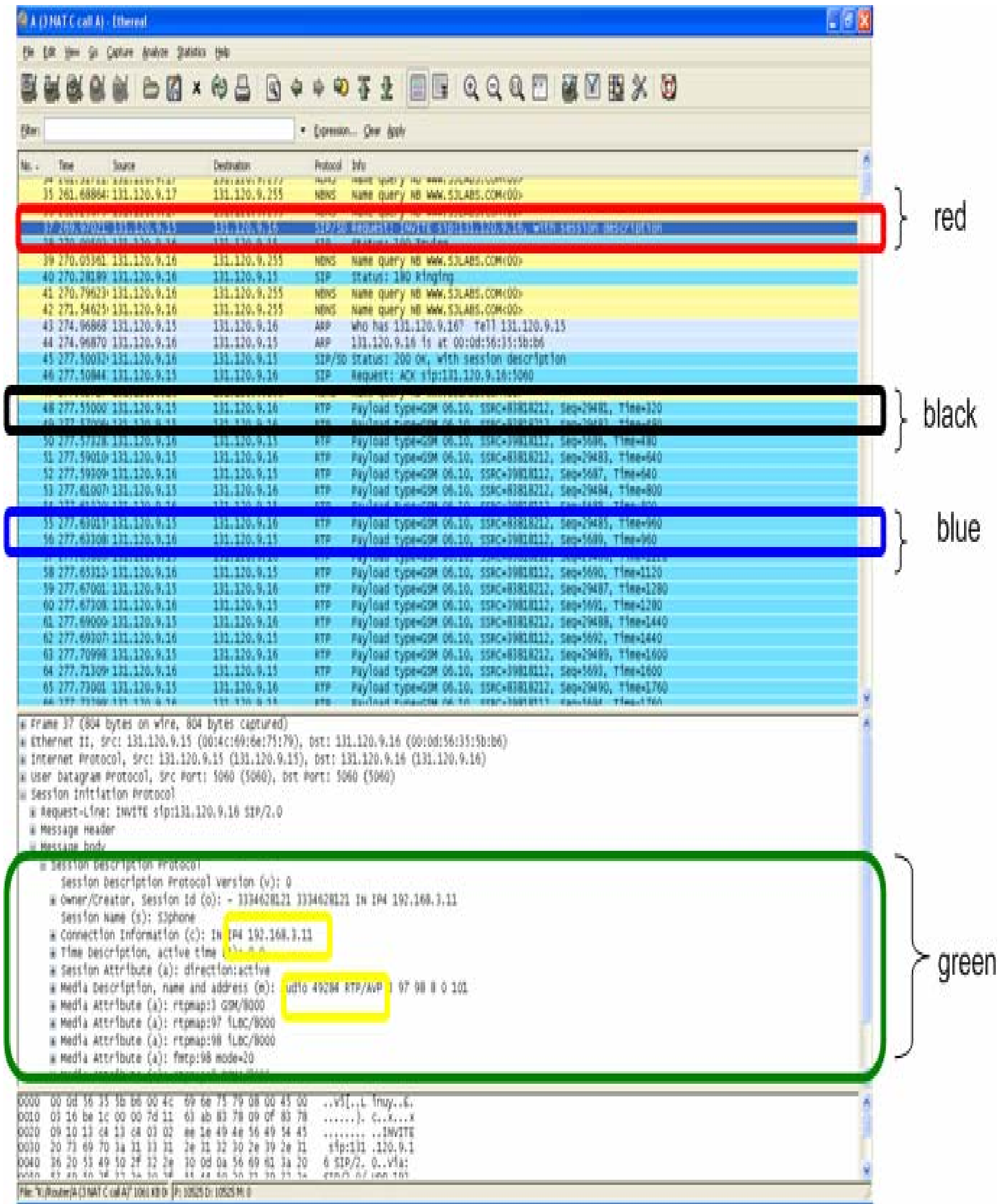



Figure 44. Test 5: Packet Capture on Client A

E.2. Client C

The following is a snapshot of the packets captured on Client C when it calls Client A.

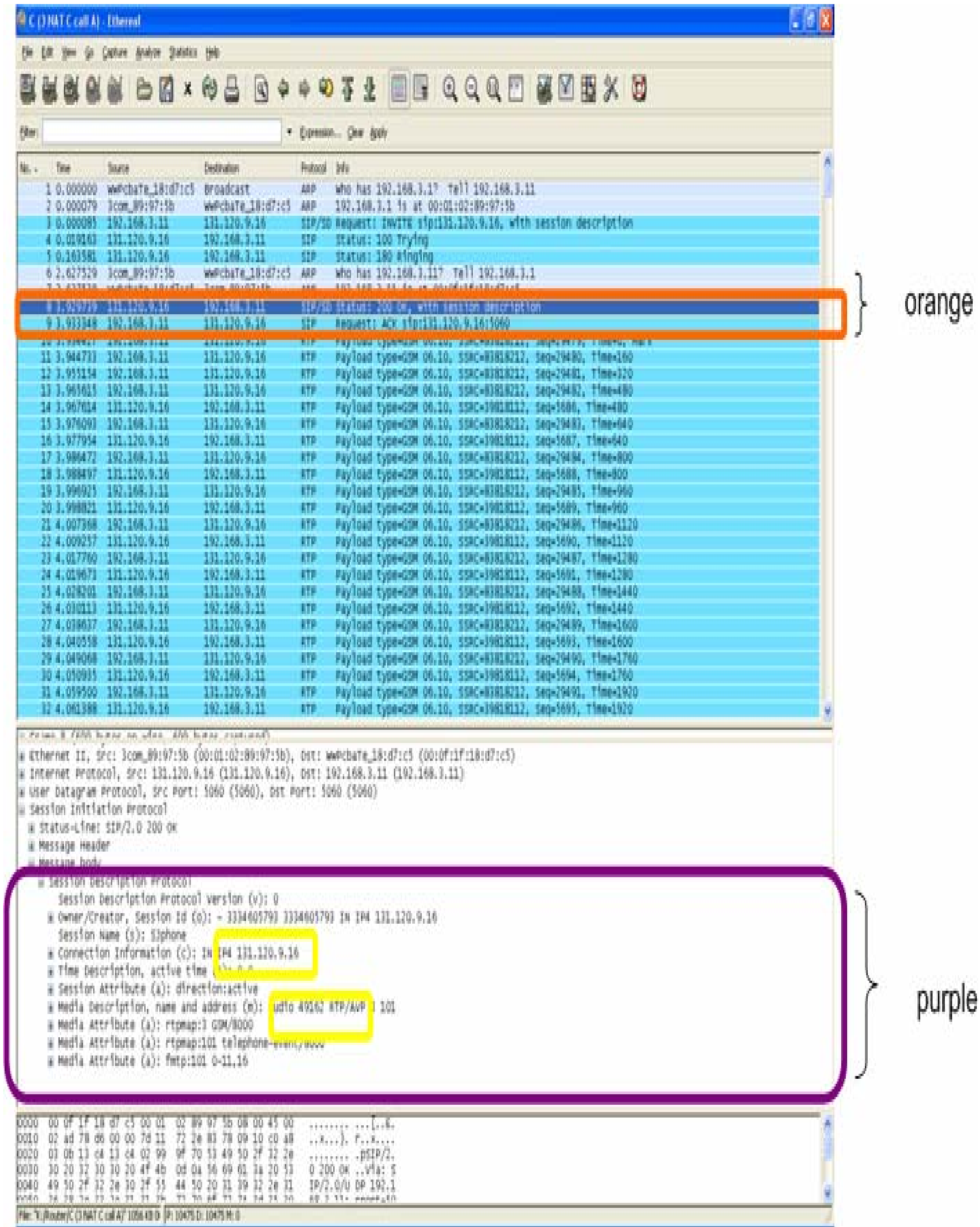



Figure 45. Test 5: Packet Capture on Client C

E.3. NAT 2 Eth0

The following is a snapshot of the packets captured on the first interface (eth0) of NAT 2.

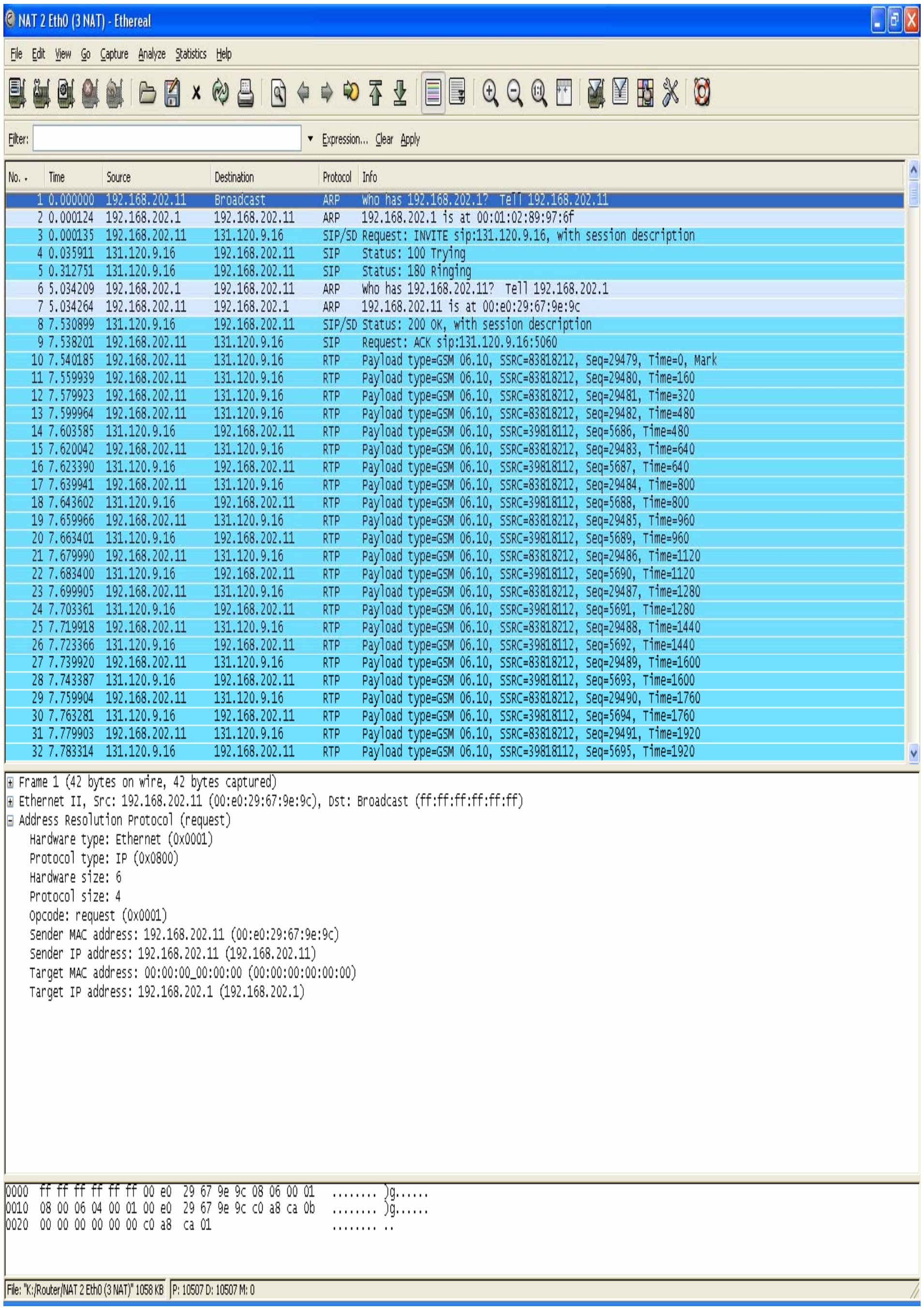


Figure 46.  Test 5: Packet Capture on eth0 of NAT 2

E.4. NAT 2 Eth1

The following is a snapshot of the packets captured on the second interface (eth1) of NAT 2.

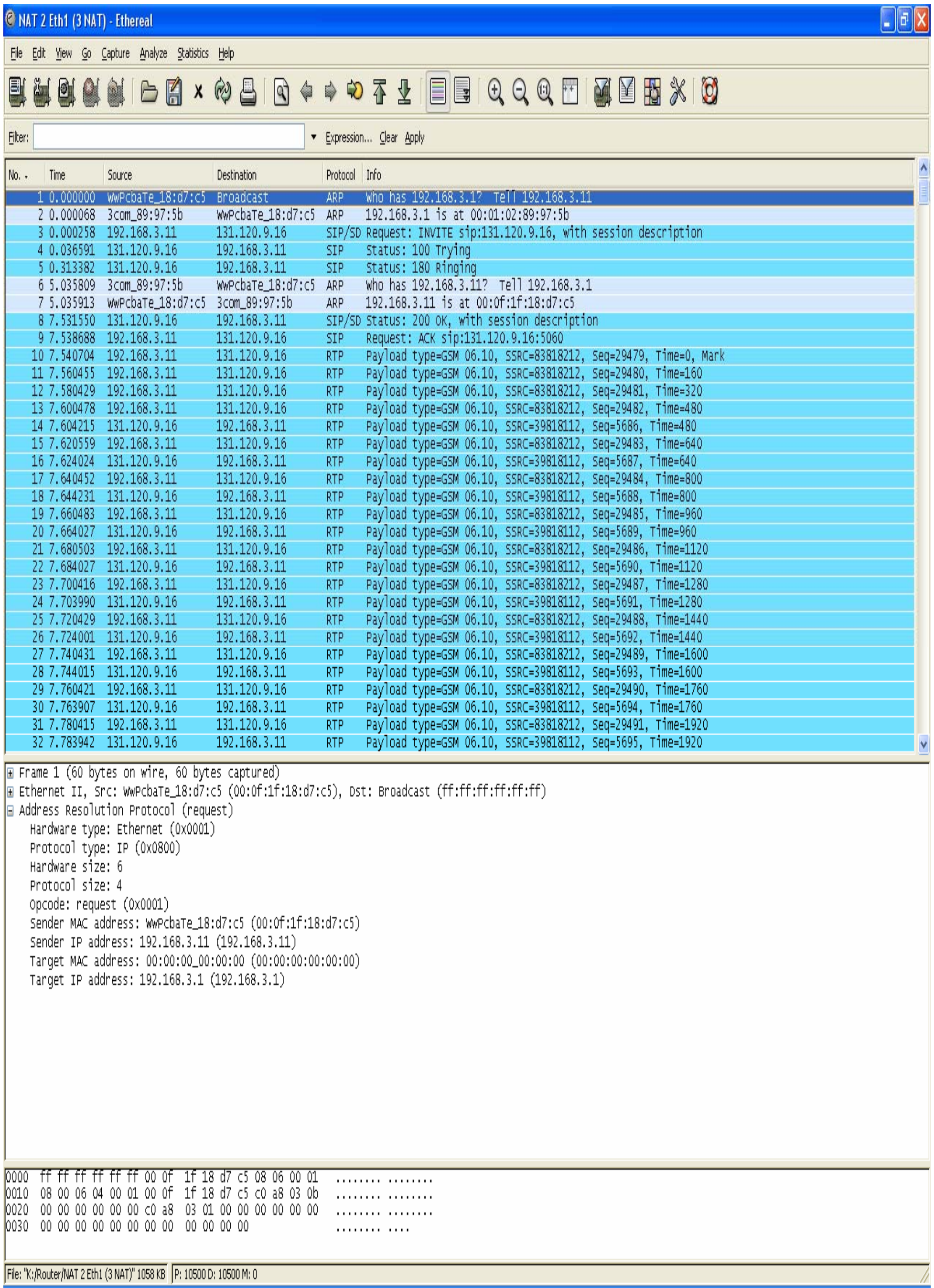



Figure 47. Test 5: Packet Capture on eth1 of NAT 2

E.5. Router Eth0

The following is a snapshot of the packets captured on the first interface (eth0) of Router.

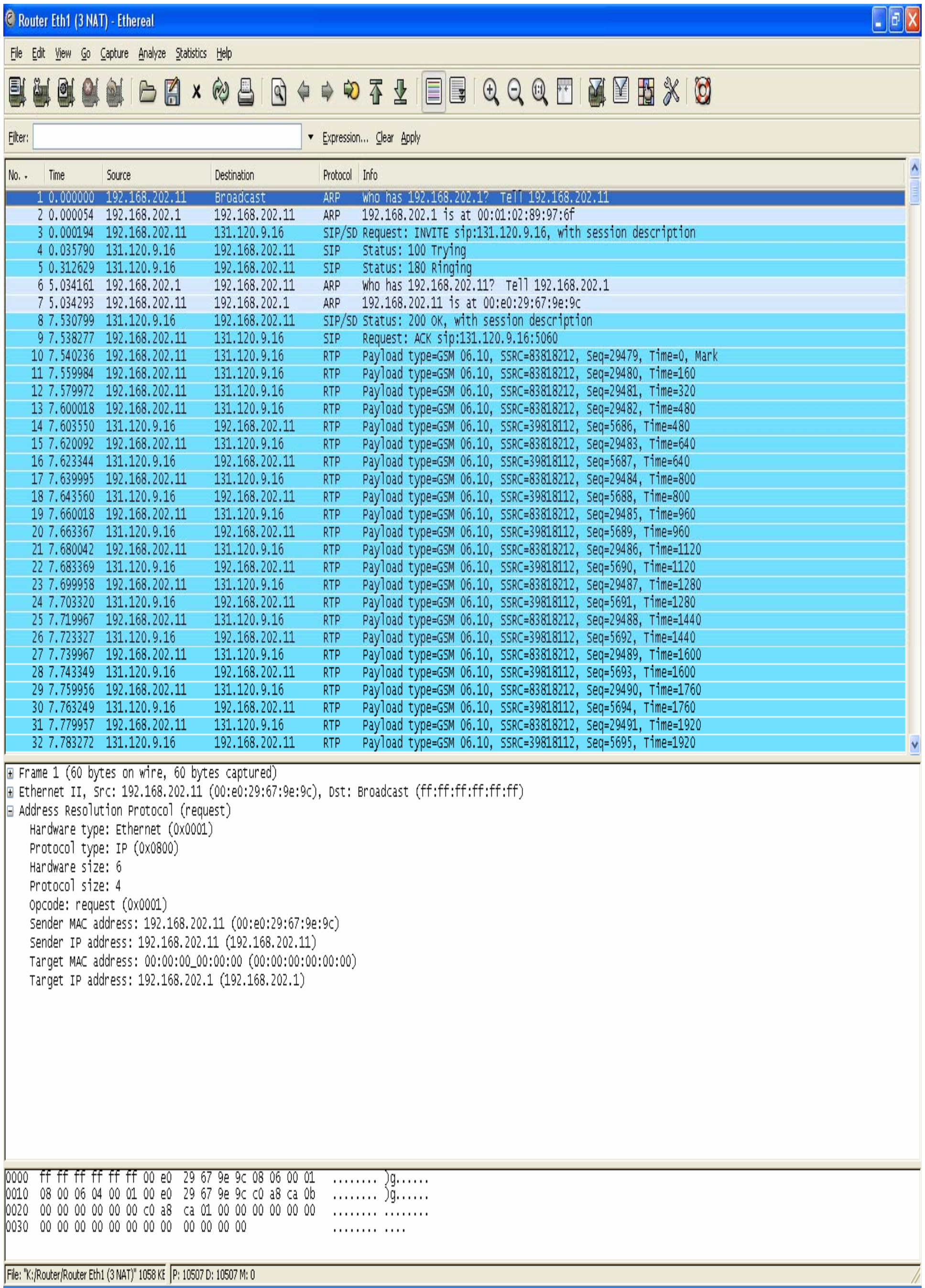



Figure 48. Test 5: Packet Capture on eth0 of Router

E.6. Router Eth1

The following is a snapshot of the packets captured on the second interface (eth1) of Router.

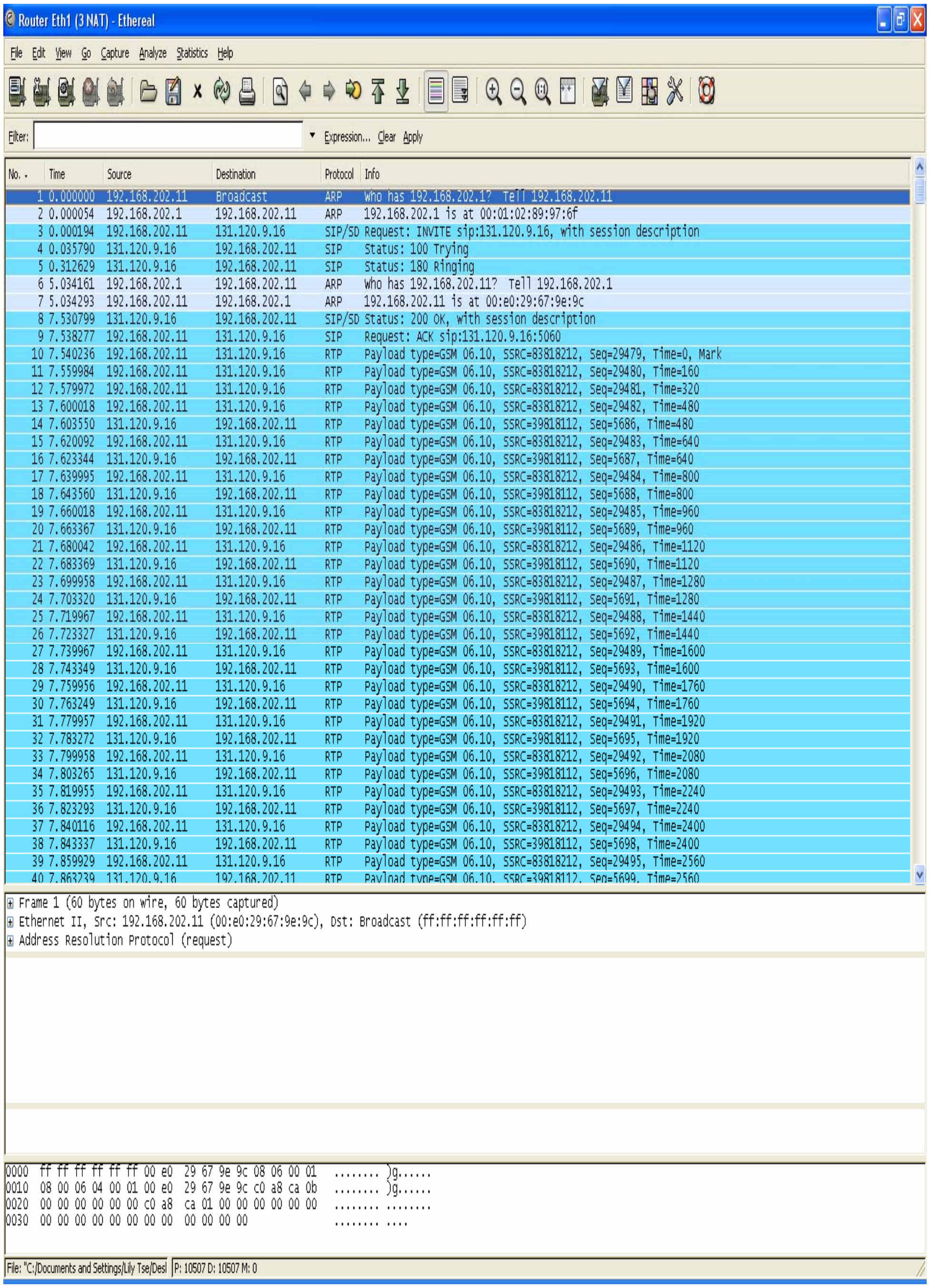


Figure 49. Test 5: Packet Capture on eth1 of Router

E.7. Router Eth2

The following is a snapshot of the packets captured on the third interface (eth2) of Router.

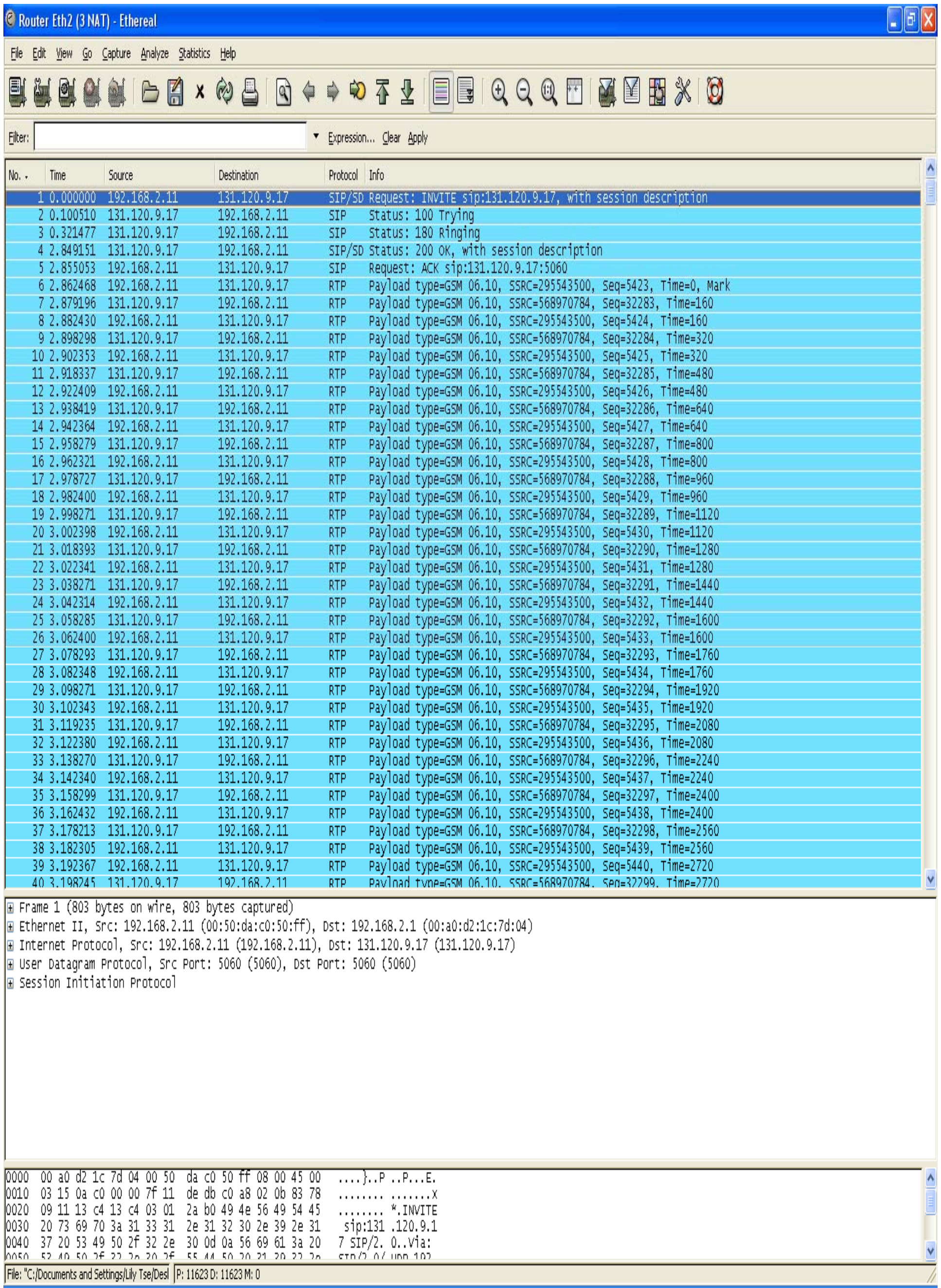



Figure 50. Test 5: Packet Capture on eth2 of Router

E.8. Client B

The following is a snapshot of the packets captured on Client B when it receives a call from Client D.

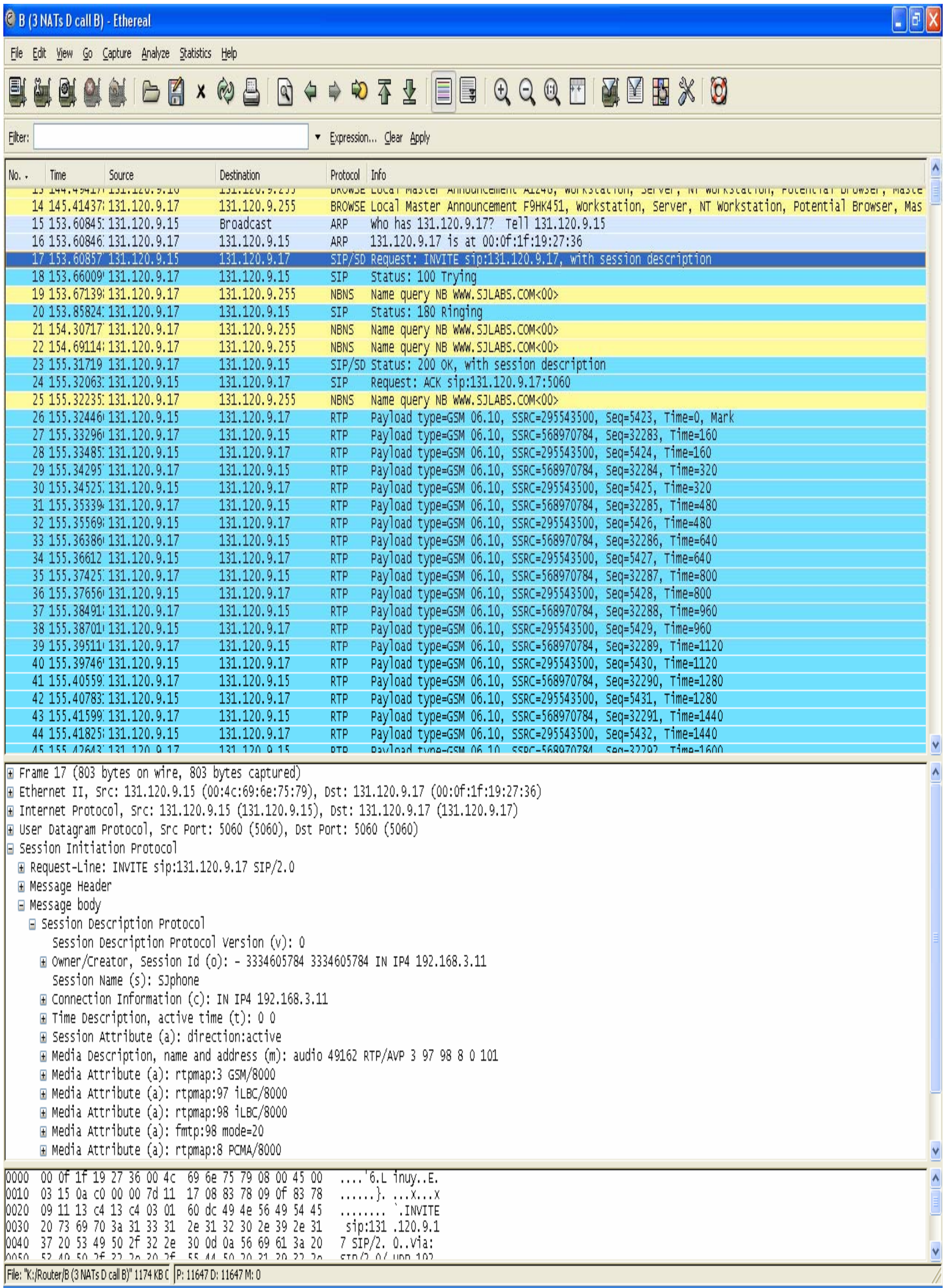


Figure 51. Test 5: Packet Capture on Client B

## E.9. Client D

The following is a snapshot of the packets captured on Client D when it calls Client B.

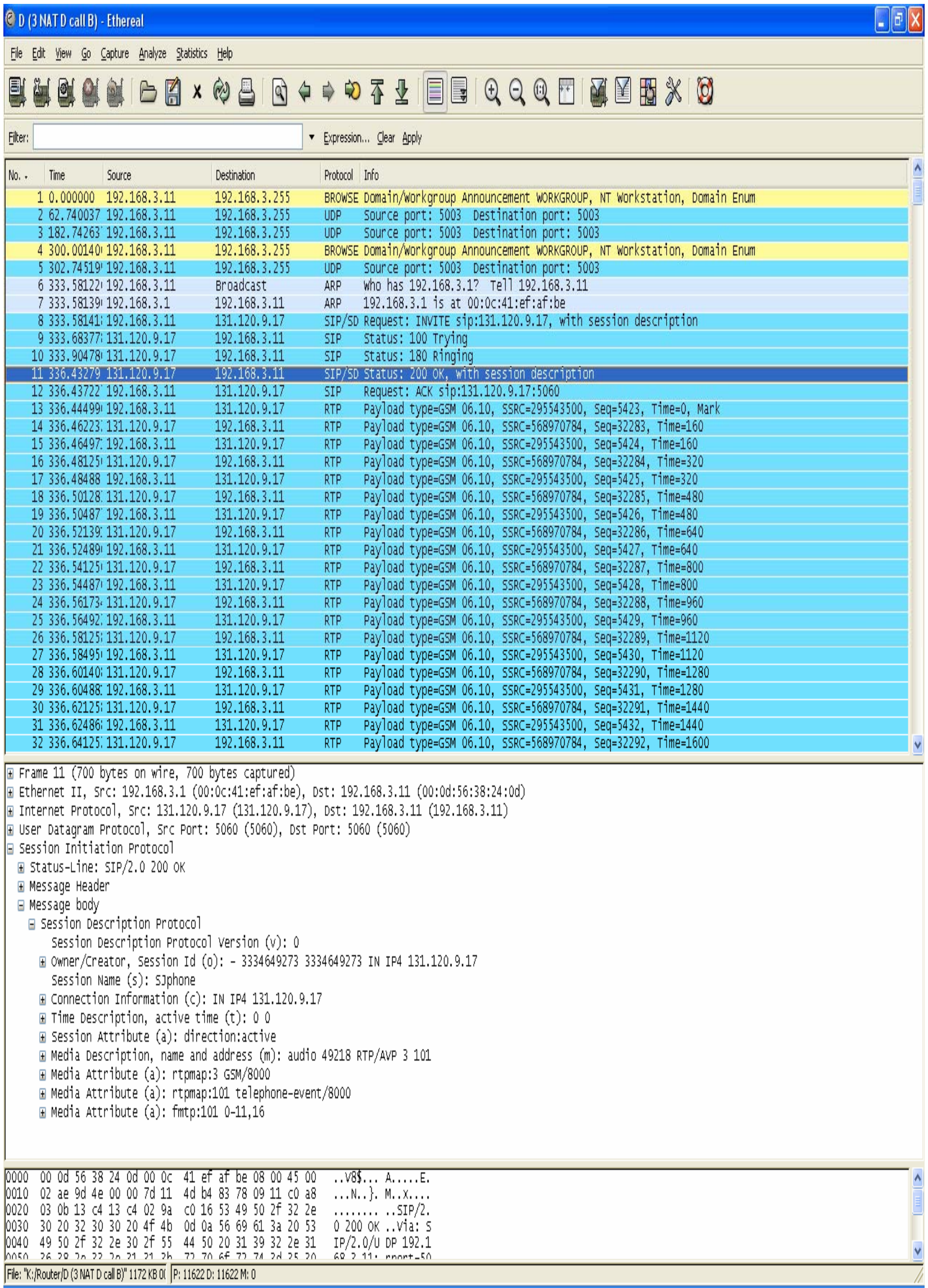



Figure 52. Test 5: Packet Capture on Client D

### E.10. NAT 3 Eth0

The following is a snapshot of the packets captured on the first interface (eth0) of NAT 3.

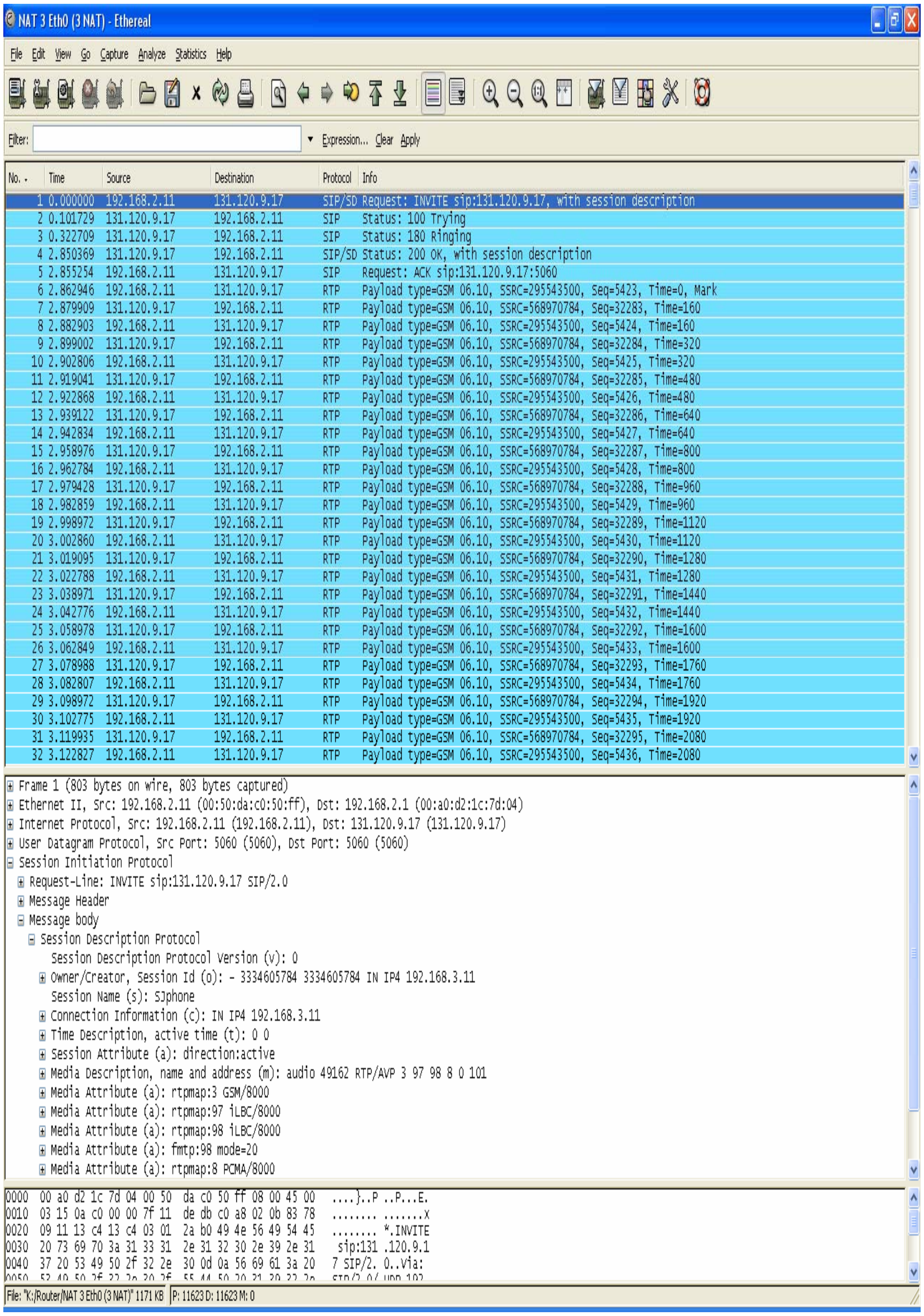


Figure 53. Test 5: Packet Capture on eth0 of NAT 3

E.11. NAT 3 Eth1

The following is a snapshot of the packets captured on the second interface (eth1) of NAT 3.

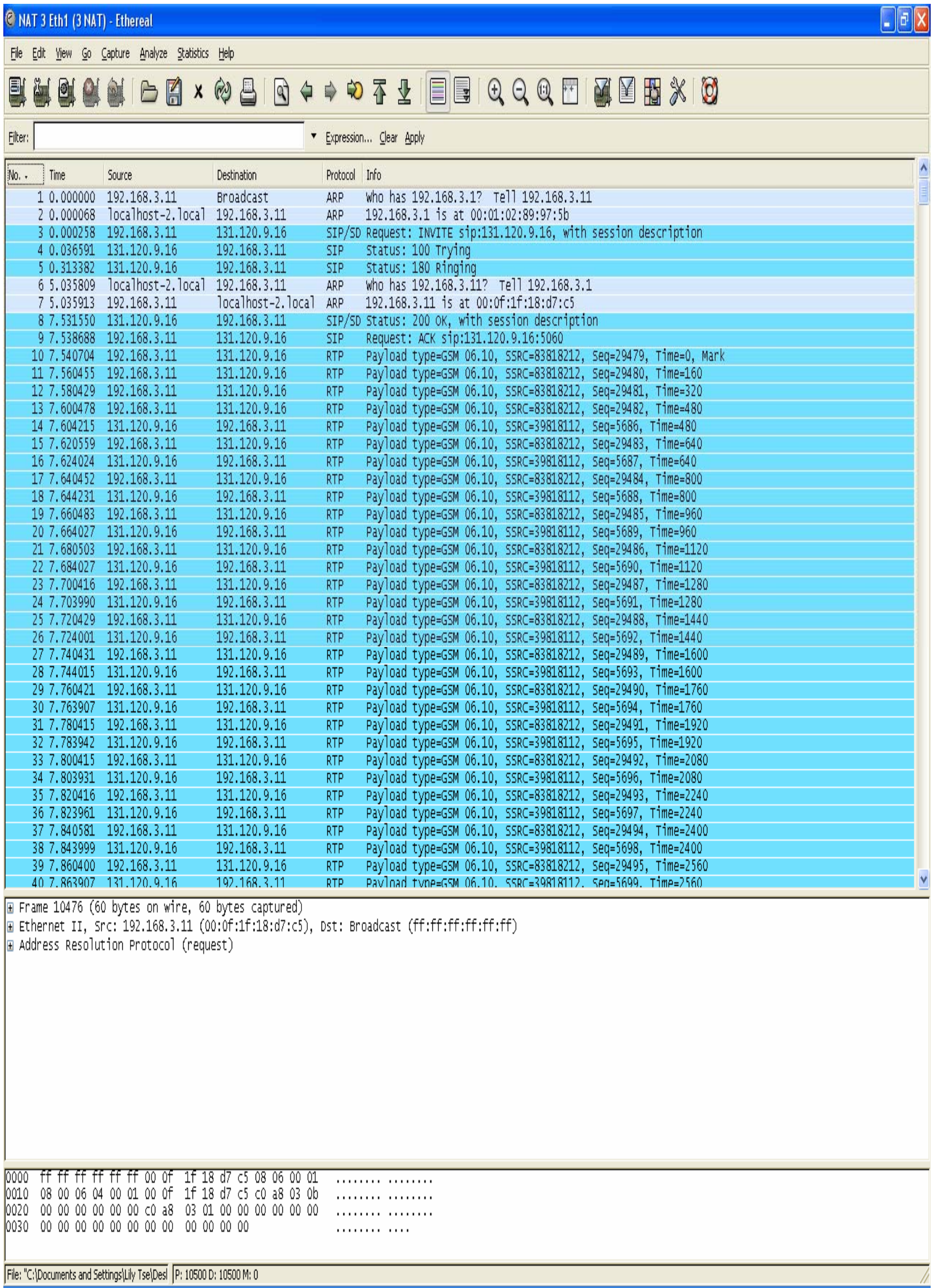



Figure 54. Test 5: Packet Capture on eth1 of NAT 3

## E.12. NAT 1 Eth0

The following is a snapshot of the packets captured on the first interface (eth0) of NAT 1.


NAT1_ETH0 - WordPad

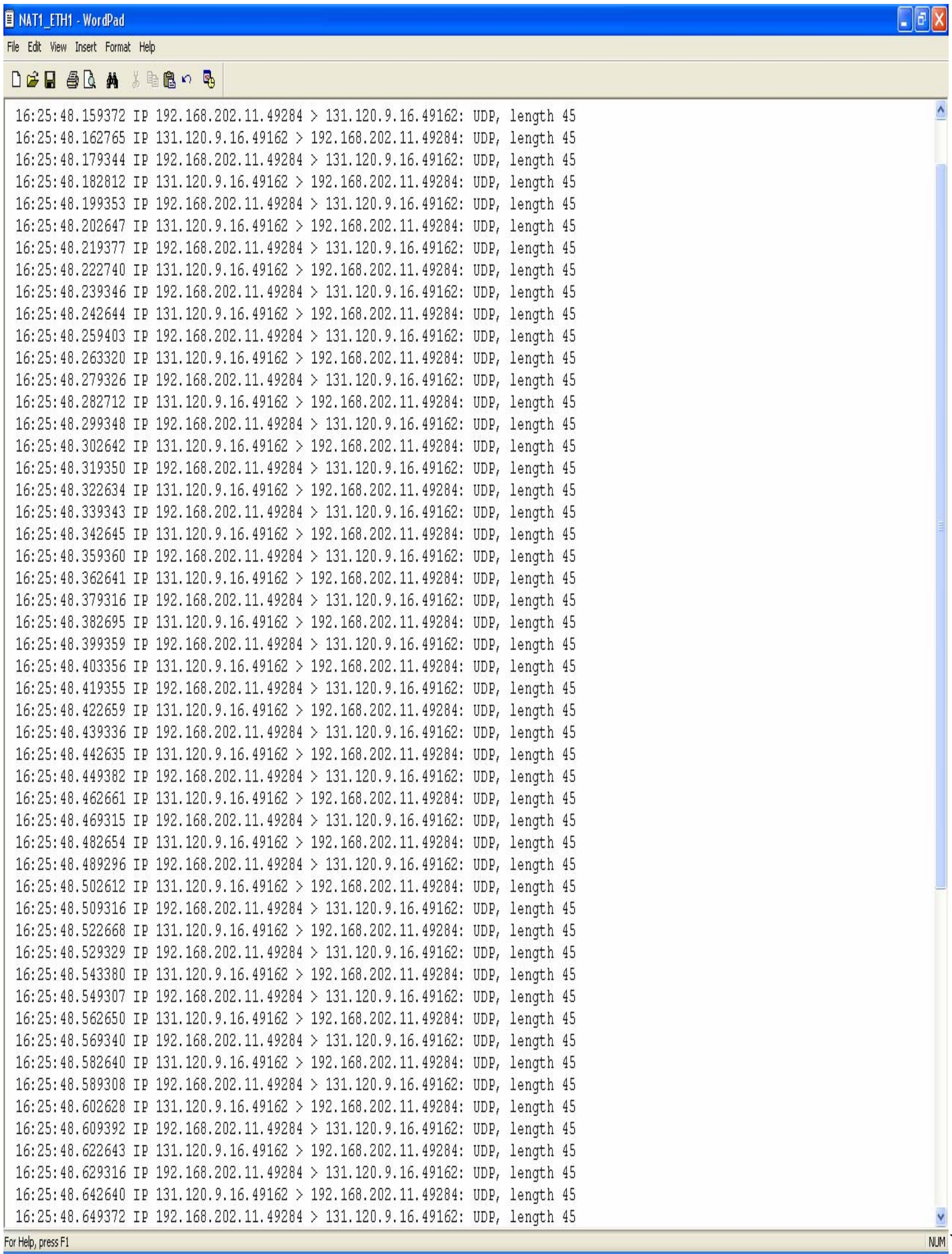
```
16:25:49.519295 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.532642 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.539430 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.552142 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.559306 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.572149 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.579251 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.582311 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.599409 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.602181 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.619294 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.622193 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.639268 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.642178 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.659367 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.662143 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.679254 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.682173 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.699316 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.702296 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.719246 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.722171 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.739302 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.742157 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.759326 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.762154 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.779284 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.782164 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.799300 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.802151 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.819233 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.822164 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.839311 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.842186 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.859399 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.862166 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.879374 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.882136 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.899275 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.902156 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.919290 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.922157 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.939367 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.942159 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.959288 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.962159 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.979303 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:49.982129 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:49.999288 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
16:25:50.002150 IP 131.120.9.16.49162 > 131.120.9.15.49284: UDP, length 45
16:25:50.019301 IP 131.120.9.15.49284 > 131.120.9.16.49162: UDP, length 45
```

For Help, press F1

Figure 55. Test 5: Packet Capture on eth0 of NAT 1

## E.13. NAT 1 Eth1

The following is a snapshot of the packets captured on the second interface (eth1) of NAT 1.




```
16:25:48.159372 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.162765 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.179344 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.182812 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.199353 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.202647 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.219377 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.222740 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.239346 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.242644 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.259403 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.263320 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.279326 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.282712 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.299348 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.302642 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.319350 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.322634 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.339343 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.342645 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.359360 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.362641 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.379316 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.382695 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.399359 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.403356 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.419355 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.422659 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.439336 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.442635 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.449382 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.462661 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.469315 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.482654 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.489296 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.502612 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.509316 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.522668 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.529329 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.543380 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.549307 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.562650 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.569340 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.582640 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.589308 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.602628 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.609392 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.622643 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.629316 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
16:25:48.642640 IP 131.120.9.16.49162 > 192.168.202.11.49284: UDP, length 45
16:25:48.649372 IP 192.168.202.11.49284 > 131.120.9.16.49162: UDP, length 45
```

Figure 56. Test 5: Packet Capture on eth1 of NAT 1

## E.14. Analysis

As soon as Client C dialed the IP address of A, Client C sent out an “INVITE” message to Client A (red outline in Figure 44). The message had embedded SDP information to inform Client A that Client C would be sending and receiving RTP packets at 192.168.3.11 on port 49284 (purple outline in Figure 45). To acknowledge the invitation, Client A sent a “200 OK” packet to Client B with embedded SDP information indicating that it would send and receive RTP packets at 131.120.9.16 on port 49284 (green outline in Figure 44). The exchange of the “INVITE” and “200 OK” messages also occurred for the communication between Clients B and D. Client D informed Client B that it would send and receive RTP packets at 192.168.3.11 on 49162. On the other hand, Client B sent and received RTP packets at 131.120.9.17 on 49128. Figures 44 shows that Client A sent and received RTP packet directly to/from the public IP address of NAT 1. As explained in Appendix C, SJPhone will first attempt to send RTP media packets to the IP address indicated in the SDP messages (or the private IP address of Clients C and D). Since the firewall rules on Client A and Client B were configured to drop packets destined to the private IP address of Clients C and D, none of the initial packets sent out by Clients A or B could reach Clients C or D. Therefore, Clients A and B sent subsequent RTP packets to the IP address where it received Client C and D’s RTP media packets from (blue outline in Figure 44). The packet captures on Clients C indicate that the first RTP packet in the communication was sent by Client C. Even though NAT 1 was not explicitly configured to rewrite the destination IP address of incoming packets to 192.168.1.2 (public IP address of NAT 2), NAT 1 was able to intelligently determine this because *iptables* has a mechanism to maintain connection states of packets that are initiated from the local network. In our scenario, when the first RTP packet sent by Client C reached NAT 1, the packet was processed get changed from 192.168.202.11 to 192.168.120.9. At the same time, NAT 1 created an entry in its connection tracking table to store essential information (such as source and destination IP addresses and ports) that would allow it to associate incoming packets with Client C. This was also true for the communication between Clients D and B (refer to Figures 49 through 54).

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# APPENDIX G. TEST 6: MYSEA VOIP CONFIGURATION

The objective of Test 6 is to confirm that the MYSEA server could support simultaneous VoIP sessions from multiple MLS LAN clients. As described in Chapter IV, each network interface on the MLS server, currently an XTS-400, currently can only have one static route entry. However, two different static routes are needed to route packets destined for Client C and Client D. The test scenarios described here were intended to be workarounds to the XTS-400 routing problem. The goal was to forward packets to the Router if the packets were received on the single-level interface of the MLS server or forward packets to the NAT 1 if the packets were received on the MLS LAN interface of the MLS server. In other words, the goal was to configure XTS-400 to forward packets to its adjacent components, namely NAT 1 and Router, based on the network interface it receives the packets. Table 9 lists network configurations that were applied to the MLS server for the four test scenarios.

| Test Scenario | Device Name | Gateway Address |
|---|---|---|
| 1 | /dev/ether0 | 192.168.0.27 |
| | /dev/ether1 | 192.168.100.88 |
| 2 | /dev/ether0 | 192.168.100.88 |
| | /dev/ether1 | 192.168.0.27 |
| 3 | /dev/ether1 | 192.168.100.88 |
| | /dev/ether0 | 192.168.0.27 |
| 4 | /dev/ether1 | 192.168.0.27 |
| | /dev/ether0 | 192.168.100.88 |

Table 9. Test 6: Test Scenario Configurations

After each set of network configurations was applied to the MLS server, the Unix network utility *ping* was used to confirm the correctness of the network routing. In particular, each of the four IP addresses listed in Table 10 was pinged sequentially for four times from both the Router and NAT 1. Packets were captured using Ethereal at the MLS LAN interface of the Router (eth0, 192.168.0.27) and the single-level interface of NAT 1 (eth1, 192.168.100.88) for post-test analysis.

| Interface | Device | IP Address |
|---|---|---|
| MLS LAN (eth0) | MLS server | 192.168.0.130 |
| single-level (eth1) | MLS server | 192.168.100.130 |
| single-level (eth1) | NAT 1 | 192.168.100.88 |
| public (eth0) | NAT 1 | 131.120.9.15 |

Table 10. Test 6: Ping Operations

None of the four tests was completed successfully, i.e., there was at least one interface on the MLS server and/or the public NAT that the Router or NAT 1 was unable to ping. See the next four sections for the results.

A. Network Topology

Refer to Figure 14 and Figure 15 for the physical and logical network topology. Note that the clients and NAT 3 were not used in the test scenarios.

B. Equipment Requirements

B.1. NAT 1, NAT 2 and Router

B.1.1. Linux Operating System (Fedora Core 4)

B.1.2. netfilter and iptables

B.1.3. Ethereal

B.1.4. Two network cards (for NAT 1 and NAT 2)

B.1.5. Three network cards (for Router only)

B.2. MLS Server

B.2.1. XTS-400

B.3. Additional Equipment

B.3.1. Cross-over cables and a switch or hub to implement the network architecture illustrated in Figure 14.

C. Installation and Configuration

C.1. MLS Server

C.1.1. Configure two network interfaces to be at the same level as the MLS LAN by entering:

`min` as the security level

max as the integrity level

C.2. Router

C.2.1. Configure eth0 by editing /etc/sysconfig/network-scripts/ifcfg-eth0 to include the following:

```
DEVICE=eth0
BOOTPROTO=NONE
IPADDR=192.168.0.27
NETMASK=255.255.255.0
GATEWAY=192.168.0.130
```

C.2.2. Activate eth0 by running:

```
ifup eth0
```

C.2.3. Configure eth1 by editing /etc/sysconfig/network-scripts/ifcfg-eth1 to include the following:

```
DEVICE=eth1
BOOTPROTO=NONE
IPADDR=192.168.202.1
NETMASK=255.255.255.0
```

C.2.4. Activate eth1 by running:

```
ifup eth1
```

C.2.5. Configure eth2 by editing and saving /etc/sysconfig/network-scripts/ifcfg-eth2 to include the following:

```
DEVICE=eth2
BOOTPROTO=NONE
IPADDR=192.168.2.1
NETMASK=255.255.255.0
```

C.2.6. Activate eth2 by running:

```
ifup eth2
```

C.2.7. Enable IP Forwarding by running:

```
echo 1 > /proc/sys/net/ipv4/ip_forward
```

C.2.8. Flush any existing firewall and NAT rules by running:

```
iptables -F
```

iptables -t nat -F

C.3. NAT 1

C.3.1. Configure eth0 by editing /etc/sysconfig/network-scripts/ifcfg-eth0 to include the following:

```
DEVICE=eth0
BOOTPROTO=NONE
IPADDR=131.120.9.15
NETMASK=255.255.255.0
GATEWAY=131.120.9.17
```

C.3.2. Activate eth0 by running:

```
ifup eth0
```

C.3.3. Configure eth1 by editing and saving /etc/sysconfig/network-scripts/ifcfg-eth1 to include the following:

```
DEVICE=eth1
BOOTPROTO=NONE
IPADDR=192.168.100.88
NETMASK=255.255.255.0
```

C.3.4. Activate eth1 by running:

```
ifup eth1
```

C.3.5. Configure static routes by running:

```
route add –net 192.168.202.0 netmask 255.255.255.0 gw 192.168.100.130 eth1
route add –net 192.168.2.0 netmask 255.255.255.0 gw 192.168.100.130 eth1
route add –net 192.168.0.0 netmask 255.255.255.0 gw 192.168.100.130 eth1
```

C.3.6. Enable IP Forwarding by running:

```
echo 1 > /proc/sys/net/ipv4/ip_forward
```

C.3.7. Flush any existing firewall and NAT rules by running:

```
iptables -F
iptables -t nat -F
```

C.3.8. Configure NAT rule by running:

```
iptables -t nat -A POSTROUTING -o eth0 -j SNAT --to 131.120.9.15
```

C.4. NAT 2

C.4.1. Configure eth0 by editing and saving /etc/sysconfig/network-scripts/ifcfg-eth0 to include the following:

```
DEVICE=eth0
BOOTPROTO=NONE
IPADDR=196.168.202.11
NETMASK=255.255.255.0
GATEWAY=198.168.202.1
```

C.4.2. Activate eth0 by running:

```
ifup eth0
```

C.4.3. Configure eth1 by editing and saving /etc/sysconfig/network-scripts/ifcfg-eth1 to include the following:

```
DEVICE=eth1
BOOTPROTO=NONE
IPADDR=192.168.3.1
NETMASK=255.255.255.0
```

C.4.4. Activate eth1 by running:

```
ifup eth1
```

C.4.5. Enable IP Forwarding by running:

```
echo 1 > /proc/sys/net/ipv4/ip_forward
```

C.4.6. Flush any existing firewall and NAT rules by running:

```
iptables -F
iptables -t nat -F
```

C.4.7. Configure NAT rules by running:

```
iptables -t nat -A POSTROUTING -o eth0 -j SNAT --to 192.168.202.11
iptables -t nat -A PREROUTING -i eth0 -j DNAT --to 192.168.3.11
```

D. Scenario 1

D.1. Description

The MLS server is configured in order as follows: any packets received on its MLS LAN (eth0, 192.168.0.130) is forwarded to the MLS LAN interface (eth0, 192.168.0.27) of the Router and any packets received on its single-level (eth1, 192.168.100.130) is forwarded to the single-level interface (eth1, 192.168.100.88) of NAT 1.

D.2. Operations

First, NAT 2 pings:

1. eth0 of MLS server
2. eth1 of MLS server
3. eth1 of NAT 1
4. eth0 of NAT 1

Then, Router pings:

5. eth0 of MLS server
6. eth1 of MLS server
7. eth1 of NAT 1
8. eth0 of NAT 1

D.3. Network Configuration on MLS Server

D.3.1. Type the following answers when prompted:

SAK

Enter command? tcpip_edit

| | |
|---|---|
| Enter editor request? | add |
| Enter TCP/IP daemon name? | tcpip_mls |
| Enter TCPIP/IP daemon description? | TCP/IP for MLS LAN network |
| Enter domain name? | cisrlabmlstestbed1.com |
| Enter host name? | mlsserver |
| Enable the subnets local flag? | n |
| Enable the IP forwarding flag? | y |
| Enable the IP send redirect flag? | y |
| Enable the shutdown on failure flag? | n |
| Use default TCP maximum retransmission? | y |
| Add the network interface configuration? | y |
| Enter TCP/IP device name? | /dev/ether0 |
| Enter interface address? | 192.168.0.130 |

| | |
|---|---|
| Enter destination address? | 0.0.0.0 |
| Enter broadcast address? | 192.168.0.255 |
| Enter network mask? | 255.255.255.0 |
| Add another network interface entry? | y |
| Enter TCP/IP device name? | /dev/ether1 |
| Enter interface address? | 192.168.100.130 |
| Enter destination address? | 0.0.0.0 |
| Enter broadcast address? | 192.168.100.255 |
| Enter network mask? | 255.255.255.0 |
| Add another network interface entry? | n |
| Add the route configuration? | y |
| Enter TCP/IP device name | /dev/ether0 |
| Is this route a default route | n |
| Enter destination address | 0.0.0.0 |
| Is destination address a host | n |
| Enter gateway address | 192.168.0.27 |
| Enter route metric | 1 |
| Add another network route entry | y |
| Enter TCP/IP device name | /dev/ether1 |
| Is this route a default route | n |
| Enter destination address | 0.0.0.0 |
| Is destination address a host | n |
| Enter gateway address | 192.168.100.88 |
| Enter route metric | 1 |
| Add another network route entry | n |
| Add the resolver configuration? | n |

D.4. Preparation and Testing

D.4.1. On NAT 1,

D.4.1.1. Launch Ethereal

D.4.1.2. Go to the Capture menu

D.4.1.3. Go to Interfaces

D.4.1.4. Click on Capture 192.168.100.88

D.4.2. On Router,

D.4.2.1. Launch Ethereal

D.4.2.2. Go to the Capture menu

D.4.2.3. Go to Interfaces

D.4.2.4. Click on Capture 192.168.0.27

D.4.3. On NAT 2,

D.4.3.1. Run the following commands:

```
ping –c 4 192.168.0.130
ping –c 4 192.168.100.130
ping –c 4 192.168.100.88
ping –c 4 131.120.9.15
```

D.4.4. Repeat the above commands on the Router

D.4.5. Stop Ethereal captures on both NAT 1 and Router

D.5. Result

Table 11 lists the result of the Scenario 1. The first column shows where the ping was initiated and the first row shows what hosts/IP addresses were pinged. Neither NAT 2 nor the Router was able to ping the public interface of the Public NAT.

| from \ to | 192.168.0.130 (eth0, MLS LAN interface of MLS server) | 192.168.100.130 (eth1, single-level interface of MLS server) | 192.168.100.88 (eth1, single-level interface of NAT 1) | 131.120.9.15 (eth0, public interface of NAT 1) |
|---|---|---|---|---|
| NAT 2 | Successful | Successful | Successful | Failed |
| Router | Successful | Successful | Successful | Failed |

Table 11. Test 6: Scenario 1 Result

D.6. Packet Capture when pinged from NAT 2

D.6.1. Router

The following two figures are snapshots of the packets captured on eth0 of Router when the four interfaces were pinged from NAT 2.

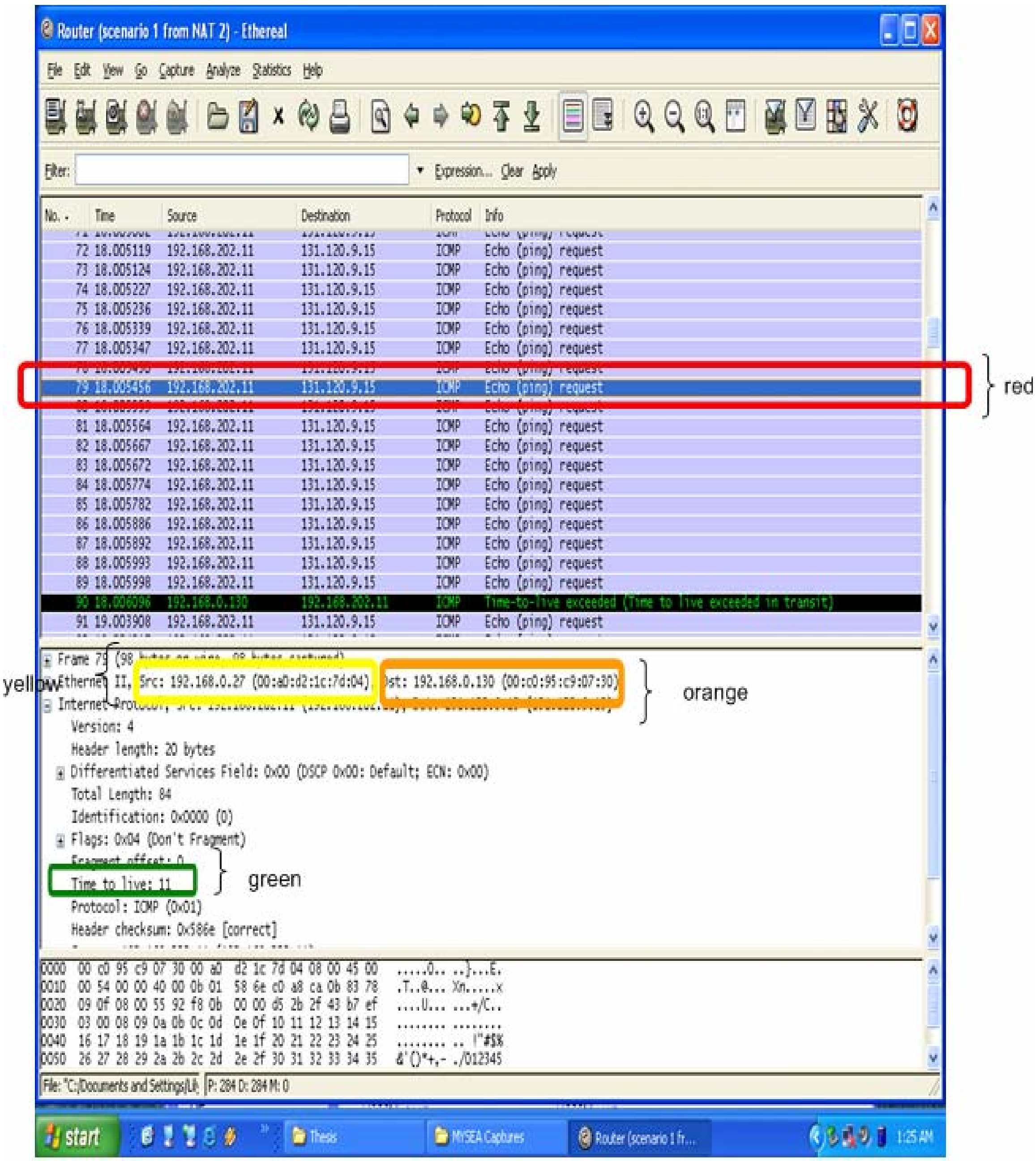


Figure 57. Test 6: Scenario 1 Packet Capture on Router (pinged from NAT 2), Part 1

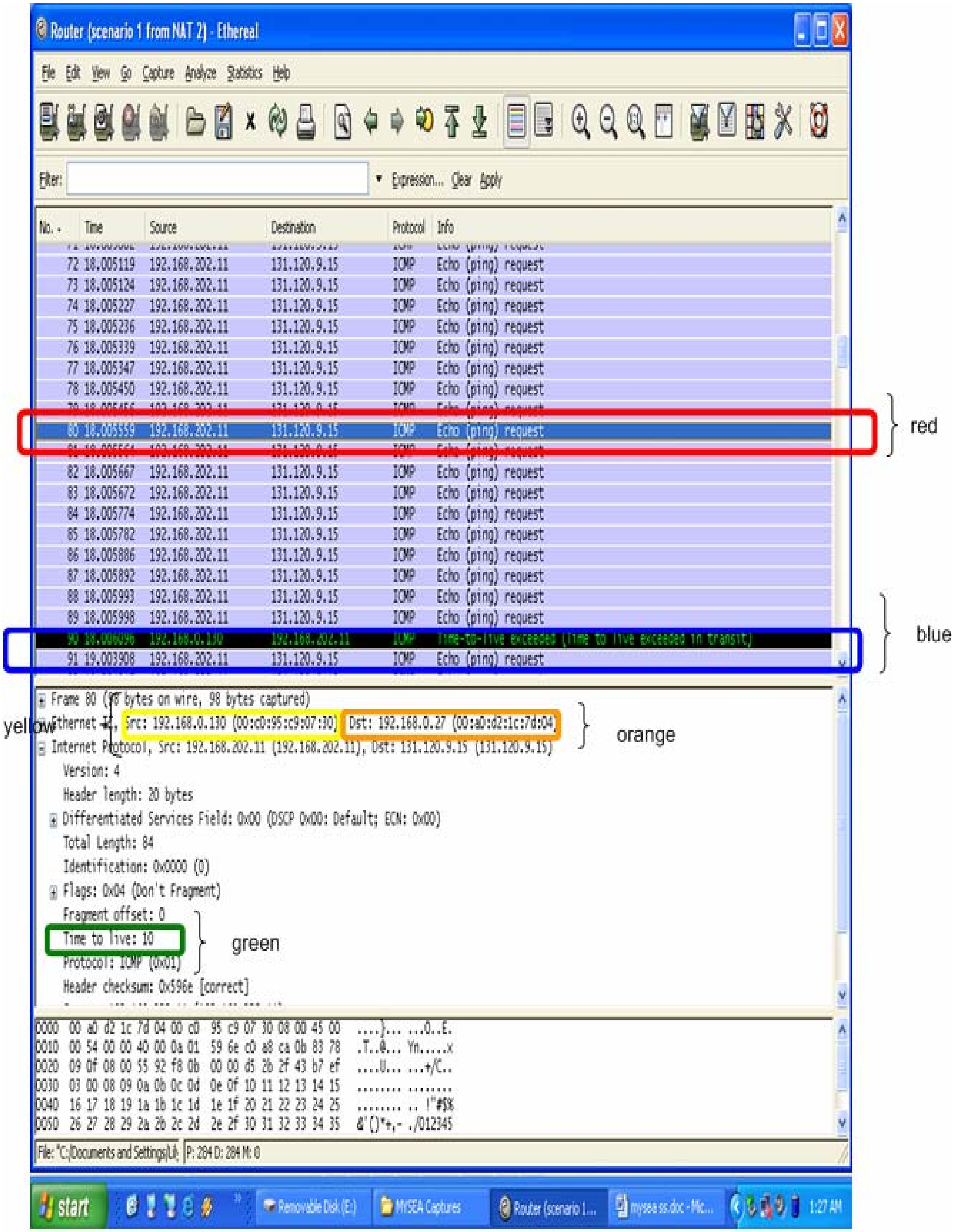


Figure 58. Test 6: Scenario 1 Packet Capture on Router (pinged from NAT 2), Part 2

D.6.2. NAT 1

The following is a snapshot of the packets captured on the eth1 of NAT 1 when the four interfaces were pinged from NAT 2.

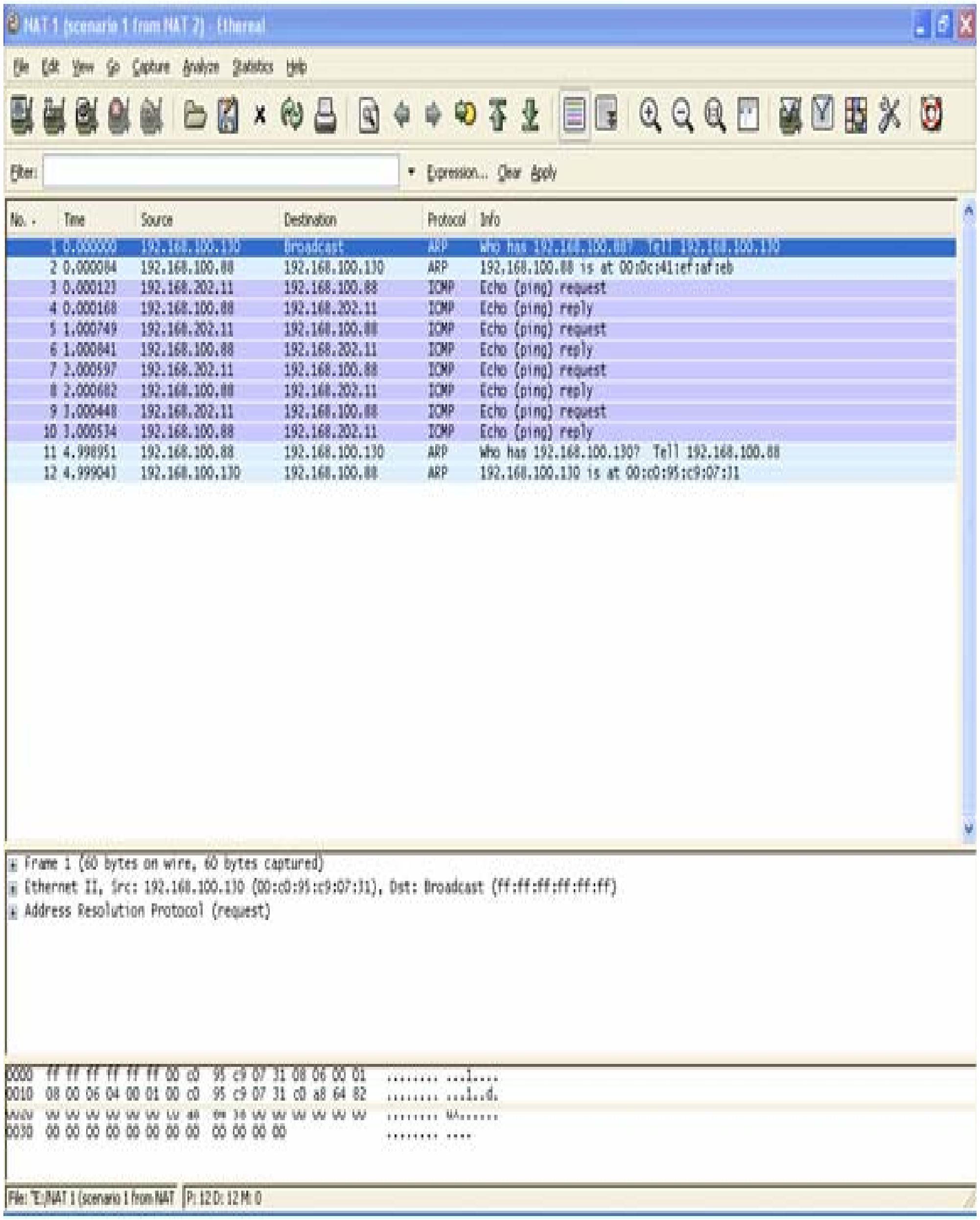


Figure 59. Test 6: Scenario 1 Packet Capture on NAT 1 (pinged from NAT 2)

### D.6.3. Analysis

NAT 2 was able to ping 192.168.0.130 and 192.168.100.130 because the MLS server shared a peer-to-peer relationship with the Router and the Router had logic to route packets between the MLS server and NAT 2. It was also able to ping 192.168.100.88 because XTS-400 was capable of routing the Echo requests and replies to its immediate peers that in turn, routed the packets to the destination. Pinging 131.120.9.15 was unsuccessful from NAT 2. The Router saw ICMP requests when NAT 2 pinged 131.120.9.15 (red outline in Figure 57). The ICMP requests were routed from 192.168.0.27 (yellow outline in Figure 57) to 192.168.0.130 (orange outline in Figure 57). As soon as the MLS server received the requests, the XTS-400 bounced them back to the IP address from which they came (yellow and orange outlines in Figure 58). Note that the time-to-live field was decremented from 11 (green outline in Figure 57) to 10 (green outline in Figure 58) indicating that the two requests seen in the packet capture were, in fact, the same packet. This sequence of events continued until the time-to-live was exceeded in transit (blue outline in Figure 57). Thus, the ICMP was never able to reach NAT 1 as shown in Figure 59.

D.7. Packet Capture when pinged from Router

D.7.1. Router

The following two pictures are snapshots of the packets captured on eth0 of the Router when the four interfaces were pinged from Router.

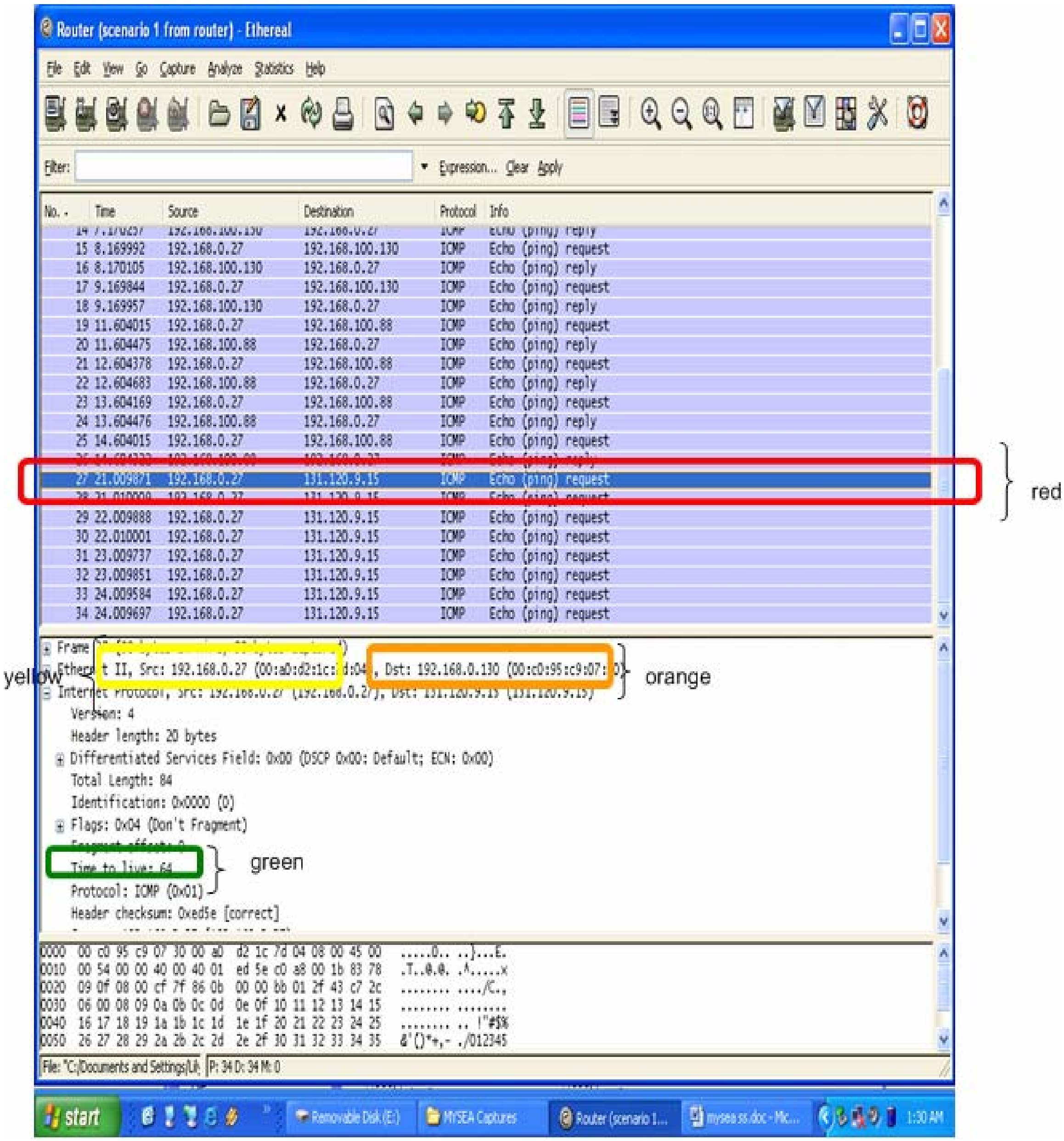


Figure 60. Test 6: Scenario 1 Packet Capture on Router (pinged from Router), Part 1

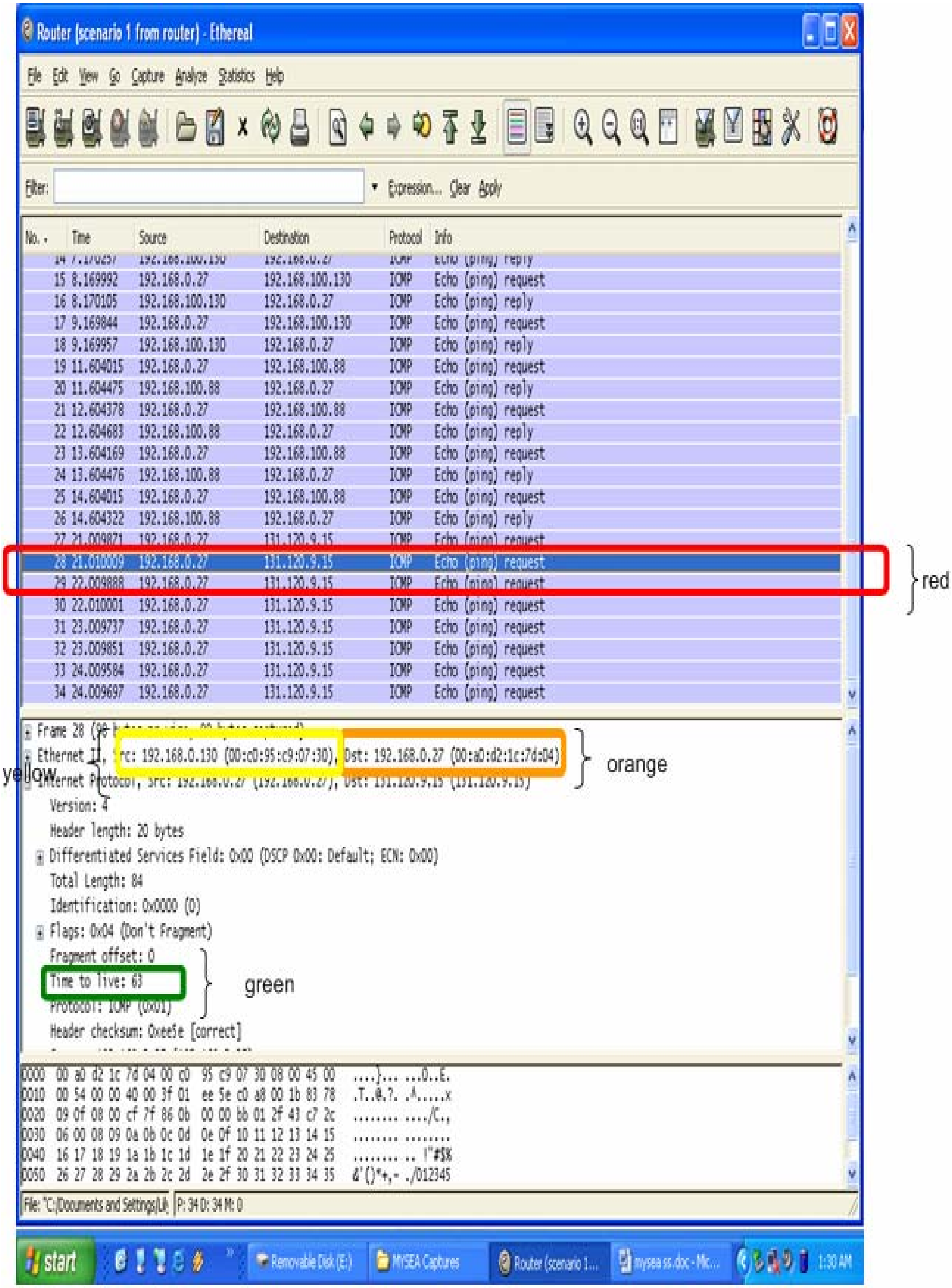


Figure 61. Test 6: Scenario 1 Packet Capture on Router (pinged from Router), Part 2

D.7.2. NAT 1

The following is a snapshot of the packets captured on the NAT 1 when the four interfaces were pinged from Router.

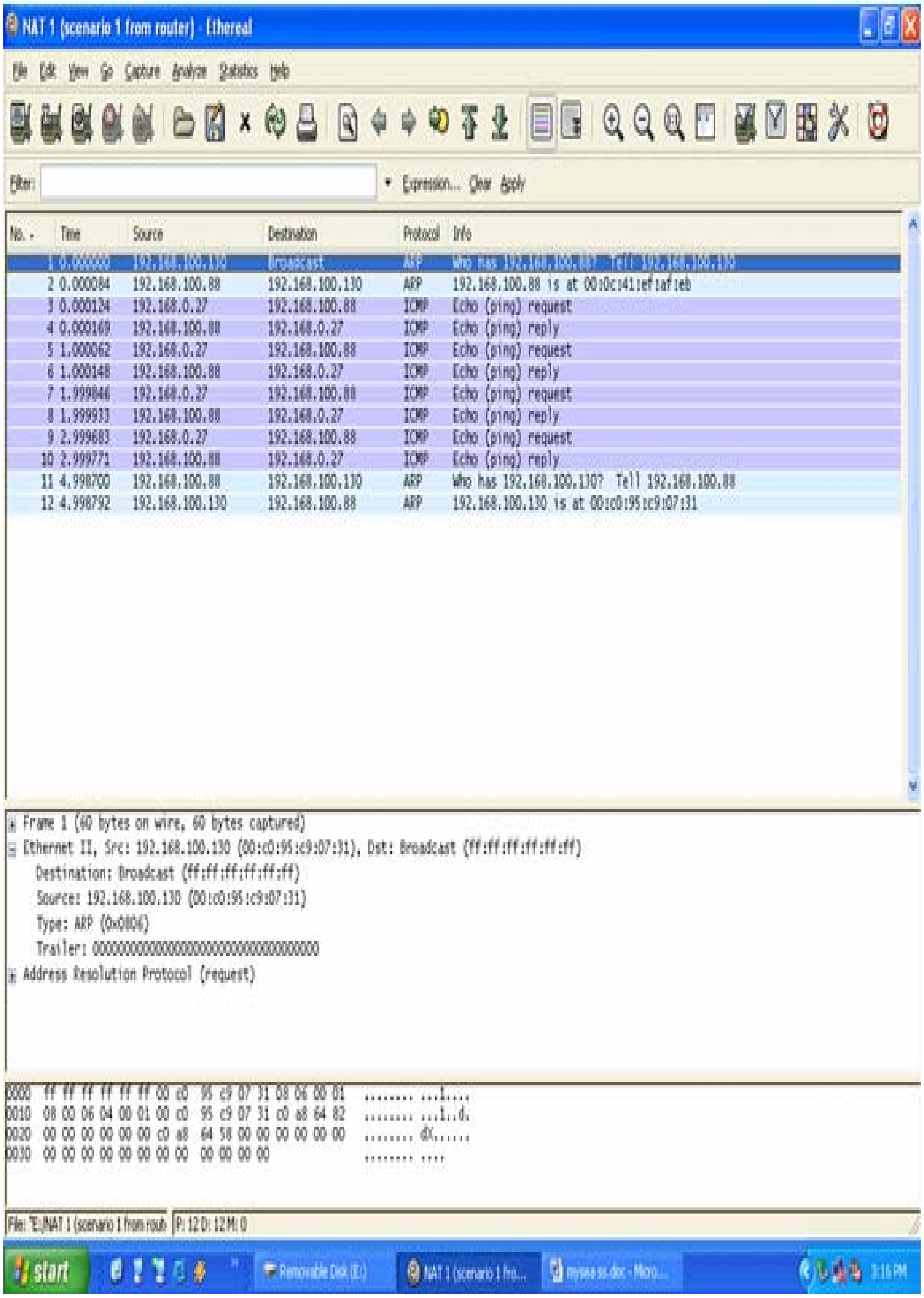


Figure 62. Test 6: Scenario 1 Packet Capture on NAT 1 (pinged from Router)

D.7.3. Analysis

Figures 60 to 62 indicate similar behaviors when the four interfaces were pinged from the Router instead of NAT 2. The Router could ping 192.168.0.130, 192.168.100.130 and 192.168.100.88 for the reason explained in D.6.3. Pinging 131.120.9.15 from the Router had a different behavior than the one seen when it was pinged from NAT 2 such that the Echo requests never had their time-to-live exceeded. Each of the four Echo requests was routed between 192.168.0.27 and 192.168.0.130 twice (yellow and orange outlines in Figure 60 and Figure 61). Each request was routed from 192.168.0.27 first and then XTS-400 routed it back to 192.168.0.130 (Router) as soon as XTS-400 received it. The decrement in the time-to-live field indicated that the same request was routed back and forth (green outlines in Figure 60 and Figure 61). The request was not routed further when the Router received it because the Router recognized its own packet and thus, stopped routing it. As a result, NAT 1 never saw the Echo requests as shown in Figure 62.

E. Scenario 2

E.1. Description

The MLS server is configured in order as follows: any packets received on its MLS LAN interface (eth0, 192.168.0.130) is forwarded to the single-level interface of NAT 1 (eth1, 192.168.100.88) and any packets received on its single-level interface (eth1, 192.168.100.130) is forwarded to the public interface of Router (eth0, 192.168.0.27).

E.2. Operations

First, NAT 2 pings:

1. eth0 of MLS server
2. eth1 of MLS server
3. eth1 of NAT 1
4. eth0 of NAT 1

Then, Router pings:

5. eth0 of MLS server
6. eth1 of MLS server
7. eth1 of NAT 1

8. eth0 of NAT 1

## E.3. Network Configuration on MLS Server

### E.3.1. Type the following answers when prompted:

SAK

| | |
|---|---|
| Enter command? | tcpip_edit |
| Enter editor request? | add |
| Enter TCP/IP daemon name? | tcpip_mls |
| Enter TCPIP/IP daemon description? | TCP/IP for MLS LAN network |
| Enter domain name? | cisrlabmlstestbed1.com |
| Enter host name? | mlsserver |
| Enable the subnets local flag? | n |
| Enable the IP forwarding flag? | y |
| Enable the IP send redirect flag? | y |
| Enable the shutdown on failure flag? | n |
| Use default TCP maximum retransmission? | y |
| Add the network interface configuration? | y |
| Enter TCP/IP device name? | /dev/ether0 |
| Enter interface address? | 192.168.0.130 |
| Enter destination address? | 0.0.0.0 |
| Enter broadcast address? | 192.168.0.255 |
| Enter network mask? | 255.255.255.0 |
| Add another network interface entry? | y |
| Enter TCP/IP device name? | /dev/ether1 |
| Enter interface address? | 192.168.100.130 |
| Enter destination address? | 0.0.0.0 |
| Enter broadcast address? | 192.168.100.255 |
| Enter network mask? | 255.255.255.0 |
| Add another network interface entry? | n |
| Add the route configuration? | y |
| Enter TCP/IP device name | /dev/ether0 |

| | |
|---|---|
| Is this route a default route | n |
| Enter destination address | 0.0.0.0 |
| Is destination address a host | n |
| Enter gateway address | 192.168.100.88 |
| Enter route metric | 1 |
| Add another network route entry | y |
| Enter TCP/IP device name | /dev/ether1 |
| Is this route a default route | n |
| Enter destination address | 0.0.0.0 |
| Is destination address a host | n |
| Enter gateway address | 192.168.0.27 |
| Enter route metric | 1 |
| Add another network route entry | n |
| Add the resolver configuration? | n |

E.4. Preparation and Testing

E.4.1. On NAT 1,

E.4.1.1. Launch Ethereal

E.4.1.2. Go to the Capture menu

E.4.1.3. Go to Interfaces

E.4.1.4. Click on Capture 192.168.100.88

E.4.2. On Router,

E.4.2.1. Launch Ethereal

E.4.2.2. Go to the Capture menu

E.4.2.3. Go to Interfaces

E.4.2.4. Click on Capture 192.168.0.27

E.4.3. On NAT 2,

E.4.3.1. Run the following commands:

```
ping –c 4 192.168.0.130
ping –c 4 192.168.100.130
ping –c 4 192.168.100.88
ping –c 4 131.120.9.15
```

E.4.4. Repeat the above commands on the Router

E.4.5. Stop Ethereal captures on both NAT 1 and Router

E.5. Result

Table 12 lists the result of the Scenario 2. The first column shows where the ping was initiated and the first row shows what hosts/IP addresses were pinged. NAT 2 was unable to ping any of the four IP addresses.

| to / from | 192.168.0.130 (eth0, MLS LAN interface of MLS server) | 192.168.100.130 (eth1, single-level interface of MLS server) | 192.168.100.88 (eth1, single-level interface of NAT 1) | 131.120.9.15 (eth0, public interface of NAT 1) |
|---|---|---|---|---|
| NAT 2 | Failed | Failed | Failed | Failed |
| Router | Successful | Successful | Successful | Successful |

Table 12. Test 6: Scenario 2 Result

E.6. Packet Capture when pinged from NAT 2

E.6.1. Router

The following is a snapshot of the packets captured on the Router when the four interfaces were pinged from NAT 2.

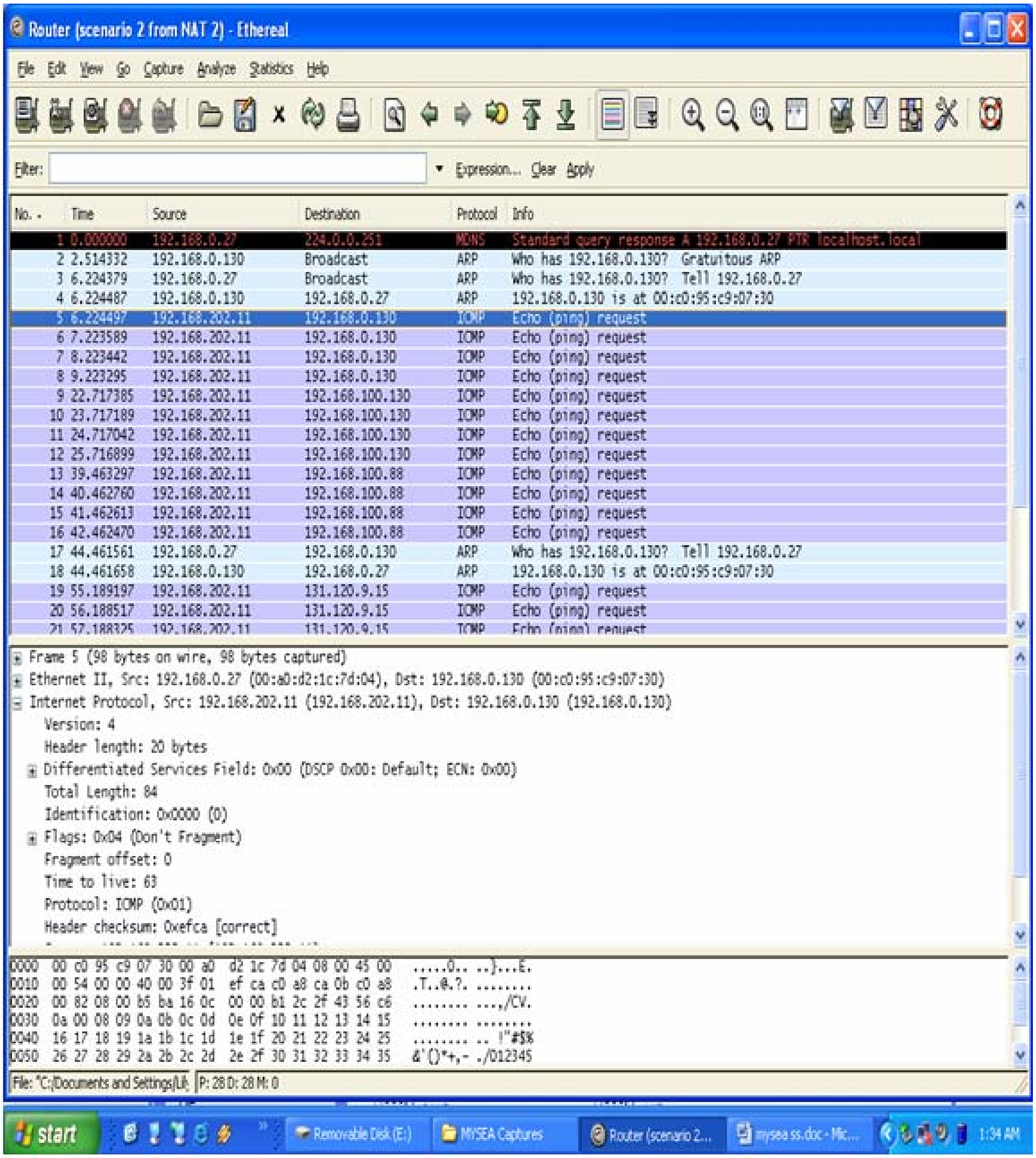

Figure 63. Test 6: Scenario 2 Packet Capture on Router (pinged from NAT 2)

E.6.2. NAT 1

The following four figures are snapshots of the packets captured on NAT 1 when the four interfaces were pinged from NAT 2.

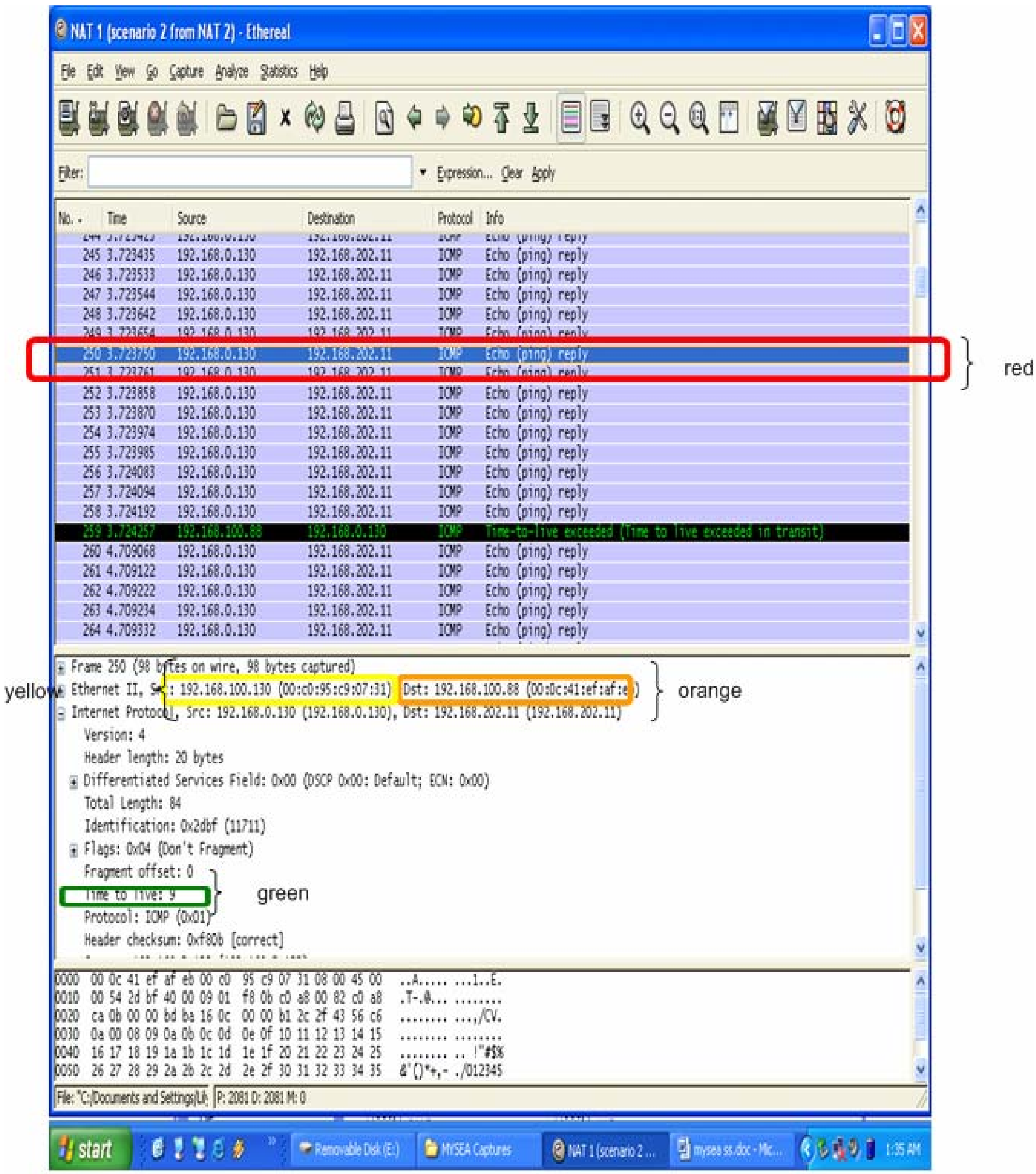


Figure 64. Test 6: Scenario 2 Packet Capture on NAT 1 (pinged from NAT 2), Part 1

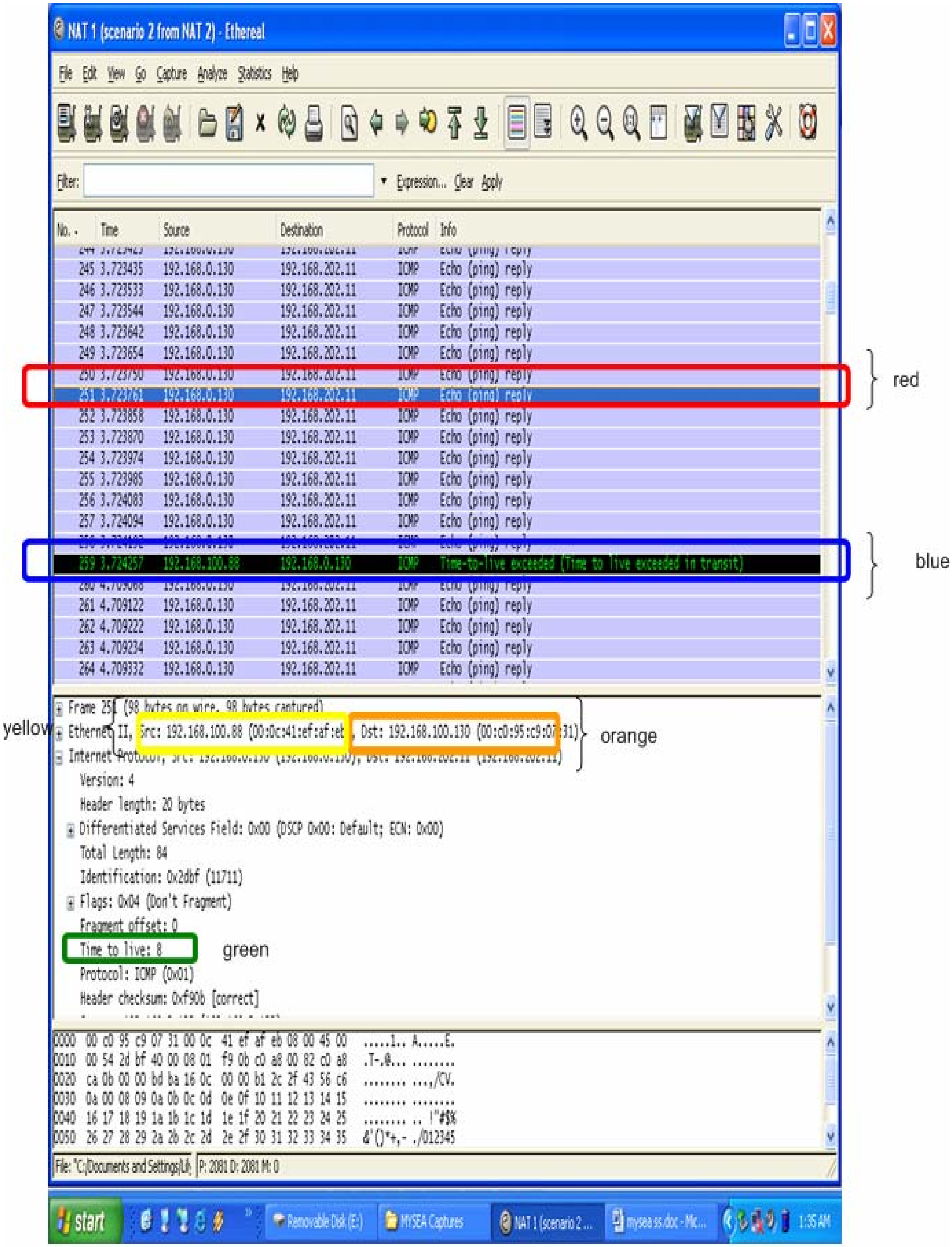


Figure 65. Test 6: Scenario 2 Packet Capture on NAT 1 (pinged from NAT 2), Part 2

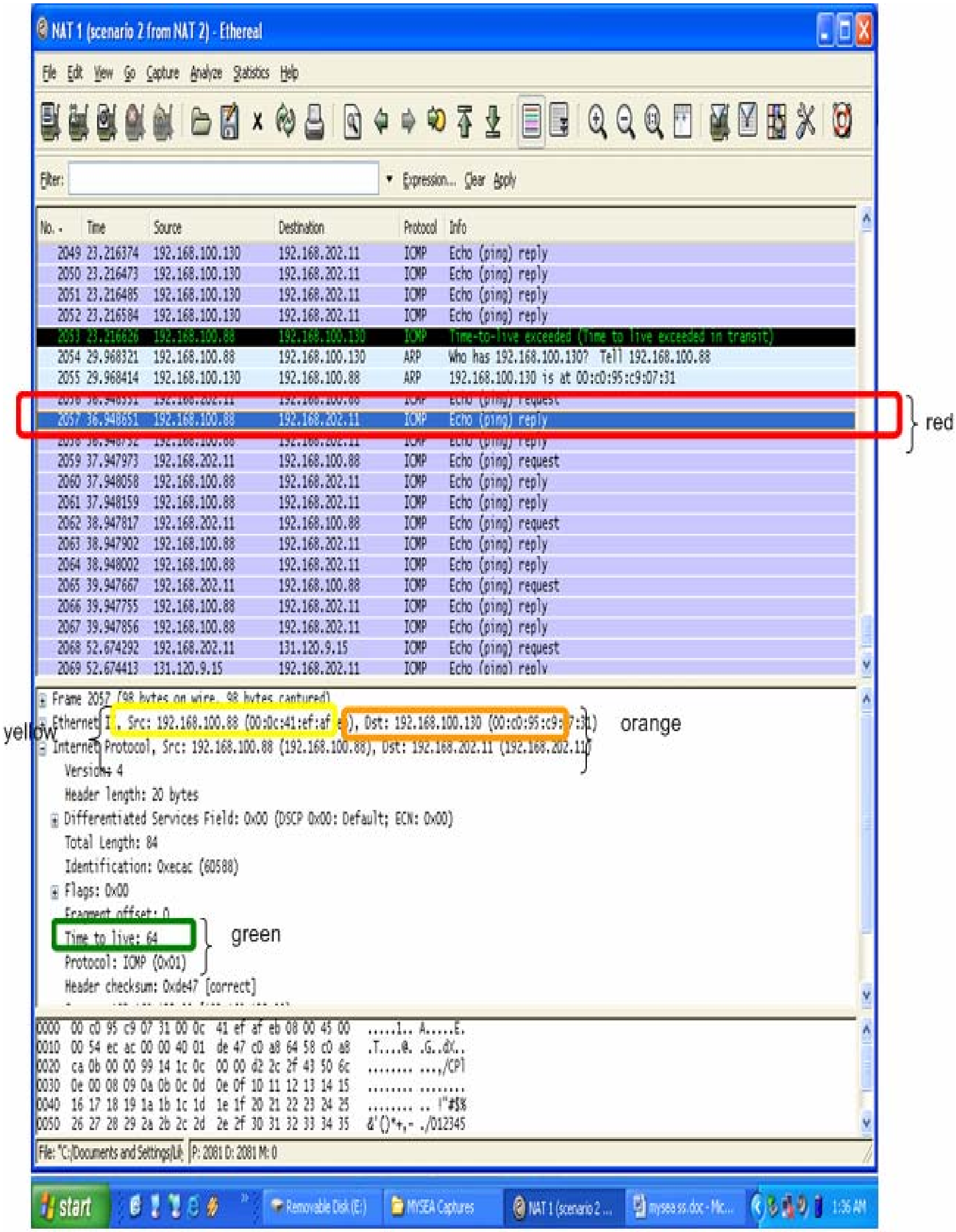


Figure 66. Test 6: Scenario 2 Packet Capture on NAT 1 (pinged from NAT 2), Part 3

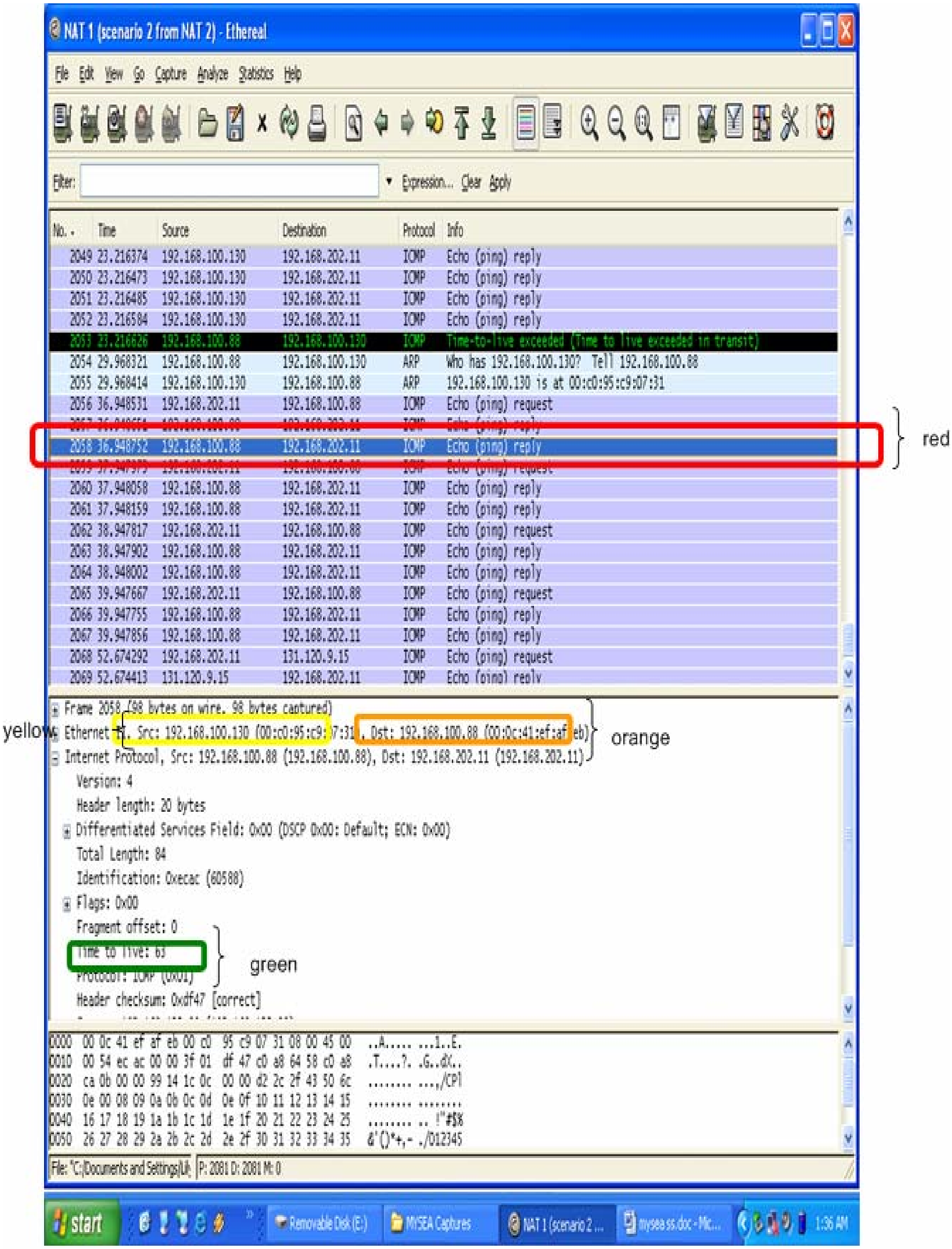


Figure 67. Test 6: Scenario 2 Packet Capture on NAT 1 (pinged from NAT 2), Part 4

E.6.3. Analysis

NAT 2 failed to ping 192.168.0.130 and 192.168.100.130.. Figure 63 show that the Router did not see any Echo replies from 192.168.0.130 or 192.168.100.130. However, it is evident that 192.168.0.130 and 192.168.100.130 replied since NAT 1 saw them (red outline in Figure 64). If routing was done correctly, the Echo replies should have been sent to the Router and then to NAT 2 instead of to NAT 1. Instead, MLS server sent the Echo replies to NAT 1. Figure 65 indicated that when NAT 1 received the Echo replies, it sent them to next hop (192.168.100.130) based on its routing table (yellow and orange outlines in Figure 65). However, XTS-400 forwarded the replies back NAT 1 at 192.168.100.88 (yellow and orange outlines in Figure 64). Hence, Echo replies were bounced back and forth between 192.168.100.130 and 192.168.100.88 until the time-to-live field reached zero (blue outline in Figure 65). This sequence of events also occurred for the Echo requests from 192.168.202.11 to 192.168.100.130.

NAT 2 also failed to ping 192.168.100.88 and 131.120.9.15. Figures 66 and 67 show that there exists two Echo replies for each Echo request destined for 192.168.100.88 and 131.120.9.15. For each Echo request destined for 192.168.100.88, NAT 1 responded with an Echo reply which is sent to its next hop at 192.168.100.130 (orange outline in Figure 66). However, the same Echo reply was bounced back by XTS-400 to where it came from when XTS-400 received it (yellow and orange outline in Figure 67). NAT 1 stopped routing the Echo reply further since it recognized its own packet. This explains why NAT 2 was never able to receive any replies. The same was true when NAT 2 pinged 131.120.9.15.

E.7. Packet Capture when pinged from Router

E.7.1. Router

The following is a snapshot of the packets captured on the Router when the four interfaces were pinged from the Router.

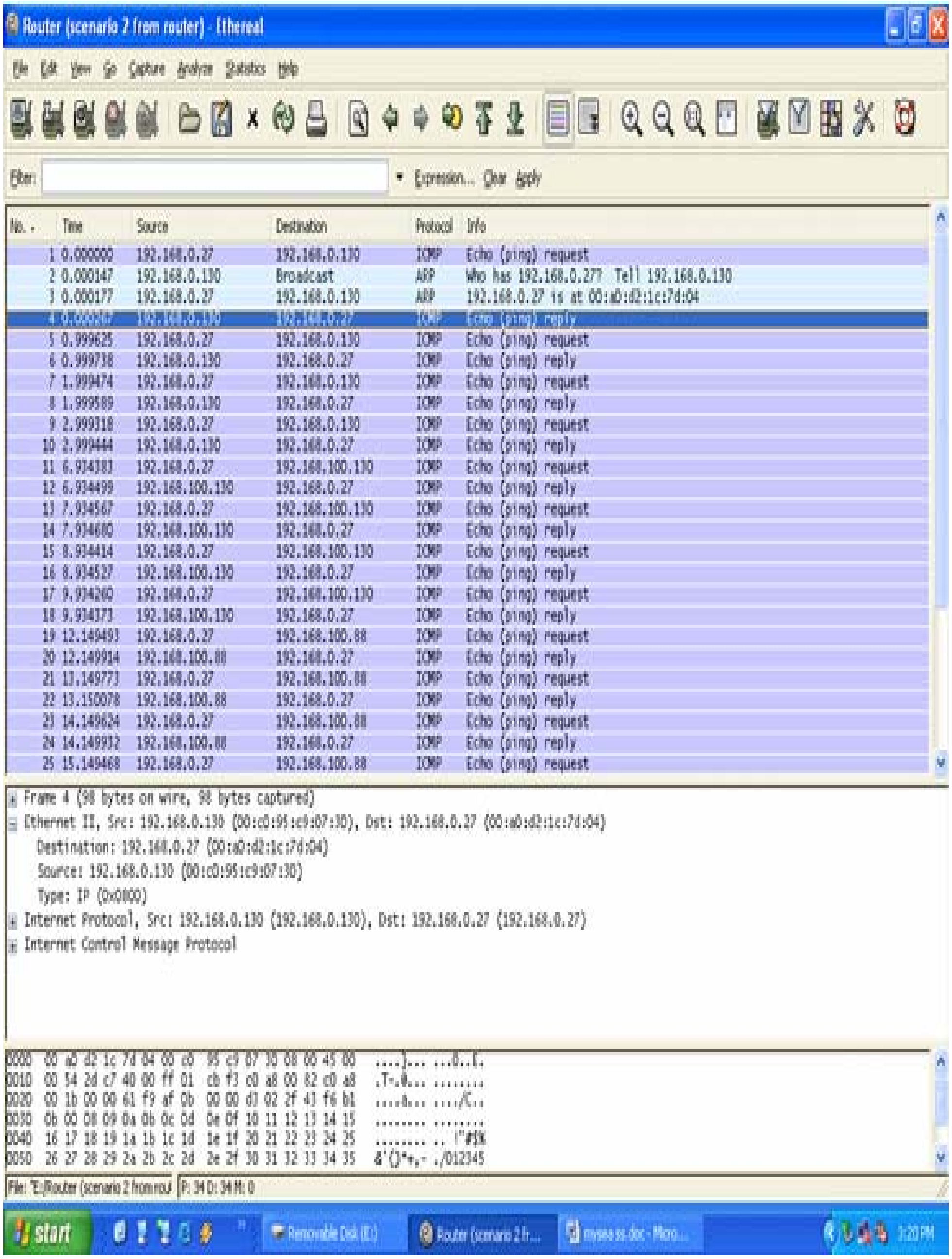


Figure 68. Test 6: Scenario 2 Packet Capture on Router (pinged from Router)

E.7.2. NAT 1

The following is a snapshot of the packets captured on NAT 1 when the four interfaces were pinged from the Router.

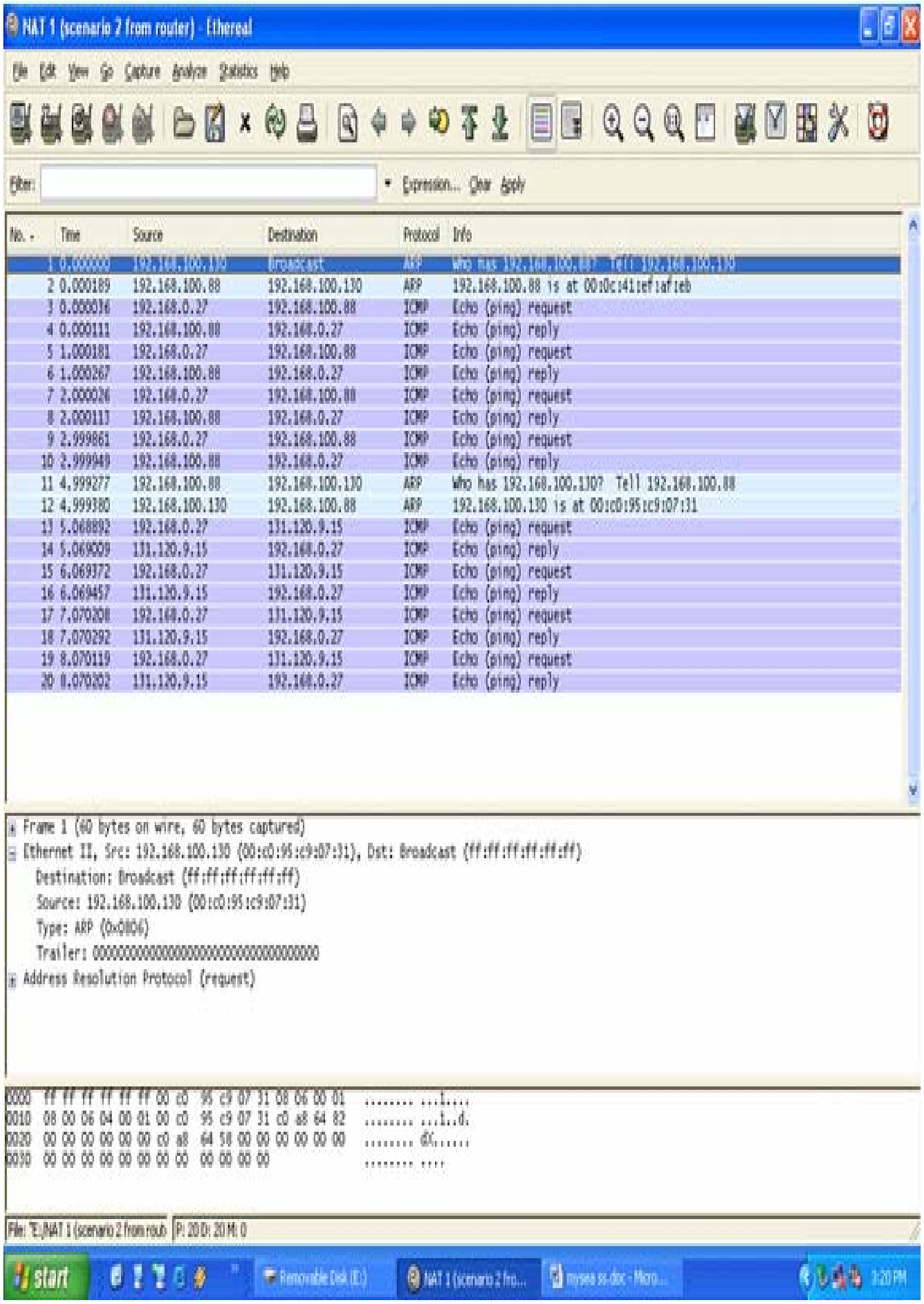

Figure 69. Test 6: Scenario 2 Packet Capture on NAT 1 (pinged from Router)

E.7.3. Analysis

The Router was able to ping all four interfaces described in D.2. The Router could ping 192.168.0.130 because it shares a peer-to-peer relationship with the MLS server. It was also able to ping 192.168.100.130 because XTS-400 knew its interfaces. The same logic applied to the case when the Router pinged 192.168.100.88. The Router successfully pinged 131.120.9.15 in this scenario but not in Scenario 1. In order for this to occur, XTS-400 would have to send the Echo requests to 192.168.100.88 in order for the requests to reach 131.120.9.15. When 131.120.9.15 received the requests, it sent Echo replies to the Router by routing the packet to the next hop or the MLS server. XTS-400 had the logic to route the replies to the Router since it knew its peers.

F. Scenario 3

F.1. Description

The MLS server is configured in order as follows: any packets received on its single-level interface (eth1, 192.168.100.130) is forwarded to the single-level interface of NAT 1 (192.168.100.88) and any packets received on its MLS LAN interface (192.168.0.130) is forwarded to the public interface of the Router (192.168.0.27).

F.2. Operations

First, NAT 2 pings:

1. eth0 of MLS server
2. eth1 of MLS server
3. eth1 of NAT 1
4. eth0 of NAT 1

Then, Router pings:

5. eth0 of MLS server
6. eth1 of MLS server
7. eth1 of NAT 1
8. eth0 of NAT 1

F.3. Network Configuration on MLS Server

F.3.1. Type the following answers when prompted:

SAK

| | |
|---|---|
| Enter command? | tcpip_edit |
| Enter editor request? | add |
| Enter TCP/IP daemon name? | tcpip_mls |
| Enter TCPIP/IP daemon description? | TCP/IP for MLS LAN network |
| Enter domain name? | cisrlabmlstestbed1.com |
| Enter host name? | mlsserver |
| Enable the subnets local flag? | n |
| Enable the IP forwarding flag? | y |
| Enable the IP send redirect flag? | y |
| Enable the shutdown on failure flag? | n |
| Use default TCP maximum retransmission? | y |
| Add the network interface configuration? | y |
| Enter TCP/IP device name? | /dev/ether0 |
| Enter interface address? | 192.168.0.130 |
| Enter destination address? | 0.0.0.0 |
| Enter broadcast address? | 192.168.0.255 |
| Enter network mask? | 255.255.255.0 |
| Add another network interface entry? | y |
| Enter TCP/IP device name? | /dev/ether1 |
| Enter interface address? | 192.168.100.130 |
| Enter destination address? | 0.0.0.0 |
| Enter broadcast address? | 192.168.100.255 |
| Enter network mask? | 255.255.255.0 |
| Add another network interface entry? | n |
| Add the route configuration? | y |
| Enter TCP/IP device name | /dev/ether1 |
| Is this route a default route | n |
| Enter destination address | 0.0.0.0 |
| Is destination address a host | n |

| | |
|---|---|
| Enter gateway address | 192.168.100.88 |
| Enter route metric | 1 |
| Add another network route entry | y |
| Enter TCP/IP device name | /dev/ether0 |
| Is this route a default route | n |
| Enter destination address | 0.0.0.0 |
| Is destination address a host | n |
| Enter gateway address | 192.168.0.27 |
| Enter route metric | 1 |
| Add another network route entry | n |
| Add the resolver configuration? | n |

F.4. Preparation and Testing

F.4.1. On NAT 1,

F.4.1.1. Launch Ethereal

F.4.1.2. Go to the Capture menu

F.4.1.3. Go to Interfaces

F.4.1.4. Click on Capture 192.168.100.88

F.4.2. On Router,

F.4.2.1. Launch Ethereal

F.4.2.2. Go to the Capture menu

F.4.2.3. Go to Interfaces

F.4.2.4. Click on Capture 192.168.0.27

F.4.3. On NAT 2,

F.4.3.1. Run the following commands:

ping –c 4 192.168.0.130

ping –c 4 192.168.100.130

ping –c 4 192.168.100.88

ping –c 4 131.120.9.15

F.4.4. Repeat the above commands on the Router

F.4.5. Stop Ethereal captures on both NAT 1 and Router

F.5. Result

Table 13 lists the result of the Scenario 3. The first column shows where the ping was initiated and the first row shows what hosts/IP addresses were pinged. The result is exactly the same as the result obtained in Scenario 2.

| to / from | 192.168.0.130 (eth0, MLS LAN interface of MLS server) | 192.168.100.130 (eth1, single-level interface of MLS server) | 192.168.100.88 (eth1, single-level interface of NAT 1) | 131.120.9.15 (eth0, public interface of NAT 1) |
|---|---|---|---|---|
| NAT 2 | Failed | Failed | Failed | Failed |
| Router | Successful | Successful | Successful | Successful |

Table 13. Test 6: Scenario 3 Result

F.6. Packet Capture when pinged from NAT 2

F.6.1. Router

The following is a snapshot of the packets captured on the Router when the four interfaces were pinged from NAT 2.

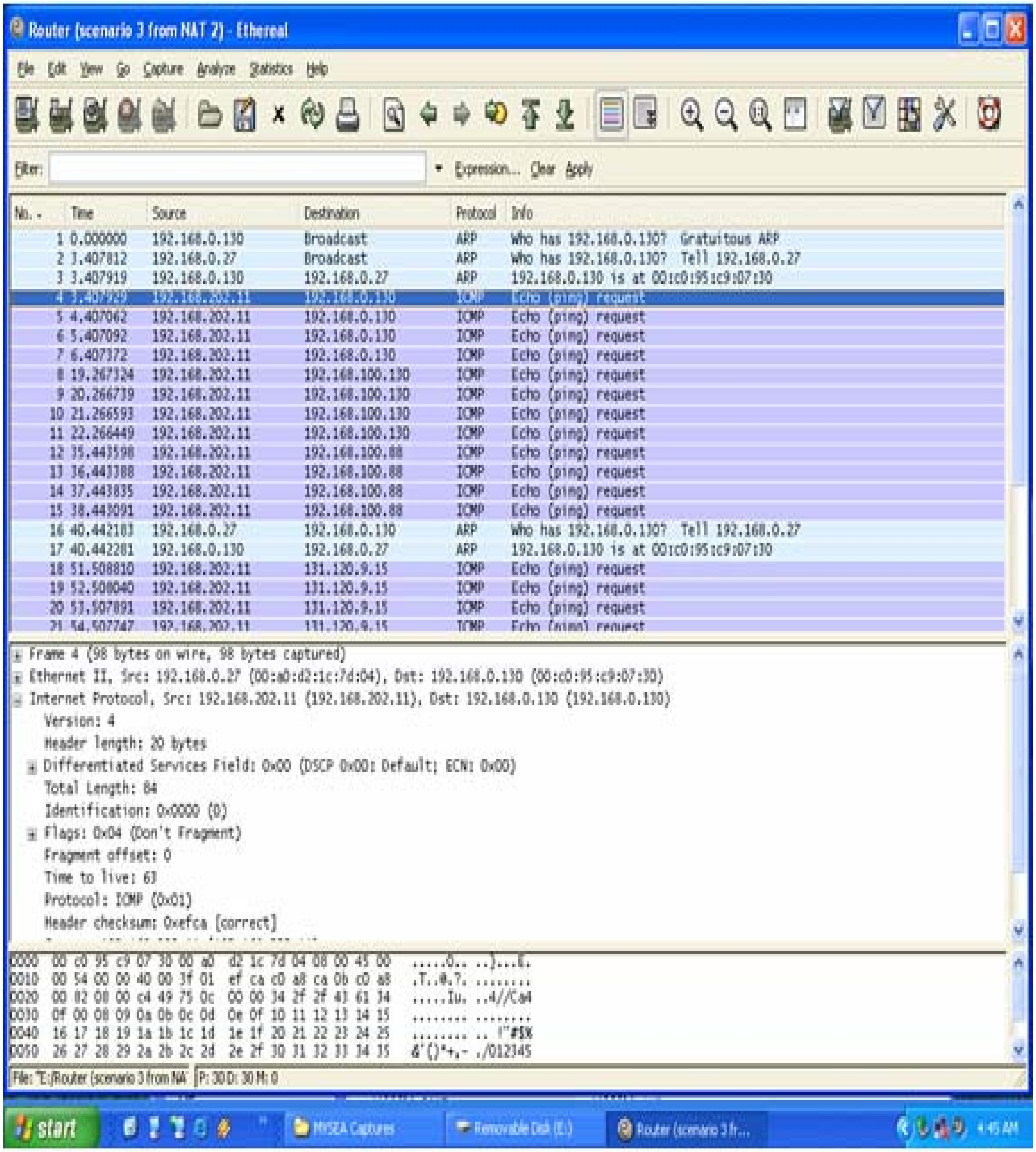

Figure 70. Test 6: Scenario 3 Packet Capture on Router (pinged from NAT 2)

F.6.2. NAT 1

The following three figures are snapshots of the packets captured on NAT 1 when the four interfaces were pinged from NAT 2.

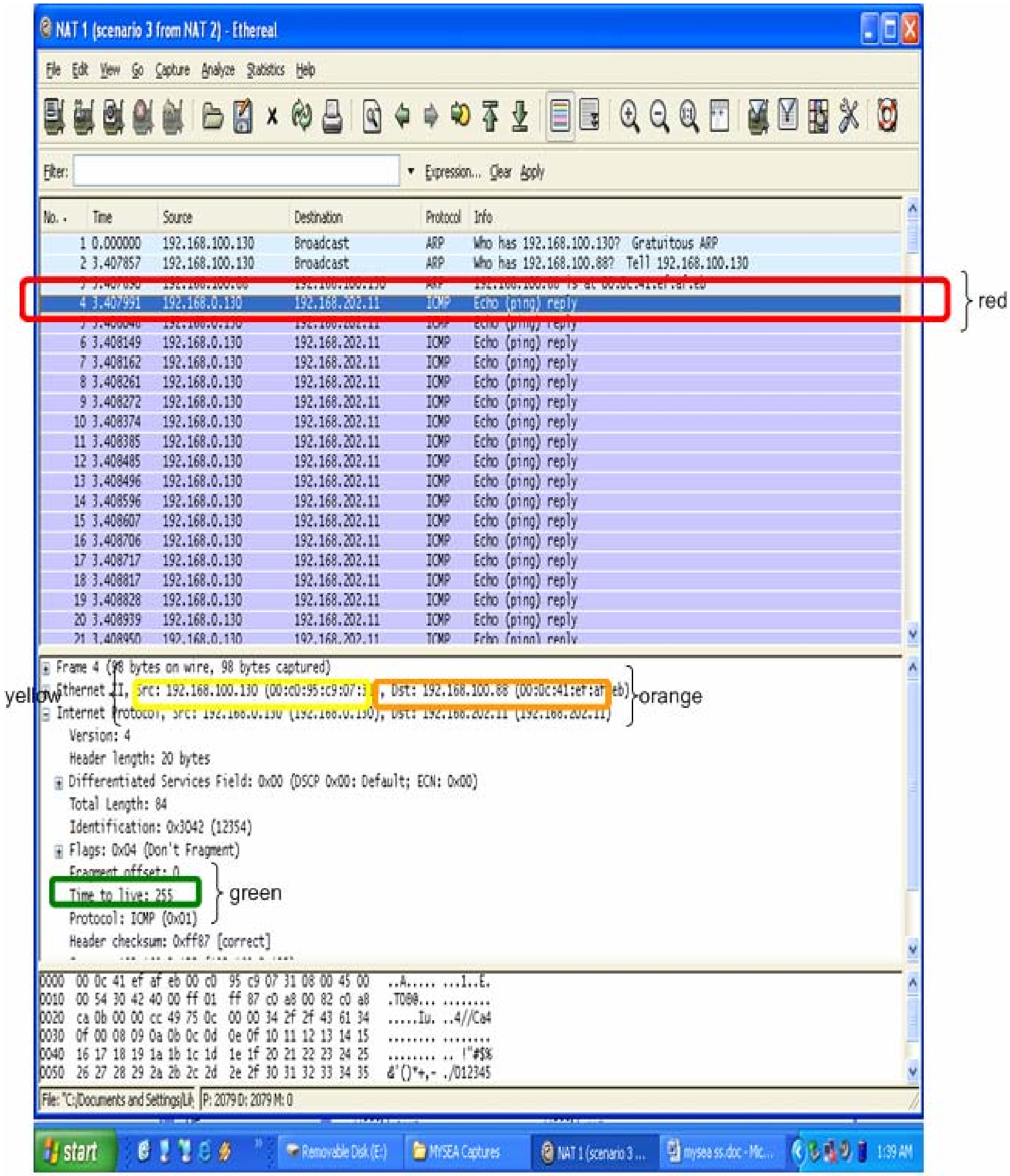


Figure 71. Test 6: Scenario 3 Packet Capture on NAT 1 (pinged from NAT 2), Part 1

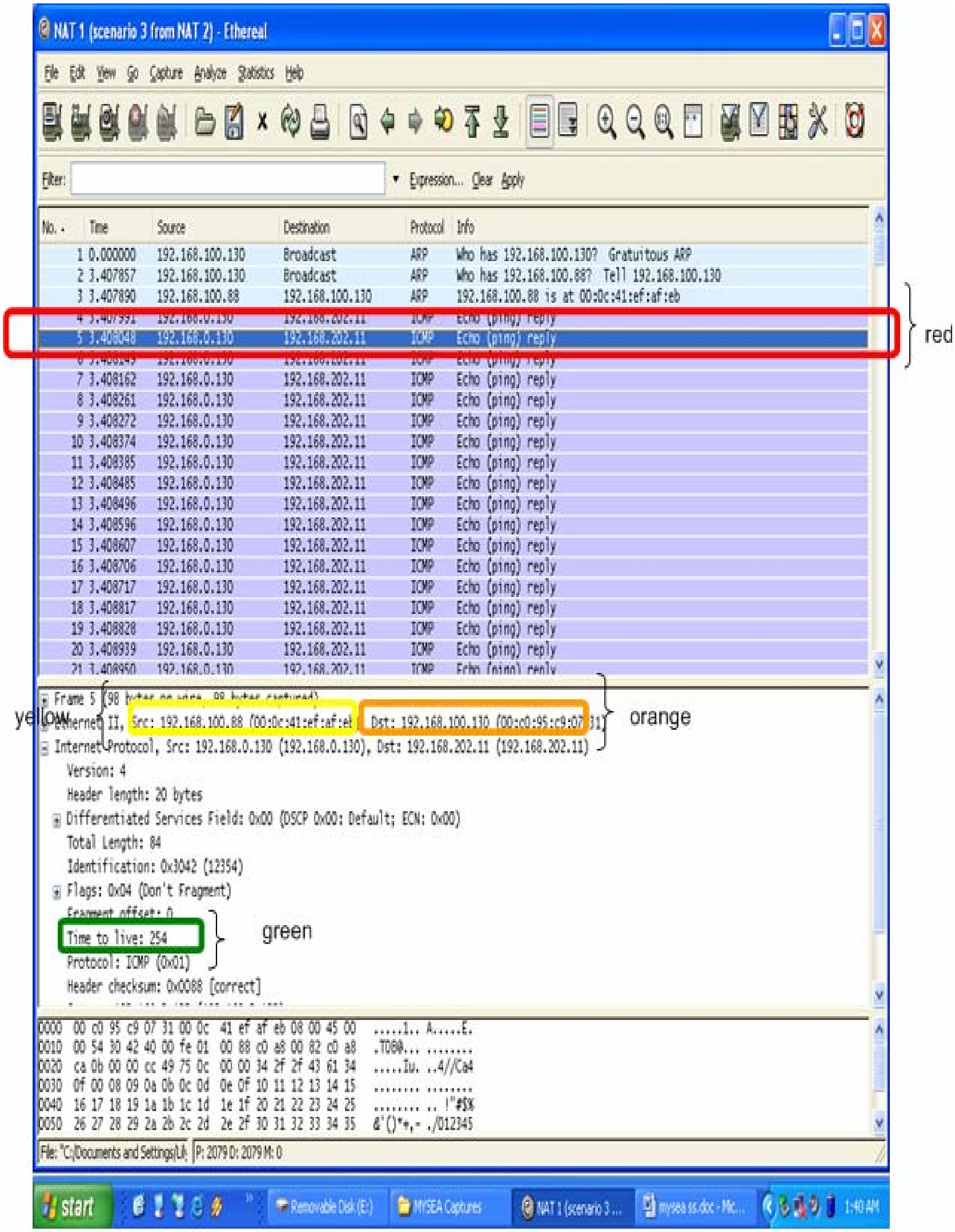


Figure 72. Test 6: Scenario 3 Packet Capture on NAT 1 (pinged from NAT 2), Part 2

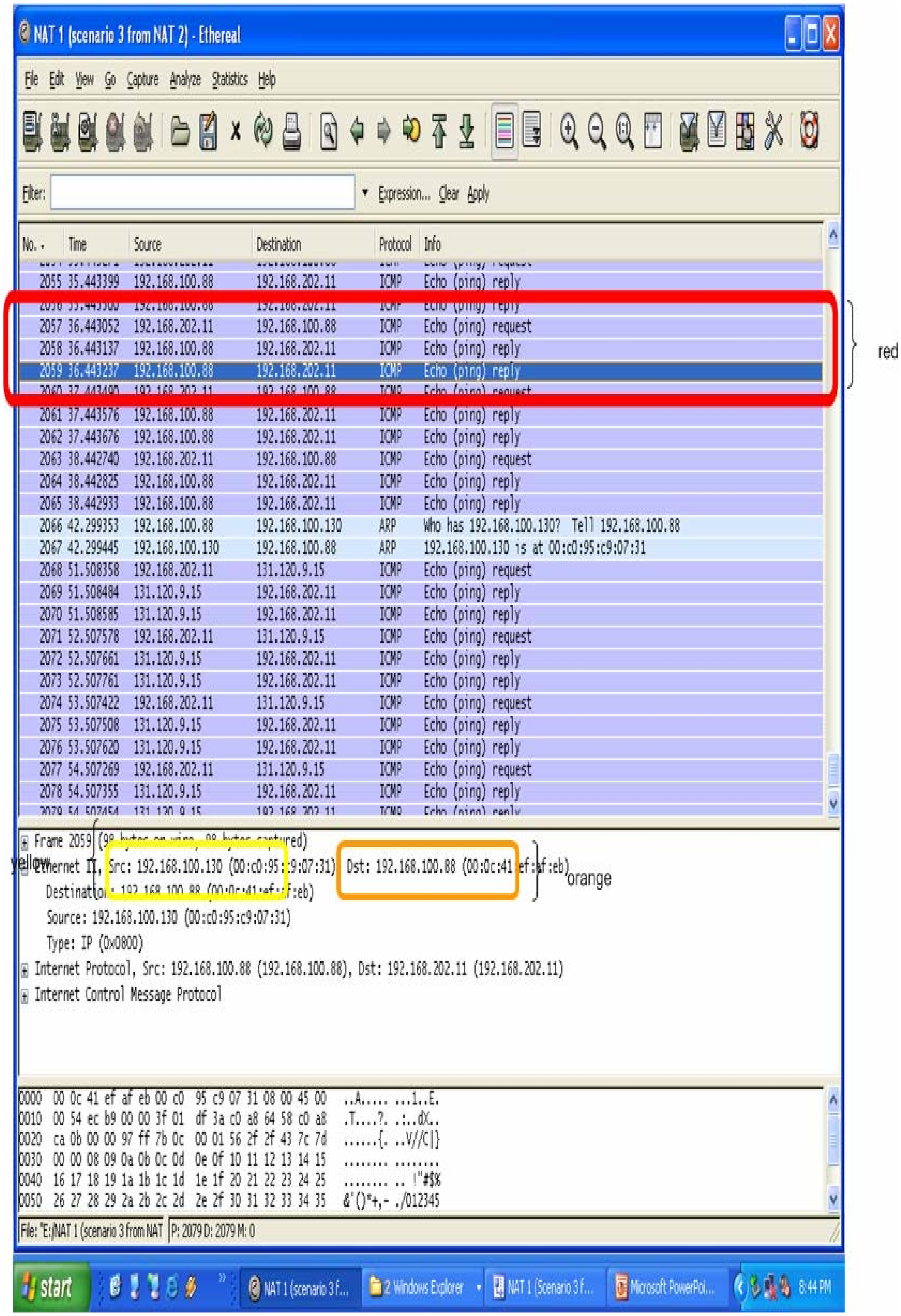


Figure 73. Test 6: Scenario 3 Packet Capture on NAT 1 (pinged from NAT 2), Part 3

### F.6.3. Analysis

NAT 2 failed to ping all four interfaces. The behaviors seen in the packet captures in this section are identical to the behaviors seen in section E.5. All the Echo replies (red outline in Figure 71) from 192.168.0.130 and 192.168.100.130 were bounced between 192.168.100.130 and 192.168.100.88 (yellow and orange outlines in Figure 71 and Figure 72). As a result, the replies never reached the Router (Figure 70). As described in F.5, every Echo request destined for 192.168.100.88 and 131.120.9.15 had two Echo replies (red outline in Figure 72). The first reply went from 192.168.100.88 to 192.168.100.130. When the MLS server received the reply, the XTS-400 routed it back to 192.168.100.88. This was the reason for seeing two Echo replies per request. Refer to E.6.3 for more detail explanation.

F.7. Packet Capture when pinged from Router

F.7.1. Router

The following is a snapshot of the packets captured on the Router when the four interfaces were pinged from Router.

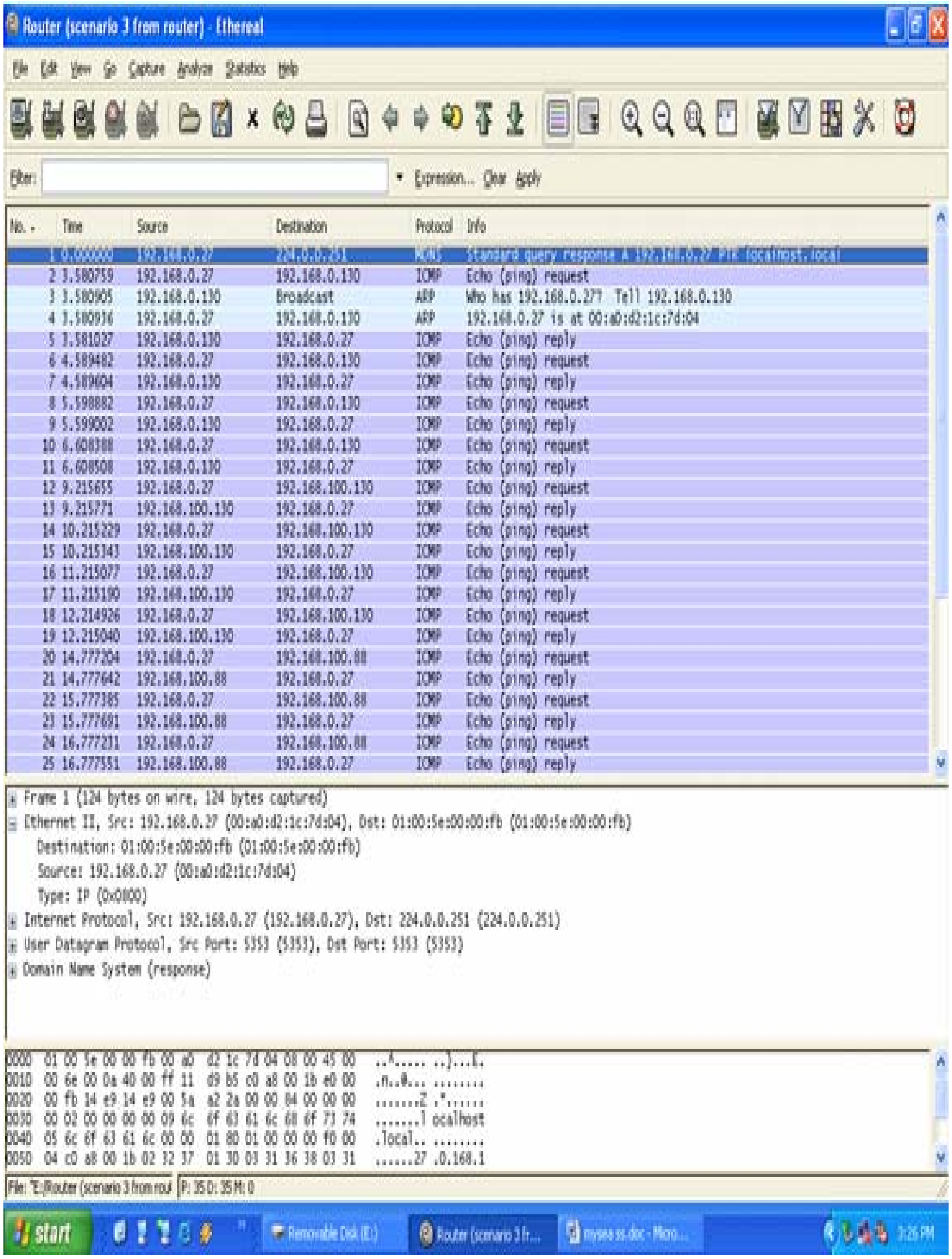


Figure 74. Test 6: Scenario 3 Packet Capture on Router (pinged from Router)

F.7.2. NAT 1

The following is a snapshot of the packets captured on NAT 1 when the four interfaces were pinged from Router.

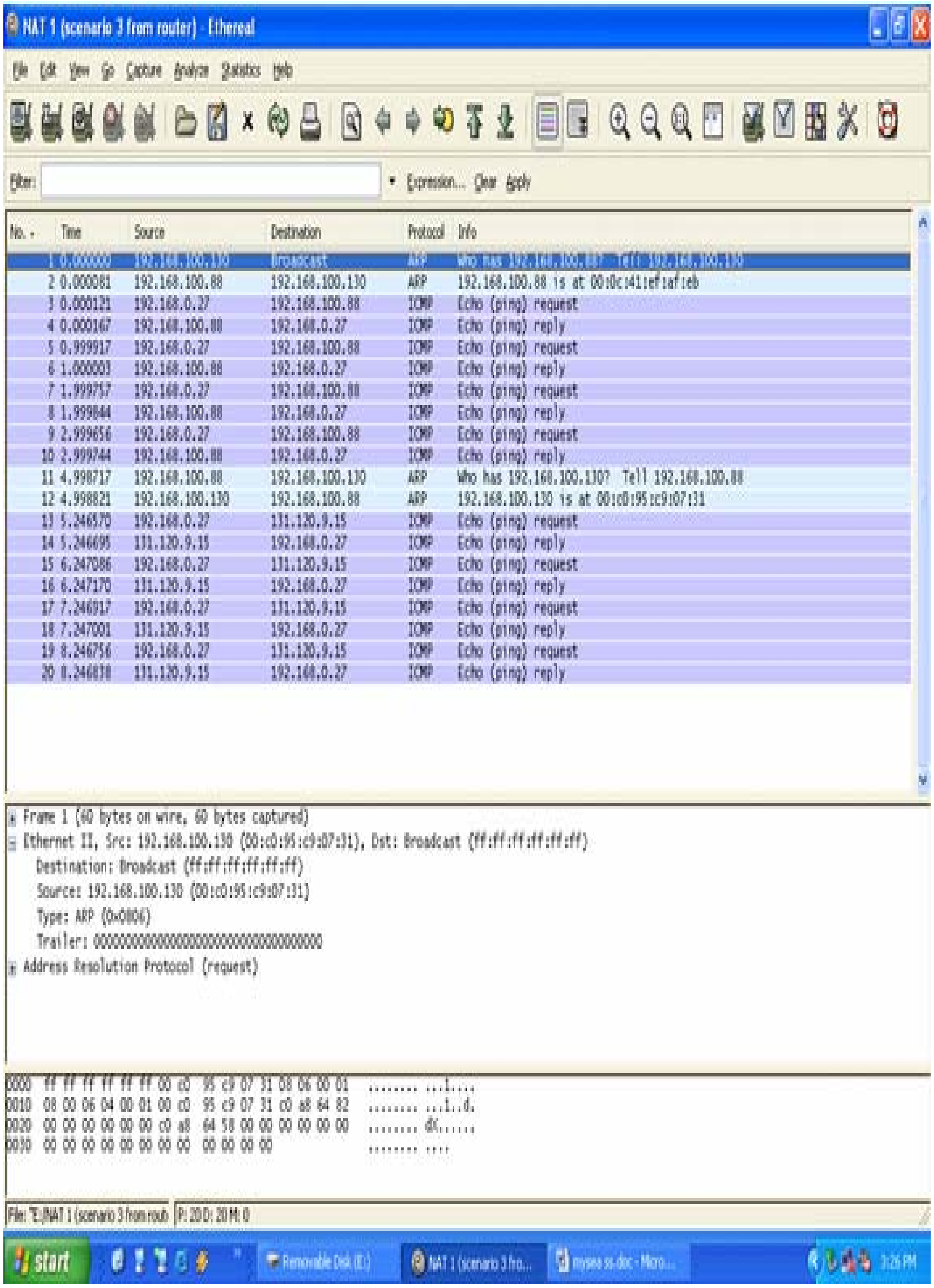


Figure 75. Test 6: Scenario 3 Packet Capture on NAT 1 (pinged from Router)

F.7.3. Analysis

The Router was able to ping all four interfaces as described in F.2. The behavior of this scenario is identical to the one in E.7. Refer to E.7.3 for a more detail analysis.

# G. Scenario 4

## G.1. Description

The MLS server is configured as follows in order: any packets received on its single-level interface (eth1, 192.168.100.130) is forwarded to the single-level interface of the Router (eth0, 192.168.0.27) and any packets received on its MLS LAN interface (eth0, 192.168.0.130) is forwarded to the single-level interface of the NAT 1 (eth1, 192.168.100.88).

## G.2. Operations

First, NAT 2 pings:

1. eth0 of MLS server
2. eth1 of MLS server
3. eth1 of NAT 1
4. eth0 of NAT 1

Then, Router pings:

5. eth0 of MLS server
6. eth1 of MLS server
7. eth1 of NAT 1
8. eth0 of NAT 1

## G.3. Network Configuration on MLS Server.

G.3.1. Type the following answers when prompted:

SAK

Enter command? tcpip_edit

Enter editor request? add

Enter TCP/IP daemon name? tcpip_mls

Enter TCPIP/IP daemon description? TCP/IP for MLS LAN network

Enter domain name? cisrlabmlstestbed1.com

| | |
|---|---|
| Enter host name? | mlsserver |
| Enable the subnets local flag? | n |
| Enable the IP forwarding flag? | y |
| Enable the IP send redirect flag? | y |
| Enable the shutdown on failure flag? | n |
| Use default TCP maximum retransmission? | y |
| Add the network interface configuration? | y |
| Enter TCP/IP device name? | /dev/ether0 |
| Enter interface address? | 192.168.0.130 |
| Enter destination address? | 0.0.0.0 |
| Enter broadcast address? | 192.168.0.255 |
| Enter network mask? | 255.255.255.0 |
| Add another network interface entry? | y |
| Enter TCP/IP device name? | /dev/ether1 |
| Enter interface address? | 192.168.100.130 |
| Enter destination address? | 0.0.0.0 |
| Enter broadcast address? | 192.168.100.255 |
| Enter network mask? | 255.255.255.0 |
| Add another network interface entry? | n |
| Add the route configuration? | y |
| Enter TCP/IP device name | /dev/ether1 |
| Is this route a default route | n |
| Enter destination address | 0.0.0.0 |
| Is destination address a host | n |
| Enter gateway address | 192.168.0.27 |
| Enter route metric | 1 |
| Add another network route entry | y |
| Enter TCP/IP device name | /dev/ether0 |
| Is this route a default route | n |
| Enter destination address | 0.0.0.0 |
| Is destination address a host | n |

| | |
|---|---|
| Enter gateway address | 192.168.100.88 |
| Enter route metric | 1 |
| Add another network route entry | n |
| Add the resolver configuration? | n |

G.4. Preparation and Testing

G.4.1. On NAT 1,

G.4.1.1. Launch Ethereal

G.4.1.2. Go to the Capture menu

G.4.1.3. Go to Interfaces

G.4.1.4. Click on Capture 192.168.100.88

G.4.2. On Router,

G.4.2.1. Launch Ethereal

G.4.2.2. Go to the Capture menu

G.4.2.3. Go to Interfaces

G.4.2.4. Click on Capture 192.168.0.27

G.4.3. On NAT 2,

G.4.3.1. Run the following commands:

ping –c 4 192.168.0.130

ping –c 4 192.168.100.130

ping –c 4 192.168.100.88

ping –c 4 131.120.9.15

G.4.4. Repeat the above commands on the Router

G.4.5. Stop Ethereal captures on both NAT 1 and Router

G.5. Result

Table 14 lists the result of the Scenario 4. The first column shows where the ping was initiated and the first row shows what hosts/IP addresses were pinged. The result from Scenario 4 is exactly the same as the result obtained in Scenario 1.

| from \ to | 192.168.0.130 (eth0, MLS LAN interface of MLS server) | 192.168.100.130 (eth1, single-level interface of MLS server) | 192.168.100.88 (eth1, single-level interface of NAT 1) | 131.120.9.15 (eth0, public interface of NAT 1) |
|---|---|---|---|---|
| NAT 2 | Successful | Successful | Successful | Failed |
| Router | Successful | Successful | Successful | Failed |

Table 14. Test 6: Scenario 4 Result

G.6. Packet Capture when pinged from NAT 2

G.6.1. Router

The following is a snapshot of the packets captured on the Router when the four interfaces were pinged from NAT 2.

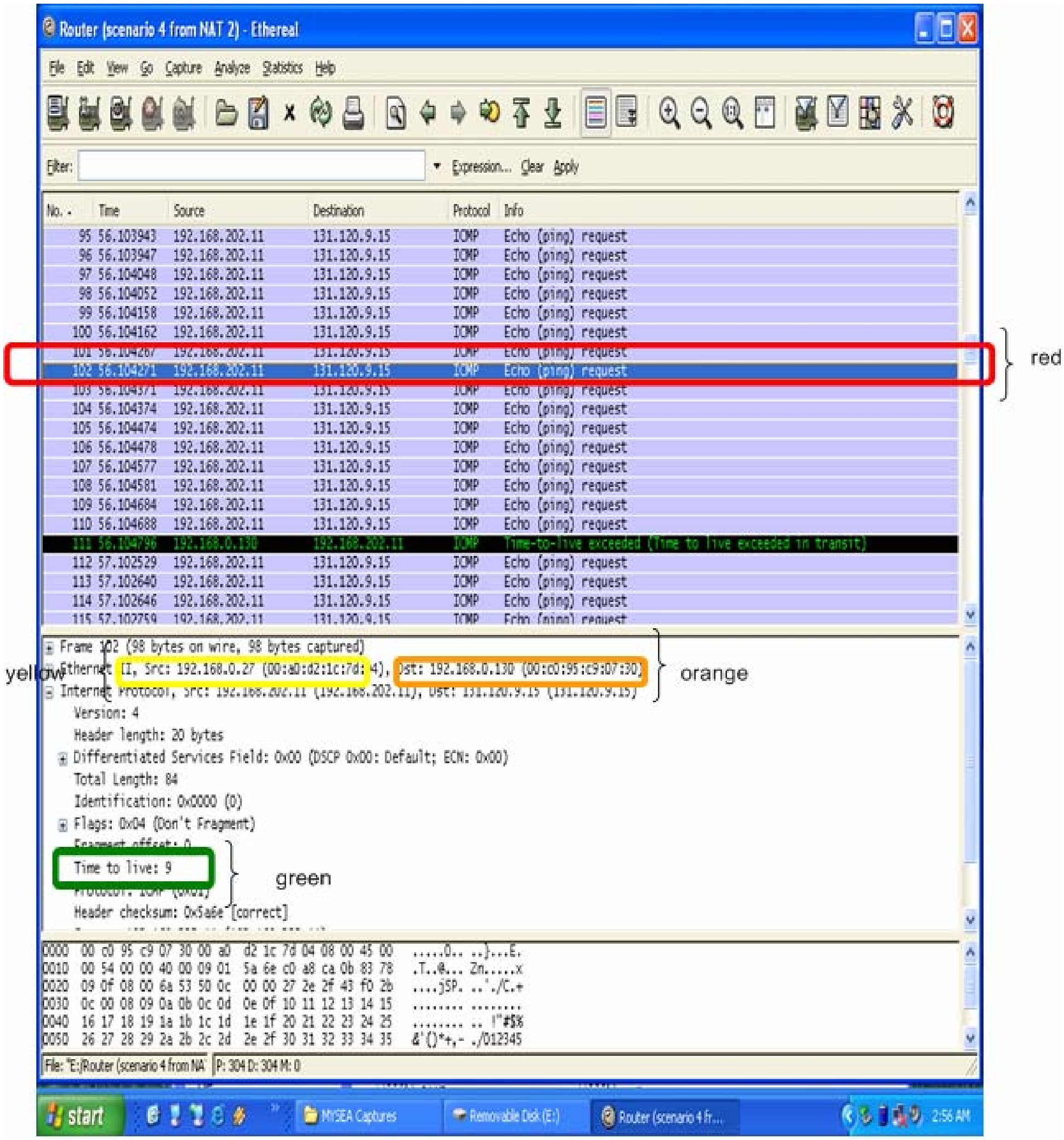


Figure 76. Test 6: Scenario 4 Packet Capture on Router (pinged from NAT 2), Part 1

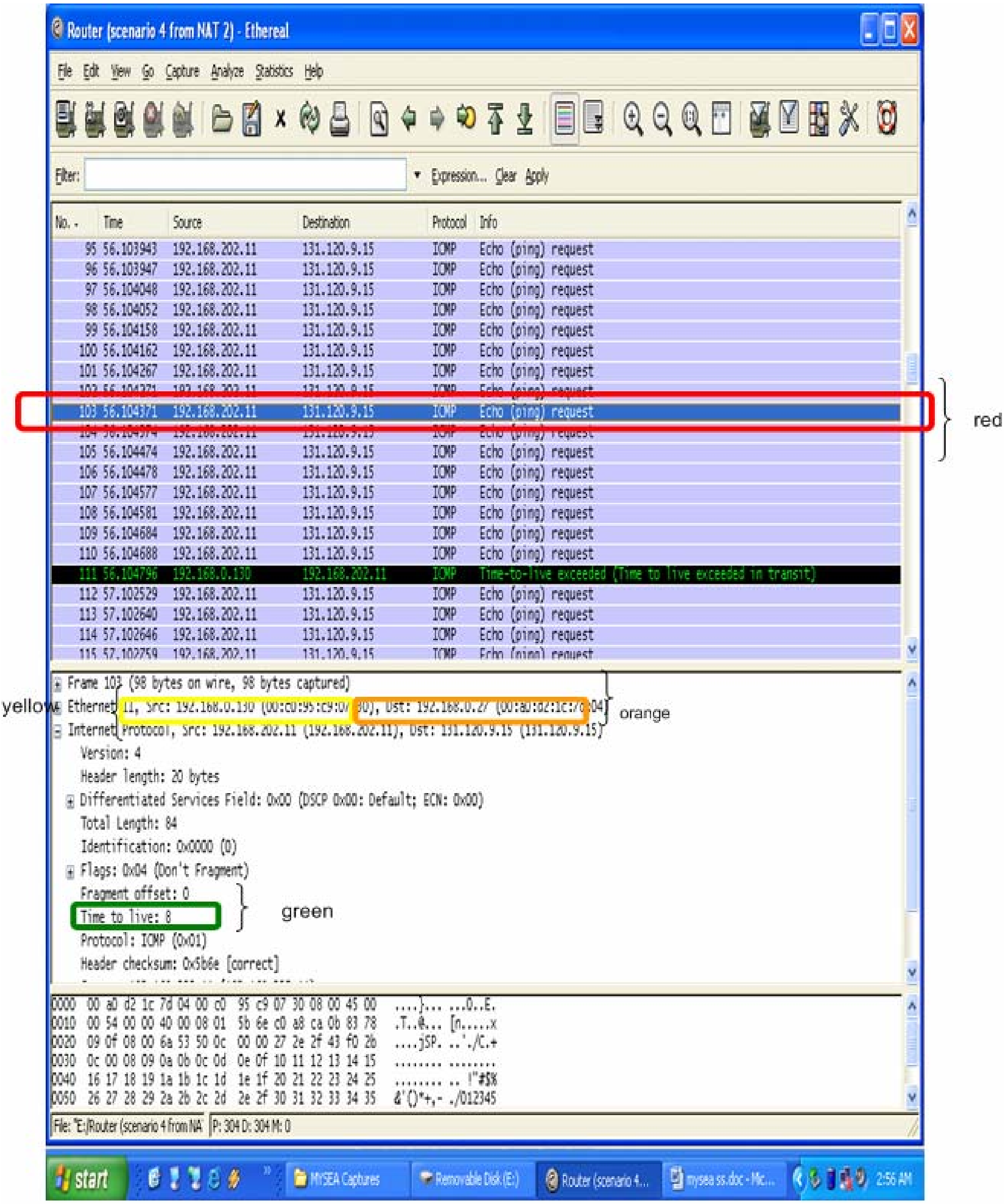


Figure 77. Test 6: Scenario 4 Packet Capture on Router (pinged from NAT 2), Part 2

G.6.2. NAT 1

The following is a snapshot of the packets captured on the NAT 1 when the four interfaces were pinged from NAT 2.

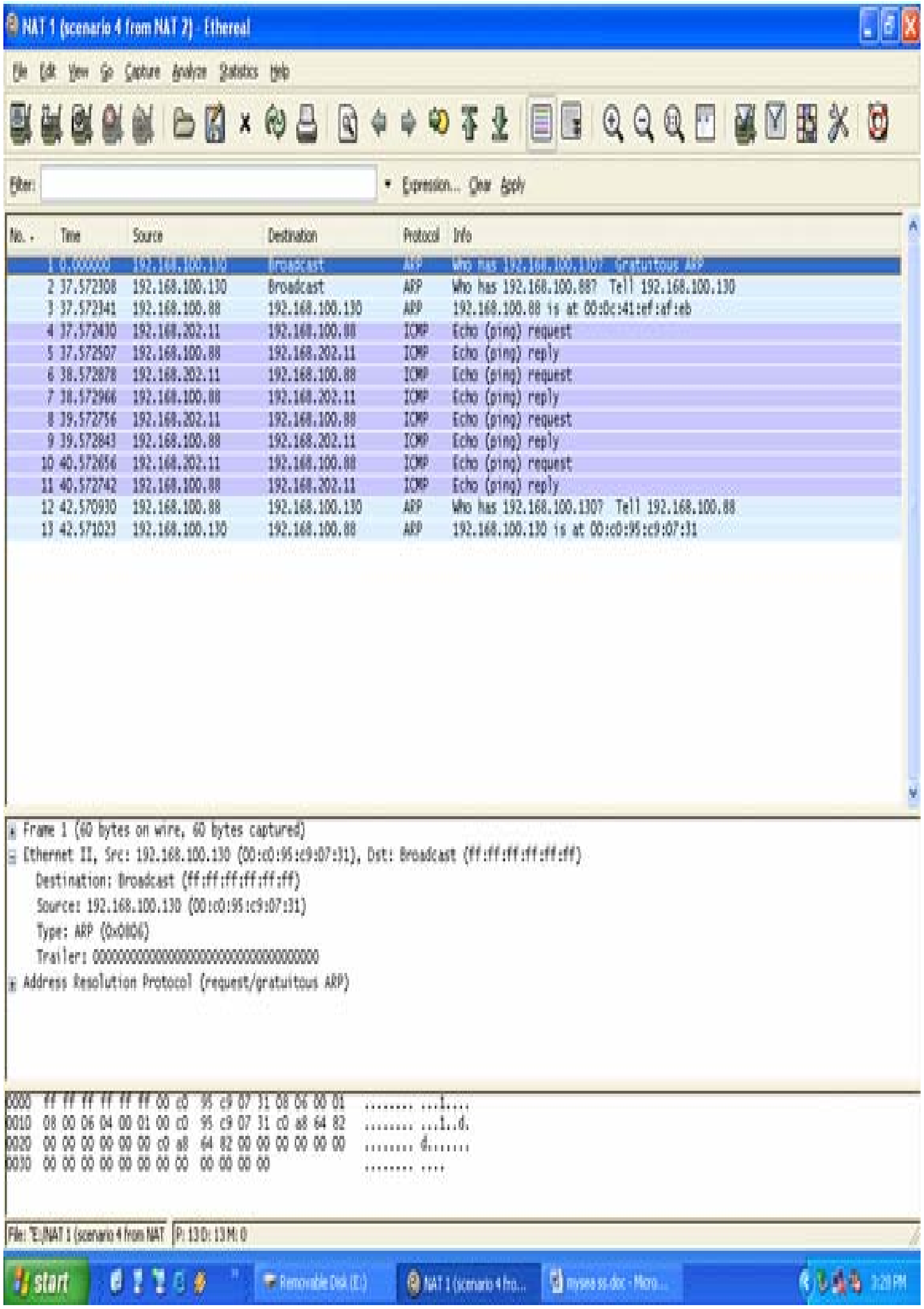


Figure 78. Test 6: Scenario 4 Packet Capture on NAT 1 (pinged from NAT 2)

### G.6.3. Analysis

Scenario 4 had the same result as Scenario 1. In other words, NAT 2 was able to ping 192.168.0.130, 192.168.100.130, and 192.168.100.88 only. It failed to ping 131.120.9.15. The Echo request destined for 131.120.9.15 was routed from 192.168.0.27 to 192.168.0.130 (yellow and orange outlines in Figure 76). However, XTS-400 forwarded the packet back to 192.168.0.27. This sequence of events occurred until the time-to-live exceeded. As a result, the Echo requests never reached NAT 1 (Figure 77). Refer to D.6.3 for more details.

G.7. Packet Capture when pinged from Router

G.7.1. Router

The following is a snapshot of the packets captured on the Router when the four interfaces were pinged from Router.

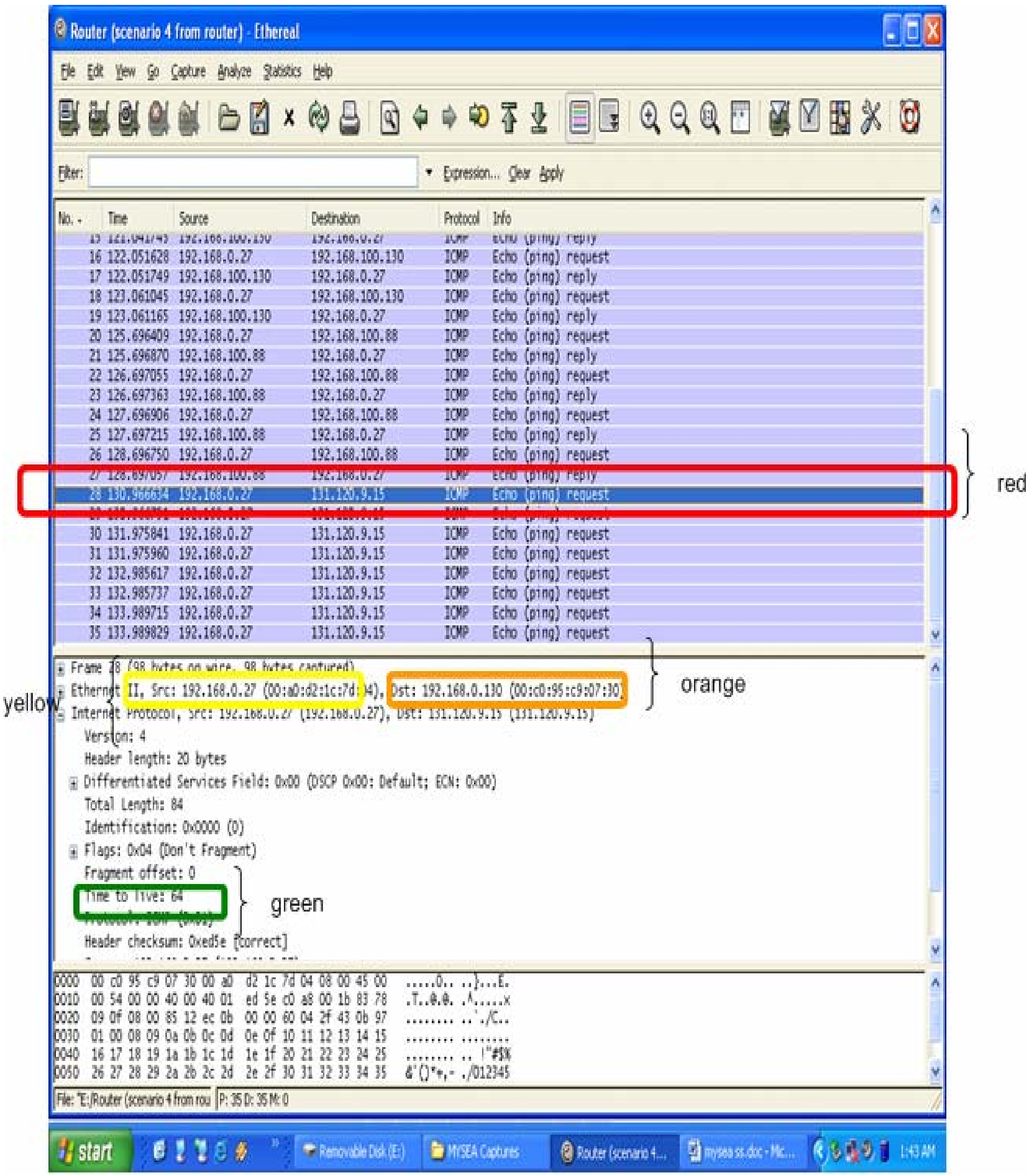


Figure 79. Test 6: Scenario 4 Packet Capture on Router (pinged from Router), Part 1

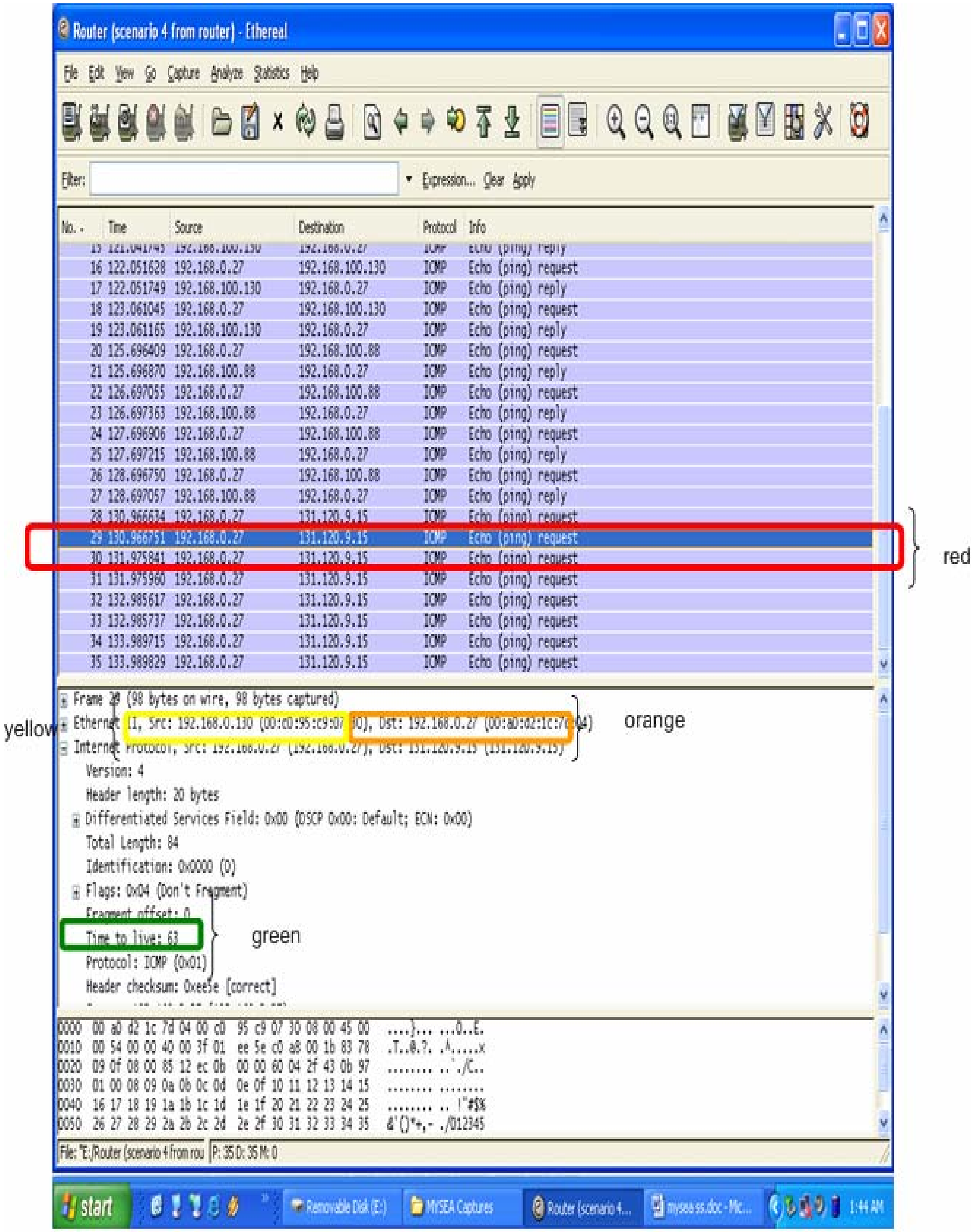


Figure 80. Test 6: Scenario 4 Packet Capture on Router (pinged from Router), Part 2

G.7.2. Router

The following is a snapshot of the packets captured on the NAT 1 when the four interfaces were pinged from Router.

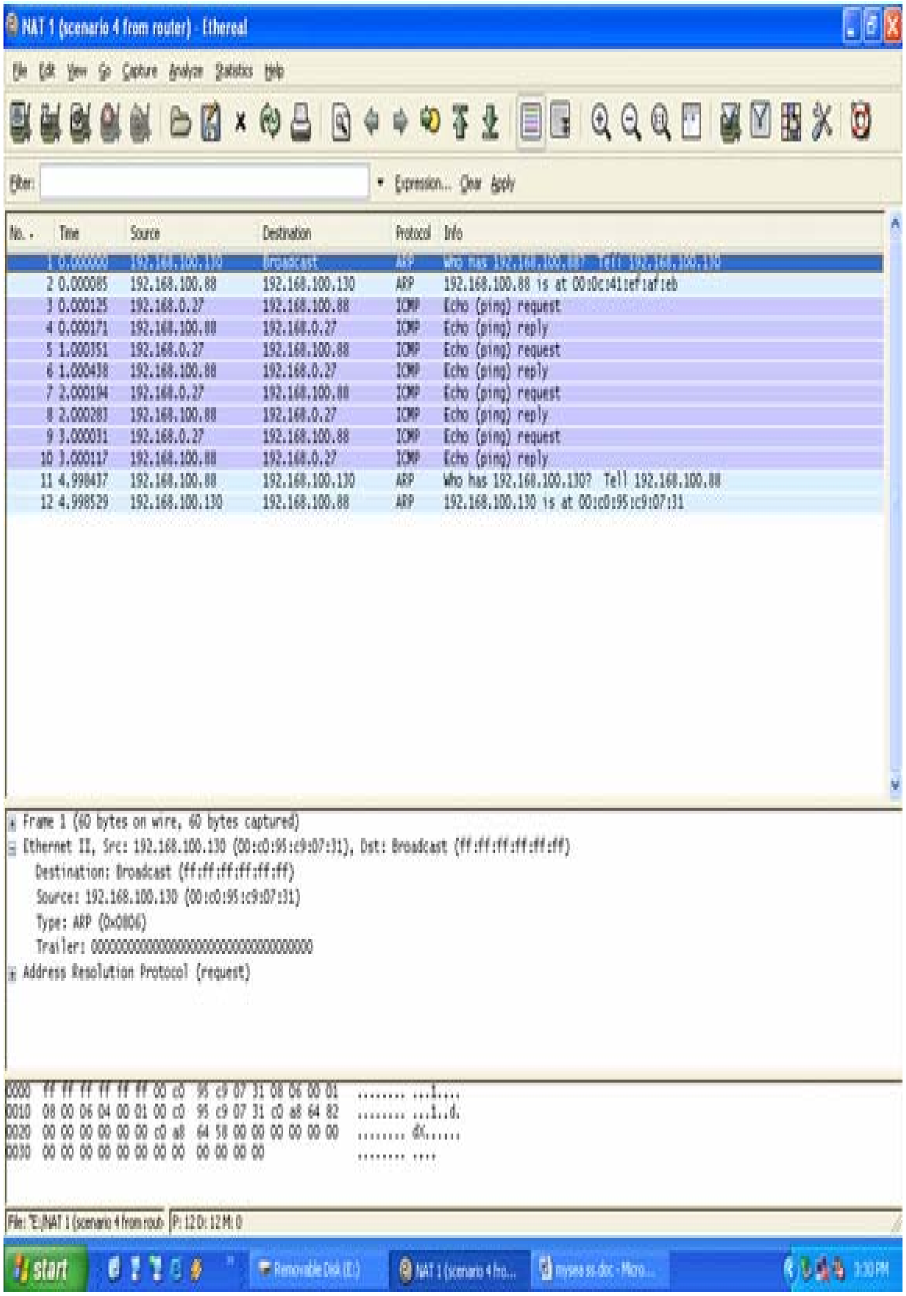

Figure 81. Test 6: Scenario 4 Packet Capture on NAT 1 (pinged from Router)

### G.7.3. Analysis

The Router could ping both interfaces of the MLS server because the Router and the MLS server share a peer-to-peer relationship. The Router was also able to ping NAT 1 since XTS-400 has routing capabilities to route packets to its immediate peers. However, pinging 131.120.9.15 was unsuccessful. Echo requests destined (red outline in Figure 79) for 131.129.9.15 were sent out by the Router (yellow and orange outlines in Figure 79). However, XTS-400 sent those requests back to the Router when it received them (yellow and orange outlines in Figure 80). The Router stopped routing the requests further as it recognized them. Therefore, the Echo requests never reached NAT 1 (Figure 81).

## H. Observation

A number of observations can be made from analyzing the results and packet captures for the four scenarios. First, there were two different sets of results from running the four test scenarios. Scenarios 1 and 4 generated the first set and scenarios 2 and 3 have generated the other set of results (refer to sections D.5, E.5, F.5, and G.5). Second, each scenario that shared the same gateway address sequence in its routing configuration yielded identical results. In other words, the results were dependent on the order of the gateway address and were independent of the device name in the routing configuration (refer to Table 9). Third, XTS-400 seemed to always forward packets destined for unknown networks to the gateway indicated in the first static route in its routing table. In scenarios 1 and 4, XTS-400 bounced Echo requests it received from 131.120.9.15 to 192.168.0.27 instead of forwarding them onto NAT 1. Also in scenarios 1 and 4, similar behavior of XTS-400 forwarding packets to 192.168.0.27 was seen when the Router pinged 131.120.9.15. In both cases, the XTS-400 did not have routing information for the 131.120.9.x network and the gateway for its first static route is 192.168.0.27. The XTS-400 always routed packets destined for unknown networks to 192.168.100.88 instead in scenarios 2 and 3 where the 192.168.100.88 was the gateway in its first static route.

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